

## RECOMMENDATION ITU-R BO.1294

**COMMON FUNCTIONAL REQUIREMENTS FOR THE RECEPTION OF DIGITAL  
MULTIPROGRAMME TELEVISION EMISSIONS BY SATELLITES  
OPERATING IN THE 11/12 GHz FREQUENCY RANGE\***

(Question ITU-R 217/11)

(1997)

The ITU Radiocommunication Assembly,

*considering*

- a) that there is an increasing interest worldwide in the introduction of satellite digital multiprogramme television services due to advantages in video and sound quality, flexibility of use, spectrum efficiency and emission robustness;
- b) that Recommendation ITU-R BO.1211 recommends the consideration of a digital multiprogramme television system, along with other digital existing systems, in converging to a worldwide standard;
- c) that three digital multiprogramme television systems, including the one described in the Recommendation ITU-R BO.1211, have been developed and are in widespread use supporting satellite television services in the 11/12 GHz band;
- d) that these three systems provide the same functions using very similar architectures;
- e) that it may be advantageous to encourage the maximum possible commonality in the implementation of these systems, in order to allow for maximum economies of scale for integrated circuits;
- f) that the ITU-R, in cooperation with the industry, has analysed the commonalities between existing systems, defined the functions of a generic reference model and identified the universal elements of the various subsystems;
- g) the proposal of the World Broadcasting Union for increasing the usefulness of ITU-R Recommendations and the fact that detailed feasibility analysis and data concerning cost and complexity are provided in Report ITU-R BO.2008,

*recommends*

- 1** that one of the transmission systems described in Annex 1 be selected when implementing digital multiprogramme television services via satellite;
- 2** that the universal elements of the common functional requirements of a satellite integrated receiver-decoder (IRD), as described in § 3 of Annex 1, serve as a basis for implementation of the services in those areas where more than one system coexist or may coexist in the future,

*further recommends*

- 1** that further studies be carried out to address the benefits of including other essential satellite IRD functions that are not specified in this Recommendation.

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\* This Recommendation should be brought to the attention of the International Electrotechnical Commission/International Standardization Organization (IEC/ISO).

## ANNEX 1

**Common functional requirements for the reception of digital multiprogramme television emissions by satellite operating in the 11/12 GHz frequency range**

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## 1 Introduction

Satellite digital TV systems have shown their advantages with respect to the analogue TV allowing a more efficient use of the satellite frequency spectrum available and establishing a more robust scenario with respect to interference protection.

With the aim to promote the convergence on a worldwide standard for satellite digital multi-programme reception systems for television, sound and data services, the common functional requirements for the reception of digital multiprogramme television emissions by satellite are described. These common functional requirements configure the universal elements of the satellite integrated receiver decoder (IRD). Although the present Annex addresses satellites operating in the 11/12 GHz frequency range, other ranges are not excluded.

The universal elements of the satellite IRD are capable of receiving emissions from System A, System B and System C.

The common and specific elements of each system have been analysed and it has been concluded that the implementation of the universal elements of a Satellite IRD is feasible. This Annex analyses the common elements among existing systems, defines and describes the functions of a generic system model and identifies the processes and the minimum set of parameters of the various sub-systems of the universal elements of a satellite IRD.

The feasibility of the implementation of the common elements in a satellite IRD has been demonstrated in consultation with the industry.

## 2 Generic reference model for the common functional requirements of a satellite IRD

A generic reference model for the common functional requirements of a satellite IRD has been produced in order to analyse the feasibility of the universal elements of a satellite IRD, identifying the applicability of the generic reference model to the three systems currently in use.

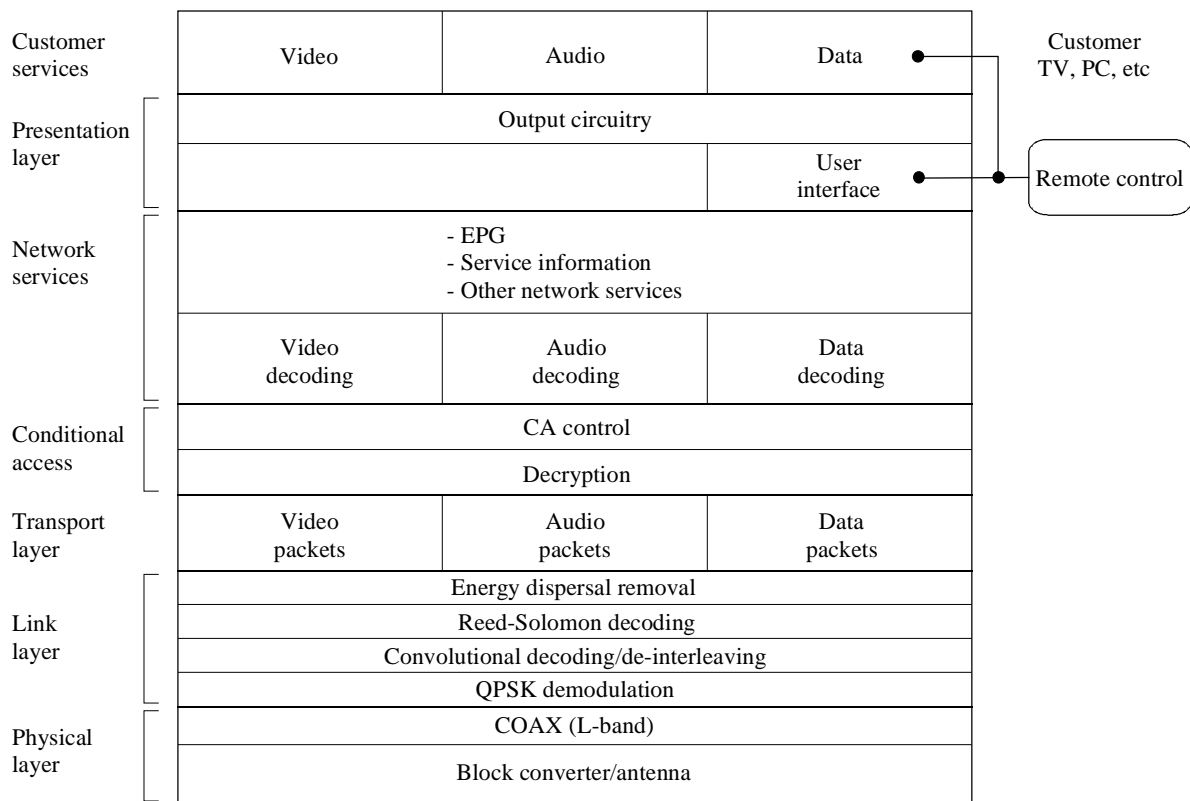
This generic model refers to domestic type IRDs and does not intend to specify the functions required for professional IRDs.

The generic reference model has been defined based on the functions required for covering all layers of a typical IRD protocol stack. For reference, Figure 1 presents the typical IRD protocol stack which is based on the following layers:

- *Physical and link layers* covering the typical front-end functions: tuner, QPSK demodulator, convolutional decoding, de-interleaving, Reed-Solomon decoding and energy dispersal removal.
- *Transport layer* responsible for the demultiplexing of the different programs and components as well as the depacketization of the information (video, audio and data).
- *Conditional access* functions which control the operation of external decoder functions (Common Interface for Conditional Access as an option).

- *Network services* performing the video and audio decoding as well as the management of electronic program guide (EPG) functions and service information and, optionally, data decoding.
- *Presentation Layer* responsible, among other things, for the user interface, operation of the remote control, etc.
- *Customer services* covering the different applications based on video, audio and data.

FIGURE 1  
Typical IRD protocol stack



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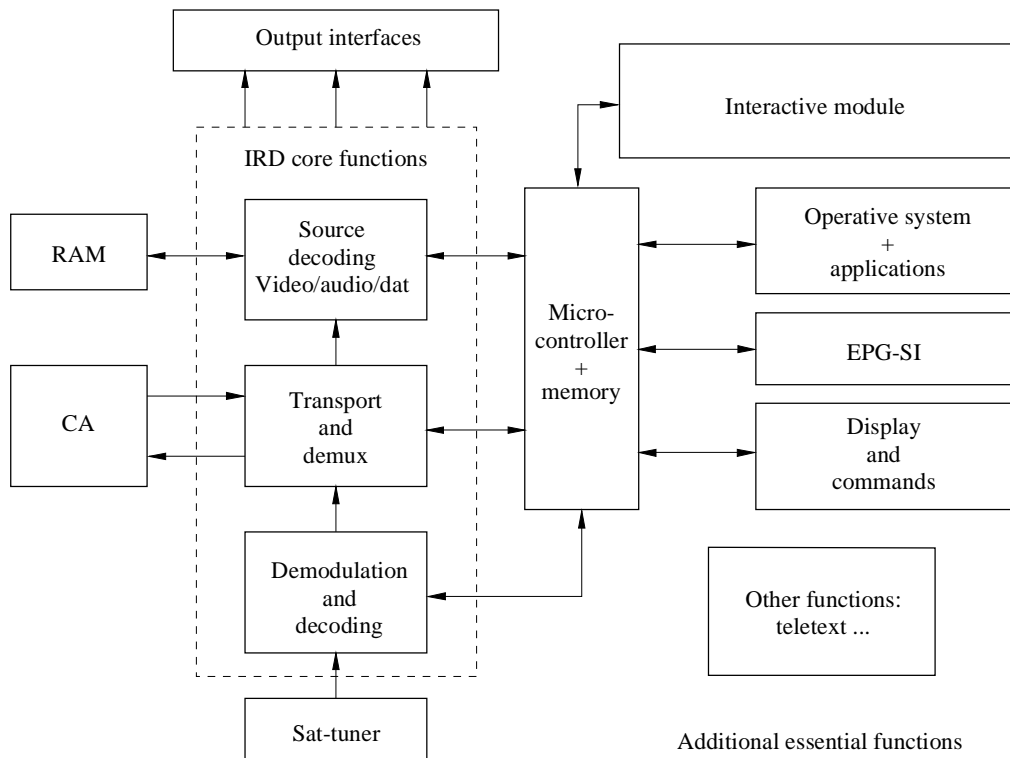
Based on the protocol stack, the generic reference model for the satellite IRD (Figure 2) is derived.

Two types of functions are identified in the generic reference model: IRD core functions and other additional essential functions.

- *The IRD core functions* cover the key IRD functions which define the digital TV system. IRD core functions include:
  - Demodulation and decoding,
  - Transport and demultiplexing,
  - Source decoding video, audio and data.
- *The additional essential functions* are required to perform the operation of the system and upgrade it with additional and/or complementary features. These functions are closely related to the service provision. The following functions and blocks could be considered as the additional essential functions and may differentiate one IRD from another:
  - Satellite tuner,
  - Output interfaces,
  - Operative system and applications,

- EPG,
- SI (service/system information),
- CA (conditional access),
- Display, remote control and different commands,
- ROM, RAM and FLASH memory,
- Interactive module,
- Microcontroller,
- Other functions as teletext, subtitling, etc.

FIGURE 2  
Generic reference model for a satellite IRD



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### 3 Universal elements of a satellite IRD

An analysis of the core and essential functions of the three systems, as provided in § 4, validated the feasibility of defining universal elements for a satellite IRD.

The universal elements of a satellite IRD perform the following functions:

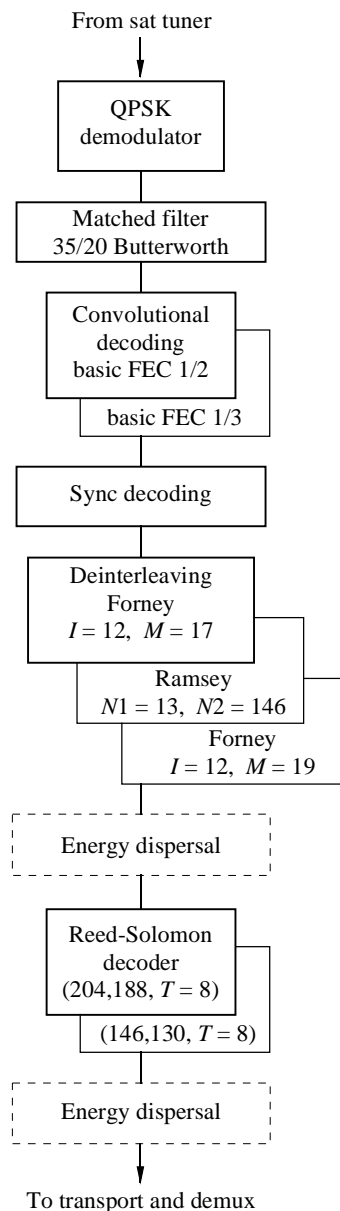
- Demodulation and decoding.
- Transport and demultiplexing.
- Source decoding of video, audio and data.

It is understood that definition of additional essential functions is out of the scope of this Recommendation because these functions would be specific to each service and very close to the specific implementation by each manufacturer, subject to a number of external and service conditions. Therefore the potential diversity of additional essential functions among satellite IRDs does not impact on the universal elements of the satellite IRD.

### 3.1 Demodulation and decoding

The block diagram of the demodulation and decoding functions for the universal elements of a satellite IRD is presented in Figure 3. Overlapped blocks represent functions with common elements for the three systems although with different characteristics. Dashed blocks represent functions not utilized by all three systems.

FIGURE 3  
Block diagram for demodulation and channel decoding



FEC: forward error correction

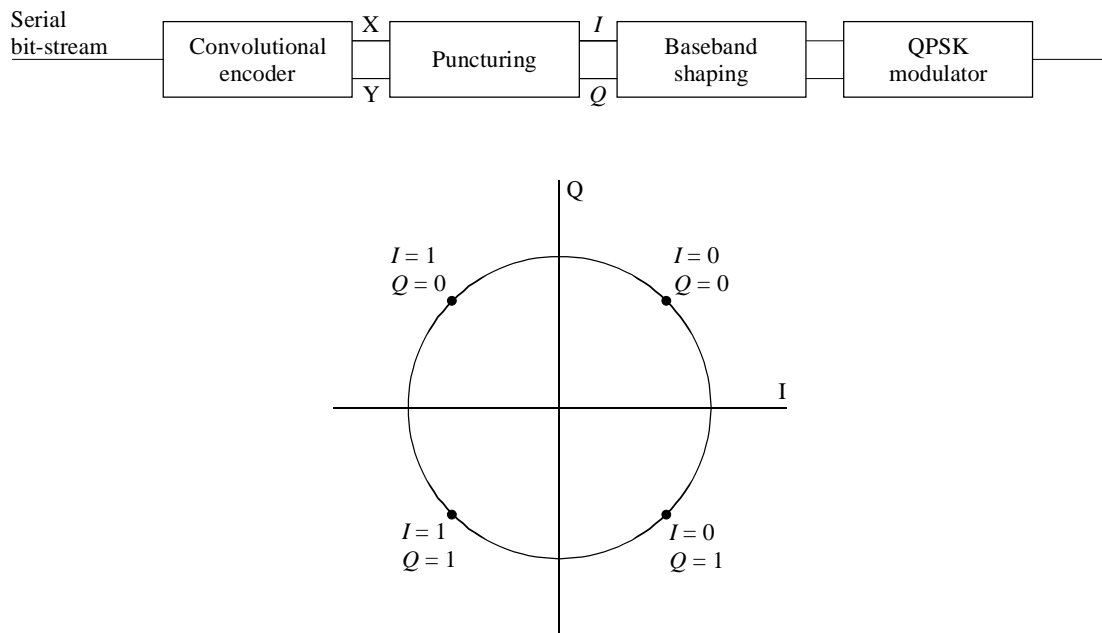
### 3.1.1 QPSK demodulator

This universal element of a satellite IRD performs the quadrature coherent demodulation function and the analogue to digital conversion, providing “soft decision”  $I$  and  $Q$  information to the inner decoder.

This universal element of a satellite IRD will be capable of demodulating a signal employing conventional Gray-coded QPSK modulation with absolute mapping (no differential coding).

Bit mapping in the signal as given in Fig. 4 will be used.

FIGURE 4  
QPSK constellation



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### 3.1.2 Matched filter

This universal element of a satellite IRD performs the complementary pulse shaping filtering type according to the roll-off. The use of a finite impulse response (FIR) digital filter could provide equalization of the channel linear distortions in the IRD.

This universal element of a satellite IRD will be capable of processing signal with the following shaping and roll-off factors:

Square root raised cosine:  $\alpha = 0.35$  and  $0.20$

Band-limited 4th order Butterworth: Standard and truncated-spectrum modes

Information about the template for the signal spectrum at the modulator output is given in § 5.1.

### 3.1.3 Convolutional decoding

This universal element of a satellite IRD performs first level error protection decoding. It will operate at an input equivalent “hard decision” BER of the order of between  $1 \times 10^{-1}$  and  $1 \times 10^{-2}$  (depending on the adopted code rate), and will produce an output BER of about  $2 \times 10^{-4}$  or lower. This output BER corresponds to QEF service after outer code correction. It is possible that this unit makes use of “soft decision” information. This unit is in a position to try each of the code rates and puncturing configurations until lock is acquired. Furthermore, it is in a position to resolve  $\pi/2$  demodulation phase ambiguity.

The inner code has the following characteristics:

- Viterbi and puncturing.
- Code constraint length  $K = 7$ .

This universal element of a satellite IRD will be capable of decoding the three different convolutional codes. The system will allow convolutional decoding with code rates based on a rate of either 1/2 or 1/3:

- Based on a basic rate 1/2: FEC = 1/2, 2/3, 3/4, 5/6, 6/7 and 7/8.
- Based on a basic rate 1/3: FEC = 5/11, 1/2, 3/4, 2/3, 3/5, 4/5, 5/6 and 7/8.

Specific characteristics are provided in § 5.2.

### 3.1.4 Sync byte decoder

This universal element will decode the sync bytes. This decoder provides synchronization information for the de-interleaving. It is also in a position to recover  $\pi$  ambiguity of QPSK demodulator (not detectable by the Viterbi decoder).

Specific characteristics are provided in § 5.3.

### 3.1.5 Convolutional de-interleaver

This universal element allows the error bursts at the output of the inner decoder to be randomized on a byte basis in order to improve the burst error correction capability of the outer decoder.

This universal element of a satellite IRD will be capable of receiving Ramsey Type II ( $N_1 = 13$ ,  $N_2 = 146$ ) and Ramsey Type III (Forney approach) ( $I = 12$ ,  $M = 17$  and 19) convolutional interleaver systems, as specifically defined in § 5.4.

### 3.1.6 Reed-Solomon decoder

This universal element of a satellite IRD provides second level error protection. It is in a position to provide QEF output (i.e. BER of about  $1 \times 10^{-10}$  and  $1 \times 10^{-11}$ ) in the presence of input error bursts at a BER of about  $7 \times 10^{-4}$  or better with infinite byte interleaving. In the case of interleaving depth  $I = 12$ , BER =  $2 \times 10^{-4}$  is assumed for QEF.

This universal element of a satellite IRD will decode the following characteristics:

- Reed-Solomon generator: (255,239,  $T = 8$ )
- Reed-Solomon Code generator polynomial:

$$(x + \alpha^0) + (x + \alpha^1) + \dots + (x + \alpha^{15})$$

or

$$(x + \alpha^1) + (x + \alpha^2) + \dots + (x + \alpha^{16})$$

where:

$$\alpha = 02_{\text{h}}$$

- Reed-Solomon field generator polynomial:

$$x^8 + x^4 + x^3 + x^2 + x$$

Specific characteristics are provided in § 5.5.

### 3.1.7 Energy dispersal removal

This universal element of a satellite IRD recovers the user data by removing the randomizing pattern used for energy dispersal purposes. It can be implemented in such a way as to be capable of derandomizing signals where the derandomizing process has been placed before or after the Reed-Solomon decoder. This universal element of a satellite IRD may implement a bypass to this feature.

Specific characteristics are provided in § 5.6.



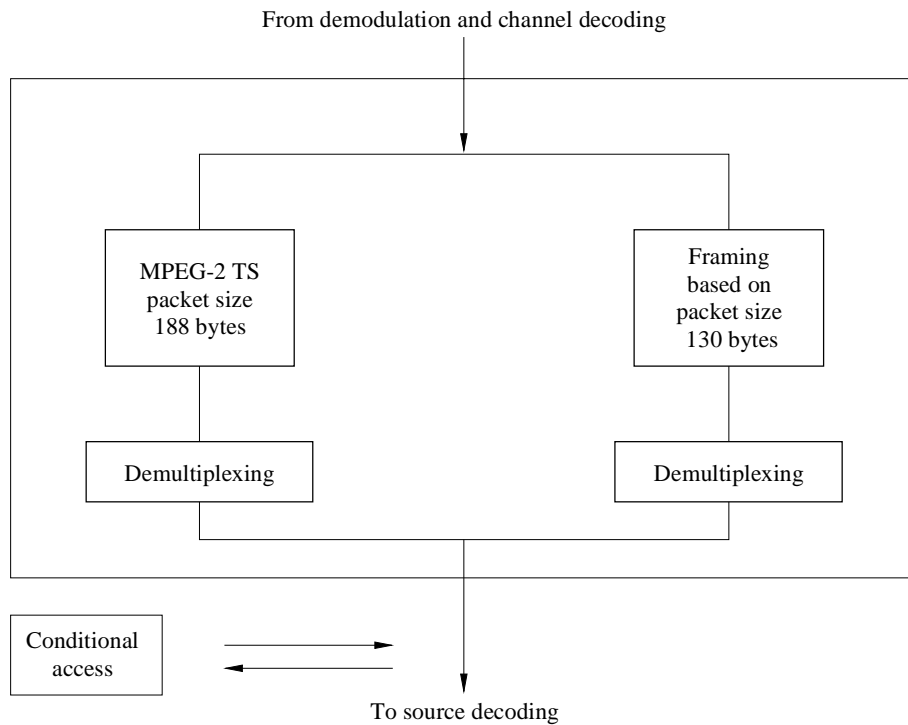
### 3.2 Transport and demultiplexing

The block diagram of the transport and demultiplexing functions for the satellite IRD is presented in Fig. 5.

The system will be capable of receiving and demultiplexing packets following MPEG-2 transport multiplexer (see ISO/IEC 13818-1) as well as transport stream specific characteristics defined in § 5.7.

Conditional access is outside the scope of this Recommendation.

FIGURE 5  
Block diagram for transport and demultiplexing



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### 3.3 Source decoding of video, audio and data

The block diagram of the source decoding of video, audio and data functions for the satellite IRD is presented in Fig. 6.

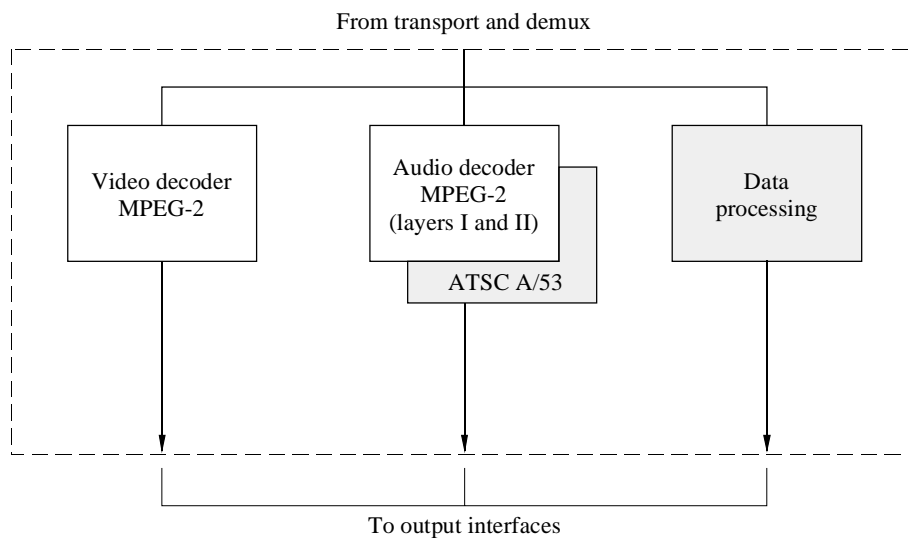
#### 3.3.1 Video

This universal element of a satellite IRD will require, as a minimum, the decoding of video formats following the main profile main level MPEG-2 signals which have been coded as specified in ISO/IEC 13818-2.

#### 3.3.2 Audio

This universal element of a satellite IRD will require the decoding of audio signals following the MPEG-2 Layers I and II (ISO/IEC 13818-3) and ATSC-A/53 Annex B (Recommendation ITU-R BS.1196, Annex 2) formats.

FIGURE 6  
Block diagram for source decoding



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### 3.3.3 Data

This block addresses the functions to process the data associated to the transport multiplex. This item is outside the scope of the Recommendation.

## 4 Summary characteristics of digital multiprogramme TV systems by satellite

The following Table 1 provides information on relevant parameters which characterize the systems in use in the world. It includes core functions as well as additional essential functions.

TABLE 1

## Summary characteristics of digital multiprogramme TV systems by satellite

<i>Demodulation and channel decoding</i>			
Function	System A	System B	System C
Randomization for energy dispersal	PRBS: $1 + x^{14} + x^{15}$	None	PRBS: $1 + x + x^3 + x^{12} + x^{16}$ truncated for a period of 4 894 bytes
Synchronous randomization	Yes	N.A.	Yes
Loading sequence into pseudo random binary sequence (PRBS) register	100101010000000	N.A.	0001 <sub>h</sub>
Place for derandomization application at the IRD	After the Reed-Solomon decoder	N.A.	Before the Reed-Solomon decoder
Reed-Solomon outer code	(204,188, $T = 8$ )	(146,130, $T = 8$ )	(204,188, $T = 8$ )
Reed-Solomon generator	(255,239, $T = 8$ )		
Reed-Solomon generator polynomial.	$(x + \alpha^0)(x + \alpha^1) \dots (x + \alpha^{15})$ where, $\alpha = 02_h$		$(x + \alpha^1)(x + \alpha^2) \dots (x + \alpha^{16})$ where, $\alpha = 02_h$
Reed-Solomon field generator polynomial.	$x^8 + x^4 + x^3 + x^2 + 1$		
Interleaving	Convolutional, $I = 12, M = 17$ (Forney)	Convolutional, $N1 = 13, N2 = 146$ (Ramsey II)	Convolutional, $I = 12, M = 19$ (Forney)
Inner coding	Convolutional		
Code constraint length	$K = 7$		
Basic code	1/2		1/3
Generator polynomial	171, 133 (octal)		117, 135, 161 (octal)
FEC	1/2, 2/3, 3/4, 5/6 and 7/8	1/2, 2/3 and 6/7	1/2, 2/3, 3/4, 3/5, 4/5, 5/6, 5/11 and 7/8
Signal modulation	QPSK		
Symbol rate	Variable	Fixed	Variable
Symbol rate range	Not specified	20 MBd	19.5 and 29.3 MBd
Occupied bandwidth (-3 dB)	Variable	20 MHz	y MHz where y = symbol rate
Allocated bandwidth (-25 dB)	Variable	25.1 MHz	1.33 y (optionally 1.55 y) (MHz) for y (symbol rate) = 19.5 and 29.3 MBd
Baseband shaping roll-off	0.35 (square root raised cosine)	0.20 (square root raised cosine)	Bandwidth limited 4th Butterworth, standard and truncated-spectrum modes approximately equivalent to $\alpha = 0.55$ and $\alpha = 0.33$ , respectively

TABLE 1 (Continued)

<i>Transport and Demultiplexing</i>			
Function	System A	System B	System C
Transport layer	MPEG-2	System B	MPEG-2
Packet size (bytes)	188	130	188
Identification ID (bit)	13 (PID)	12 (SCID)	13 (PID)
Continuity counter (bit)	4 CC		
Adaptation flag (bit)	2 (AD)	4 (Aux)	2 (AD)
Scrambling flag (bit)	2(S)		
Priority (bit)	1(P)	None	1(P)
Bundle boundary (bit)	1 (PE)		
Error indicating (bit)	1 (E)	4 byte media error field <sup>(1)</sup>	1 (E)
Payload (bytes)	184	127	184
Sync word byte	47 <sub>h</sub>	None	47 <sub>h</sub>
Sync byte inversion	From 47 <sub>h</sub> to B8 <sub>h</sub>	None	From 47 <sub>h</sub> to B8 <sub>h</sub>
Statistical multiplexing	Not restricted	Capable	Capable
Master reference clock	27 MHz		
Method of synchronization for video and audio	Time stamping		
<i>Source Decoding</i>			
Video source decoding	Syntax	MPEG-2	
	Levels	at least main level	
	Profiles	at least main profile	
Audio source decoding	MPEG-2, Layers I and II	MPEG-1, Layer II (included in MPEG-2)	ATSC A/53 or MPEG-2 Layers I and II
Data source			

TABLE 1 (Continued)

<i>Other Characteristics</i>			
Function	System A	System B	System C
Typical transponder bandwidth (MHz)	Not specified	24/27 MHz	24/27/36 MHz
Satellite downlink frequency range	Originally designed for 11/12 GHz, not excluding other satellite frequency ranges		Originally designed for 11/12 GHz and 4 GHz satellite frequency ranges
Compatibility with SMATV	Yes	Some processing is required at the headend when transmodulating to QAM	Yes
Compatibility with existing telecommunications hierarchies	The transport stream could be defined to be accommodated within the existing hierarchies		
Selectable conditional access	Yes		
Service information	ETS 300 468 (see [3], § 7)	System B	ATSC A/56 + SCTE DVS/011
EPG	ETS 300 707 (see [4], § 7)	System B	User selectable
Teletext	Supported	Not specified	
Subtitling	Supported		
Closed caption	Not specified	Yes	
Delivered TV standards	Not specified	NTSC and PAL M	NTSC and PAL
Aspect ratios	4:3 16:9 (2.12:1 Optionally)	4:3 16:9	4:3 16:9
Video resolution formats	Not restricted, Recommended: 720 × 576 704 × 576 544 × 576 480 × 576 352 × 576 352 × 288	720 × 480 704 × 480 544 × 480 480 × 480 352 × 480 352 × 240	720(704) × 576 720(704) × 480 528 × 480 528 × 576 352 × 480 352 × 576 352 × 288 352 × 240
Frame rates	Not specified	29.97	25 (PAL) 29.97 (NTSC)
Compatibility with other MPEG-2 delivery systems	ISO/IEC 13818	Some processing required	ISO/IEC 13818

(1) Sent with a redundant picture header to indicate uncorrectable error within a picture. Uses ISO MPEG defined sequence error code.

5 Specific characteristics

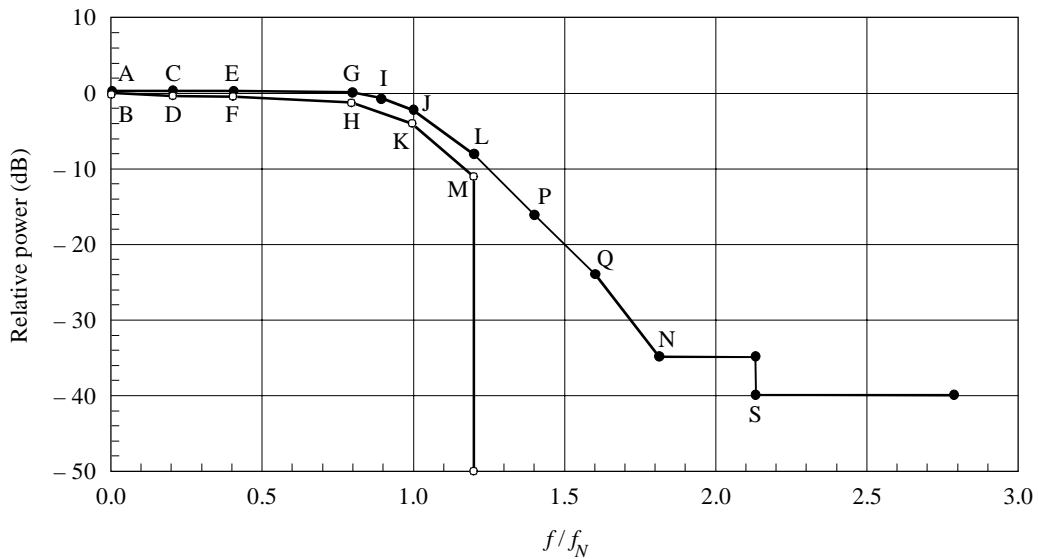
5.1 Signal spectrum of the different systems at the modulator output

5.1.1 Signal spectrum for System A

System A uses a square root raised cosine roll-off factor of 0.35.

Figure 7 gives a template for the signal spectrum at the modulator output.

FIGURE 7  
 Template for the signal spectrum mask at the modulator output represented in the baseband frequency domain



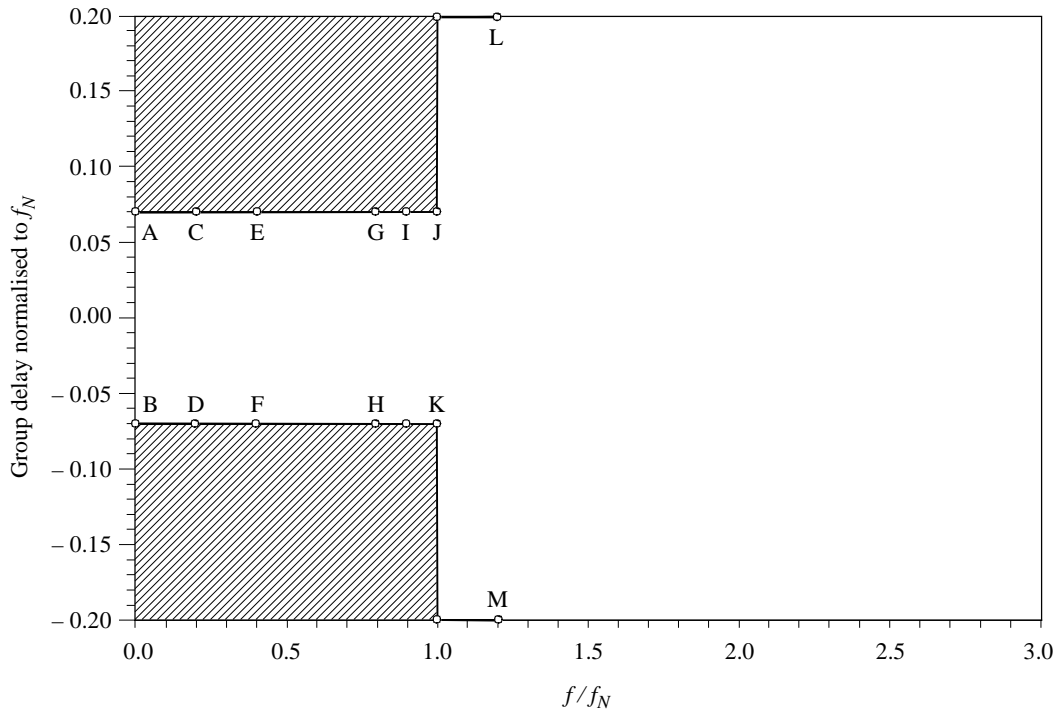
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Figure 7 also represents a possible mask for a hardware implementation of the Nyquist modulator filter. The points A to S shown in Figs. 7 and 8 are defined in Table 2. The mask for the filter frequency response is based on the assumption of ideal Dirac delta input signals, spaced by the symbol period  $T_s = 1/R_s = 1/2f_N$ , while in the case of rectangular input signals a suitable  $x/\sin x$  correction shall be applied on the filter response.

Figure 8 gives a mask for the group delay for the hardware implementation of the Nyquist modulator filter.

FIGURE 8

Template of the modulator filter group delay



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TABLE 2

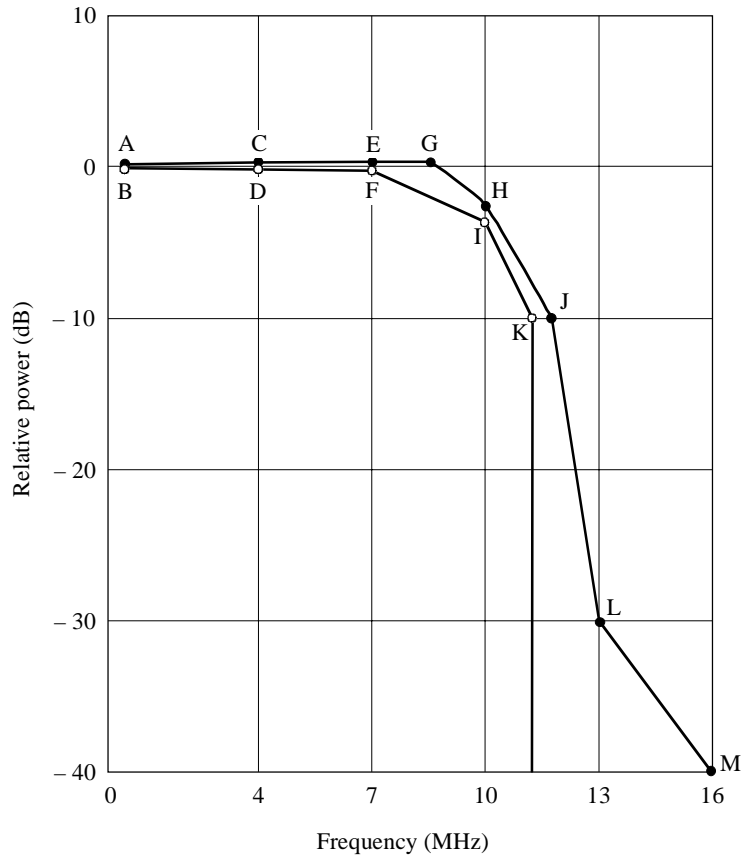
Coordinates of points given in Figs. 7 and 8

Point	Frequency	Relative power (dB)	Group delay
A	$0.0 f_N$	+0.25	$+0.07/f_N$
B	$0.0 f_N$	-0.25	$-0.07/f_N$
C	$0.2 f_N$	+0.25	$+0.07/f_N$
D	$0.2 f_N$	-0.40	$-0.07/f_N$
E	$0.4 f_N$	+0.25	$+0.07/f_N$
F	$0.4 f_N$	-0.40	$-0.07/f_N$
G	$0.8 f_N$	+0.15	$+0.07/f_N$
H	$0.8 f_N$	-1.10	$-0.07/f_N$
I	$0.9 f_N$	-0.50	$+0.07/f_N$
J	$1.0 f_N$	-2.00	$+0.07/f_N$
K	$1.0 f_N$	-4.00	$-0.07/f_N$
L	$1.2 f_N$	-8.00	-
M	$1.2 f_N$	-11.00	-
N	$1.8 f_N$	-35.00	-
P	$1.4 f_N$	-16.00	-
Q	$1.6 f_N$	-24.00	-
S	$2.12 f_N$	-40.00	-

5.1.2 Signal spectrum for System B

System B uses a square root raised cosine roll-off factor of 0.2.

FIGURE 9  
Signal spectrum for System B



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TABLE 3  
Coordinates of points

Point	Relative power (dB)	Frequency (MHz)
A	0.2	0.05
B	-0.2	0.05
C	0.25	3.5
D	-0.25	3.5
E	0.3	7
F	-0.3	7
G	0.3	8.5
H	-2.5	10
I	-3.5	10
J	-10	11.75
K	-10	11.25
L	-30	13
M	-40	16



**5.1.3 Signal spectrum for System C**

This section defines System C design recommendations for baseband signal shaping and the modulator output spectrum.

**5.1.3.1 Baseband signal shaping**

System C uses bandlimited 4th-order Butterworth filtering in standard or truncated-spectrum mode, depending on the system requirements.

**5.1.3.1.1 Amplitude response**

Figures 10a and 10b show the recommended standard and truncated-spectrum mode design goals for baseband signal shaping spectral density as normalized to the transmit symbol rate. Table 4a and Table 4b tabulate the corresponding breakpoints for standard and truncated-spectrum modes, respectively.

FIGURE 10a  
Spectral density mask for standard mode

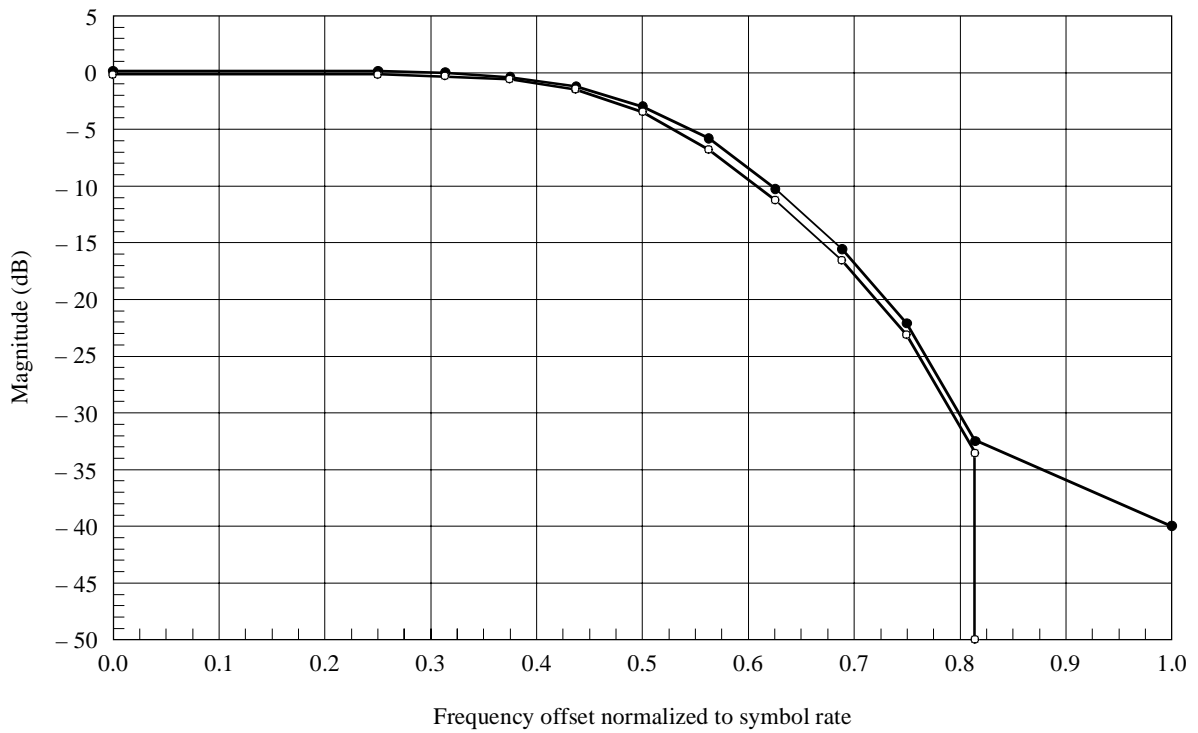


TABLE 4a

Spectral density mask breakpoints for standard mode

Frequency offset normalized to transmit symbol rate	Upper mask breakpoints (dB)	Lower mask breakpoints (dB)
0.0	0.1	-0.1
0.25	0.1	-0.1
0.3125	0.0	-0.2
0.375	-0.35	-0.55
0.4375	-1.25	-1.45
0.50	-3.0	-3.50
0.5625	-5.85	-6.85
0.625	-10.25	-11.25
0.6875	-15.55	-16.55
0.75	-22.05	-23.05
0.8125	-32.3	-33.3
0.8125		-50.0
1.0	-40.0	

FIGURE 10b

Spectral density mask for truncated-spectrum mode

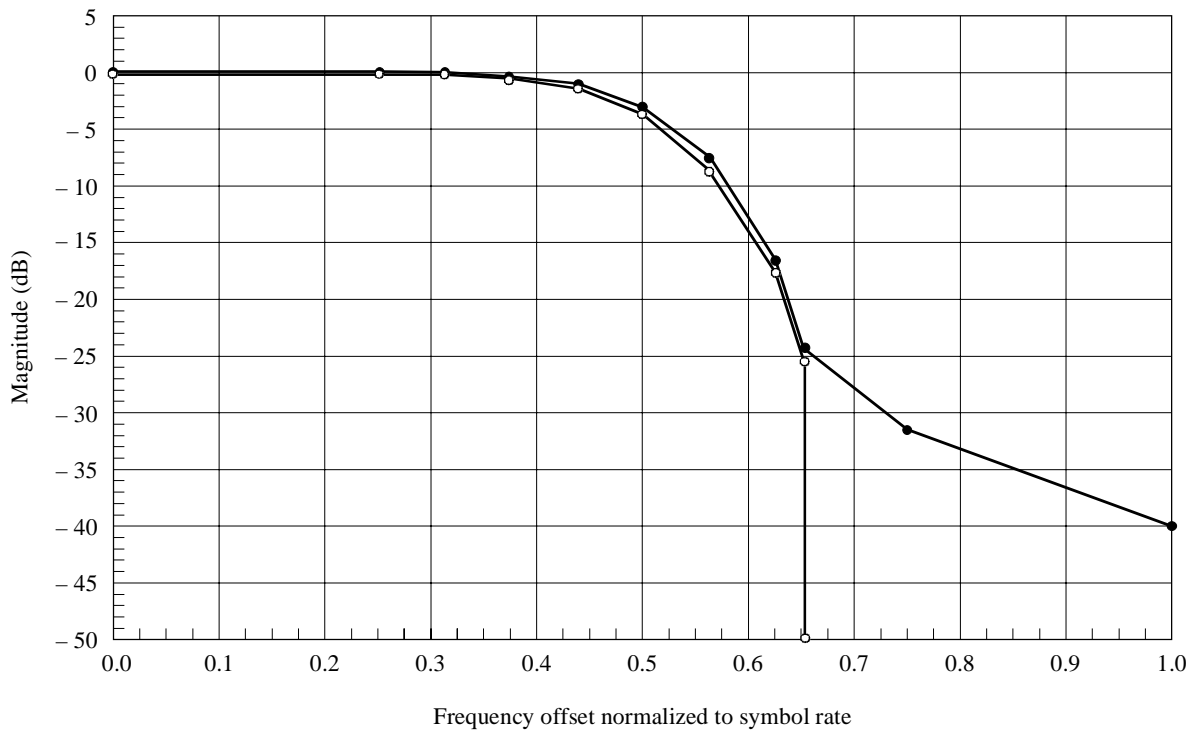


TABLE 4b

Spectral density mask breakpoints for truncated-spectrum mode

Frequency offset normalized to transmit symbol rate	Upper mask breakpoints (dB)	Lower mask breakpoints (dB)
0.0	0.1	-0.1
0.25	0.1	-0.1
0.3125	-0.15	-0.35
0.375	-0.35	-0.55
0.4375	-1.0	-1.2
0.50	-2.9	-3.4
0.5625	-7.4	-8.4
0.625	-16.6	-17.6
0.654	-24.5	-25.5
0.654		-50.0
0.75	-31.8	
1.0	-40.0	

5.1.3.1.2 Group delay response

Figures 11a and 11b show the recommended standard and truncated-spectrum mode design goals for baseband signal shaping group delay as normalized to the transmit symbol rate. Table 5a and Table 5b tabulate the corresponding breakpoints for standard and truncated-spectrum modes, respectively. The actual required group delay can be obtained by dividing the table values by the symbol rate in Hz; for example, for 29.27 Msymbol/s operation the standard mode upper mask point at a frequency offset of  $0.3 \times 29.27 \text{ MHz} = 8.78 \text{ MHz}$  is found from Table 5a to be  $(0.20/29.27 \times 10^6 \text{ Hz}) = 6.8 \times 10^{-9} \text{ s} = 6.8 \text{ ns}$ .

FIGURE 11a

Normalized group delay mask for standard mode

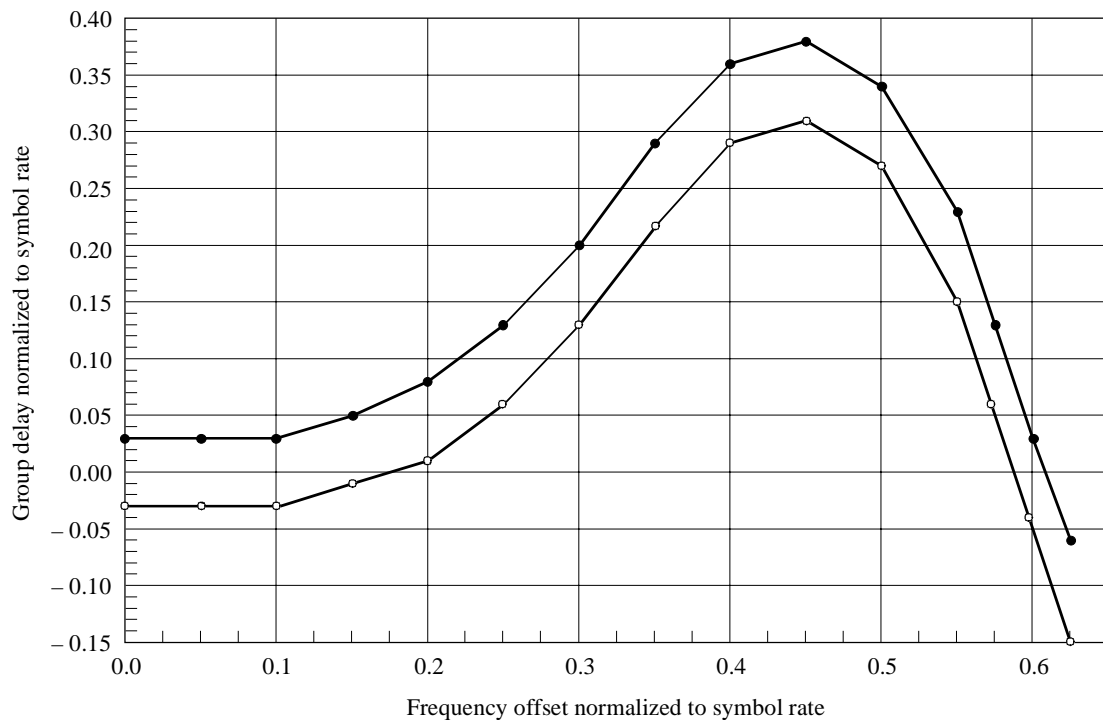


TABLE 5a

Normalized group delay breakpoints for standard mode

Frequency offset normalized to symbol rate ( $f_{sym}$ )	Upper mask group delay normalized to symbol rate (delay $\times$ ( $f_{sym}$ (Hz)))	Lower mask group delay normalized to symbol rate (delay $\times$ ( $f_{sym}$ (Hz)))
0.0	0.03	-0.03
0.05	0.03	-0.03
0.10	0.03	-0.03
0.15	0.05	-0.01
0.20	0.08	0.01
0.25	0.13	0.06
0.30	0.20	0.13
0.35	0.29	0.22
0.40	0.36	0.29
0.45	0.38	0.31
0.50	0.34	0.27
0.55	0.23	0.15
0.575	0.13	0.06
0.60	0.03	-0.04
0.625	-0.06	-0.15

FIGURE 11b

Normalized group delay mask for truncated-spectrum mode

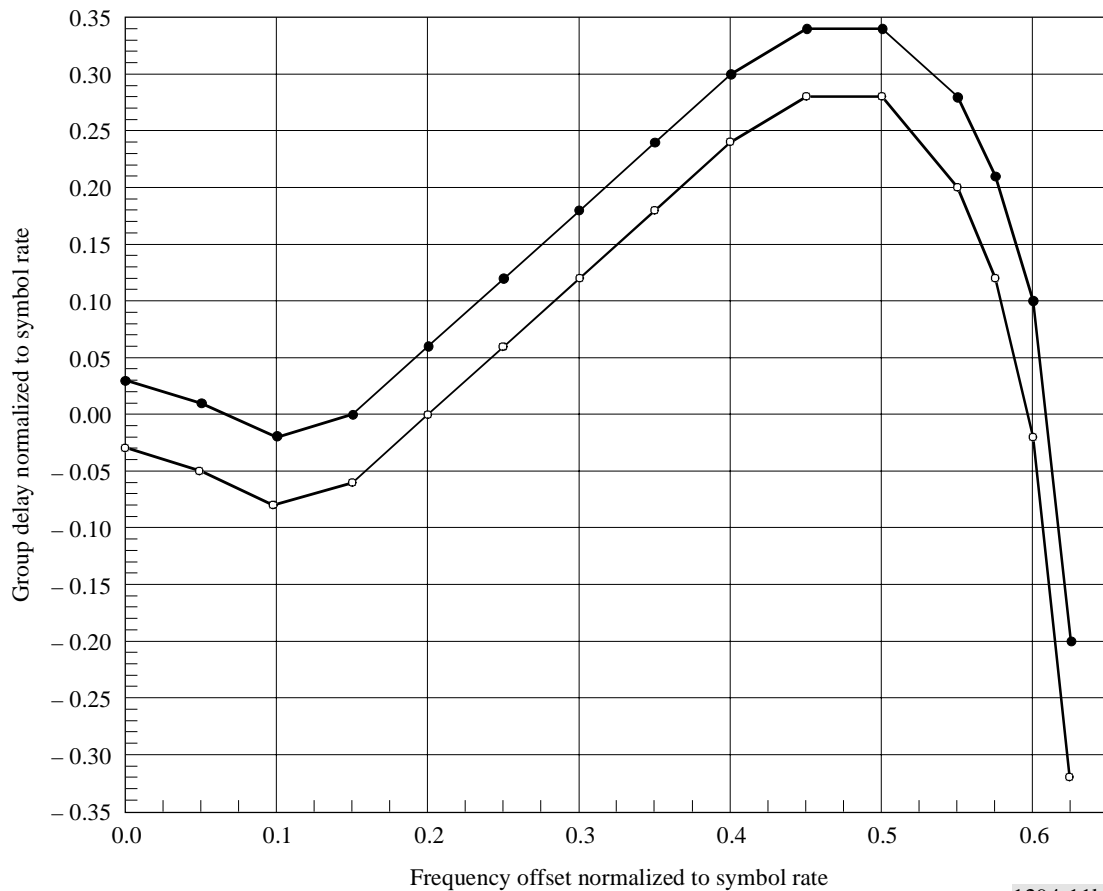


TABLE 5b

Normalized group delay breakpoints for truncated-spectrum mode

Frequency offset normalized to symbol rate ( $f_{sym}$ )	Upper mask group delay normalized to symbol rate (delay $\times (f_{sym} \text{ (Hz)})$ )	Lower mask group delay normalized to symbol rate (delay $\times (f_{sym} \text{ (Hz)})$ )
0.0	0.03	-0.03
0.05	0.01	-0.05
0.10	-0.02	-0.08
0.15	0.00	-0.06
0.20	0.06	0.0
0.25	0.12	0.06
0.30	0.18	0.12
0.35	0.24	0.18
0.40	0.30	0.24
0.45	0.34	0.28
0.50	0.34	0.28
0.55	0.28	0.20
0.575	0.21	0.12
0.60	0.10	-0.02
0.625	-0.20	-0.32

5.1.3.2 Modulator response

The recommended modulator output spectral response for System C is shown in Fig. 11c and tabulated in Table 5c.

FIGURE 11c  
System C spectral mask

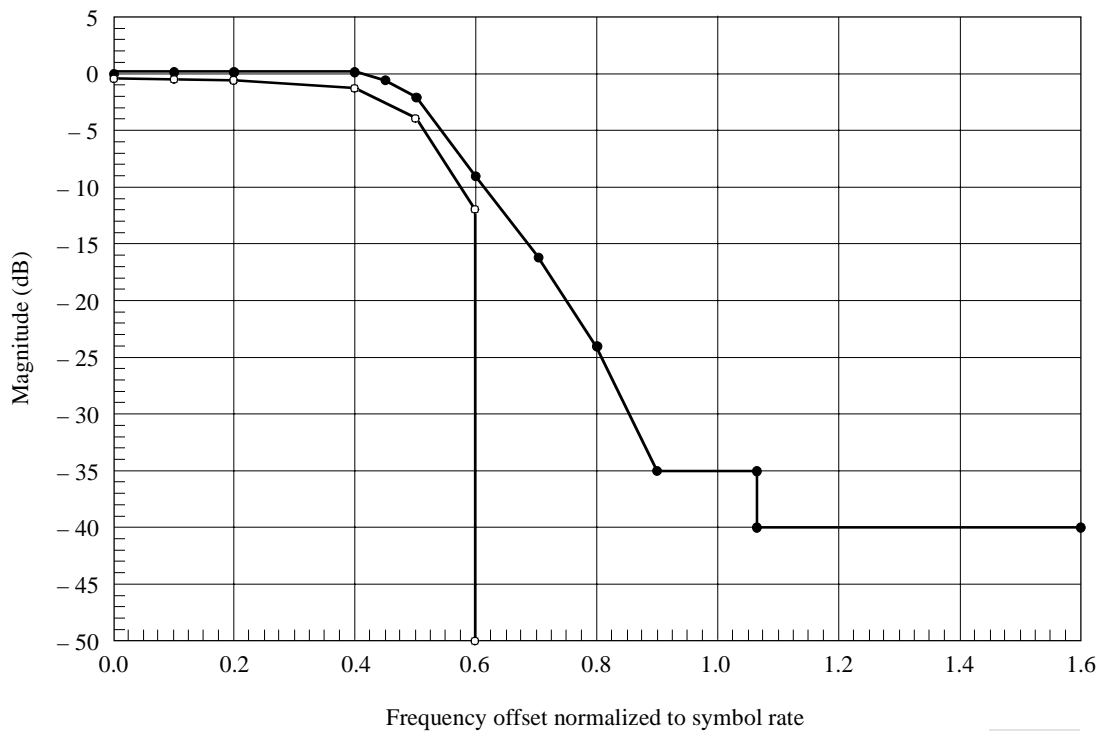


TABLE 5c

## System C spectral mask

Frequency offset normalized to transmit symbol rate	Upper mask breakpoints (dB)	Lower mask breakpoints (dB)
0.0	0.25	-0.25
0.1		-0.4
0.2		-0.4
0.4	0.25	-1.0
0.45	-0.5	
0.5	-2.0	-4.0
0.6	-9.0	-12.0
0.6		-50.0
0.7	-16.0	
0.8	-24.0	
0.9	-35.0	
1.06	-35.0	
1.06	-40.0	
1.6	-40.0	

## 5.2 Convolutional coding

## 5.2.1 Convolutional coding characteristics for System A

Table 6a defines the punctured code definition for System A based on basic code 1/2:

TABLE 6a

## Convolutional coding characteristics for system A

Original code			Code rates									
			1/2		2/3		3/4		5/6		7/8	
$K$	$G_1(X)$	$G_2(Y)$	$P$	$d_{free}$	$P$	$d_{free}$	$P$	$d_{free}$	$P$	$d_{free}$	$P$	$d_{free}$
7	$171_o$	$133_o$	$X = 1$ $Y = 1$  $I = X_1$ $Q = Y_1$	10	$X = 10$ $Y = 11$  $I = X_1 Y_2 Y_3$ $Q = Y_1 X_3 Y_4$	6	$X = 101$ $Y = 110$  $I = X_1 Y_2$ $Q = Y_1 X_3$	5	$X = 10101$ $Y = 11010$  $I = X_1 Y_2 Y_4$ $Q = Y_1 X_3 X_5$	4	$X = 1000101$ $Y = 1111010$  $I = X_1 Y_2 Y_4 Y_6$ $Q = Y_1 Y_3 X_5 X_7$	3

1: transmitted bit

0: non transmitted bit

$P$ : puncture.

## 5.2.2 Convolutional coding characteristics for System B

Table 6b defines the punctured code definition for System B.

TABLE 6b

Convolutional coding characteristics for System B

Original code			Code rates					
			1/2		2/3		6/7	
<i>K</i>	$G_1(X)$	$G_2(Y)$	<i>P</i>	$d_{free}$	<i>P</i>	$d_{free}$	<i>P</i>	$d_{free}$
7	171 <sub>o</sub>	133 <sub>o</sub>	$X = 1$ $Y = 1$ $I = X_1$ $Q = Y_1$	10	$X = 10$ $Y = 11$ $I = X_1 Y_2 Y_3$ $Q = Y_1 X_3 Y_4$	6	$X = 100101$ $Y = 111010$ $I = X_1 Y_2 X_4 X_6$ $Q = Y_1 Y_3 Y_5 Y_7$	To be determined

*P*: puncture.

5.2.3 Convolutional coding characteristics for System C

The punctured code definition for system C based on basic code 1/3 is as follows:

The following convolutional coding characteristics are included in the coding layer:

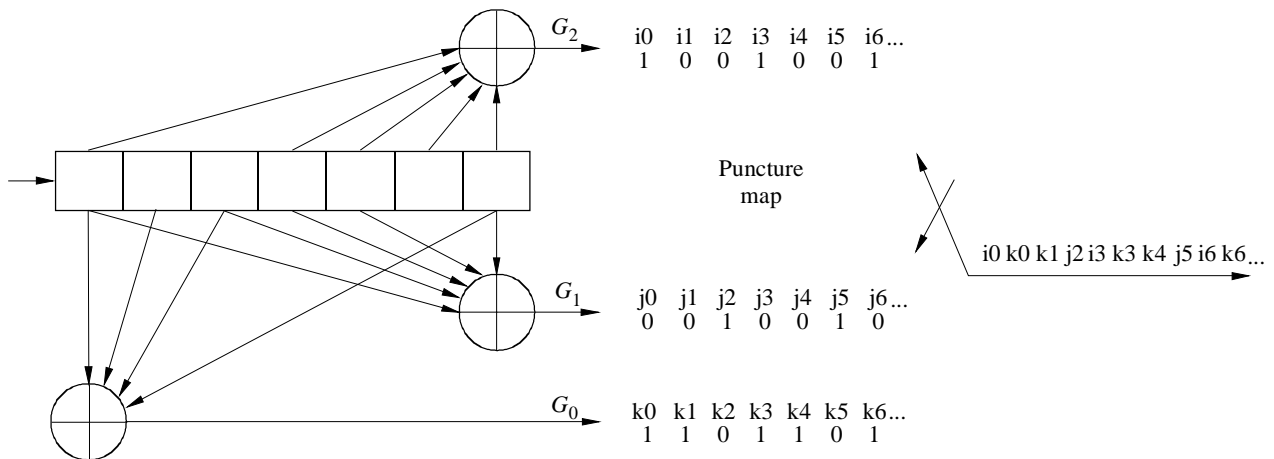
- Transmission of bit-by-bit interleaved *I* and *Q* multiplex channels is supported by the convolutional encoder.
- The IRD performs convolutional code node and puncture synchronization.
- The convolutional code is punctured from a constraint length 7, rate 1/3 code. The code generators for the rate 1/3 code are  $G(2) = 1001111$  binary (117 octal),  $G(1) = 1011101$  binary (135 octal), and  $G(0) = 1110001$  binary (161 octal). The code generators are defined from the least delayed to the most delayed input bit (see Fig. 12).
- The puncture matrices are as follows:
  - The rate 3/4 puncture matrix is  $p_2 = [100]$ ,  $p_1 = [001]$ ,  $p_0 = [110]$  (binary). For output 1, every second and third bit in a sequence of three is deleted, for output 2, every first and second bit is deleted and for output 3 every third output bit is deleted.
  - The rate 1/2 puncture matrix is  $[0]$ ,  $[1]$ ,  $[1]$  (binary).
  - The rate 5/11 puncture matrix is  $[00111]$ ,  $[11010]$ ,  $[11111]$  (binary).
  - The rate 2/3 puncture matrix is  $[11]$ ,  $[00]$ ,  $[01]$  (binary).
  - The rate 4/5 puncture matrix is  $[0111]$ ,  $[0010]$ ,  $[1000]$  (binary).
  - The rate 7/8 puncture matrix is  $[0000000]$ ,  $[0000001]$ ,  $[1111111]$  (binary).
  - The rate 3/5 puncture matrix is  $[001]$ ,  $[010]$ ,  $[111]$  (binary).
  - The rate 5/6 puncture matrix is  $[00111]$ ,  $[00000]$ ,  $[11001]$  (binary).
- The output ordering from the convolutional encoder is punctured  $G_2$  output, followed by punctured  $G_1$  output, followed by punctured  $G_0$ .
- The first bit of the puncture sequence out of the encoder is applied to the *I* channel of the QPSK signal in a combined MUX mode of operation; e.g., in the following diagram (Fig. 12),  $i_0, k_1, i_3, k_4, \dots$  are applied to the *I* channel while  $k_0, j_2, k_3, j_5, \dots$  are applied to the *Q* channel.

5.3 Synchronization characteristics

5.3.1 Synchronization characteristics for System A

The system input stream shall be organized in fixed length packets, following the MPEG-2 transport multiplexer (see ISO/IEC DIS 13818-1 see [1] § 6). The total packet length of the MPEG-2 transport multiplex (MUX) packet is 188 bytes. This includes 1 sync-word byte (i.e. 47<sub>h</sub>). The processing in order at the transmitting side shall always start from the MSB (i.e. "0") of the sync word-byte (i.e. 01000111).

FIGURE 12  
Convolutional encoder (rate 3/4 example)



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### 5.3.2 Synchronization characteristics for System B

A single synchronization byte is added to each encoded block (146 bytes). The synchronization byte is added after interleaving is performed. The synchronization byte is the binary value 00011101 and is appended to the beginning of each encoded block.

### 5.3.3 Synchronization characteristics for System C

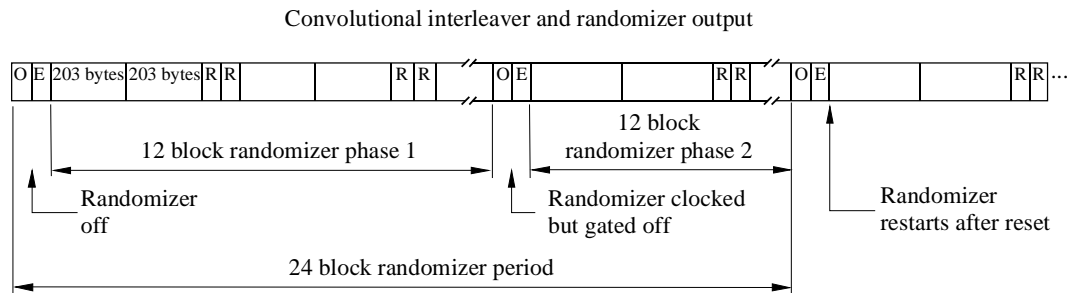
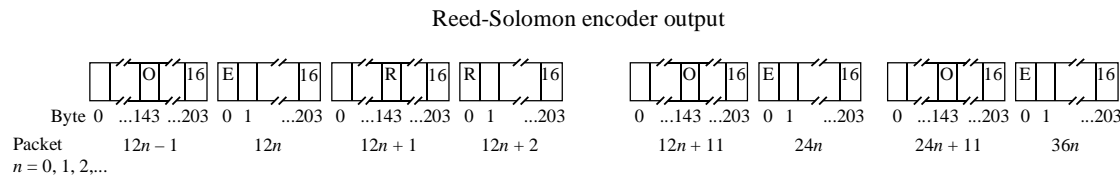
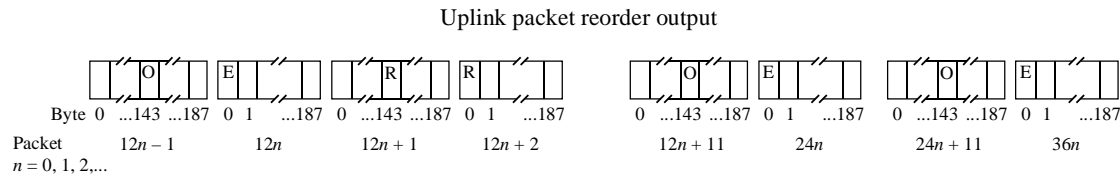
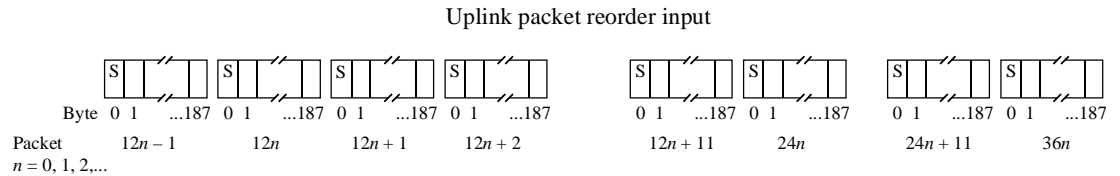
The uplink transmission processing facilitates downlink synchronization of the FEC code system by performing MPEG-2 packet reordering and 16 bit frame sync and reserved word formatting. Figure 13 shows the uplink processing required to ensure that the 16 bit frame sync pattern appears at the Viterbi decoder output in consecutive byte locations every 12 Reed-Solomon block intervals.

The following functions are performed by the encoder for synchronization purposes:

- The uplink packet reorder input is a stream of 188 byte MPEG-2 transport packets here byte numbered 0 to 187. The MPEG-2 transport packets can be numbered  $n = 0, 1, 2,$
- For transport packets numbered 0 modulo-12, the MPEG-2 sync byte number 0 is replaced by the even frame sync byte 00110110 numbered from left-to-right as MSB to LSB. The MSB is transmitted first on the channel. If the current MPEG transport stream is a  $Q$ -channel MUX in a split MUX mode, the even sync byte is 10100100.
- For transport packets numbered 11 modulo-12, the MPEG-2 sync byte number 0 is discarded, byte numbers 1 through 143 are shifted, the odd frame sync byte 01011010 (MSB to LSB, MSB first on the channel) is inserted following MPEG-2 byte 143 (for the  $Q$ -Channel MUX in a split MUX mode, the odd sync byte is 01111110), and MPEG-2 bytes 144 through 187 are appended to complete the packet structure. Figure 14 shows this odd numbered packet processing.
- For even numbered transport packets not equal 0 modulo-12, the MPEG-2 sync byte number 0 is replaced by a reserved byte.
- For odd numbered transport packets not equal 11 modulo-12, the MPEG-2 sync byte number 0 is discarded, byte numbers 1 through 143 are shifted, the reserved byte is inserted following MPEG-2 byte 143 and MPEG bytes 144 through 187 are appended to complete the packet structure.
- The randomizer is initialized at transport packets numbered 0 modulo-24; the randomizer is gated off during the 16 bit occurrences of odd and even sync bytes at the convolutional interleaver output every 12 Reed-Solomon block times.
- For split MUX operation the  $Q$  stream data is delayed one symbol time relative to the  $I$  stream data when applied to the QPSK modulator. This allows for rapid reacquisition during downlink fades or cycle slips.



FIGURE 13  
Uplink processing



S: MPEG sync byte = 01000111

O: odd frame sync byte = 01011010 (*I*-channel in split MUX) = 01111110 (*Q*-channel in split MUX)

E: even frame sync byte = 00110110 (*I*-channel in split MUX) = 10100100 (*Q*-channel in split MUX)

R: reserved byte.

(All bytes transmitted MSB first, *I*-channel frame sync used in combined MUX.)

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This uplink processing produces a 16 bit sync word at the interleaver output every 12 Reed-Solomon block intervals. The corresponding sync word for *I*-channel MUX or combined MUX modes of operation is:

*I*-channel or combined MUX sync: 0101, 1010, 0011, 0110

MSB                      LSB

where the MSB is transmitted first on the channel.

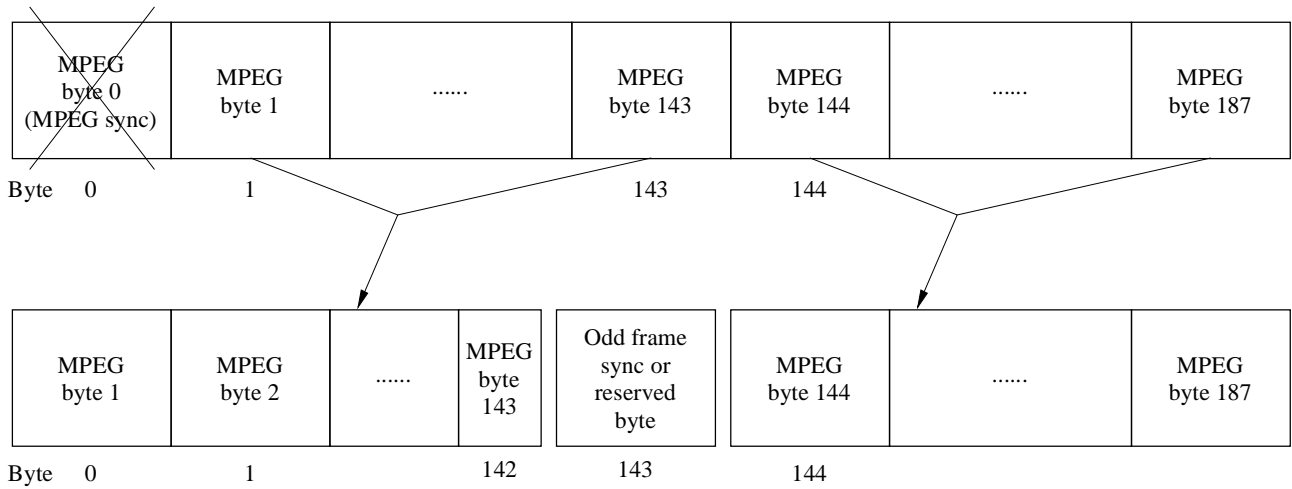
The corresponding *Q*-channel MUX sync word for split MUX modes of operation is:

*Q*-channel for split MUX sync: 0111, 1110, 1010, 0100

MSB                      LSB

A pair of reserved bytes covered by the randomizer sync sequence appears every 2 Reed-Solomon block intervals; this gives 10 reserved words per truncated randomizer period.

FIGURE 14  
Uplink packet reorder for odd numbered packets



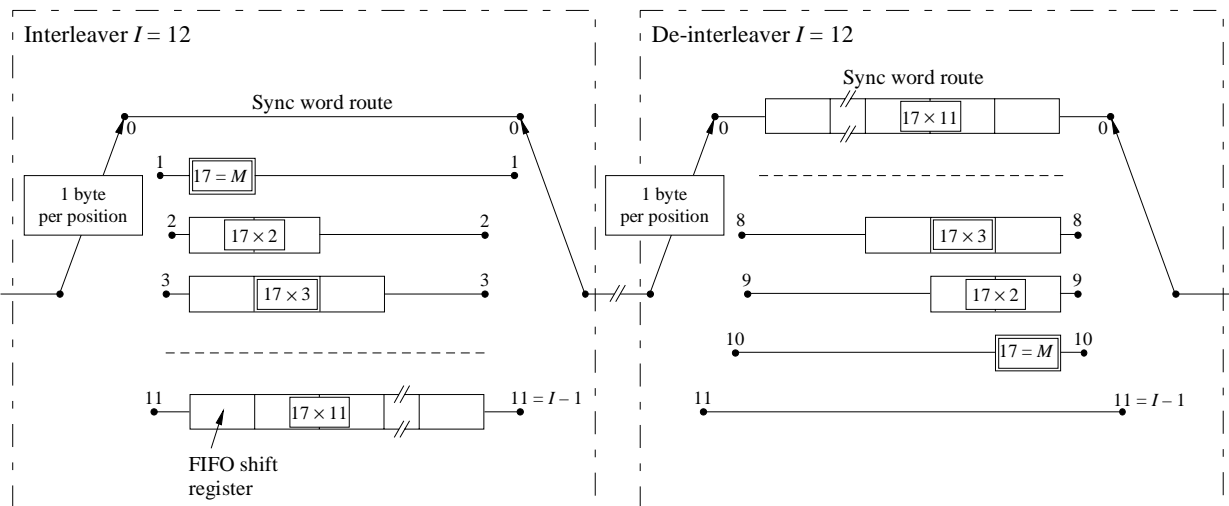
1294-14

## 5.4 Convolutional interleaver

### 5.4.1 Convolutional interleaver for System A

Following the conceptual scheme of Fig. 15a, convolutional interleaving with depth  $I = 12$  shall be applied to the error protected packets (see Fig. 20c). This results in an interleaved frame.

FIGURE 15a  
Conceptual diagram of the convolutional interleaver and de-interleaver



1294-15a

The convolutional interleaving process shall be based on the Forney approach which is compatible with the Ramsey type III approach, with  $I = 12$ . The interleaved frame shall be composed of overlapping error protected packets and shall be delimited by inverted or non-inverted MPEG-2 sync bytes (preserving the periodicity of 204 bytes).

The interleaver may be composed of  $I = 12$  branches, cyclically connected to the input byte-stream by the input switch. Each branch shall be a First-In, First-Out (FIFO) shift register, with depth  $(M - j)$  cells (where  $M = 17 = N/I$ ,  $N = 204 =$  error protected frame length,  $I = 12 =$  interleaving depth,  $j =$  branch index). The cells of the FIFO shall contain 1 byte, and the input and output switches shall be synchronized.

For synchronization purposes, the sync bytes and the inverted sync bytes shall be always routed in the branch “0” of the interleaver (corresponding to a null delay).

NOTE 1 – The de-interleaver is similar, in principle, to the interleaver, but the branch indexes are reversed (i.e.,  $j = 0$  corresponds to the largest delay). The de-interleaver synchronization can be carried out by routing the first recognized sync byte in the “0” branch.

**5.4.2 Convolutional interleaver for System B**

System B uses a convolutional interleaver defined by the block diagram in Fig. 15b. This interleaver is a Ramsey Type II interleaver (see Note 1) with the following parameters:

$I = 146$  interleaver block length and

$D = 13$  interleaving depth.

NOTE 1 – RAMSEY J. [May 1970] Realization of optimum interleavers. *IEEE Trans. Inform. Theory*, Vol. IT-16, 338-345.

The convolutional interleaving introduces an absolute read to write delay which increments linearly with the byte index within a block of  $I$  bytes:

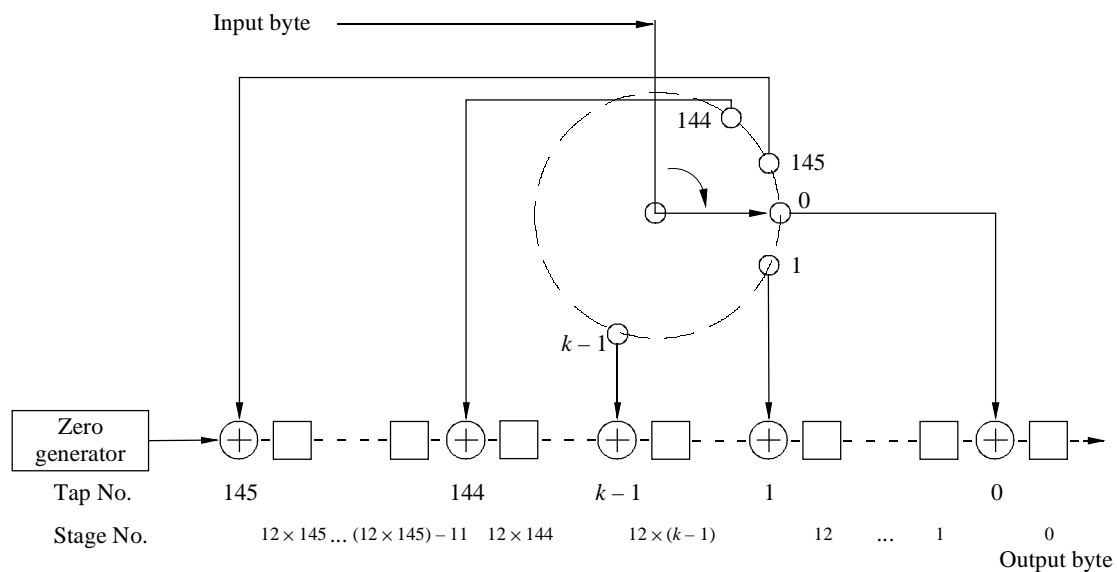
$$\text{Read/write delay (bytes)} = (D - 1)k \quad \text{with } k = 0, \dots, I - 1.$$

The interleaver does not add overhead data to the data stream. It consists of a commutator and a tapped shift register. The interleaver starts at commutator position 0 at the beginning of each data packet and functions according to the following steps.

For each input byte:

- Step 1:* add the input byte at the tap at the current location of the commutator (0 is present at the tap when not selected by the commutator),
- Step 2:* shift the shift register to the right one byte,
- Step 3:* move the commutator to the next commutator position,
- Step 4:* sample the output byte at shift register location 0.

FIGURE 15b  
Block diagram of the convolutional interleaver of System B

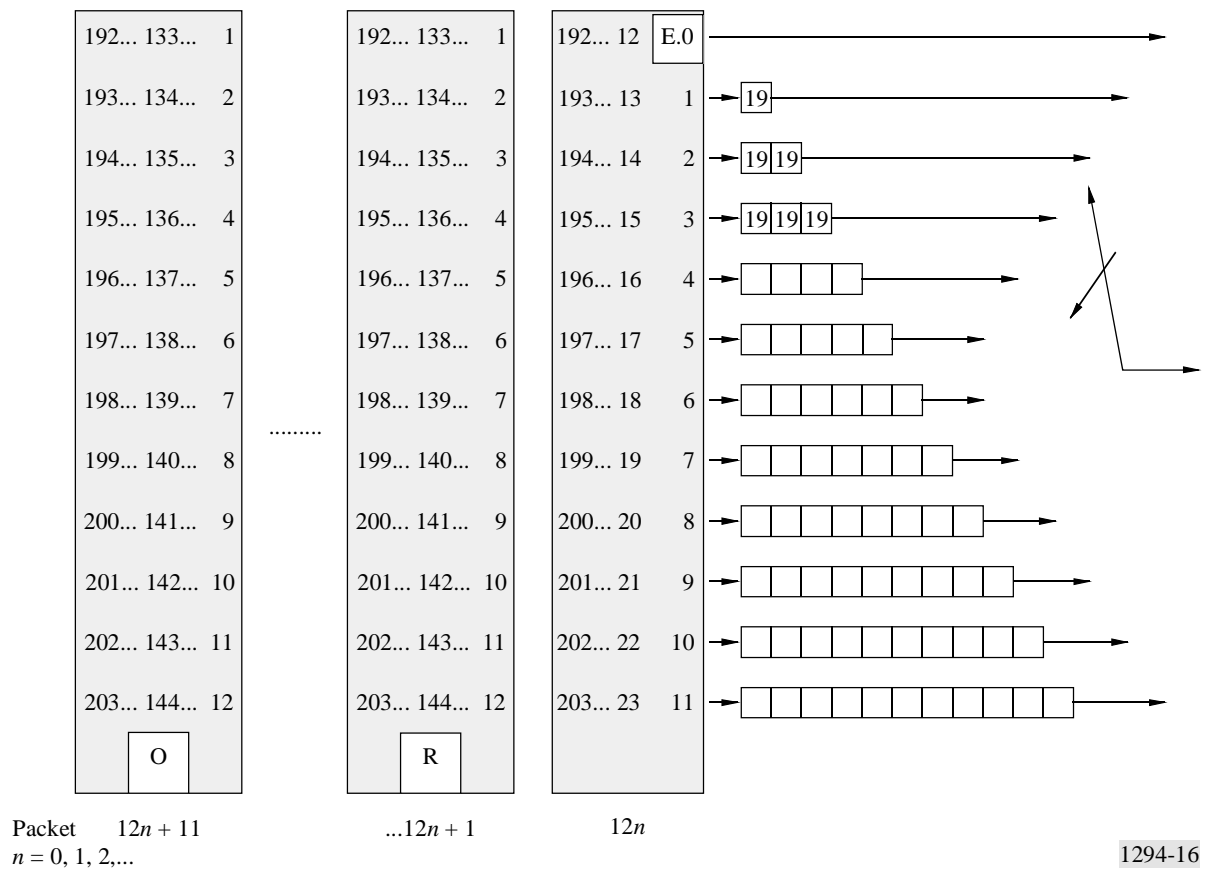


5.4.3 Convolutional interleaver for System C

The coding layer provides convolutional interleaving of 8-bit Reed-Solomon encoder output symbols. The following characteristics define the convolutional interleaving:

- The depth  $I = 12$ ,  $J = 19$  interleaver consists of an  $I(I - 1)J/2 = 1254$  Reed-Solomon symbol memory. The interleaver structure will be compatible with the commutator type as shown in Fig. 16.
- The first byte of a Reed-Solomon encoded output block is input and output on the zero-delay interleaver commutator arm.
- The  $k^{th}$  commutator arm consists of  $k \cdot J$  byte delays for  $k = 0, 1, \dots, 11$  and  $J = 19$ . An output byte is read from the  $k^{th}$  FIFO or circular buffer, an input byte is written or shifted into the  $k^{th}$  buffer, and the commutator arm advances to the  $k + 1$  interleaver arm. After reading and writing from the last commutator arm, the commutator advances to the zero-delay arm for its next output.

FIGURE 16  
Convolutional interleaver



5.5 Reed-Solomon encoder

The Reed-Solomon decoder will be capable of working with the following shortened parameters:

- (204,188,  $T = 8$ )
- (146,130,  $T = 8$ ).

The shortened Reed-Solomon codes may be implemented by adding bytes (51 for (204,188), and 109 for (146,130)), all set to zero, before the information bytes at the input of a (255,239) encoder. After the Reed-Solomon coding procedure these null bytes shall be discarded.

**5.5.1 Reed-Solomon encoder characteristics for System A**

System A uses: (204,188,  $T = 8$ )

**5.5.2 Reed-Solomon encoder characteristics for System B**

System B uses: (146,130,  $T = 8$ )

**5.5.3 Reed-Solomon encoder characteristics for System C**

System C uses: (204,188,  $T = 8$ )

The Reed-Solomon code is a (204,188,  $T = 8$ ) code with 8-bit symbols, shortened from a block length of 256 symbols, and correcting up to  $t = 8$  symbols per block.

The finite field GF(256) is constructed from the primitive polynomial  $p(x) = x^8 + x^4 + x^3 + x^2 + 1$ .

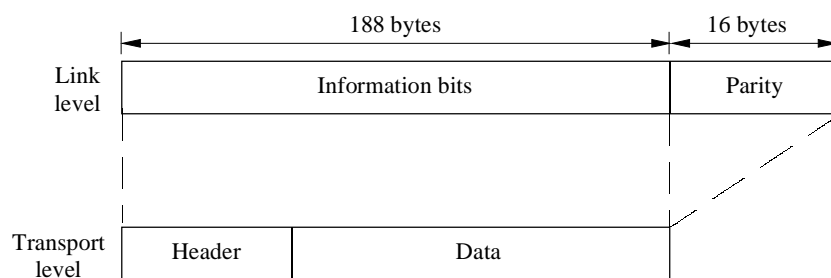
The generator polynomial for the  $t$ -error correcting code has roots at  $x = a^i, i = 1, 2, \dots, 2t, g(x) = \prod_{i=1}^{i=2t} (x + a^i)$ .

For  $t = 8$  the generator polynomial is  $g(x) = x^{16} + a^{121}x^{15} + a^{106}x^{14} + a^{110}x^{13} + a^{113}x^{12} + a^{107}x^{11} + a^{167}x^{10} + a^{83}x^9 + a^{11}x^8 + a^{100}x^7 + a^{201}x^6 + a^{158}x^5 + a^{181}x^4 + a^{195}x^3 + a^{208}x^2 + a^{240}x + a^{136}$ .

For an  $(N, N - 2t)$  code, an  $N$ -symbol codeword is generated by inputting the data symbols in the first  $N - 2t$  clock cycles, then running the circuit to generate the  $2t$  parity symbols. This encoder is clearly systematic, since the output is identical to the data symbol input for the first  $N - 2t$  cycles. Algebraically, the symbol sequence  $d_{N-2t-1}, d_{N-2t-2}, \dots, d_0$  input into the encoder represents the polynomial  $d(x) = d_{N-2t-1}x^{N-2t-1} + d_{N-2t-2}x^{N-2t-2} + \dots + d_1x + d_0$ . The encoder forms the codeword  $c(x) = x^{2t}d(x) + \text{rmd}[d(x) / g(x)]$ , and outputs the coefficients from the highest to lowest order.

The convention of parallel-to-serial conversion from data bits to symbols is that of a left-to-right shift register with the oldest bit forming the LSB and the most recent bit forming the MSB. The Reed-Solomon code is applied to packets as shown in Fig. 17.

FIGURE 17  
Reed-Solomon code applied to a packet



1294-17

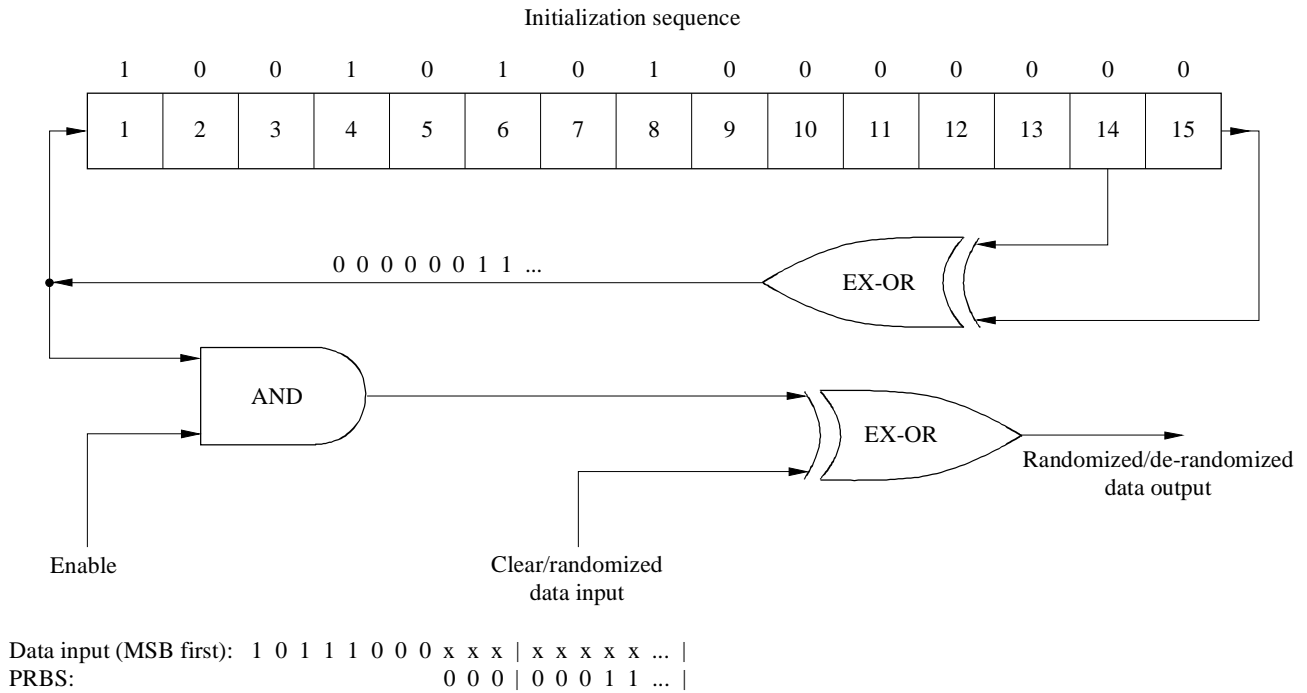
**5.6 Energy dispersal**

**5.6.1 Energy dispersal for System A**

System A removes the randomization pattern after Reed-Solomon decoding. The polynomial for the PRBS generator shall be  $1 + x^{14} + x^{15}$  with a loading sequence “10010101000000”.

In order to comply with ITU Radio Regulations and to ensure adequate binary transitions, the data of the input MPEG-2 multiplex shall be randomized in accordance with the configuration depicted in Fig. 18.

FIGURE 18  
Randomizer/de-randomizer schematic diagram



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The polynomial for the PRBS generator shall be:

$$1 + x^{14} + x^{15}$$

Loading of the sequence “10010101000000” into the PRBS registers, as indicated in Fig. 18, shall be initiated at the start of every eight transport packets. To provide an initialization signal for the descrambler, the MPEG-2 sync byte of the first transport packet in a group of eight packets is bit-wise inverted from 47<sub>h</sub> to B8<sub>h</sub>. This process is referred to as the “Transport Multiplex Adaptation”.

The first bit at the output of the PRBS generator shall be applied to the first bit (i.e. MSB) of the first byte following the inverted MPEG-2 sync byte (i.e. B8<sub>h</sub>). To aid other synchronization functions, during the MPEG-2 sync bytes of the subsequent 7 transport packets, the PRBS generation shall continue, but its output shall be disabled, leaving these bytes unrandomized. Thus, the period of the PRBS shall be 1 503 bytes.

The randomization process shall be active also when the modulator input bit-stream is non-existent, or when it is non-compliant with the MPEG-2 transport stream format (i.e. 1 sync byte + 187 packet bytes). This is to avoid the emission of an unmodulated carrier from the modulator.

### 5.6.2 Energy dispersal for System B

System B does not use randomization pattern.

### 5.6.3 Energy dispersal for System C

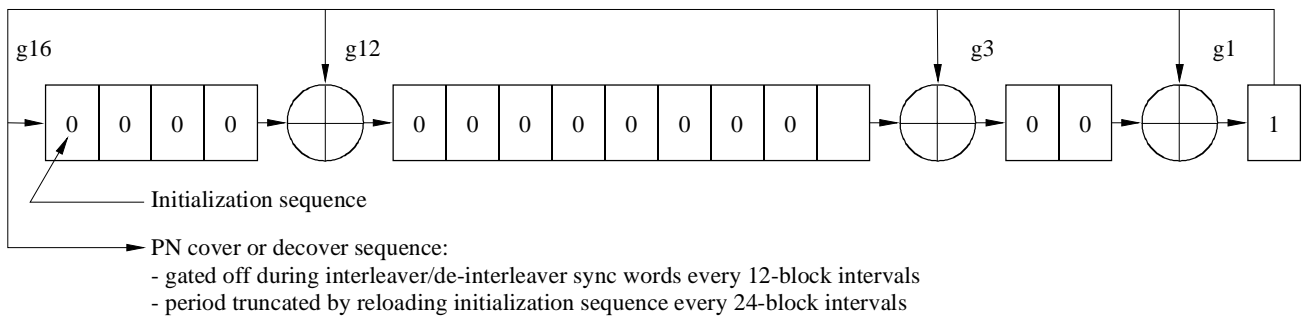
System C applies randomization functions after convolutional decoding. The polynomial for the PRBS generator shall be  $1 + x + x^3 + x^{12} + x^{16}$ , with a loading sequence “0001<sub>h</sub>”.

The coding layer uses data randomization (scrambling) at the interleaver output and de-interleaver input for energy dispersal and to ensure a high data transition density for bit timing recovery purposes. The following characteristics define the data randomization:

- The transmit data prior to convolutional coding is randomized via an EXCLUSIVE-OR operation with a truncated  $2^{16} - 1$  maximal length pseudo-random (PN) sequence that is restarted every 24 Reed-Solomon encoder block intervals, as shown in Fig. 19.

- The 16 bit FEC sync patterns occurring every 12 Reed-Solomon block intervals are not randomized. The randomizer is clocked during the 16 bit times that FEC sync patterns are inserted, but the randomizer output is not used in the EXCLUSIVE-or operation with the transmit data.
- The PN sequence is generated from a 16 stage linear feedback shift register with taps at stages 16, 12, 3, and 1 as shown in Fig. 19. The randomizer input is defined as the PN randomization sequence.
- The randomizer is initialized with the value  $0001_h$  at the first bit following the odd-byte/even-byte FEC frame sync word output from the interleaver every 24 blocks intervals.

FIGURE 19  
Randomizer block diagram



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## 5.7 Framing and transport stream characteristics

### 5.7.1 Framing and transport stream characteristics for System A

The framing organization shall be based on the input packet structure (see Fig. 20a)).

### 5.7.2 Framing and transport stream characteristics for System B

See Appendix 1.

### 5.7.3 Framing and transport stream characteristics for System C

See synchronization characteristics (§ 5.3.3).

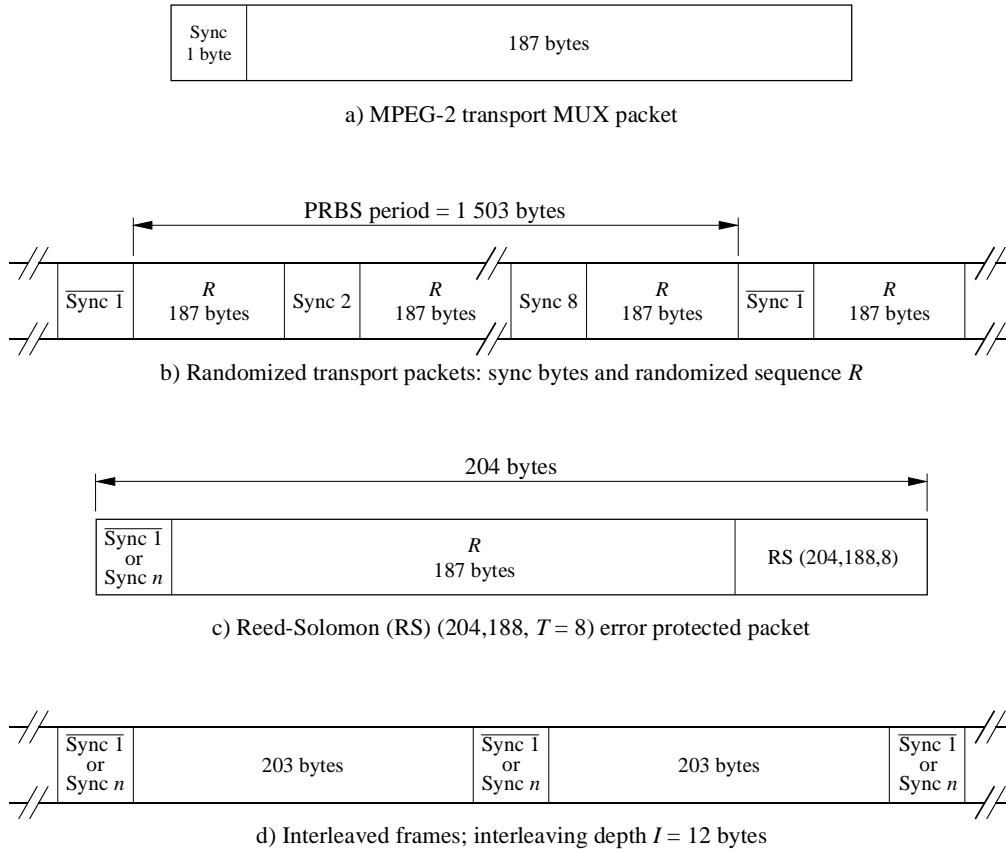
## 6 Normative references

- [1] ISO/IEC [November 1994]: Standard ISO/IEC DIS 13818. Coding of moving pictures and associated audio, Parts 1, 2 and 3.
- [2] Standard ATSC/A53, Annex B. Recommendation ITU-R BS.1196, Annex 2.

## 7 Informative references

- [3] Standard ETS 300 468. Digital broadcasting systems for television, sound and data services; Specification for Service Information (SI) in Digital Video Broadcasting (DVB) systems.
- [4] Standard ETS 300 707. Electronic Programme Guide (EPG); Protocol for a TV-guide using electronic data.

FIGURE 20  
Framing structure



Sync 1: not randomized complemented sync byte  
 Sync  $n$ : not randomized sync byte,  $n = 2, 3, \dots, 8$

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## 8 List of acronyms

- AD Auxiliary data
- ATM Asynchronous transfer mode
- ATSC Advanced television systems committee
- CA Conditional access
- ETS European telecommunication standard
- FEC Forward error correction
- IRD Integrated receiver-decoder
- MPEG Motion pictures experts group
- MPEG-2 TS MPEG-2 transport stream



PID	Program identification
PRBS	Pseudo-random binary sequence
QAM	Quadrature amplitude modulation
QEF	Quasi error-free
QPSK	Quadrature phase-shift keying
RAM	Random access memory
ROM	Read only memory
RS	Reed-Solomon
SCID	Service channel identification
SCTE	Society of cable and telecommunication engineers

APPENDIX 1

TO ANNEX 1

**System B transport stream characteristics\***

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\* The transport stream characteristics of Systems A and C are provided in Reference [1] in § 6 of Annex 1.

## 1 Introduction

This Annex defines the transport protocol of System B bit streams. It has a fixed length packet structure which provides the basis for error detection, logical resynchronization and error concealment at the receiver. The System B transport protocol consists of two distinct sub-layers: a “data-link/network” sub-layer, prefix, and a transport “adaptation” sub-layer specific to each service. The data-link/network sub-layer provides generic transport services such as scrambling control flags, asynchronous cell multiplexing, and error control. The adaptation layer is designed for efficient packing of variable length MPEG data into fixed length cells, while providing for rapid logical resynchronization and error concealment support at the decoder after uncorrectable error events.

The transport protocol format defines fixed length cells (or packets) of data where each cell includes a prefix and a transport block. The prefix consists of four bits of control information and twelve bits for service channel identification. Service multiplexing capabilities provide support of a mix of video, audio, and data services. The transport block includes auxiliary data containing timing and scrambling information, and service-specific data, e.g. for MPEG video services: redundant MPEG headers and standard MPEG data.

Provided within this protocol are mechanisms to facilitate rapid decoder recovery after detecting the loss of one or more cells on the channel. By identifying specific information and redundantly transmitting key MPEG data, the decoder can control the region of the image affected by errors.

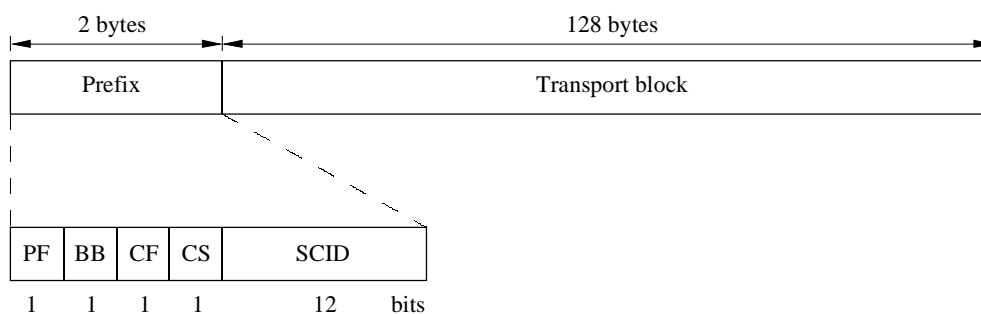
Section 2 of this Appendix describes the prefix part of the transport structure in detail. Two special purpose transport packets, the null packets and the ranging packets are described in § 3. Sections 4 and 5 describe the details of video application packets, and audio application packets, respectively. Program guide related packets are described in § 6. This Recommendation is concluded with § 7 of this Appendix with description of multiplexing constraints for transport buffer management.

Note that within this specification the term “scrambling” is used generically and means encryption when applied to digital systems.

## 2 Prefix

The System B transport packets shall consist of 130 bytes. Of these, the first two bytes shall be reserved for prefix bits. The prefix contains several link layer control flags as well as the channel identities for many different video, audio, and data services. Figure 21 illustrates the logical structure of a transport cell in which the prefix and its relationship to the transport block are identified.

FIGURE 21  
System B transport packet structure



The Semantic definition of the fields in prefix is given below in Table 7:

TABLE 7

**Prefix fields**

PF	Packet framing	This bit toggles between 0 and 1 with each packet.
BB	Bundle boundary	This bit has significance for video service only: BB bit is set to 1 in the first packet containing a redundant video sequence header, and 0 in all other packets. The decoder should ignore this bit.
CF	Control flag	CF = 1: the transport block of this packet is not scrambled CF = 0: the transport block of this packet is scrambled.
CS	Control sync	For scrambled transport packets (i.e. CF=0), this bit indicates the key to be used for descrambling. In Auxiliary packets, if the Aux packet payload contains control word packet (CWP), this bit indicates which CWP is sent (CS = 0 or CS = 1). The de-scrambling key information, derived from the CWP, is used to de-scramble the service packets with the same CS (i.e., the key obtained from Aux packet with CS = 0 is used for de-scrambling transport packets with CS = 0).
SCID	Service channel ID	This 12 bit field (unsigned integer, MSB first) uniquely identifies the application for which the information in the transport packet's transport block is intended. The following SCIDs are reserved for specific purposes: SCID = 0x000 - NULL packet SCID = 0xFFFF - Reserved (do not use!).
	Transport block	This is the application data (128 bytes) to be processed by the application addressed by the SCID.

### 3 Null and Ranging Packets

There are two special transport packets defined in the System B system: null packets and ranging packets.

The null packets and the ranging packets shall be unencrypted. (i.e., CF = 1).

The packet structure of these packets is as follows:

*For the null packets:*

PF = x (Toggles between packets)

BB = 0

CF = 1

CS = 0

SCID = 0x 000

Therefore, the first 2 bytes (prefix) of the null packets reads in hexadecimal notation; 0x 20 00, or 0x A0 00 depending on the value of the PF bit.

*For the ranging packets:*

PF = x (Toggles between packets)

BB = 0

CF = 1

CS = 0

SCID: Determined by the multiplex equipment.

The 128 bytes (transport block) of the null packets and the ranging packets are identical, and are described below in Table 8. (The content is designed to be spectrally neutral in order to maintain tuning lock.)

TABLE 8

**Null and ranging packet transport block**

Byte No.	Value	Byte No.	Value	Byte No.	Value	Byte No.	Value
1 <sup>(1)</sup>	4 <sup>(1)</sup>	33	48	65	38	97	125
2	9	34	124	66	137	98	137
3	180	35	121	67	99	99	212
4	6	36	26	68	57	100	61
5	149	37	179	69	113	101	187
6	240	38	128	70	146	102	96
7	167	39	88	71	191	103	192
8	88	40	113	72	245	104	141
9	169	41	223	73	71	105	69
10	6	42	82	74	194	106	15
11	78	43	75	75	159	107	108
12	175	44	112	76	212	108	80
13	172	45	18	77	55	109	184
14	129	46	242	78	154	110	106
15	134	47	249	79	235	111	159
16	185	48	172	80	227	112	231
17	162	49	112	81	129	113	224
18	181	50	199	82	200	114	157
19	137	51	214	83	197	115	197
20	118	52	50	84	13	116	198
21	8	53	93	85	230	117	57
22	149	54	159	86	112	118	60
23	57	55	218	87	19	119	134
24	198	56	180	88	246	120	61
25	147	57	223	89	86	121	11
26	97	58	65	90	128	122	218
27	2	59	141	91	182	123	100
28	83	60	123	92	122	124	50
29	64	61	64	93	127	125	214
30	38	62	184	94	197	126	95
31	41	63	0	95	176	127	53
32	20	64	54	96	233	128	184

<sup>(1)</sup> Note that this byte corresponds to the CC/HD byte in other packets i.e., CC = 0 HD = 0100<sub>b</sub>.

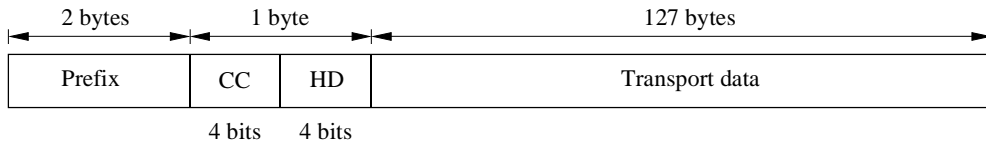
#### 4 Video application packets

The general structure of the video transport packets is illustrated in Fig. 22. Within the video application packets there are 4 types of transport cells, characterized by the type of video service related data transported through them:

- Auxiliary data packets (time stamps, encryption control word packets)
- Basic video service packets (MPEG video data)
- Redundant data packets (redundant MPEG headers, and non-redundant MPEG video data)
- Non-MPEG video data packets (non-MPEG data and non-redundant MPEG video data).

To indicate different cell types and associated counters, the video transport layer format has 4 bits for a continuity counter (CC) and 4 bits for a header designator (HD), as shown in Fig. 22. A detailed description of these fields is given below in Table 9. Note that, of the 130 byte long packet, the first 2 bytes are used for prefix, the third byte contains the CC and HD fields, and the remaining 127 bytes carry the payload.

FIGURE 22  
General video application packet structure



1294-22

TABLE 9  
The Semantic definition of fields in the CC HD byte

CC	Continuity counter	This 4 bit field (unsigned integer, MSB first) is incremented by one with each packet with the same SCID. After CC reaches its maximum value 15 (1111 <sub>b</sub> ), the CC wraps around to 0. The CC is set to 0 (0000 <sub>b</sub> ) and shall not be incremented when the HD field contains "0x 00" (i.e., Auxiliary packets). Note that from the definition of the null and ranging packets, the CC field in null and ranging packets is set to 0.  The CC allows a receiver to detect cell discontinuity (due to cell errors) for a particular transport service.
HD	Header designator	This 4 bit field indicates the 4 video application packet types as: HD 0000 <sub>b</sub> Auxiliary data packets 01x0 <sub>b</sub> Basic video service packets 10x0 <sub>b</sub> Redundant data packets 11x0 <sub>b</sub> Non-MPEG video data packets x: this bit can be 0 or 1. All other values are reserved for future use.

#### 4.1 Auxiliary data packets

Auxiliary data packets (Aux packets) are used for the transmission of auxiliary data groups (ADGs) and are identified by HD = 0000<sub>b</sub>.

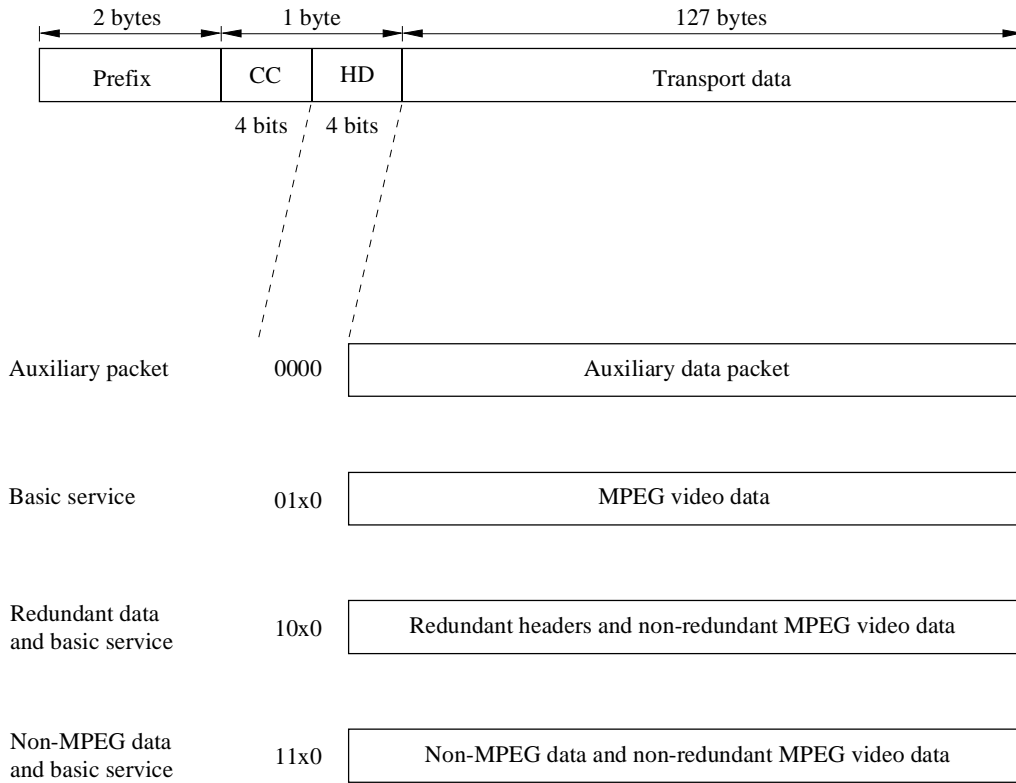
These packets are transmitted in clear (not scrambled) and the control flag (CF) bit in the prefix is set to 1 to indicate this.

The ADG may contain:

- reference time codes and stamps;
- encryption control word packets (CWPs).

An ADG consists of 2 parts: auxiliary data prefix (ADP) of 2 bytes and auxiliary data block (ADB) of variable length. An Aux packet may contain one or more data groups placed next to each other. If the 127 byte payload is not completely filled with ADG data, the remaining (unused) bytes are filled with zeros. Also the CFF bit in each ADP field indicates whether the corresponding ADB contains defined, valid data. If this bit is set to zero, the remainder of the packet starting immediately after that CFF bit, shall be ignored. This means that the AFID, AFS, and ADB of the ADG with a zero CFF bit shall be ignored. Also, no valid ADG can be transmitted in the remainder of the packet.

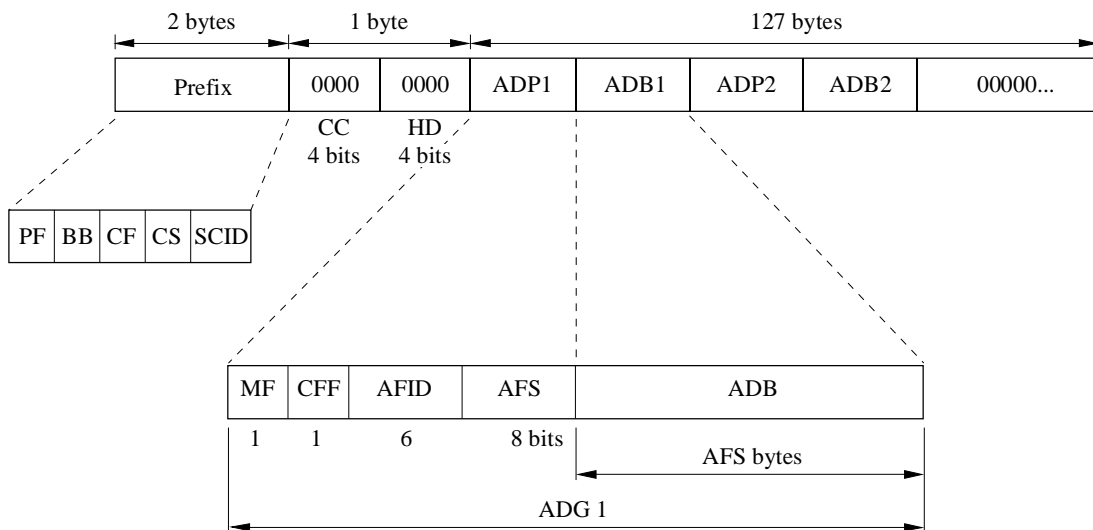
FIGURE 23  
Video application packet structures



1294-23

An example of auxiliary data packet structure with two ADG fields is illustrated in Fig. 24. The semantic definition of the (relevant) fields in the auxiliary data packet are given in Table 10.

FIGURE 24  
Auxiliary data packet structure



1294-24

TABLE 10

**The Semantic definition of the (relevant) fields in the auxiliary data packet**

BB	Bundle boundary	BB = 0 for Aux packets
CF	Control flag	CF = 1 for Aux packets (not scrambled)
CS	Control sync	If the Aux packet payload contains control word packet (CWP), this bit indicates which CWP is sent (CS = 0 or CS = 1). The scrambling key information, that is derived from the CWP, is used to de-scramble the service packets with the same CS (i.e., key obtained from Aux packet CS = 0 is used for de-scrambling transport packets with CS = 0)
CC	Continuity counter	CC = 0000 <sub>b</sub> for Aux packets
HD	Header designator	HD = 0000 <sub>b</sub> for Aux packets
MF	Modifiable flag	MF = 1: the following ADB can be modified MF = 0: the following ADB cannot be modified The decoder shall ignore this flag.
CFF	Current field flag	CFF = 1: this field contains a valid ADG CFF = 0: this field does not contain a valid ADG
AFID	Aux field ID	This 6 bit field identifies the auxiliary data information carried in this auxiliary data group. Three different auxiliary data groups are defined. AFID            Definition of ADG 000000 <sub>b</sub> Reference time stamp only 000001 <sub>b</sub> Encryption control word packet (CWP) only 000011 <sub>b</sub> Reference time stamp and CWP 000010 <sub>b</sub> , and 000100 <sub>b</sub> to 111111 <sub>b</sub> : reserved for future definition
AFS	Auxiliary field size	This one byte field (unsigned integer, MSB first) contains the length of the following auxiliary data block in bytes.
ADB	Auxiliary data block	Auxiliary data information of size AFS bytes.

There are three ADGs defined in System B, as identified by the AFID field in the auxiliary data prefix.

*Reference time stamp only*

AFID = 000000<sub>b</sub>

AFS = 5 (0x05)

ADB = byte time stamp: A byte of all 0s followed by 32 bits representing a sample from the 27 MHz system reference counter at the encoder. This sample is taken at the time the auxiliary data packet left the encoder. Please note that this is different than the reference time stamps used by MPEG. An increment of one in the System B reference time stamps equals one cycle of the 27 MHz clock. An increment of one in the MPEG reference time stamps equals 300 cycles of the 27 MHz clock, or one increment of a 90 kHz clock. This sample is taken at the time the auxiliary data packet left the encoder.

*Encryption CWP only*

AFID = 000001<sub>b</sub>

AFS = 120 (0x78)

ADB = 120 bytes of control word packet: Information required for managing encryption and conditional access.

Note that the CS bit in the prefix indicates which CWP is sent in the payload (CS = 0 or CS = 1). The de-scrambling key information, derived from the CWP, is used to de-scramble the service packets with the same CS (i.e., key obtained from Aux packet with CS = 0 is used for de-scrambling transport packets with CS = 0).

*Reference time stamp and CWP*AFID = 000011<sub>b</sub>

AFS = 125 (0x7D)

ADB = 5 byte time stamp followed by 120 bytes of CWP

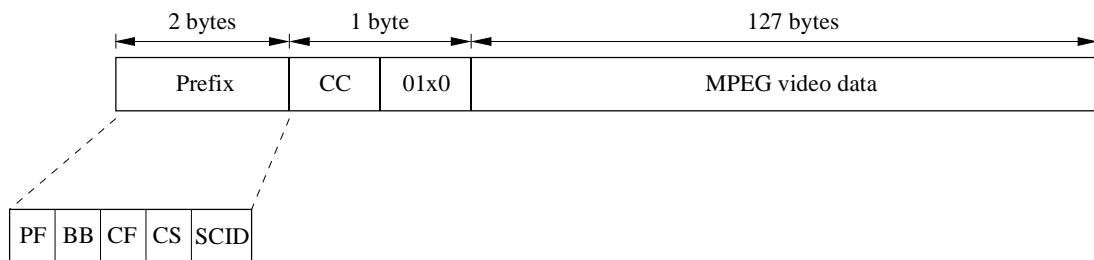
NOTE 1 – For “multi-service” programs, i.e. those containing two or more combinations of audio, and video, and data services, it is usual (but not required) that auxiliary data will occur on only one of these services. As a result, timing and/or conditional access information received in a single auxiliary data packet may apply to more than one service within the given program. This is possible because:

- the system clock reference is common for all services within a given program;
- from the CWP, the conditional access system may indicate authorization for up to three services within a given program.

**4.2 Basic video service packets**

The transport packets of a video service with HD field set to 01x0 carry basic video service (i.e., MPEG video bits) information. The structure of the basic video service packet is illustrated in Fig. 25. The semantic definition of the (relevant) fields in basic video service packet structure is given in Table 11.

FIGURE 25

**Video basic service packet structure**

1294-25

TABLE 11

**The Semantic definition of the (relevant) fields in basic video service packet structure**

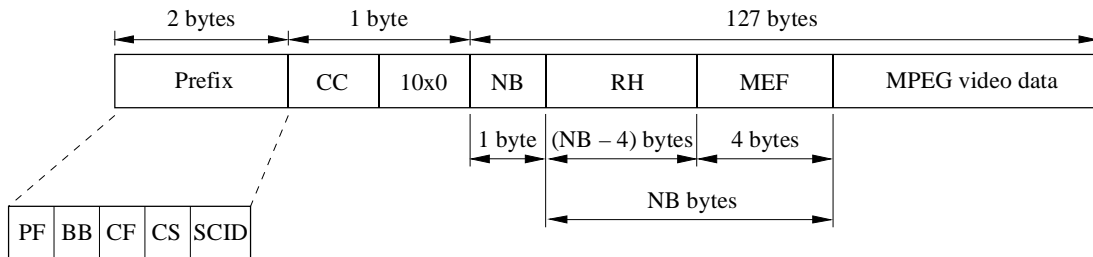
BB	Bundle boundary	BB bit is set to 1 in first basic video packet containing a redundant video sequence header, and 0 in all other packets. The decoder should ignore this bit.
CF	Control flag	CF = 1: The transport block of this packet is not scrambled CF = 0: The transport block of this packet is scrambled
CS	Control sync	For scrambled transport packets (i.e. CF=0), this bit indicates the key to be used for de-scrambling.
HD	Header designator	HD = 01x0 <sub>b</sub> for basic video service packets The HD(1) bit, indicated by x in HD = 01x0 <sub>b</sub> , toggles with each basic video service packet containing a non-redundant picture header start code. For these packets, the picture header start code is packet-aligned to be the first four bytes of the MPEG video data payload following the CC/HD fields. No other packets will toggle the HD(1) bit.
	MPEG video data	127 bytes of MPEG video data.



### 4.3 Redundant data packets

A special packet type with HD = 10x0 is defined to contain redundant GOP and picture headers. Redundant GOP and picture headers may or may not exist in a video bitstream. Therefore, redundant data packets may or may not exist. The structure of the redundant data packet is illustrated in Fig. 26. The semantic definition of the (relevant) fields in the redundant data packet is given in Table 12.

FIGURE 26  
Redundant data packet structure



1294-26

TABLE 12

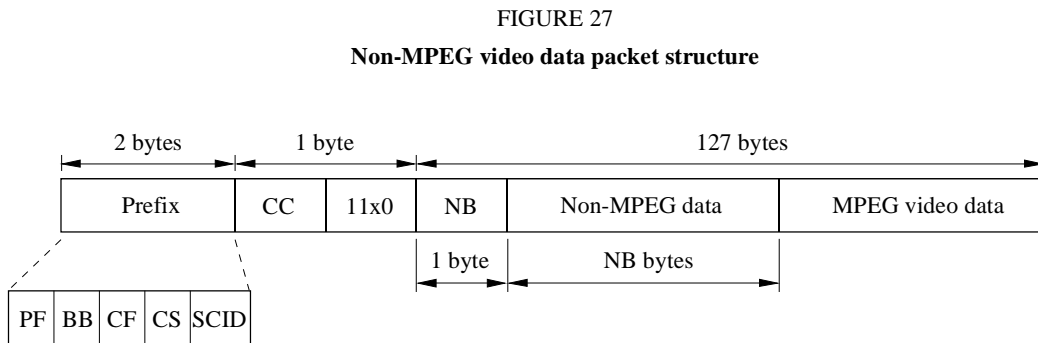
The Semantic definition of the (relevant) fields in the redundant data packet

BB	Bundle boundary	BB = 0 for redundant video service packets. The decoder should ignore this bit.
CF	Control flag	CF = 1: the transport block of this packet is not scrambled CF = 0: the transport block of this packet is scrambled
CS	Control sync	For scrambled transport packets (i.e. CF=0), this bit indicates the key to be used for de-scrambling.
HD	Header designator	HD = 10x0 <sub>b</sub> for redundant data packets The HD(1) bit, indicated by x in HD = 10x0 <sub>b</sub> , reflects the toggle state of the HD of the last basic video service packet (x value in HD = 01x0 <sub>b</sub> ) of the same SCID containing the original picture header start code.
NB	Number of bytes	This one byte field (unsigned integer, MSB first) represents the total length in bytes of the RH and MEF. The number of bytes indicated in the NB field has to be greater than or equal to 5 and less than or equal to 126 bytes, i.e., 5 ≤ NB ≤ 126.
RH	Redundant headers	This (NB - 4) byte field consists of redundant GOP and/or picture headers.
MEF	Media error field	This 4 byte MEF field is set equal to ISO MPEG defined sequence error code: 0x 00 00 01 B4 The intended use is that the transport processor sends the redundant GOP and picture headers and the media error field bytes to the MPEG video decoder whenever a packet error is detected (by the FEC decoder or by CC discontinuity). At other times the GOP and picture headers and the media field are not sent to the MPEG video decoder. The MPEG video decoder detects the presence of the Media error bytes and activates an error concealment procedure.
	MPEG data	The remainder of the data packet is filled with standard MPEG video data (non-redundant), which is a continuation of the video data stream from the previous packet of the same SCID having video data.

#### 4.4 Non-MPEG video data packets

The non-MPEG data packets are not used in normal operation. An exception is allowed only in the case of the first packet issued from an encoder changing from “backup” to “operational” mode.

The structure of a non-MPEG data packet is illustrated in Fig. 27. The semantic definition of the (relevant) fields in the non-MPEG video data packet is given in Table 13.



1294-27

TABLE 13

**The Semantic definition of the (relevant) fields in the non-MPEG video data packet**

BB	Bundle boundary	BB = 0 for non-MPEG video data packet The decoder should ignore this bit.
CF	Control flag	CF = 1: The transport block of this packet is not scrambled CF = 0: The transport block of this packet is scrambled
CS	Control sync	For scrambled transport packets (i.e. CF = 0), this bit indicates the key to be used for de-scrambling.
HD	Header designator	HD = 11x0 <sub>b</sub> for non-MPEG video data packets The HD(1) bit, indicated by x in HD = 11x0 <sub>b</sub> , reflects the toggle state of the HD of the last basic video service packet (x value in HD = 01x0 <sub>b</sub> ) of the same SCID.
NB	Number of bytes	This one byte field (unsigned integer, MSB first) represents the length in number of bytes of the following non-MPEG data field. The number of bytes indicated in the NB field has to be greater than or equal to 5 and less than or equal to 126 bytes, i.e., 5 ≤ NB ≤ 126.
	Non-MPEG data	This NB byte field consists of non-MPEG data, that cannot be interpreted by an MPEG video decoder.
	MPEG data	The remainder of the non-MPEG data packet is filled with standard MPEG video data (non-redundant).

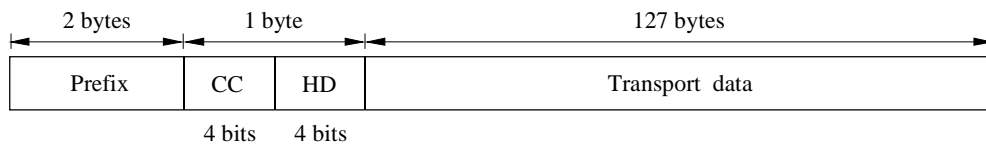
## 5 Audio application packets

The general structure of the audio transport packets is illustrated in Fig. 28. Within the audio application packets there are 3 types of transport cells, characterized by the type of audio service related data transported through them:

- Auxiliary data packets (time stamps, encryption control work packets)
- Basic audio service packets (MPEG audio data)
- Non-MPEG audio data packets (non-MPEG data and MPEG audio data)

To indicate different cell types and associated counters, the audio transport layer format has 4 bits for CC and 4 bits for a HD. A detailed description of these fields is given below in Table 14. Note that, of the 130 byte long packet, the first 2 bytes are used for prefix, the third byte contains CC and HD fields, and the remaining 127 bytes carry the payload.

FIGURE 28  
General audio application packet structure



1294-28

TABLE 14

The Semantic definition of elements in the CC HD byte

CC	Continuity counter	This 4 bit field (unsigned integer, MSB first) is incremented by one with each packet with the same SCID. After it reaches the maximum value of 15 (1111 <sub>b</sub> ), the continuity counter wraps around to 0. The continuity counter is set to 0 (0000 <sub>b</sub> ) and shall not be incremented when the HD field is equal to “0x 00” (Auxiliary packets). The CC allows a receiver to detect cell discontinuity (due to cell errors) for a particular transport service.
HD	Header designator	This 4 bit field indicates the 3 audio application packet types as: HD 0000 <sub>b</sub> Auxiliary data packets 0100 <sub>b</sub> Basic audio service packets 1100 <sub>b</sub> Non-MPEG audio data packets All other values are reserved.

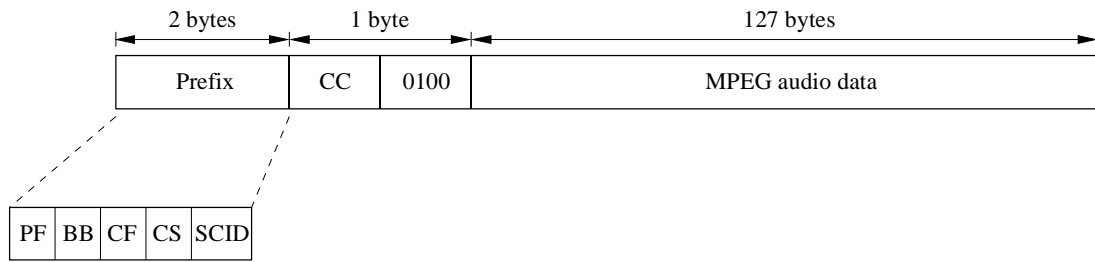
### 5.1 Auxiliary data packets

Auxiliary data packets for audio services have the same structure (syntax and semantics) as auxiliary data packets for video services as explained in § 4.1.

### 5.2 Basic audio service packets

The transport packets of an audio service with HD field set to 0100<sub>b</sub> carry basic audio service (i.e., MPEG audio bits) information. The structure of the basic audio service packet is illustrated in Fig. 29 and the semantic definition of the (relevant) fields is given in Table 15.

FIGURE 29  
Basic audio service packet structure



1294-29

TABLE 15  
The semantic definition of the (relevant) fields in the basic audio service packet

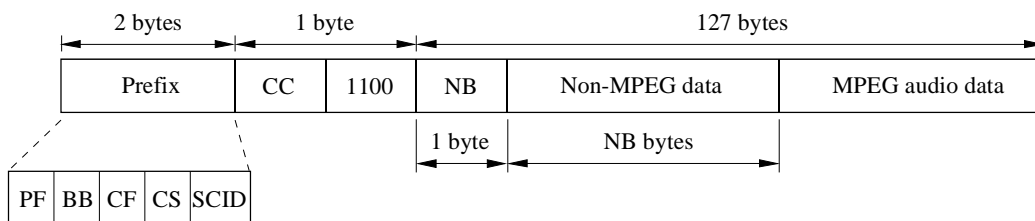
BB	Bundle boundary	BB = 0 for basic audio service packets.
CF	Control flag	CF = 1: The transport block of this packet is not scrambled. CF = 0: The transport block of this packet is scrambled.
CS	Control sync	For scrambled transport packets (i.e. CF=0), this bit indicates the key to be used for de-scrambling.
HD	Header designator	HD = 0100 <sub>b</sub> for basic audio service packets.
	MPEG audio data	127 bytes of standard MPEG audio data.

### 5.3 Non-MPEG audio data packets

The non-MPEG data packets are not used in normal operation. An exception is allowed only in the case of the first packet issued from an encoder changing from “backup” to “operational” mode.

The structure of a non-MPEG audio data packet is illustrated in Fig. 30 and the semantic definition of the (relevant) fields is given in Table 16.

FIGURE 30  
Non-MPEG audio data packet structure



1294-30

TABLE 16

The semantic definition of the (relevant) fields in the non-MPEG audio data packet

BB	Bundle boundary	BB = 0 for non-MPEG audio data packets.
CF	Control flag	CF = 1: the transport block of this packet is not scrambled. CF = 0: the transport block of this packet is scrambled.
CS	Control sync	For scrambled transport packets (i.e. CF=0), this bit indicates the key to be used for de-scrambling.
HD	Header designator	HD = 1100 <sub>b</sub> for non-MPEG audio data packets.
NB	Number of bytes	This one byte field (unsigned integer, MSB first) represents the length in number of bytes of the following non-MPEG data field. The number of bytes indicated in the NB field has to be greater than or equal to 5 and less than or equal to 126 bytes, i.e., $5 \leq NB \leq 126$ .
	Non-MPEG data	This (NB) byte field consists of non-MPEG data, that cannot be interpreted by an MPEG audio decoder.
	MPEG audio data	The remainder of the non-MPEG data packet is filled with standard MPEG audio data.

## 6 Program guide packets

The program guide packets consists of all data necessary to tune channels and display available program information to the viewers. The program guide streams defined in System B are:

Master program Guide (MPG), special program guide (SPG), purchase information parcel (PIP) and description information parcel (DIP) streams. These streams are carried in packets that have the same structure as illustrated in Fig. 31. The CF bit in the prefix field is set to 1 for all these streams (i.e., not scrambled). The SCID of the master program guide packets is always a fixed value that is predefined by the user.

FIGURE 31  
Program guide packet structure

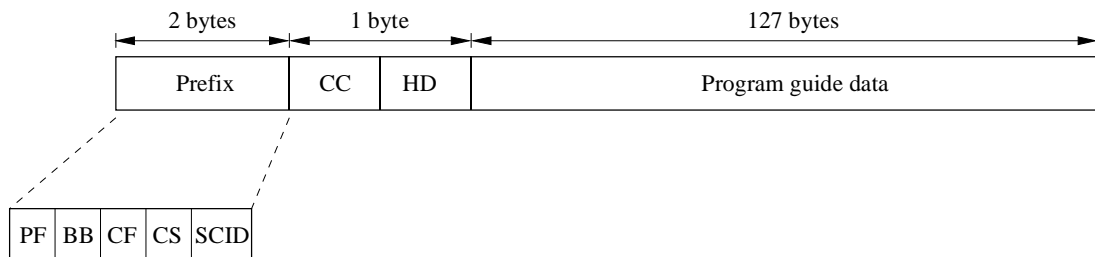


TABLE 17

## The semantic definition of the (relevant) fields in the program guide packet

BB	Bundle boundary	BB = 0 for program guide packets.
CF	Control flag	CF = 1 for program guide packets (not scrambled).
SCID	Service channel ID	SCID: this is a fixed value predefined by the user to identify master program guide data; format is a 12 bit field (unsigned integer, MSB first). Typical value is 0x001.
HD	Header designator	HD = 0100 <sub>b</sub> for program guide packets.

## 7 Transport multiplex constraints

Multiplex constraints for packet scheduling are identified for all transport packets on a transport multiplex. NULL packets are defined to fill otherwise unscheduled slots in the transport multiplex such that a constant transport multiplex rate is maintained over any interval of time.

### 7.1 Elementary stream multiplex constraint definition

The constraints identified in this section apply to transport packets of a given SCID having payload of the following elementary stream data types: video, audio, conditional access (CA), master program guide (MPG), special guides (SPG), description information parcels (DIP), purchase information parcels (PIP), “low speed” serial data (both “continuous” and “session”), and “high speed” wideband data (both buffered and unbuffered).

The nature of the constraint is to limit the frequency of occurrence for packets of a given SCID on the transport multiplex, such that packets carrying payload of a lower elementary stream rate are scheduled with less frequency than packets carrying payload of a higher elementary stream rate. The transport multiplex constraint essentially binds the peak rate of elementary stream data delivered to a decoder versus elementary stream source rate delivered from an encoder output.

A transport multiplex is considered valid if and only if each of the specified transport stream data types, per SCID, continuously satisfies the test of the *multiplex constraint* for the rates specified.

*Multiplex constraint:*

For each SCID of the specified data types, the transport packet delivery rate of elementary stream data is considered to be valid for rate “R” if and only if the following condition is continuously satisfied:

Elementary stream data is delivered from the “payload” field of transport packets of the selected SCID into a 508 byte buffer. Given that data is removed from said buffer at a constant rate “R” when data is available, transport packets of the given SCID should be scheduled such that said buffer does not overflow. Said buffer is allowed to be empty.

---