



Recommendation ITU-R BO.1516-1
(01/2012)

**Digital multiprogramme television systems
for use by satellites operating in the
11/12 GHz frequency range**

BO Series
Satellite delivery

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SF	Frequency sharing and coordination between fixed-satellite and fixed service systems
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SNG	Satellite news gathering
TF	Time signals and frequency standards emissions
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Note: This ITU-R Recommendation was approved in English under the procedure detailed in Resolution ITU-R 1.

Electronic Publication
Geneva, 2011

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RECOMMENDATION ITU-R BO.1516-1

Digital multiprogramme television systems for use by satellites operating in the 11/12 GHz frequency range

(Question ITU-R 285/4)

(2001-2012)

Scope

This Recommendation proposes common functional requirements for four satellite digital multiprogramme reception systems for television, sound and data services. Annex 1 provides the common functional requirements for television transmissions through satellites operating in the 11/12 GHz frequency range.

The ITU Radiocommunication Assembly,

considering

- a) that digital multiprogramme systems have been designed for use by satellites in the 11/12 GHz frequency range;
- b) that these systems, being digital, provide significant advantages in service quality of video, sound and data, flexibility of use, spectrum efficiency and emission robustness;
- c) that these systems provide for a multiplicity of services such as TV programmes, multimedia elements, data services, audio channels and the like in a single multiplex;
- d) that these systems are either in widespread operational use or are planned to be in operational use in the near future;
- e) that significant advances have been made in digital multiprogramme system technology following the development of former Recommendation ITU-R BO.1294, and these advances are embodied in the system described in Recommendation ITU-R BO.1408;
- f) that integrated circuits compatible with some or all of the common elements of two or three of these systems have been designed, manufactured, and are in widespread use;
- g) that these systems have various distinguishing features that may make one or other of these systems more appropriate for the needs of an administration;
- h) that Resolution ITU-R 1 states that “When Recommendations provide information on various systems relating to one particular radio application, they should be based on criteria relevant to the application, and should include, where possible, an evaluation of the recommended systems, using those criteria”,

recommends

- 1** that administrations wishing to implement digital multiprogramme television services via satellite should refer to the characteristics described in Annex 1, § 4 as an aid in the selection of a specific system;
- 2** that one of the transmission systems described in Annex 1 should be selected when implementing digital multiprogramme television services via satellite;

3 that the common elements of the common functional requirements of a digital multiprogramme transmission system, as described in § 3 of Annex 1, should serve as a basis for implementation of the services in those areas where more than one system coexists or may coexist in the future.

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

Since their introduction, satellite digital TV systems have continued to demonstrate their ability to efficiently use the satellite frequency spectrum and the ability to deliver high quality services to consumers. Four of these systems have been described in former Recommendations ITU-R BO.1211, ITU-R BO.1294 and Recommendation ITU-R BO.1408.

With the aim to promote the convergence on a worldwide standard for satellite digital multiprogramme reception systems for television, sound and data services, the common functional requirements for the reception of digital multiprogramme television emissions by satellite were described in former Recommendation ITU-R BO.1294. In this Recommendation, common functional requirements and common elements were defined for a satellite integrated receiver-decoder (IRD) operating in the 11/12 GHz frequency range. Use in other frequency ranges was not and is not excluded. Former Recommendation ITU-R BO.1294 took into account the single system described in former Recommendation ITU-R BO.1211.

The common elements of the satellite IRD as defined in former Recommendation ITU-R BO.1294 are capable of receiving emissions from three digital multiprogramme transmission systems. These systems were identified as Systems A, B and C. The common and unique elements of each of these systems were analysed, and it was concluded that practical implementation of the common elements of a satellite IRD was feasible. Since that time, the continued development of integrated circuits for use in these systems has clearly demonstrated this finding, with many integrated circuits now available that are compatible with the common elements of two or all three of these systems.

A fourth system has since been developed, and is described in Recommendation ITU-R BO.1408. It too shares the same common elements described in former Recommendation ITU-R BO.1294. It represents an advancement of the technology of these digital multiprogramme systems. It provides such added features as the ability to simultaneously support multiple modulation types, an hierarchical modulation scheme, and the ability to handle multiple Moving Picture Experts Group (MPEG) transport streams within a given carrier.

In the following sections of this Annex the common functional requirements and elements of these systems are briefly reviewed, and the functions of a generic digital multiprogramme transmission system are briefly described.

A summary and detailed system level characteristics of each of these four systems are also provided. These system level parameters are applicable to the implementation of either the transmission equipment or the integrated receiver decoder.

2 The generic reference model of digital multiprogramme transmission systems

2.1 Generic reference model

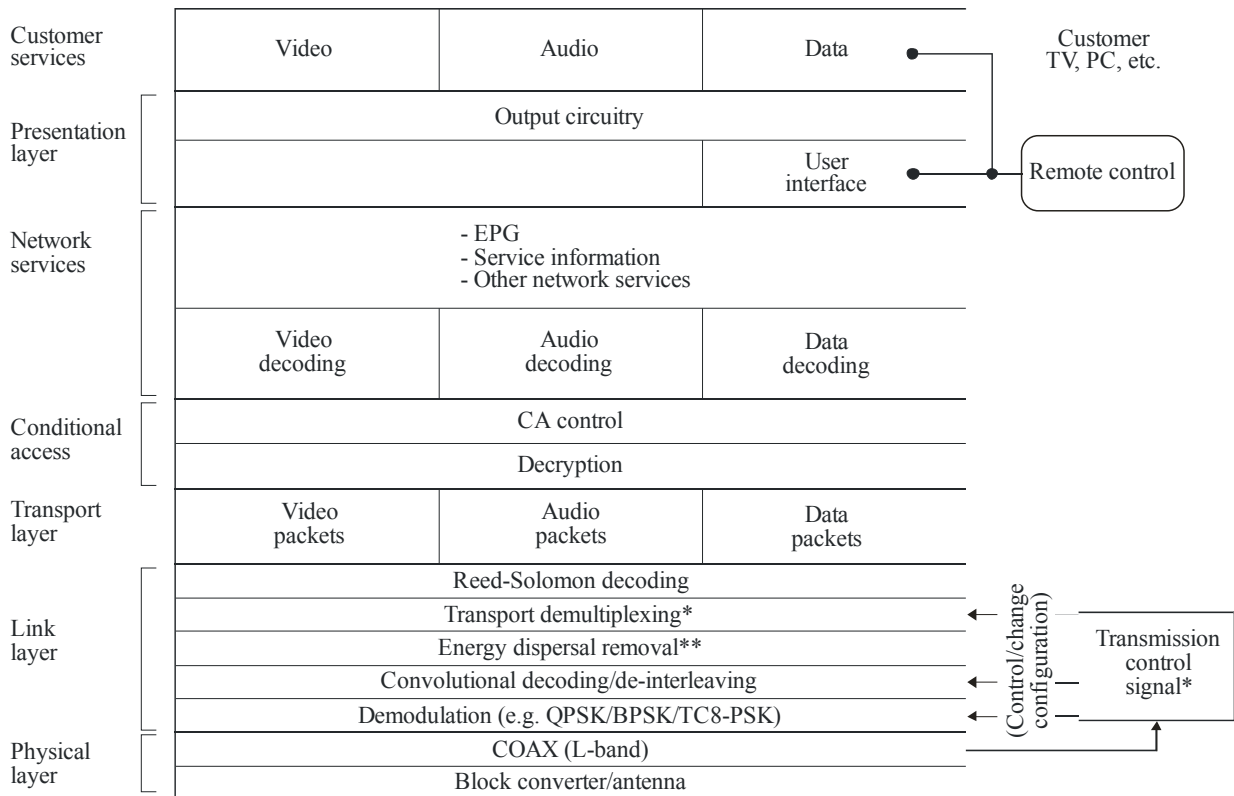
A generic reference model for the common functional requirements of a digital multiprogramme transmission system has been produced. This generic reference model has been shown to be applicable to all of the four systems described herein.

The generic reference model has been defined based on the common functions required over all layers of a digital multiprogramme transmission system protocol stack. It can be used to define the common functions required in an IRD for the reception of these transmissions.

For reference, Fig. 1 presents the typical IRD protocol stack which is based on the following layers:

- *Physical and link layers* covering the typical front-end functions: carrier generation and carrier reception (tuning), quadrature phase shift keying (QPSK) modulation and demodulation, convolutional encoding and decoding, interleaving and de-interleaving, Reed-Solomon encoding and decoding, and energy dispersal application and removal.
- *Transport layer* responsible for the multiplexing and demultiplexing of the different programs and components as well as the packetization and depacketization of the information (video, audio and data).
- *Conditional access* functions which control the operation of the external encryption and decryption functions and associated control functions (common interface for conditional access as an option).
- *Network services* performing video and audio coding and decoding as well as the management of electronic programme 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

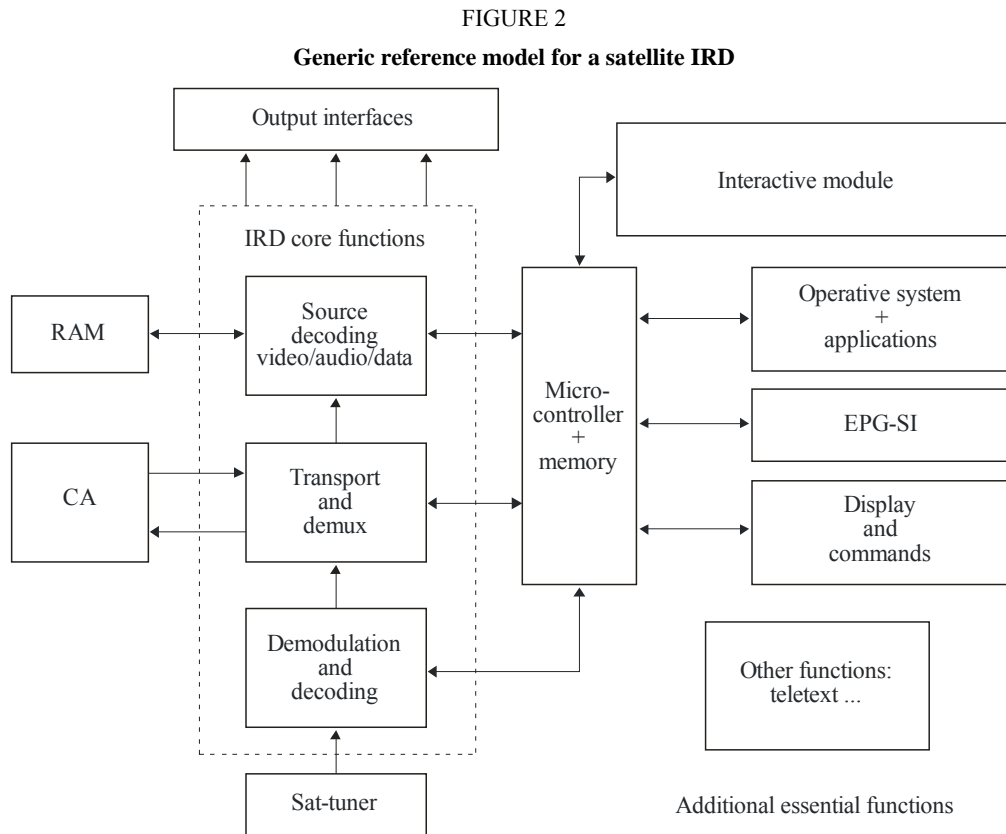


BPSK: binary PSK
 TC8-PSK: trellis-coded 8-PSK
 * Only present in System D.
 ** Not present in System B.

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2.2 Application to the satellite IRD

Based on the protocol stack, the generic block diagram for the satellite IRD (Fig. 2) can be derived. This is useful in demonstrating how the common elements are organized within the IRD.



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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
 - Service/system information (SI)
 - Conditional access (CA)
 - Display, remote control and different commands
 - Read only memory (ROM), random access memory (RAM) and FLASH memory
 - Interactive module
 - Microcontroller
 - Other functions as teletext, subtitling, etc.

3 Common elements of digital multiprogramme transmission systems

The common elements are as follows:

- Modulation/demodulation and error correction coding/decoding.
- Transport multiplex and demultiplex.
- Source encoding and decoding of video, audio and data.

3.1 Modulation/demodulation and coding/decoding

The block diagram of the modulation/demodulation and coding/decoding functions of the common elements is presented in Fig. 3. Overlapped blocks represent functions with common elements for the four systems although with different characteristics. Dashed blocks represent functions not utilized by all four systems.

3.1.1 Modulation and demodulation

This common element performs the quadrature, binary or 8 phase coherent modulation and demodulation function. In the demodulator, the demodulator provides “soft decision” I and Q information to the inner decoder.

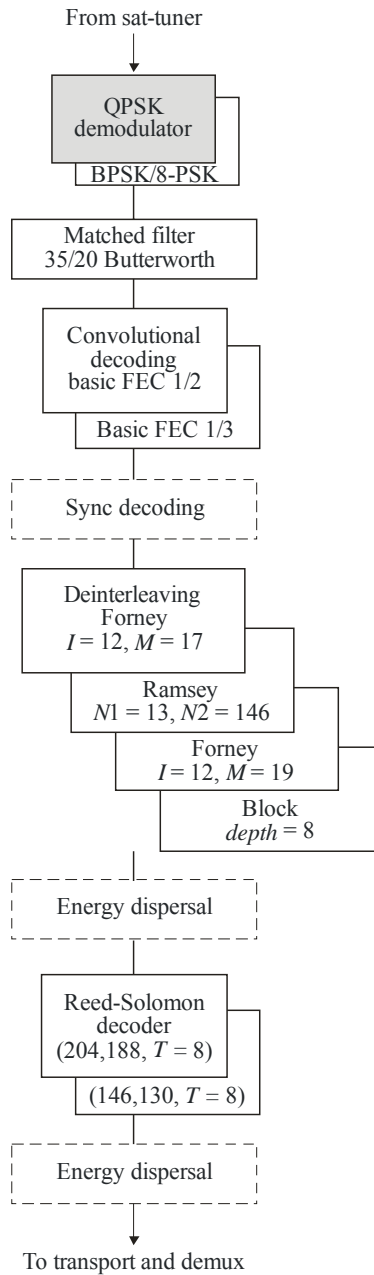
Within a satellite IRD this common element will be capable of demodulating a signal employing conventional Gray-coded QPSK modulation and TC 8-PSK modulation with absolute mapping (no differential coding).

For QPSK modulation, bit mapping in the signal as given in Fig. 4 will be used.

For the binary or 8-PSK modulation, bit mapping in the signal as described in § 5.2.4 will be used.

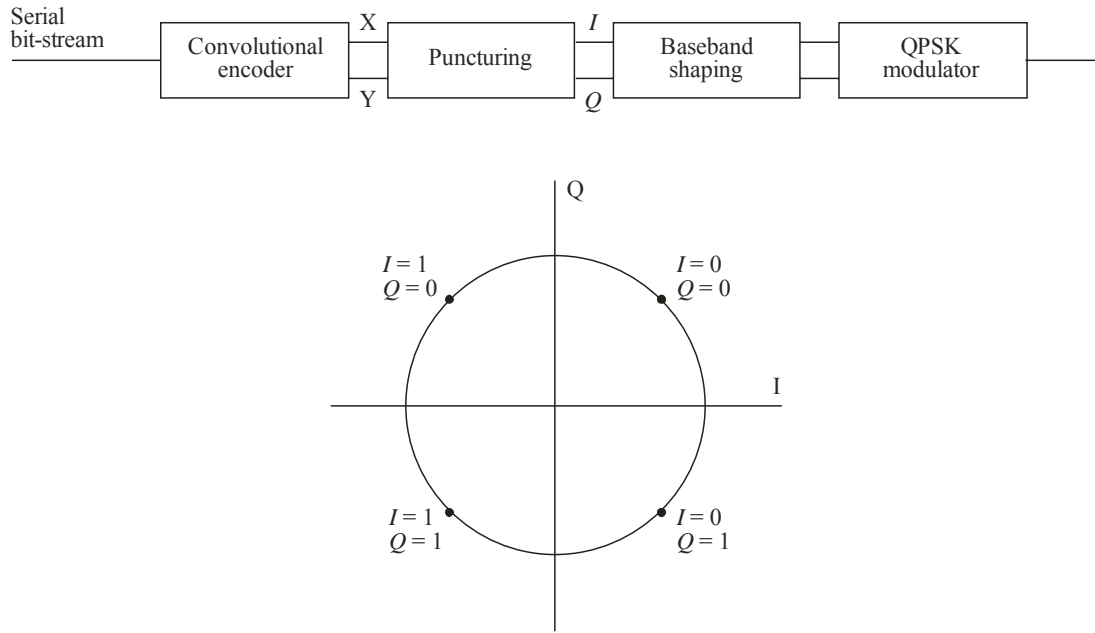
FIGURE 3

Block diagram for demodulation and channel decoding



FEC: forward error correction

FIGURE 4
QPSK constellation



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3.1.2 Matched filter

This common element within the demodulator 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.

The satellite IRD must be capable of processing the signal with the following shaping and roll-off factors:

- | | |
|---|---------------------------------------|
| Square root raised cosine: | $\alpha = 0.35$ and 0.20 |
| Band-limited 4 th 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 encoding and decoding

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

The inner code has the following characteristics:

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

The coder and decoder operate with 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 common element will decode the sync bytes. The decoder provides synchronization information for the de-interleaving. It is also in a position to recover the phase ambiguity of the demodulator (not detectable by the Viterbi decoder).

Specific characteristics are provided in § 5.3.

3.1.5 Convolutional de-interleaver

This common 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 common element utilizes Ramsey Type II ($N_1 = 13$, $N_2 = 146$) and Ramsey Type III (Forney approach) ($I = 12$, $M = 17$ and 19) convolutional interleaver systems or block interleaver system (depth = 8), as specifically defined in § 5.4.

3.1.6 Reed-Solomon coder and decoder

This common element provides second level error protection. It is in a position to provide QEF output (i.e. BER of about 1×10^{-10} and 1×10^{-11}) in the presence of input error bursts at a BER of about 7×10^{-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 common element has 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:

$$\alpha = 02_h.$$

- Reed-Solomon field generator polynomial:

$$x^8 + x^4 + x^3 + x^2 + 1$$

Specific characteristics are provided in § 5.5.

3.1.7 Energy dispersal removal

This common element adds a randomizing pattern to the transmission to ensure even energy dispersal, which when present must be removed by the demodulator. 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 common element of a satellite IRD may implement a bypass to this feature.

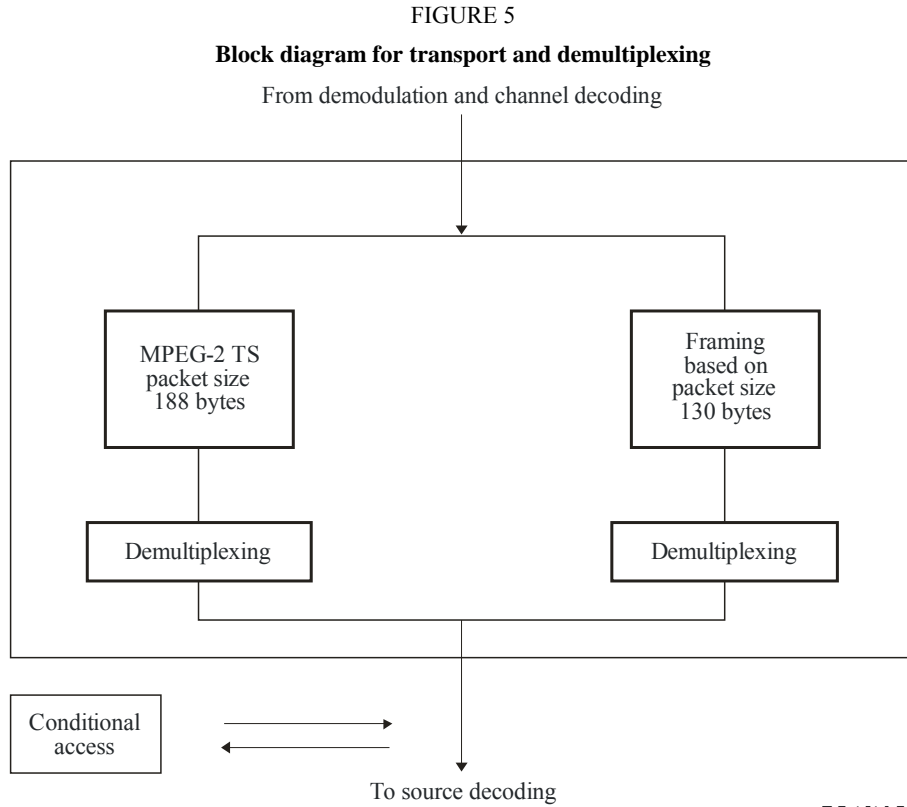
Specific characteristics are provided in § 5.6.

3.2 Transport and demultiplexing

The block diagram of the transport and multiplex/demultiplex 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.

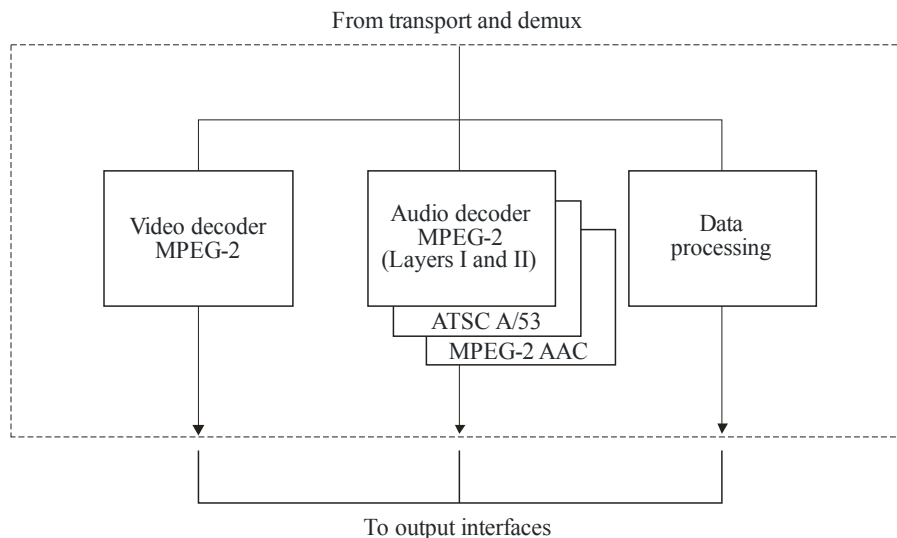
Conditional access is outside the scope of this Recommendation.



3.3 Source coding and decoding of video, audio and data

The block diagram of the source encoding or decoding of video, audio and data functions is presented in Fig. 6.

FIGURE 6

Block diagram for source decoding

ATSC: Advanced Television Systems Committee

BO.1516-06

3.3.1 Video

This common element requires, as a minimum, the source coding and decoding of video formats following the main profile main level MPEG-2 signals as specified in ISO/IEC 13818-2.

3.3.2 Audio

This common element requires the source coding and decoding of audio signals following the MPEG-2 Layers I and II (ISO/IEC 13818-3), ATSC-A/53 Annex B (Recommendation ITU-R BS.1196, Annex 2) formats, and the MPEG-2 AAC (advanced audio coding) (ISO/IEC 13818-7).

3.3.3 Data

This block addresses the functions required to process source coded data delivered to or from the transport multiplex. This item is outside the scope of the Recommendation.

4 Summary characteristics and the comparison of digital multiprogramme TV systems by satellite

As described in the introduction, this Recommendation includes the characteristics of four digital multiprogramme TV systems that share the common elements described in section 3. These systems are identified as Systems A, B, C and D. System A was first described in former Recommendation ITU-R BO.1211 and is also included in former Recommendation ITU-R BO.1294. Systems B and C were first described in former Recommendation ITU-R BO.1294. System D is described in Recommendation ITU-R BO.1408. Three of these systems are in operational use today, and the fourth is planned for operational deployment in the very near future.

These systems are designed to robustly deliver quality MPEG video and audio programming via digital satellite transmissions. The use of MPEG compression techniques provides very efficient use of the available spectrum, and the design of the transport layer allows very flexible assignment of video and audio programming to satellite transponders.

System A is based on the MPEG-2 video and sound coding algorithm and on the MPEG-2 transport multiplex. A concatenated FEC scheme using Reed-Solomon and convolutional coding, with soft-decision Viterbi decoding, allows very robust RF performance in the presence of noise and interference. Five coding rate steps in the range 1/2 to 7/8, offer different trade-offs between spectrum and power efficiency. The transmission symbol rate of the system can be chosen by the operator, to optimize the exploitation of the satellite transponder bandwidth.

System B is also based on the MPEG-2 main profile main level video coding algorithm. It uses the MPEG-1 Layer II audio syntax and the System B transport specification. As with System A, a concatenated FEC scheme using Reed-Solomon and convolutional coding, with soft-decision Viterbi decoding, allows very robust RF performance in the presence of noise and interference. Three coding rate steps in the range 1/2 to 6/7, offer different trade-offs between spectrum and power efficiency. The transmission symbol rate is fixed at 20 m symbols/s.

System C can also carry multiple digital television (and radio) services in time division multiplexed (TDM) format, and it shares the same common architectural elements as described above. The system includes renewable access control, impulse pay-per-view (IPPV), and data services. Virtual channels allow simplified viewer navigation and “surfing” between channels.

System D is a newly developed system designed for the broadcast of multimedia services. It integrates systematically various kinds of digital contents, each of which may include multiprogramme video from low definition television (LDTV) to high definition television (HDTV), multiprogramme audio, graphics, texts, etc. The proposed system can be integrated on the basis of MPEG-transport stream (MPEG-TS) which is widely used as a common container for digital contents.

In order to cover a wide range of requirements that may differ from one service to another, System D provides a series of modulation and/or error protection schemes that can be selected and combined flexibly. Introduction of multiple modulation/error correction schemes is especially useful for countries located in climatic zones experiencing high rain attenuation.

4.1 Summary system characteristics

Table 1 provides information on relevant parameters which characterize these four digital multiprogramme systems. The Table includes information on both core functions (common elements) as well as additional essential functions.

4.2 Comparison of system characteristics

The Radiocommunication Assembly in § 6.1.2 of Resolution ITU-R 1 states that: “When Recommendations provide information on various systems relating to one particular radio application, they should be based on criteria relevant to the application, and should include, where possible, an evaluation of the recommended systems, using those criteria.” Table 2 provides this evaluation. Performance criteria relevant to these systems were selected, and the associated parametric values or capabilities of each of these systems are provided.

TABLE 1

Summary characteristics of digital multiprogramme TV systems by satellite*a) Function*

	System A	System B	System C	System D
Delivered services	SDTV and HDTV	SDTV and HDTV	SDTV and HDTV	SDTV and HDTV
Input signal format	MPEG-TS	Modified MPEG-TS	MPEG-TS	MPEG-TS
Multiple input signal capability	No	No	No	Yes, 8 maximum
Rain fade survivability	Determined by transmitter power and inner code rate	Determined by transmitter power and inner code rate	Determined by transmitter power and inner code rate	Hierarchical transmission is available in addition to the transmitter power and inner code rate
Mobile reception	Not available and for future consideration	Not available and for future consideration	Not available and for future consideration	Not available and for future consideration
Flexible assignment of services bit rate	Available	Available	Available	Available
Common receiver design with other receiver systems	Systems A, B, C and D are possible	Systems A, B, C and D are possible	Systems A, B, C and D are possible	Systems A, B, C and D are possible
Commonality with other media (i.e. terrestrial, cable, etc.)	MPEG-TS basis	MPEG-ES (elementary stream) basis	MPEG-TS basis	MPEG-TS basis

TABLE 1 (continued)

b) Performance

	System A	System B	System C	System D
Net data rate (transmissible rate without parity)	Symbol rate (R_s) is not fixed. The following net data rates result from an example R_s of 27.776 Mbd: 1/2: 23.754 Mbits/s 2/3: 31.672 Mbits/s 3/4: 35.631 Mbits/s 5/6: 39.590 Mbits/s 7/8: 41.570 Mbits/s	1/2: 17.69 Mbits/s 2/3: 23.58 Mbits/s 6/7: 30.32 Mbits/s	19.5 Mbd 29.3 Mbd 5/11: 16.4 Mbits/s 24.5 Mbits/s 1/2: 18.0 Mbits/s 27.0 Mbits/s 3/5: 21.6 Mbits/s 32.4 Mbits/s 2/3: 24.0 Mbits/s 36.0 Mbits/s 3/4: 27.0 Mbits/s 40.5 Mbits/s 4/5: 28.8 Mbits/s 43.2 Mbits/s 5/6: 30.0 Mbits/s 45.0 Mbits/s 7/8: 31.5 Mbits/s 47.2 Mbits/s	Up to 52.2 Mbits/s (at a symbol rate of 28.86 Mbd)
Upward extensibility	Yes	Yes	Yes	Yes
HDTV capability	Yes	Yes	Yes	Yes
Selectable conditional access	Yes	Yes	Yes	Yes

c) Technical characteristics (Transmission)

	System A	System B	System C	System D
Modulation scheme	QPSK	QPSK	QPSK	TC8-PSK/QPSK/BPSK
Symbol rate	Not specified	Fixed 20 Mbd	Variable 19.5 and 29.3 Mbd	Not specified (e.g. 28.86 Mbd)
Necessary bandwidth (-3 dB)	Not specified	24 MHz	19.5 and 29.3 MHz	Not specified (e.g. 28.86 MHz)
Roll-off rate	0.35 (raised cosine)	0.2 (raised cosine)	0.55 and 0.33 (4 th order Butterworth filter)	0.35 (raised cosine)
Reed-Solomon outer code	(204,188, $T = 8$)	(146,130, $T = 8$)	(204,188, $T = 8$)	(204,188, $T = 8$)
Reed-Solomon generator	(255,239, $T = 8$)	(255,239, $T = 8$)	(255,239, $T = 8$)	(255,239, $T = 8$)

TABLE 1 (continued)

	System A	System B	System C	System D
Reed-Solomon code generator polynomial	$(x + \alpha^0)(x + \alpha^1)\dots(x + \alpha^{15})$ where $\alpha = 02_h$	$(x + \alpha^0)(x + \alpha^1)\dots(x + \alpha^{15})$ where $\alpha = 02_h$	$(x + \alpha^1)(x + \alpha^2)\dots(x + \alpha^{16})$ where $\alpha = 02_h$	$(x + \alpha^0)(x + \alpha^1)\dots(x + \alpha^{15})$ where $\alpha = 02_h$
Reed-Solomon field generator polynomial	$x^8 + x^4 + x^3 + x^2 + 1$	$x^8 + x^4 + x^3 + x^2 + 1$	$x^8 + x^4 + x^3 + x^2 + 1$	$x^8 + x^4 + x^3 + x^2 + 1$
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	PRBS: $1 + x^{14} + x^{15}$
Loading sequence into pseudo random binary sequence (PRBS) register	100101010000000	N.A.	0001 _h	100101010000000
Randomization point	Before RS encoder	N.A.	After RS encoder	After RS encoder
Interleaving	Convolutional, $I = 12, M = 17$ (Forney)	Convolutional, $N1 = 13, N2 = 146$ (Ramsey II)	Convolutional, $I = 12, M = 19$ (Forney)	Block (depth = 8)
Inner coding	Convolutional	Convolutional	Convolutional	Convolutional, Trellis (8-PSK: TCM 2/3)
Constraint length	$K = 7$	$K = 7$	$K = 7$	$K = 7$
Basic code	1/2	1/2	1/3	1/2
Generator polynomial	171, 133 (octal)	171, 133 (octal)	117, 135, 161 (octal)	171, 133 (octal)
Inner coding rate	1/2, 2/3, 3/4, 5/6, 7/8	1/2, 2/3, 6/7	1/2, 2/3, 3/4, 3/5, 4/5, 5/6, 5/11, 7/8	1/2, 3/4, 2/3, 5/6, 7/8
Transmission control	None	None	None	TMCC
Frame structure	None	None	None	N slot/frame (e.g. N = 48) 8 frame/super frame

TABLE 1 (continued)

	System A	System B	System C	System D
Packet size	188 bytes	130 bytes	188 bytes	188 bytes
Transport layer	MPEG-2	Non-MPEG	MPEG-2	MPEG-2
Satellite downlink frequency range	Originally designed for 11/12 GHz, not excluding other satellite frequency ranges	Originally designed for 11/12 GHz, not excluding other satellite frequency ranges	Originally designed for 11/12 GHz and 4 GHz satellite frequency ranges	Originally designed for 11/12 GHz, not excluding other satellite frequency ranges

d) Example-technical characteristics (Source coding)

		System A	System B	System C	System D
Video source coding	Syntax	MPEG-2	MPEG-2	MPEG-2	MPEG-2
	Levels	At least main level	At least main level	At least main level	From low level to high level
	Profiles	At least main profile	At least main profile	At least main profile	Main profile
Aspect ratios		4:3 16:9 (2.12:1 optionally)	4:3 16:9	4:3 16:9	4:3 16:9
Image supported 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 × 1280 1280 × 1024 1920 × 1080	720(704) × 576 720(704) × 480 528 × 480 528 × 576 352 × 480 352 × 576 352 × 288 352 × 240	1920 × 1080 1440 × 1080 1280 × 720 720 × 480 544 × 480 480 × 480 352 × 240* 176 × 120* (* for hierarchical transmission)
Frame rates at monitor (per s)		25	29.97	25 or 29.97	29.97 or 59.94

TABLE 1 (end)

	System A	System B	System C	System D
Audio source decoding	MPEG-2, Layers I and II	MPEG-1, Layer II; ATSC A/53 (AC3)	ATSC A/53 or MPEG-2 Layers I and II	MPEG-2 AAC
Service information	ETS 300 468	System B	ATSC A/56 SCTE DVS/011	ETS 300 468
EPG	ETS 300 707	System B	User selectable	User selectable
Teletext	Supported	Not specified	Not specified	User selectable
Subtitling	Supported	Supported	Supported	Supported
Closed caption	Not specified	Yes	Yes	Supported

TABLE 2

Comparison characteristics table

Modulation and coding	System A		System B		System C		System D	
Modulation modes supported individually and on the same carrier	QPSK		QPSK		QPSK		8-PSK, QPSK, and BPSK	
Performance (define quasi-error-free (QEF) required C/N (bits/s/Hz))	Spectral efficiency	C/N for QEF ⁽¹⁾	Spectral efficiency	C/N for QEF ⁽²⁾	Spectral efficiency ⁽³⁾	C/N for QEF ⁽⁴⁾	Spectral efficiency	C/N for QEF ⁽⁵⁾
Modes Inner code								
BPSK Conv. 1/2	Not used		Not used		No		0.35	0.2

TABLE 2 (continued)

Modulation and coding		System A		System B		System C		System D	
QPSK Conv.	5/11	Not used		Not used		0.54/0.63	2.8/3.0	Not used	
	1/2	0.72	4.1	0.74	3.8	0.59/0.69	3.3/3.5	0.7	3.2
	3/5	No		Not used		0.71/0.83	4.5/4.7		
	2/3	0.96	5.8	0.98	5	0.79/0.92	5.1/5.3	0.94	4.9
	3/4	1.08	6.8	Not used		0.89/1.04	6.0/6.2	1.06	5.9
	4/5	Not used		Not used		0.95/1.11	6.6/6.8	Not used	
	5/6	1.2	7.8	Not used		0.99/1.15	7.0/7.2	1.18	6.8
	6/7	Not used		1.26	7.6	Not used		Not used	
	7/8	1.26	8.4	Not used		1.04/1.21	7.7/7.9	1.24	7.4
8-PSK Trellis	Not used		Not used		Not used		1.4	8.4	
Capable of hierarchical modulation control?	No		No		No		Yes		
Symbol rate characteristics	Continuously variable		Fixed, 20 Mbd		Variable, 19.5 or 29.3 Mbd		Continuously variable		
Packet length (bytes)	188		130		188		188		
Transport streams supported	MPEG-2		System B		MPEG-2		MPEG-2		

TABLE 2 (*end*)

Transport and multiplexing	System A	System B	System C	System D
Transport stream correspondence with satellite channels	One stream/channel	One stream/channel	One stream/channel	1 to 8 streams/channel
Support for statistical multiplex of video streams	No limitation within a transport stream	No limitation within a transport stream	No limitation within a transport stream	No limitation within a transport stream. Also, may be possible across transport streams within a satellite channel

TWTA: travelling wave tube amplifier

IMUX: input multiplex

OMUX: output multiplex

- (1) At a BER $<10^{-10}$. The C/N values for System A refer to computer simulation results achieved on a hypothetical satellite chain, including IMUX, TWTA and OMUX, with modulation roll-off of 0.35. They are based on the assumption of soft-decision Viterbi decoding in the receiver. A bandwidth to symbol rate ratio of 1.28 has been adopted. The figures for C/N include a calculated degradation of 0.2 dB due to bandwidth limitations on IMUX and OMUX filters, 0.8 dB non-linear distortion on TWTA at saturation and 0.8 dB modem degradation. The figures apply to BER = 2×10^{-4} before RS (204,188), which corresponds to QEF at the RS coder output. Degradation due to interference is not taken into account.
- (2) At a BER of 1×10^{-12} .
- (3) As calculated by $2(Rc)(188/204)/1.55$ or $2(Rc)(188/204)/1.33$ for System C normal and truncated transmit spectral shaping, respectively, where Rc is the convolutional code rate.
- (4) Theory QPSK (2-bit per symbol) E_s/N_0 i.e. C/N as measured in baud rate bandwidth for normal and truncated spectral shaping, respectively. Does not include hardware implementation margin or satellite transponder loss margin.
- (5) These values were derived from computer simulations and regarded as theoretical values. The values apply to BER = 2×10^{-4} before RS (204,188) with baud rate bandwidth (Nyquist bandwidth). Do not include hardware implementation margin or satellite transponder loss margin.

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.

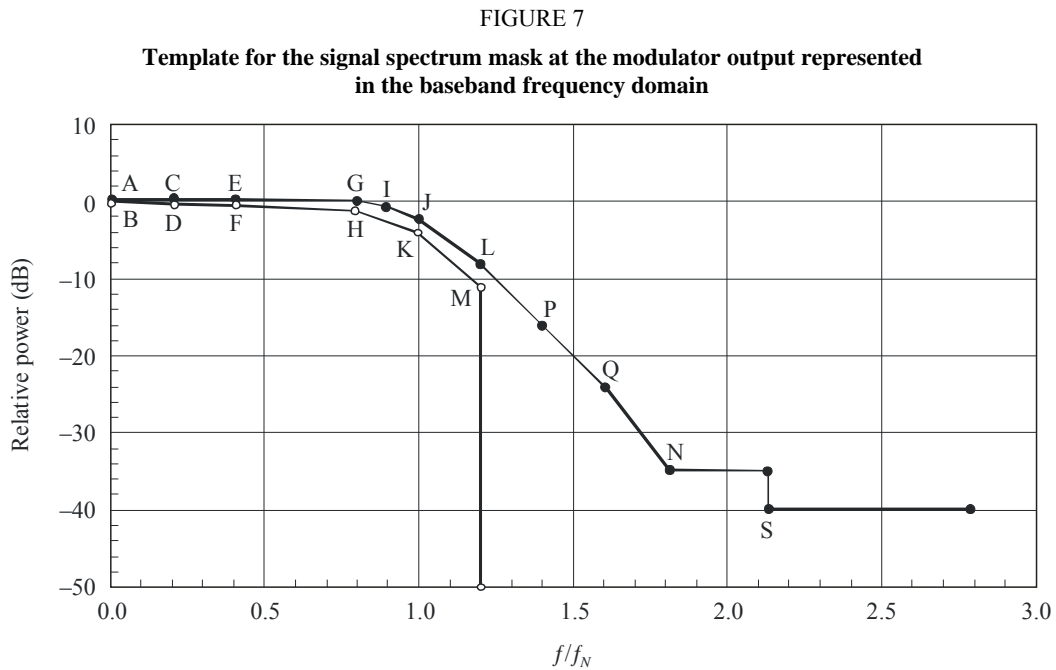
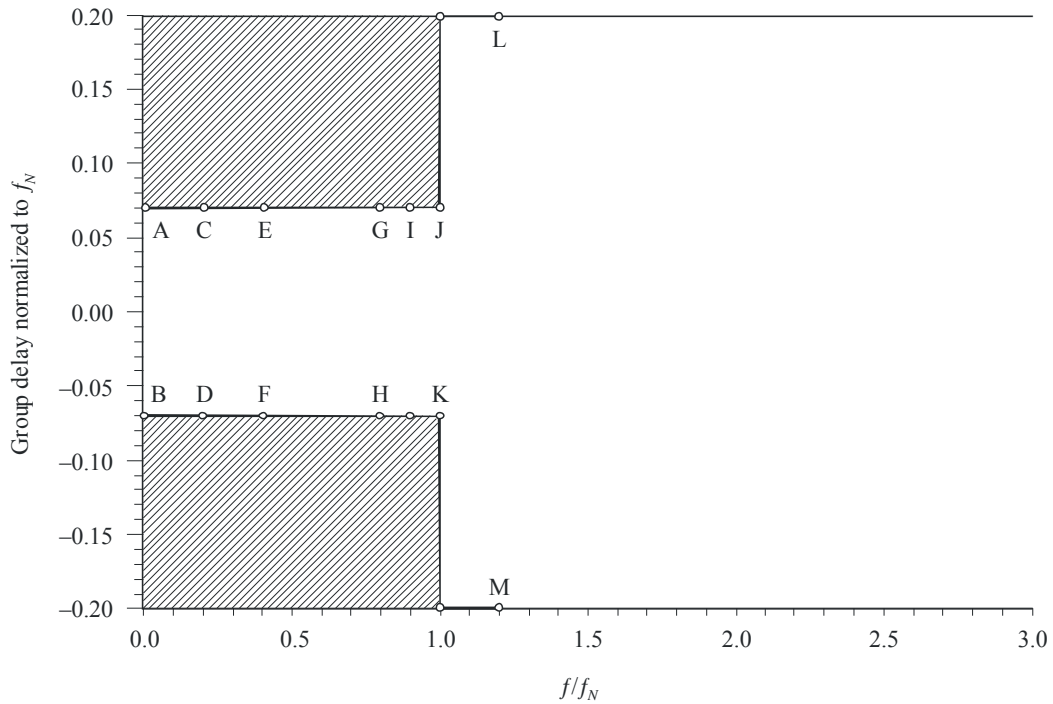


Figure 7 also represents a possible mask for a hardware implementation of the Nyquist modulator filter. The points A to S shown in Figs 7 and 8 are defined in Table 3. The mask for the filter frequency response is based on the assumption of ideal Dirac delta input signals, spaced by the symbol period $T_s = 1/R_s = 1/2f_N$, 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



BO.1516-08

TABLE 3

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	-

TABLE 3 (end)

Point	Frequency	Relative power (dB)	Group delay
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

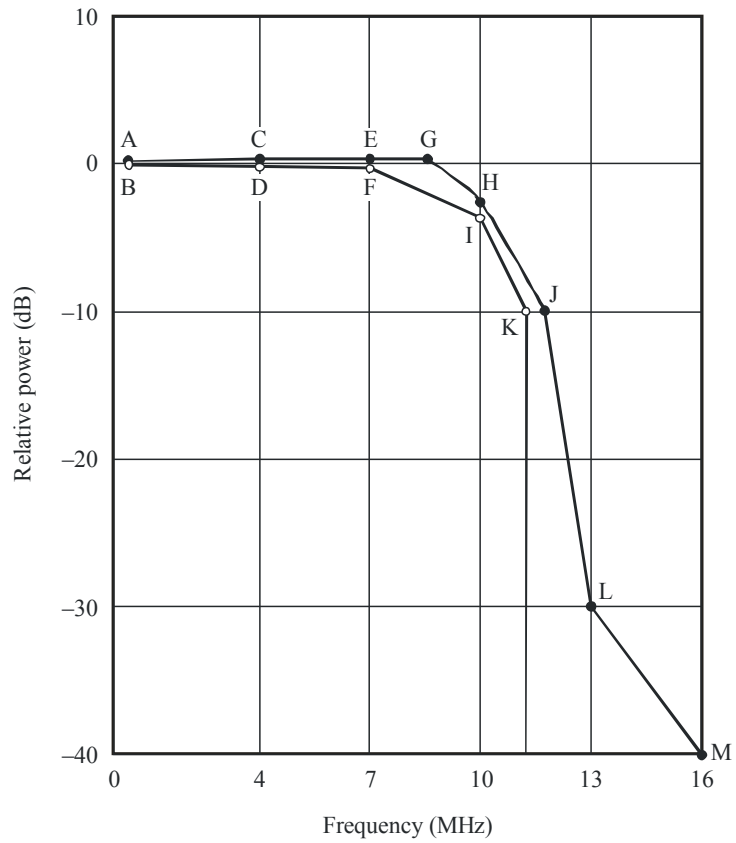


TABLE 4
Coordinates of points

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.

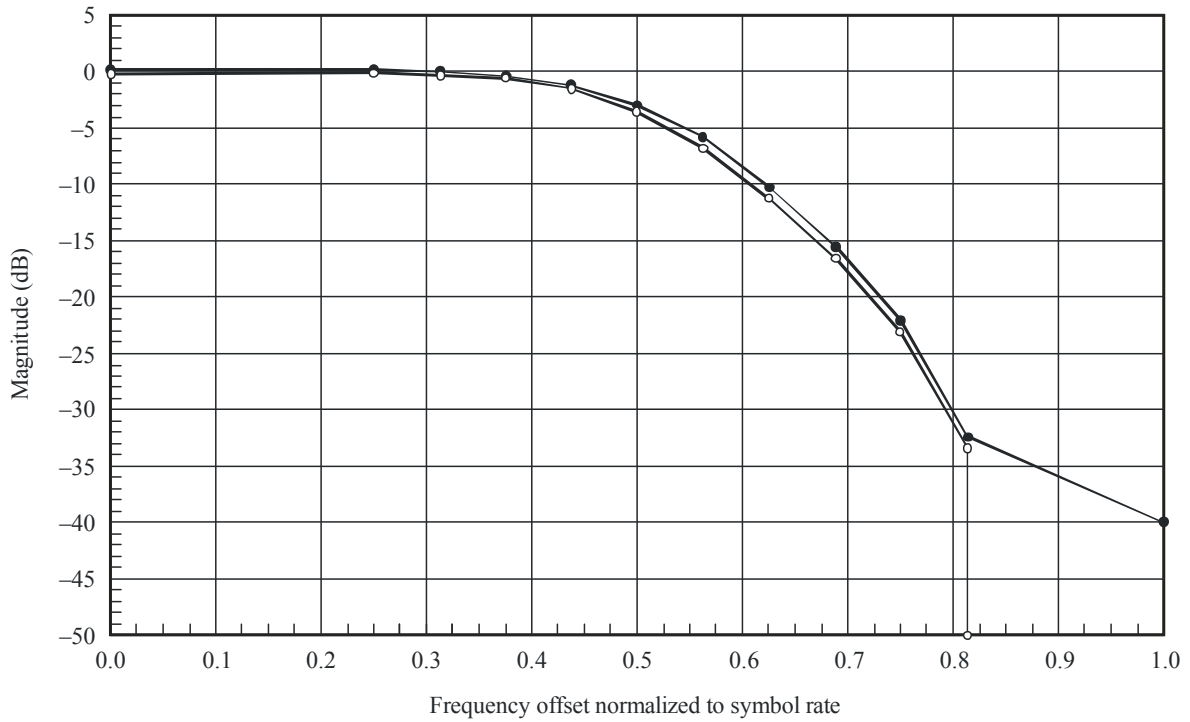
5.1.3.1 Baseband signal shaping

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. Tables 5a and 5b tabulate the corresponding breakpoints for standard and truncated-spectrum modes, respectively.

FIGURE 10a
Spectral density mask for standard mode



BO.1516-10a

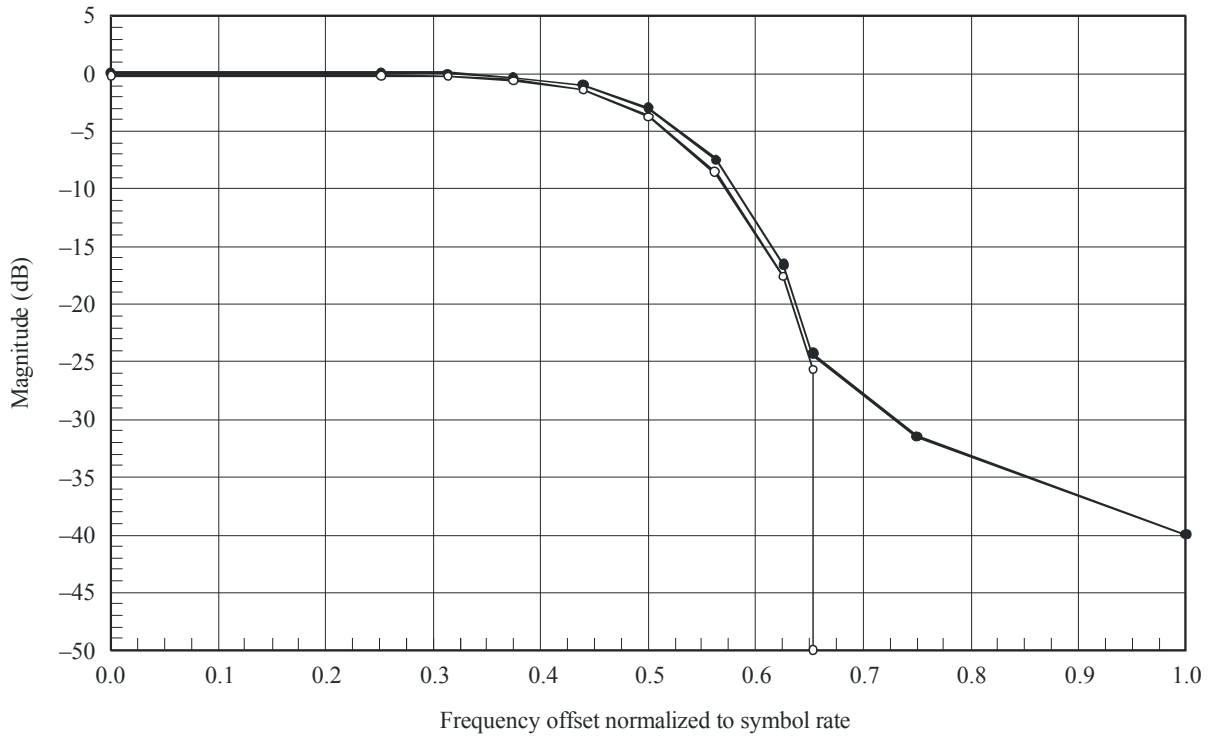
TABLE 5a

Spectral density mask breakpoints for standard mode

Frequency offset normalized to transmit symbol rate	Upper mask breakpoints (dB)	Lower mask breakpoints (dB)
0.00	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



BO.1516-10b

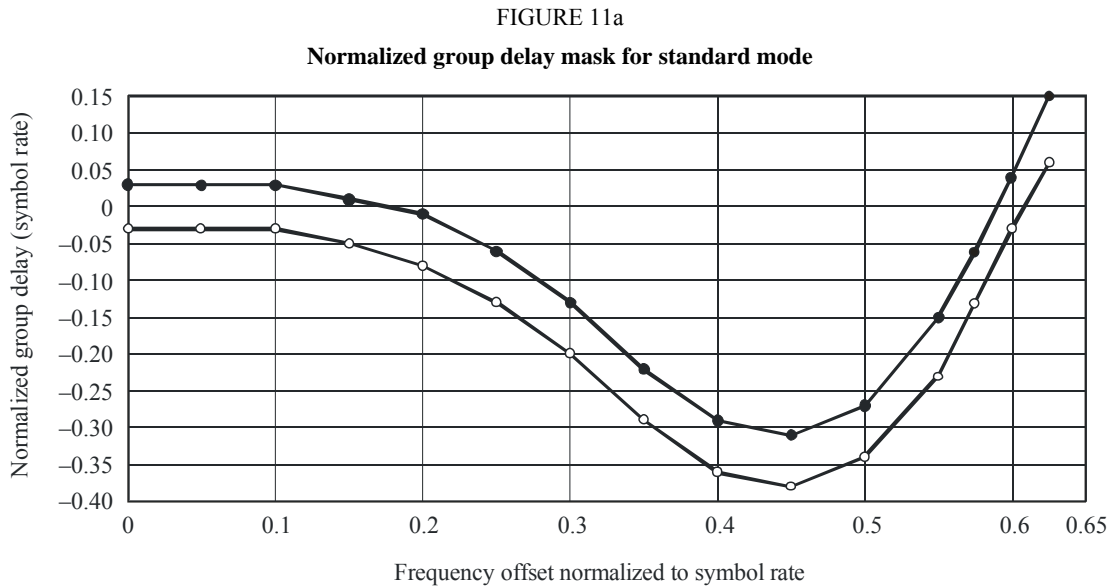
TABLE 5b

Spectral density mask breakpoints for truncated-spectrum mode

Frequency offset normalized to transmit symbol rate	Upper mask breakpoints (dB)	Lower mask breakpoints (dB)
0.00	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. Tables 6a and 6b 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 (Hz); for example, for 29.27 Msymbol/s operation the standard mode lower mask point at a frequency offset of $0.3 \times 29.27 \text{ MHz} = 8.78 \text{ MHz}$ is found from Table 6a to be $(-0.20/29.27 \times 10^6 \text{ Hz}) = -6.8 \times 10^{-9} \text{ s} = -6.8 \text{ ns}$.



BO.1516-11a

TABLE 6a

Normalized group delay breakpoints for standard mode

Frequency offset normalized to transmit symbol rate (f_{sym})	Lower mask group delay normalized to symbol rate (delay \times (f_{sym} (Hz)))	Upper mask group delay normalized to symbol rate (delay \times (f_{sym} (Hz)))
0.00	-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

TABLE 6a (end)

Frequency offset normalized to transmit symbol rate (f_{sym})	Lower mask group delay normalized to symbol rate (delay \times (f_{sym} (Hz)))	Upper mask group delay normalized to symbol rate (delay \times (f_{sym} (Hz)))
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

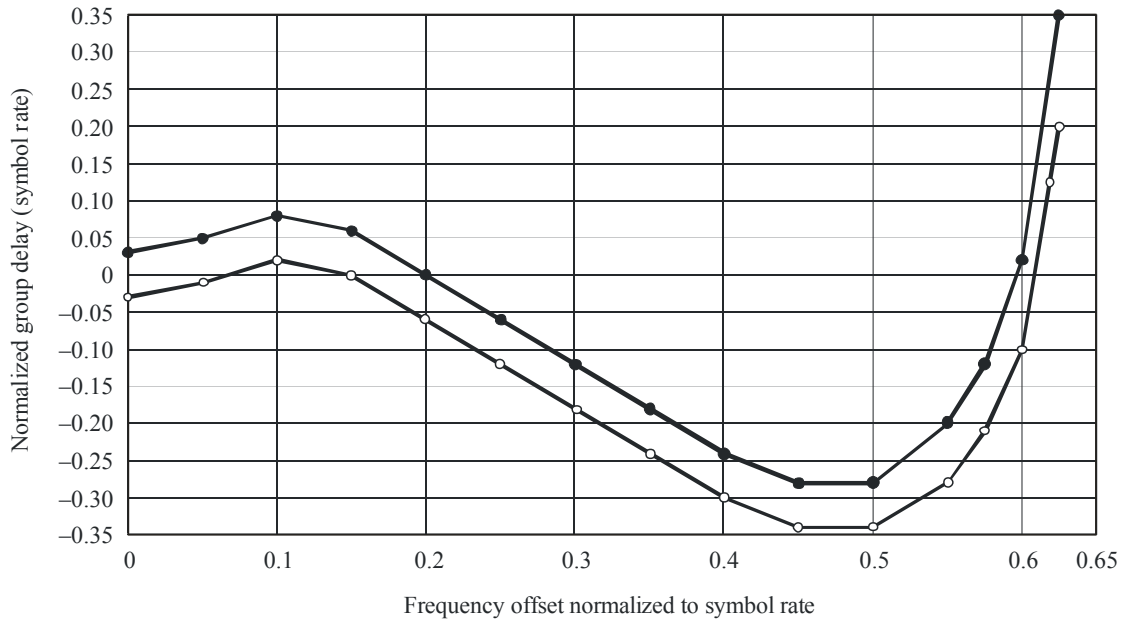


TABLE 6b

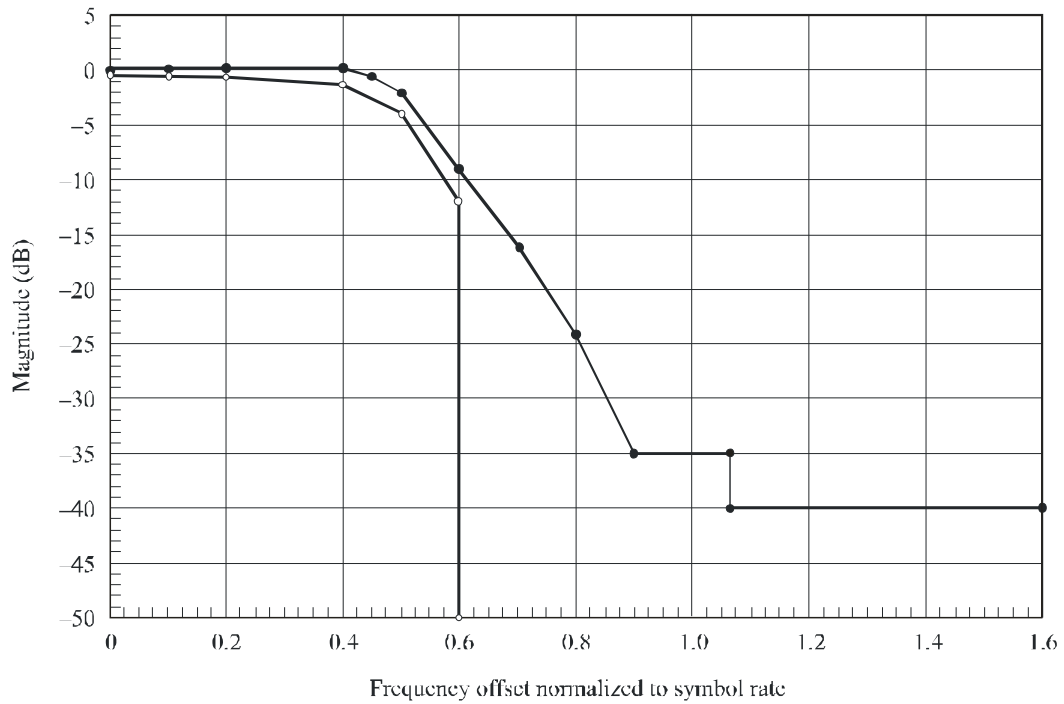
Normalized group delay breakpoints for truncated-spectrum mode

Frequency offset normalized to transmit symbol rate (f_{sym})	Lower mask group delay normalized to symbol rate (delay \times (f_{sym} (Hz)))	Upper mask group Delay normalized to symbol rate (delay \times (f_{sym} (Hz)))
0.00	-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	-0.00
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 6c.

FIGURE 11c
System C spectral mask



BO.1516-11c

TABLE 6c
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	

5.1.4 Signal spectrum for System D

Signal spectrum for System D is the same as that for System A. See § 5.1.1.

5.2 Convolutional coding

5.2.1 Convolutional coding characteristics for System A

Table 7a defines the punctured code definition for System A based on basic code 1/2:

TABLE 7a
Convolutional coding characteristics for System A

Original code			Code rates									
			1/2		2/3		3/4		5/6		7/8	
<i>K</i>	$G_1(X)$	$G_2(Y)$	<i>P</i>	d_{free}	<i>P</i>	d_{free}	<i>P</i>	d_{free}	<i>P</i>	d_{free}	<i>P</i>	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 7b defines the punctured code definition for System B.

TABLE 7b
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 _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	$X = 100101$ $Y = 111010$ $I = X_1 Y_2 X_4 X_6$ $Q = Y_1 Y_3 Y_5 Y_7$	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.

5.2.4 Convolutional coding characteristics for System D

Convolutional coding characteristics for System D are quite similar to that for System A.

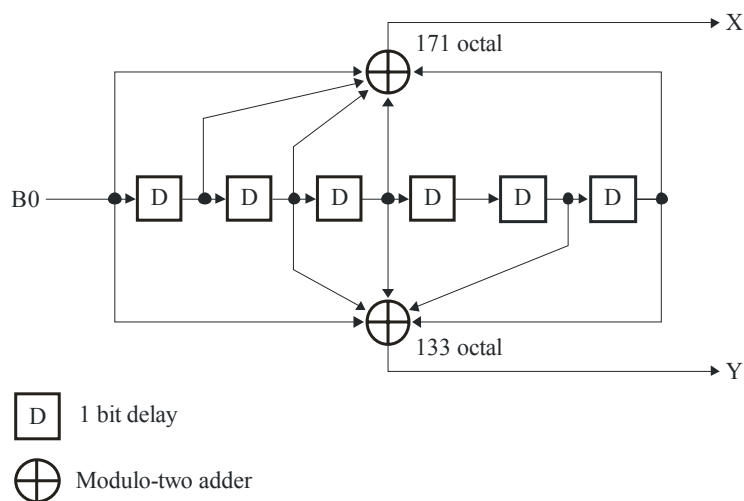
System D employs not only QPSK but also TC8-PSK and BPSK. Therefore, the characteristics for System D are expanded from that for System A.

System D allows for a variety of modulation schemes as well as a range of punctured convolutional codes based on a rate-1/2 convolutional code with a constraint length of 7. The generator polynomial is 171 octal and 133 octal (see Fig. 13). It may allow for the use of TC8-PSK, QPSK, and BPSK. When allowing these modulation schemes, the system allows for a code rate of 2/3 for TC8-PSK, code rates of 1/2, 2/3, 3/4, 5/6, and 7/8 for QPSK and 1/2 for BPSK.

Figure 12 shows the convolutional encoder while Fig. 13 shows the punctured and symbol mapping circuitry. The punctured codes are those defined in Table 8. The symbol mapping is those specified in Fig. 14. With regard to BPSK, the two encoded bits (P_0 and P_1) are transmitted in the order of P_1 and P_0 . The input bit B_1 is to be used only in the case of TC8-PSK, where B_1 and B_0 are two successive bits of a byte data (B_1 represents the higher order bit).

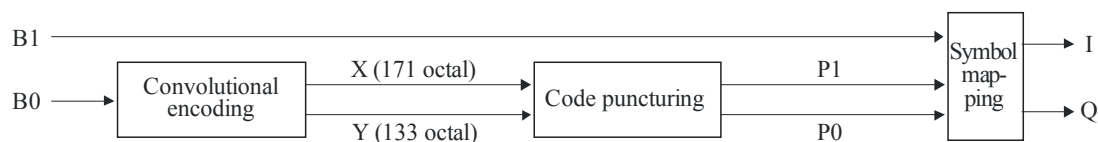
For modulations and convolutional codes other than those described above, the appropriate specifications should be applied.

FIGURE 12
Convolutional encoder



BO.1516-12

FIGURE 13
Inner coding and symbol mapping circuitry



BO.1516-13

TABLE 8
Punctured code definition

BPSK		QPSK										TC8-PSK	
1/2		1/2		2/3		3/4		5/6		7/8		2/3	
<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>	<i>P</i>	<i>d_{free}</i>
X = 1		X = 1		X = 10		X = 101		X = 10101		X = 1000101		X = 1	
Y = 1	10	Y = 1	10	Y = 11	6	Y = 110	5	Y = 11010	4	Y = 1111010	3	Y = 1	10
P1 = X ₁		P1 = X ₁		P1 = X ₁ Y ₂ Y ₃		P1 = X ₁ Y ₂		P1 = X ₁ Y ₂ Y ₄		P1 = X ₁ Y ₂ Y ₄ Y ₆		P1 = X ₁	
P0 = Y ₁		P0 = Y ₁		P0 = Y ₁ X ₃ Y ₄		P0 = Y ₁ X ₃		P0 = Y ₁ X ₃ X ₅		P0 = Y ₁ Y ₃ X ₅ X ₇		P0 = Y ₁	

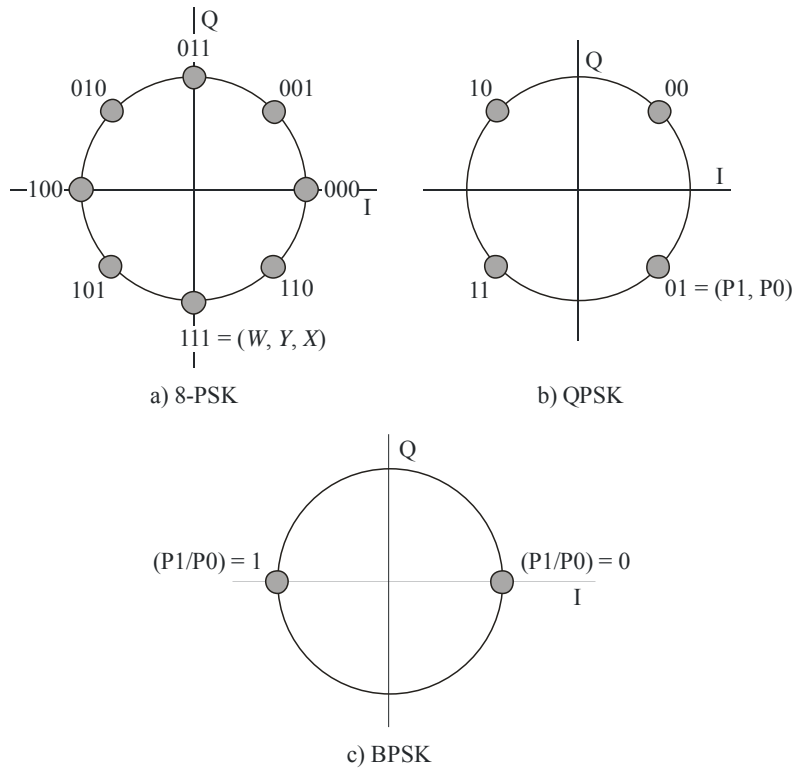
1: transmitted bit

0: non-transmitted bit

d_{free}: convolutional code free distance

NOTE 1 – The punctured code is initialized at the start of the successive slots that are assigned to the corresponding code.

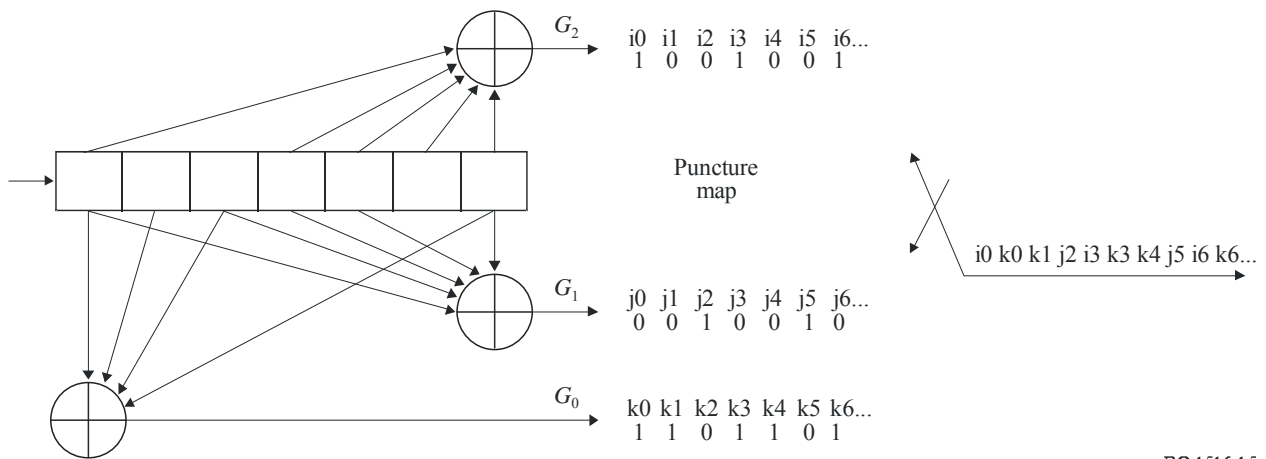
FIGURE 14
Symbol mapping



BO.1516-14

FIGURE 15

Convolutional encoder (rate 3/4 example)



BO.1516-15

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_h).

The processing in order at the transmitting side shall always start from the most significant bit (MSB) (i.e. “0”) of the sync word-byte (i.e. 01000111).

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

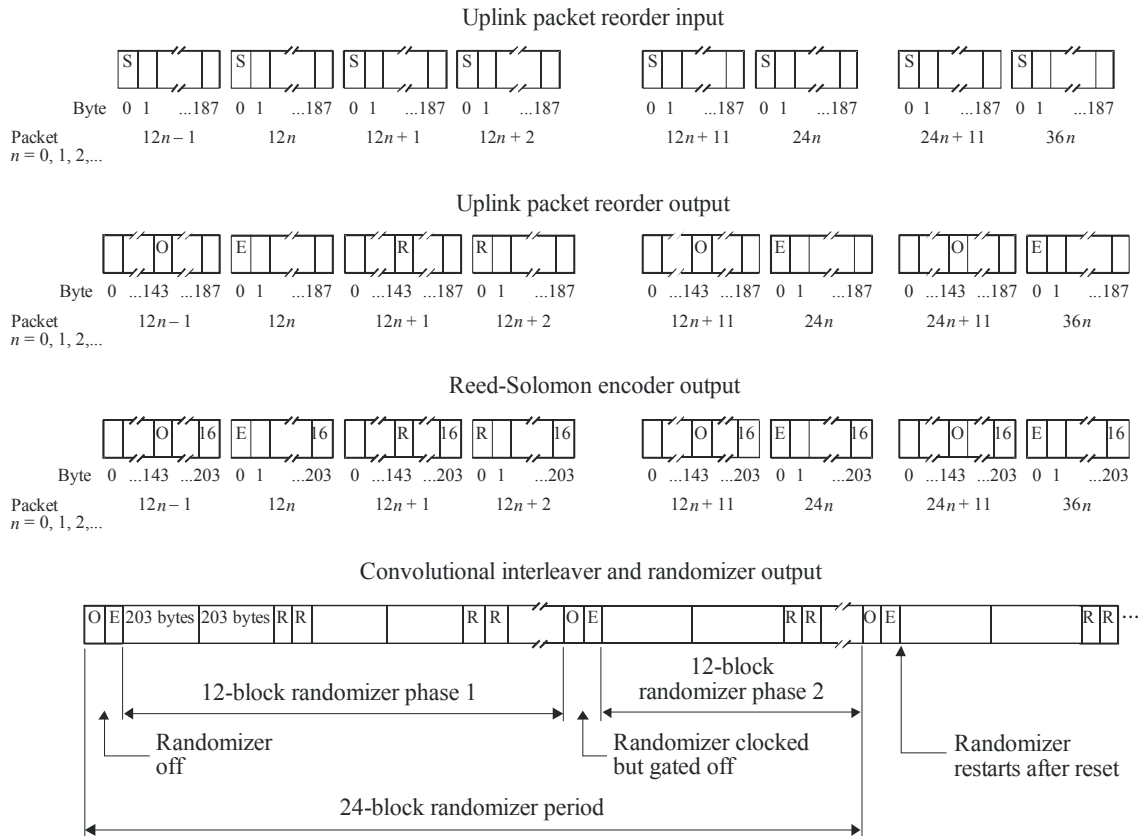
The uplink transmission processing facilitates downlink synchronization of the FEC code system by performing MPEG-2 packet reordering and 16-bit frame sync and reserved word formatting. Figure 16 shows the uplink processing required to ensure that the 16-bit frame sync pattern appears at the Viterbi 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 least significant bit (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 17 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 16

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.)

BO.1516-16

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.

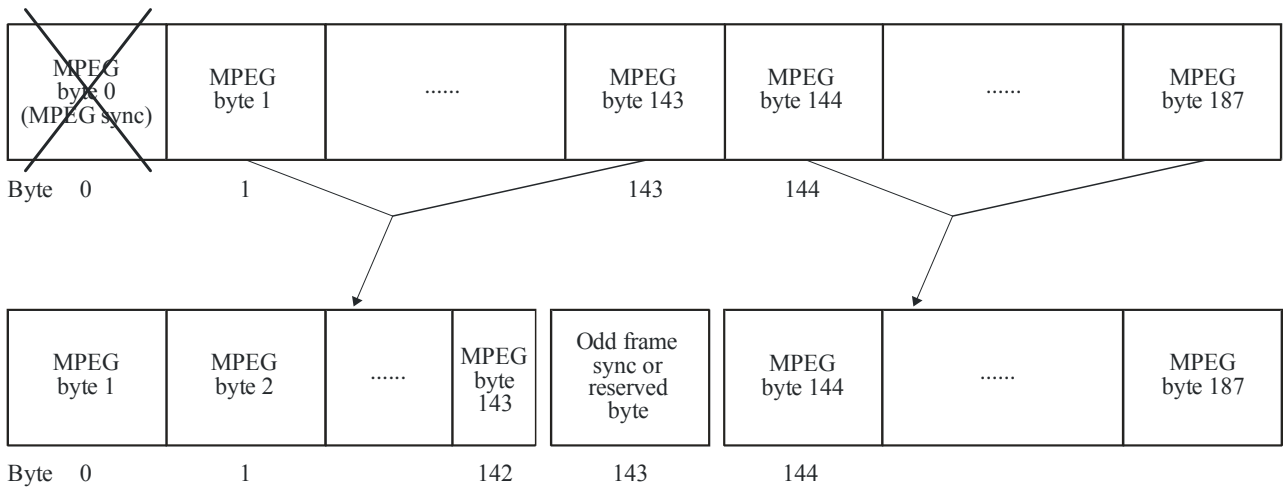
The corresponding *Q*-channel MUX sync word for split MUX modes of operation is:

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 17

Uplink packet reorder for odd numbered packets



BO.1516-17

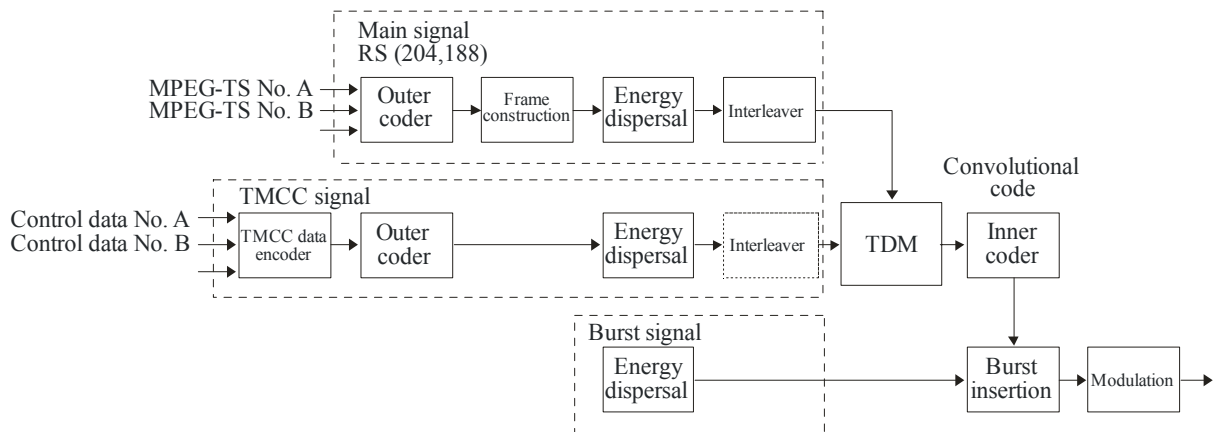
5.3.4 Synchronization characteristics for System D

A general configuration of System D is shown in Fig. 18. The system handles three kinds of signals in order to transmit multiple MPEG-TSs with various kinds of modulation schemes and in order to achieve stable and easy reception. The three signals are:

- main signal which consists of multiple MPEG-TSs and carries the programme content;
- transmission and multiplexing configuration control (TMCC) signal which informs the receiver of the modulation schemes applied, the identification of MPEG-TSs, etc.; and
- burst signal which ensures stable carrier recovery at the receiver under any reception condition (especially under low carrier-to-noise (C/N) ratio conditions).

FIGURE 18

General configuration of system



BO.1516-18

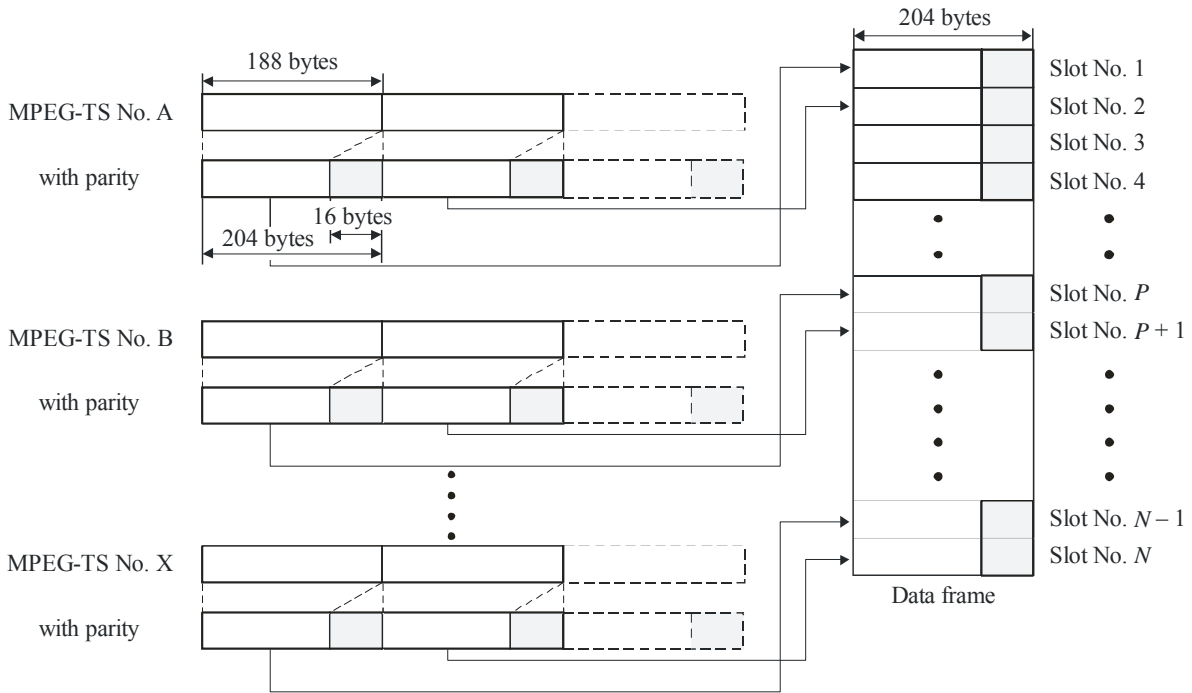
To handle multiple MPEG-TSs and to allow several modulation schemes that are used simultaneously, a frame structure is employed in System D.

To combine the MPEG-TSs, the error-protected 204-byte packets are assigned to the slots in a data frame, as shown in Fig. 19. The slot indicates the absolute position in the data frame and is used as the unit that designates the modulation scheme and MPEG-TS identification. The size of slot (the

number of bytes in a slot) is 204 bytes to keep one-to-one correspondence between slots and error-protected packets. The data frame is composed of N slots.

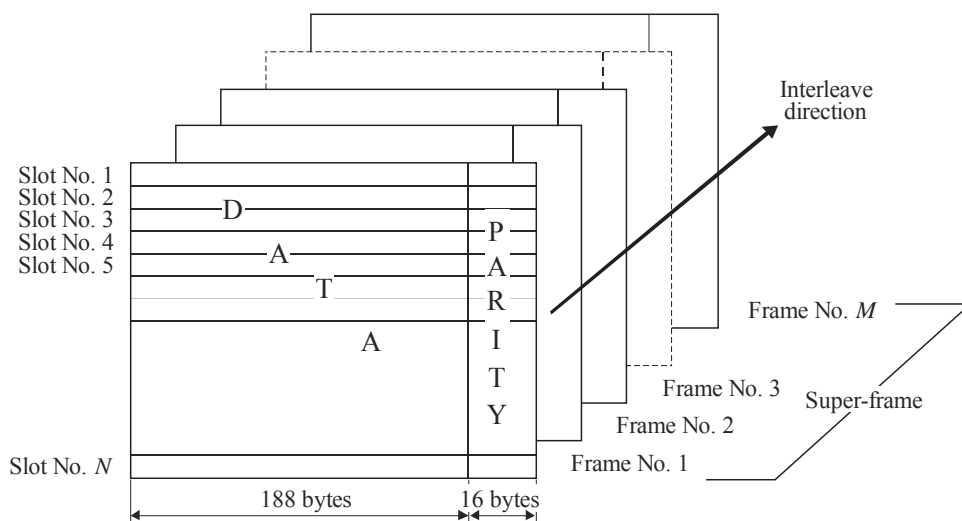
A super-frame is introduced to perform interleaving easily. Figure 20 shows the super-frame structure. The super-frame is composed of M frames, where M corresponds to the depth of interleaving.

FIGURE 19
Frame structure



BO.1516-19

FIGURE 20
Super-frame structure



BO.1516-20

As the spectrum efficiency or the transmissible bits per symbol varies with the combination of modulation and inner code rate, the number of packets being transmitted depends on the combination. Since the number of symbols to be modulated by a particular modulation scheme must

be an integer value, the relationship between the number of packets transmitted and the number of symbols for the modulation is given by equation (1).

$$I_k = \frac{8 B P_k}{E_k} \quad (1)$$

where:

I_k, P_k : integers

I_k : number of symbols transmitted with the k -th combination of the modulation scheme and inner code rate

P_k : number of packets transmitted with the k -th combination of the modulation scheme and inner code rate

E_k : spectrum efficiency of the k -th combination of the modulation scheme and inner code rate

B : number of bytes per packet (= 204).

The number of symbols per data frame, I_D , is expressed by equation (2).

$$I_D = \sum_k I_k \quad (2)$$

The number of packets transmitted during a frame duration becomes maximum when all the packets are modulated by the modulation-code combination having the highest spectrum efficiency among the possible combinations in the system. Therefore, the number of slots provided by the system is given by substituting the I_D and E_{max} for equation (1).

$$N = \frac{I_D E_{max}}{8 B} \quad (3)$$

where N denotes the number of slots that the system provides, and E_{max} denotes the highest spectrum efficiency of the modulation-coded combinations that the system provides.

When modulation-code combinations that do not have the highest spectrum efficiency are used, the number of packets being transmitted becomes lower than the number of the slots provided by the system. In this case, some of the slots shall be filled by dummy data to keep the frame size (the number of slots in a frame) constant. These slots are called “dummy slots”. The number of dummy slots S_d in a frame is obtained by the following equation (4).

$$S_d = N - \sum_k P_k \quad (4)$$

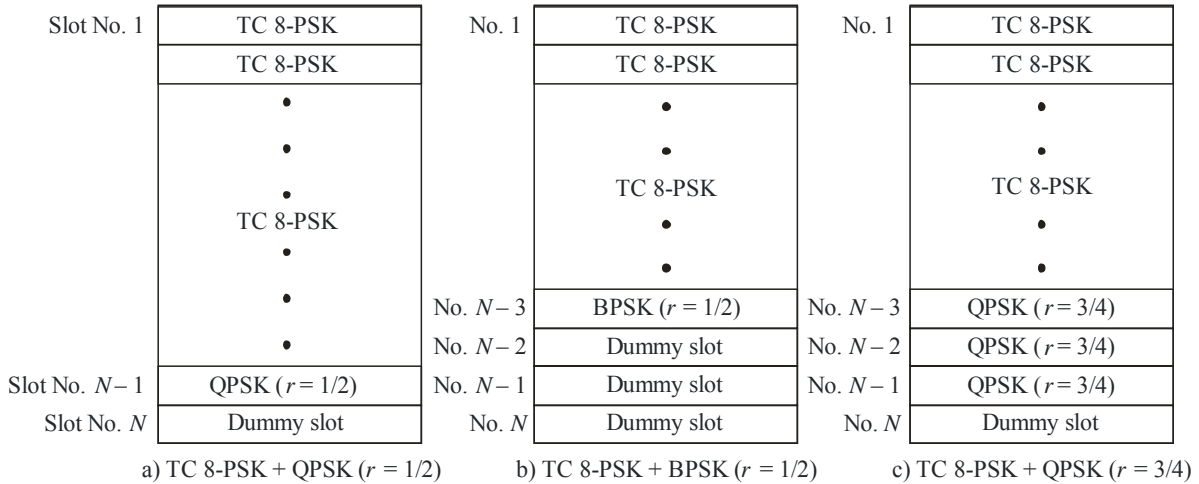
In the case where multiple modulation schemes are used simultaneously, that is, part of the slots in a frame are modulated by a particular modulation-code combination while the rest of slots are modulated by the other combinations, the data shall be modulated from the highest spectrum efficient scheme to the lowest spectrum efficient scheme among the combinations being actually used. In other words, the packets transmitted with higher efficient combinations are assigned to the lower numbered slots in a frame. This modulation order gives the minimum value in the bit error ratio (BER) after decoding the convolutional code in a low C/N reception.

Figure 21 shows some examples of slot assignment when QPSK ($r = 1/2$, r denotes code rate), BPSK ($r = 1/2$), and QPSK ($r = 3/4$) are used, respectively with trellis coded (TC) 8-PSK ($r = 2/3$). In the examples, TC 8-PSK ($r = 2/3$) is assumed as the highest spectrum efficient combination of the system. Since the spectrum efficiency of QPSK ($r = 1/2$) is half that of the TC 8-PSK, one dummy slot is inserted (Fig. 21a)); since the spectrum efficiency of BPSK ($r = 1/2$) is a quarter that of the TC 8-PSK, three dummy slots are inserted (Fig. 21b)); and since the spectrum efficiency of

QPSK ($r = 3/4$) is $3/4$ that of the TC 8-PSK, one dummy slot is inserted for three active slots (Fig. 21c)).

FIGURE 21

Example of slot assignment

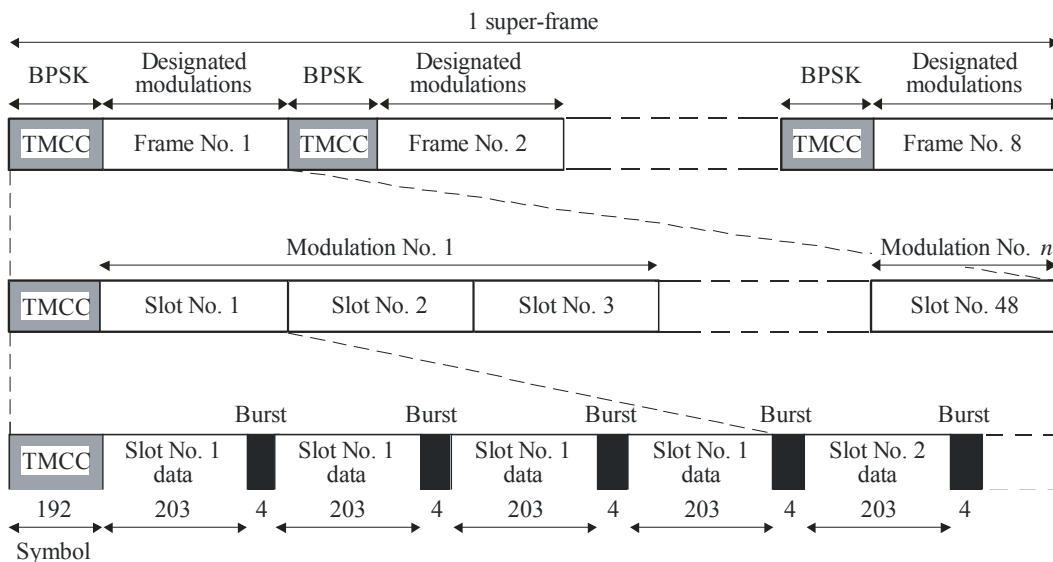


BO.1516-21

System D uses transmission and multiplexing configuration control (TMCC) signal to carry the information of the modulation schemes and the MPEG-2-TS ID, which is assigned to the slots, etc. The detailed information of TMCC is given in Appendix 2. Figure 22 illustrates outline of the transmission signal of System D.

FIGURE 22

Outline of the transmission signal



Where modulation No. 1 = TC8-PSK

BO.1516-22

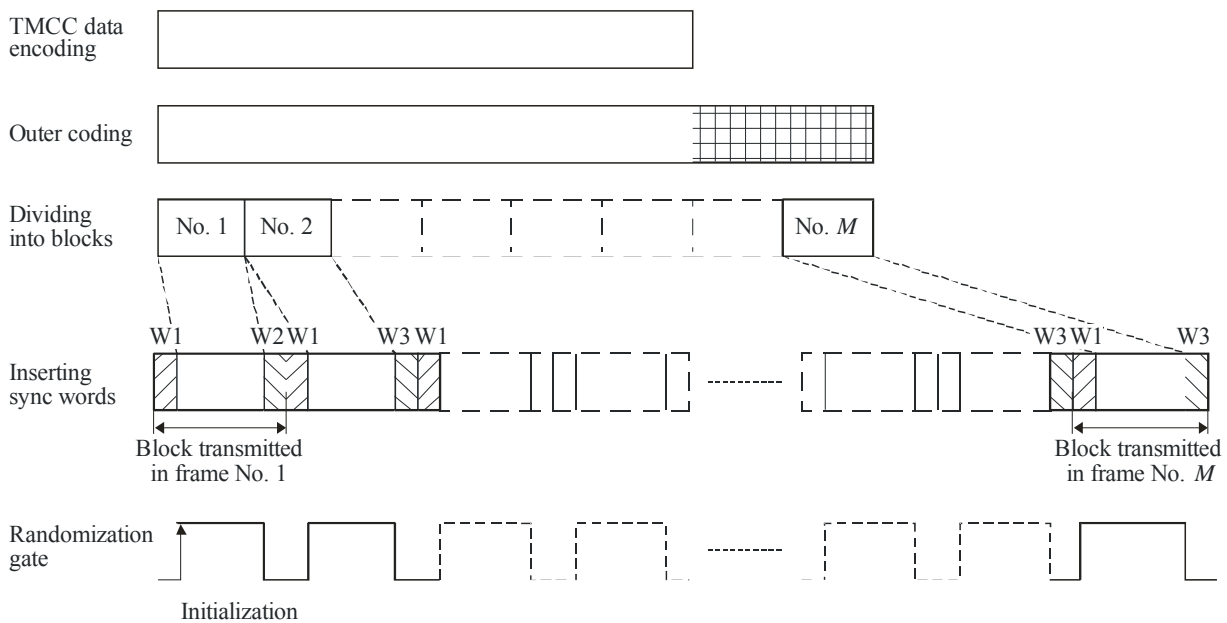
The main signal and the TMCC signal shall be time-division multiplexed at every frame. According to the modulation-code combinations designated for each slot, the time base of the multiplexed signal partially (slot basis) expands/compresses due to the convolutional coding process. By this operation, the dummy slots, if included in the main signal, shall be excluded from the transmission

signal. Figure 23 illustrates conceptual integration processes of the main, TMCC, and burst signals for forming the transmission signal.

To keep a constant interval between the successive bursts throughout a frame (see Fig. 22), a burst signal shall be inserted in every 204 symbols of the convolutional coded main signal. Note that the burst shall be inserted in every 203 symbols when the MPEG sync words are not transmitted (see § 5.4.4). The duration of the burst shall be 4 symbols. The data for burst before modulation shall be randomized with an appropriate random sequence for energy dispersal. The modulation scheme for burst signal shall be the same as that applied to the TMCC signal (the most robust scheme against transmission noise).

When carrier recovery in the receiver is carried out only from burst signals, the recovered carrier does not always lock to the right frequency. This problem (false lock of phase-locked loop (PLL)) can be solved by using the transmission signal during the TMCC duration in addition to the burst signal (when the PLL locks falsely, the number of cycles of the recovered carrier in a TMCC duration will be a different incorrect number, therefore, the PLL can be controlled by the difference in the number of cycles).

FIGURE 23
Generation of TMCC signal



BO.1516-23

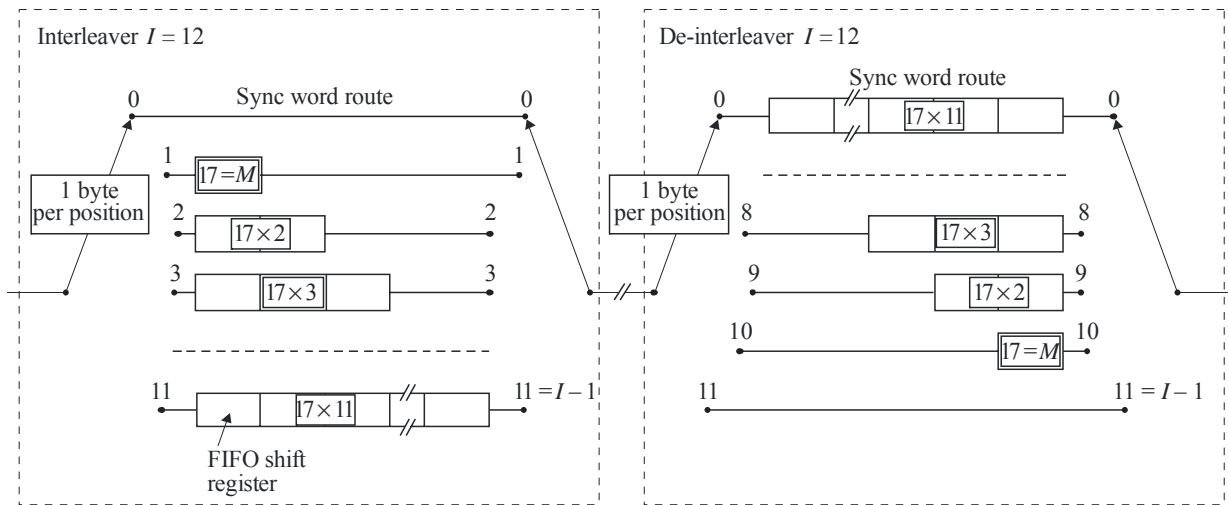
5.4 Interleaver

5.4.1 Convolutional interleaver for System A

Following the conceptual scheme of Fig. 24a, convolutional interleaving with depth $I = 12$ shall be applied to the error protected packets. This results in an interleaved frame.

FIGURE 24a

Conceptual diagram of the convolutional interleaver and de-interleaver



BO.1516-24a

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 (Mj) 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. 24b. This interleaver is a Ramsey Type II interleaver (see Note 1) with the following parameters:

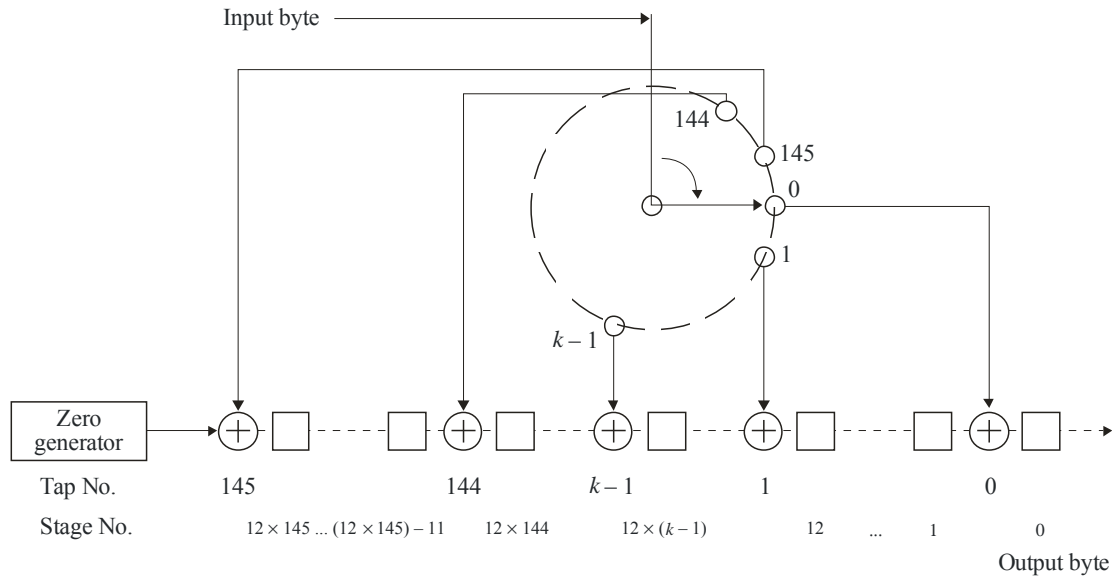
$$I = 146 \quad \text{interleaver block length and}$$

$$D = 13 \quad \text{interleaving depth.}$$

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

FIGURE 24b

Block diagram of the convolutional interleaver of System B



BO.1516-24b

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.

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. 25.
- 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.

5.4.4 Block interleaver for System D

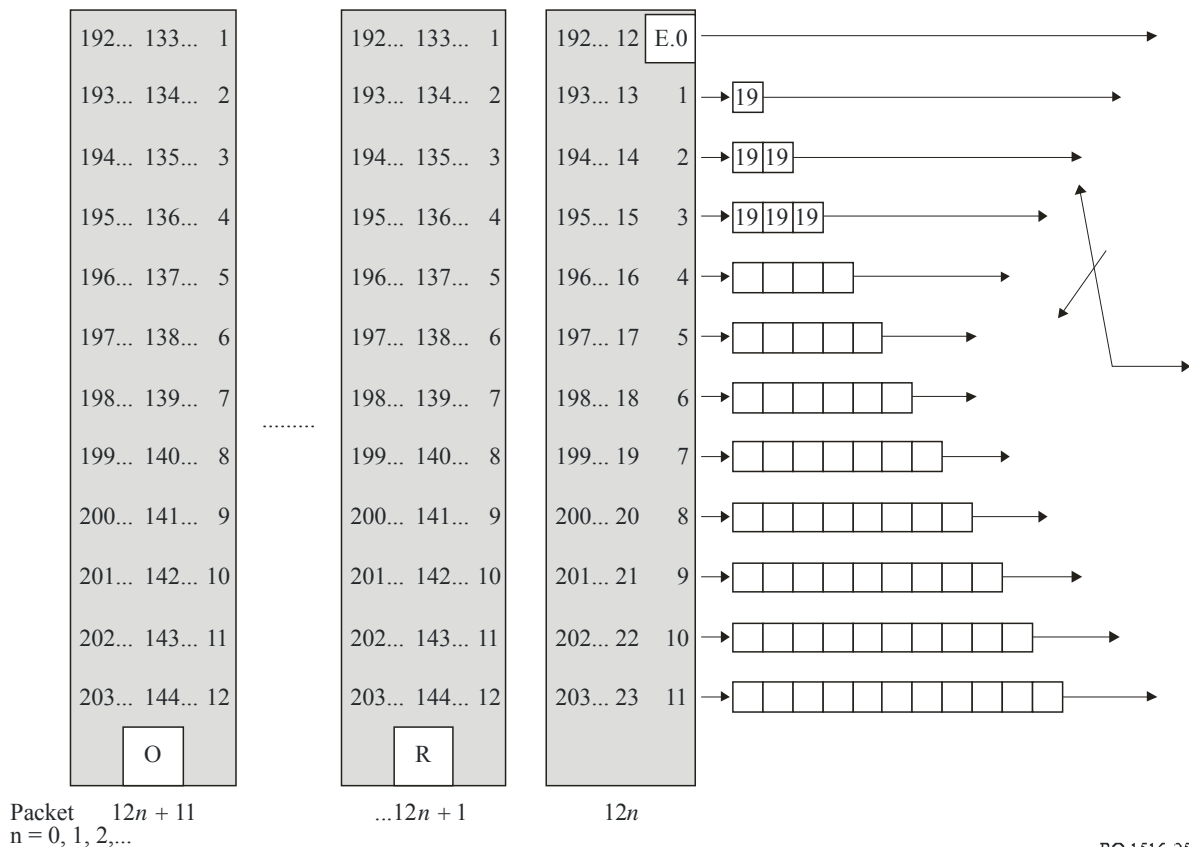
To handle multiple MPEG-TSs and to allow several modulation schemes that are used simultaneously, a frame structure is employed in System D. The framing structure is given in § 5.3.4.

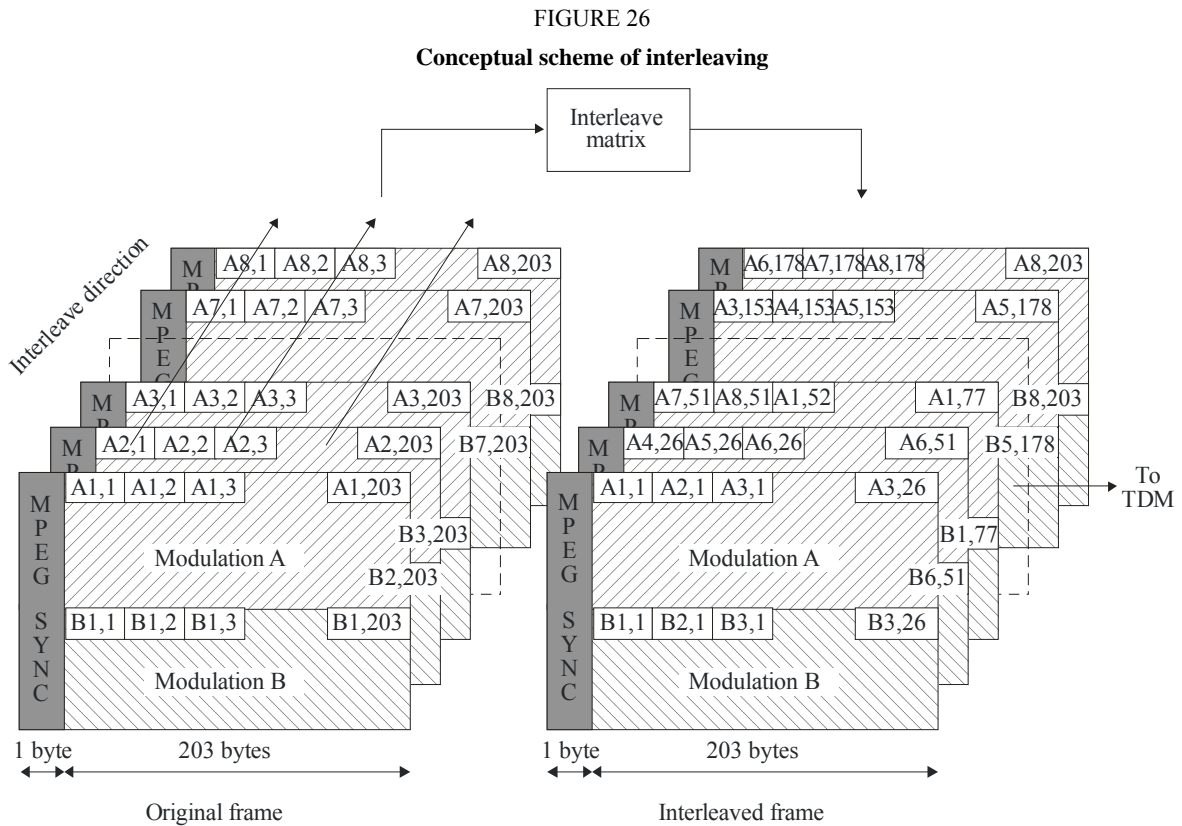
Inter-frame block interleaving with a depth of M shall be applied to the randomized data, as shown in Fig. 26. Slot assignment for every frame shall be identical throughout a super-frame, resulting in the data being interleaved only between those transmitted with the same modulation-code combination. Interleaving shall be applied except to the first byte (MPEG sync byte) of every slot.

Figure 26 illustrates an example of interleaving when the depth of interleaving is 8 (i.e. super-frame consists of 8 frames) and two kinds of modulation-code combinations are being used. The data in the original frame are read out in the inter-frame direction, i.e. in the order of $A_{1,1}, A_{2,1}, A_{3,1}, \dots$, where $A_{i,j}$ represents the byte data at j -th slot in i -th frame, to form the interleaved frame. The data in the interleaved frame are read out in the byte direction (horizontally) and fed to the TDM multiplexer.

FIGURE 25

Convolutional interleaver





It is not necessary to transmit the first byte of each packet (the MPEG sync word of 47_h) because the timing references (frame sync words) are sent by the TMCC signal. The omitted MPEG sync words have to be recovered at the receiver to perform outer decoding properly.

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, $T = 8$)

5.5.4 Reed-Solomon encoder characteristics for System D

System D 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 $p(x) = x^8 + x^4 + x^3 + x^2 + 1$.

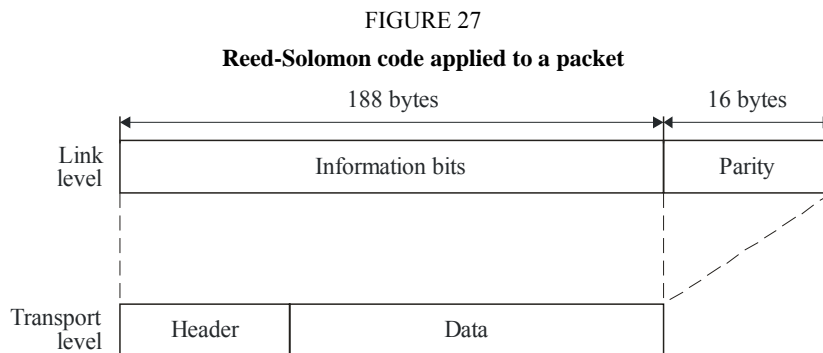
The generator polynomial for the t -error correcting code has roots at $x = a^i$, $i = 1, 2, \dots, 2t$,

$$g(x) = \prod_{i=1}^{i=2t} (x + a^i).$$

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 $d(x) = d_{N-2t-1} x^{N-2t-1} + d_{N-2t-2} x^{N-2t-2} + \dots + d_1 x + d_0$. 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. 27.



BO.1516-27

5.6 Energy dispersal

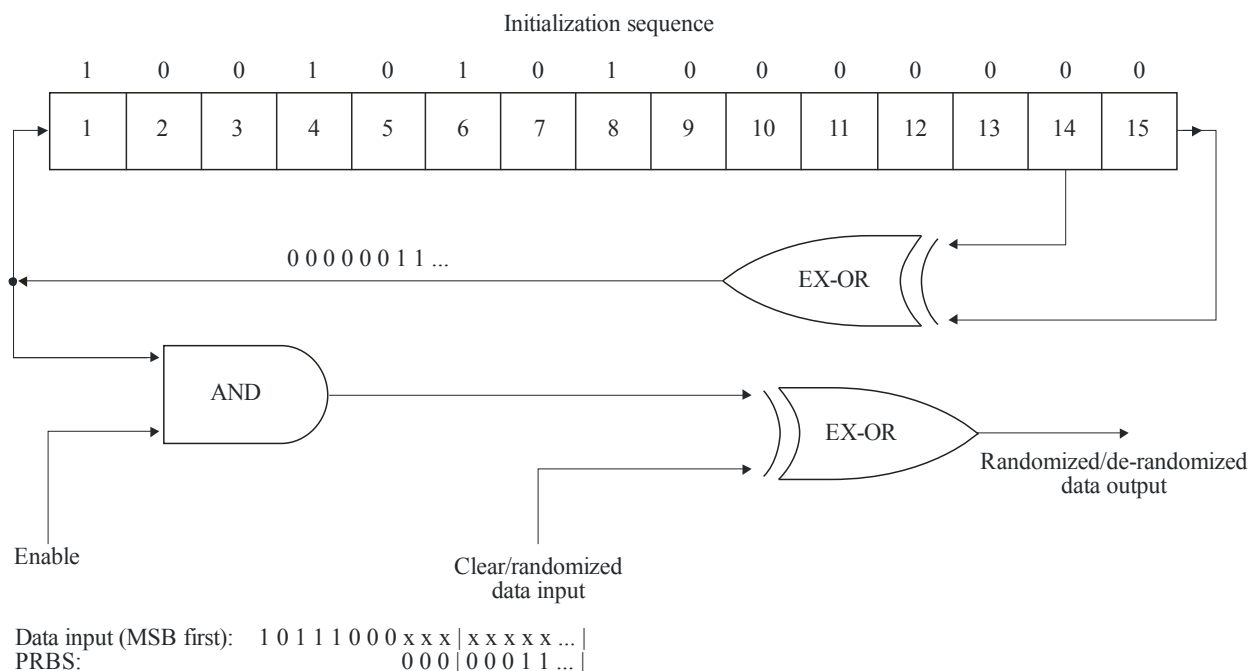
5.6.1 Energy dispersal for System A

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 “100101010000000”.

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. 28.

FIGURE 28

Randomizer/de-randomizer schematic diagram



BO.1516-23

The polynomial for the PRBS generator shall be:

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

Loading of the sequence “100101010000000” into the PRBS registers, as indicated in Fig. 28, 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_h to B8_h. 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_h). 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_h”.

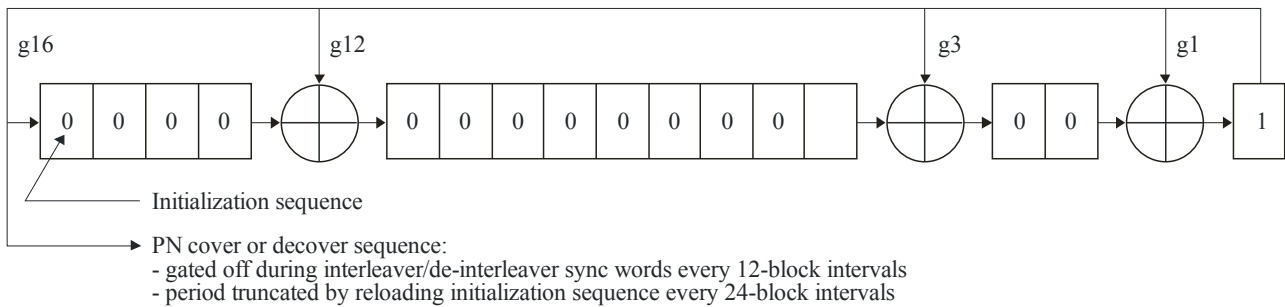
The coding layer uses data randomization (scrambling) at the interleaver output and de-interleaver input for energy dispersal and to ensure a high data transition density for bit timing recovery purposes. The following characteristics define the data randomization:

- 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. 29.

- 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. 29. 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-block intervals.

FIGURE 29

Randomizer block diagram



BO.1516-29

5.6.4 Energy dispersal for System D

In order to comply with ITU Radio Regulations and to ensure adequate binary transitions, the data of the frame shall be randomized in accordance with the configuration depicted in Fig. 30.

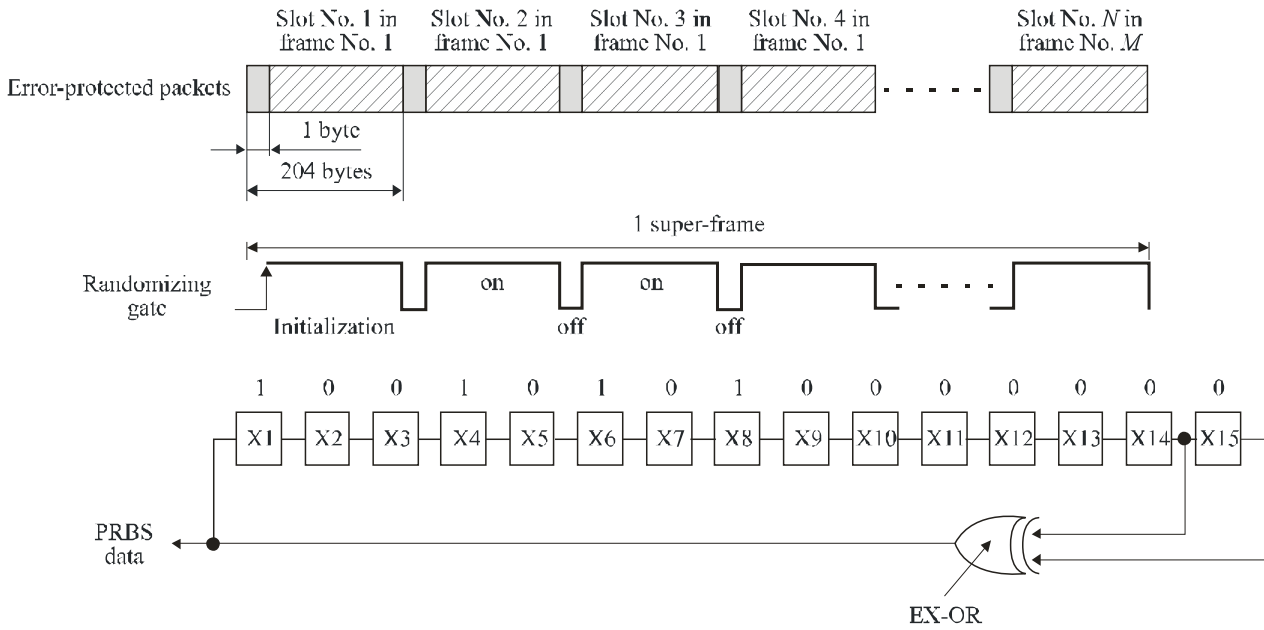
The polynomial for the PRBS generator is:

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

Loading of the sequence “100101010000000” into the PRBS registers as indicated in Fig. 30, is initiated at the second byte of every super-frame. The first bit of the output of the PRBS generator is applied to the first bit (i.e. MSB) of the second byte of slot No. 1 in frame No. 1. The PRBS is added to the data except to the first byte (MPEG sync byte) of every slot.

FIGURE 30

Randomizer schematic diagram



BO.1516-30

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. 31a)).

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).

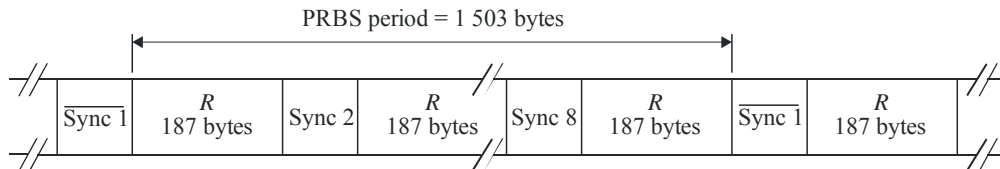
5.7.4 Framing and transport stream characteristics for System D

See synchronization characteristics (§ 5.3.4).

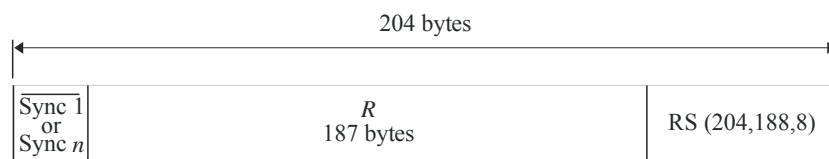
FIGURE 31
Framing structure



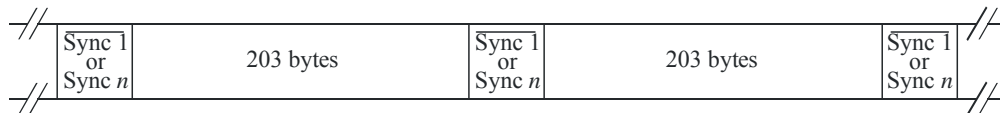
a) MPEG-2 transport MUX packet



b) Randomized transport packets: sync bytes and randomized sequence R



c) Reed-Solomon (RS) (204,188, $T = 8$) error protected packet



d) Interleaved frames; interleaving depth $F = 12$ bytes

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

BO.1516-31

5.8 Control signals

5.8.1 Control signals for System A

None.

5.8.2 Control signals for System B

None.

5.8.3 Control signals for System C

None.

5.8.4 Control signals for System D

See Appendix 2.

6 References

- [1] ISO/IEC: 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.

- [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.

7 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	Programme 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
TC8-PSK	Trellis-coded eight phase shift keying
TMCC	Transmission and multiplexing configuration control

Appendix 1

to Annex 1

System B transport stream characteristics*

CONTENTS

- 1 Introduction
- 2 Prefix
- 3 Null and ranging packets
- 4 Video application packets
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- 5 Audio application packets
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- 6 Programme guide packets
- 7 Transport multiplex constraints
 - 7.1 Elementary stream multiplex constraint definition

1 Introduction

This Appendix 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 stream characteristics of Systems A and C are provided in Reference [1], § 6 of Annex 1.

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.

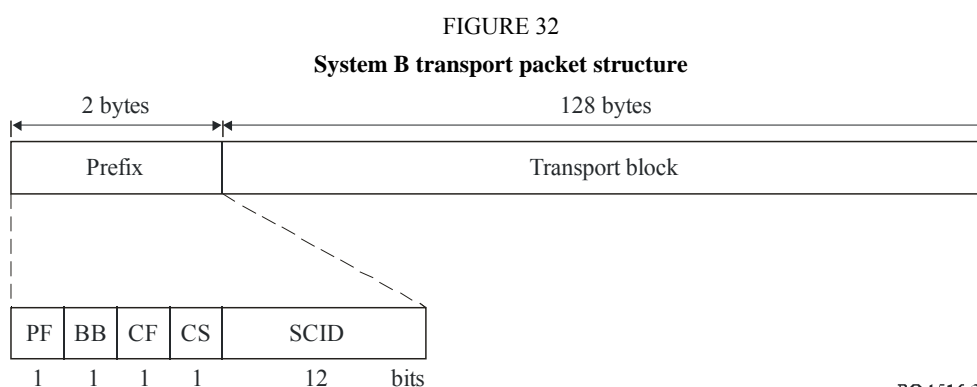
Provided within this protocol are mechanisms to facilitate rapid decoder recovery after detecting the loss of one or more cells on the channel. By identifying specific information and redundantly transmitting key MPEG data, the decoder can control the region of the image affected by errors.

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. Programme guide related packets are described in § 6. This Appendix concludes with § 7, a 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 32 illustrates the logical structure of a transport cell in which the prefix and its relationship to the transport block are identified.



The Semantic definition of the fields in prefix is given below in Table 9:

TABLE 9
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 = 0xFFF – 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 = 0x000

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 10. (The content is designed to be spectrally neutral in order to maintain tuning lock.)

TABLE 10

Null and ranging packet transport block

Byte No.	Value	Byte No.	Value	Byte No.	Value	Byte No.	Value
1 ⁽¹⁾	4 ⁽¹⁾	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

TABLE 10 (end)

Byte No.	Value	Byte No.	Value	Byte No.	Value	Byte No.	Value
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

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

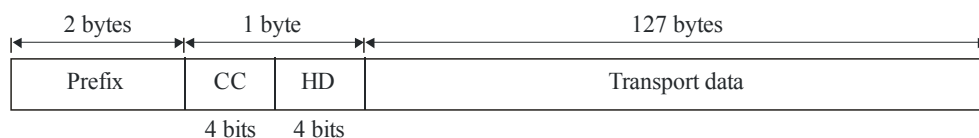
4 Video application packets

The general structure of the video transport packets is illustrated in Fig. 33. 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).

FIGURE 33

General video application packet structure



BO.1516-33

To indicate different cell types and associated counters, the video transport layer format has 4 bits for a CC and 4 bits for a HD, as shown in Fig. 33. A detailed description of these fields is given in Table 11. 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.

TABLE 11

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 _b), the CC wraps around to 0. The CC is set to 0 (0000 _b) and shall not be incremented when the HD field contains "0x00" (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 _b Auxiliary data packets 01x0 _b Basic video service packets 10x0 _b Redundant data packets 11x0 _b 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_b.

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

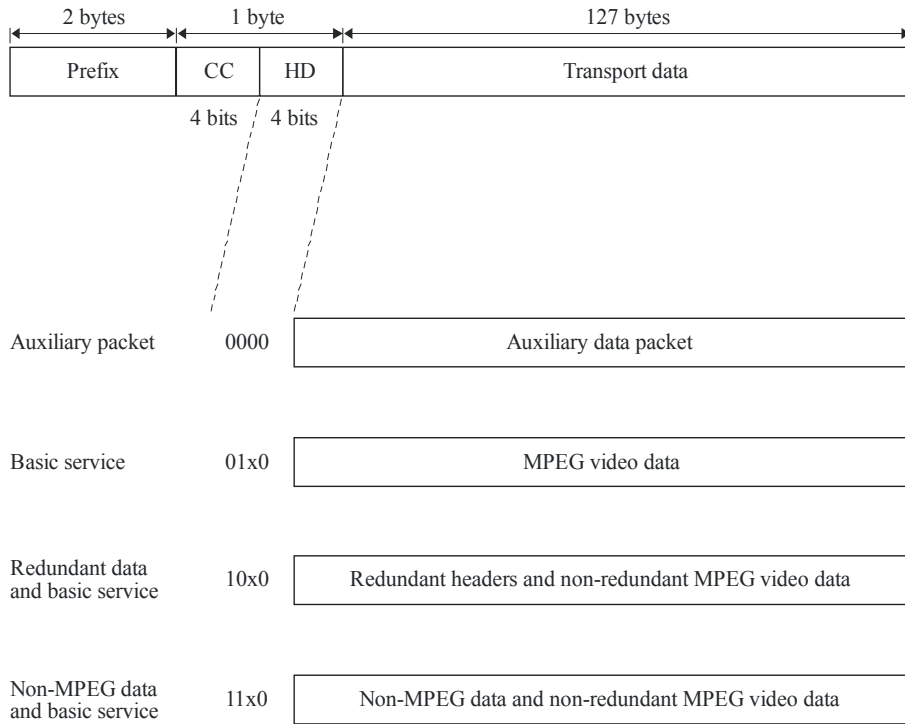
The ADG may contain:

- 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.

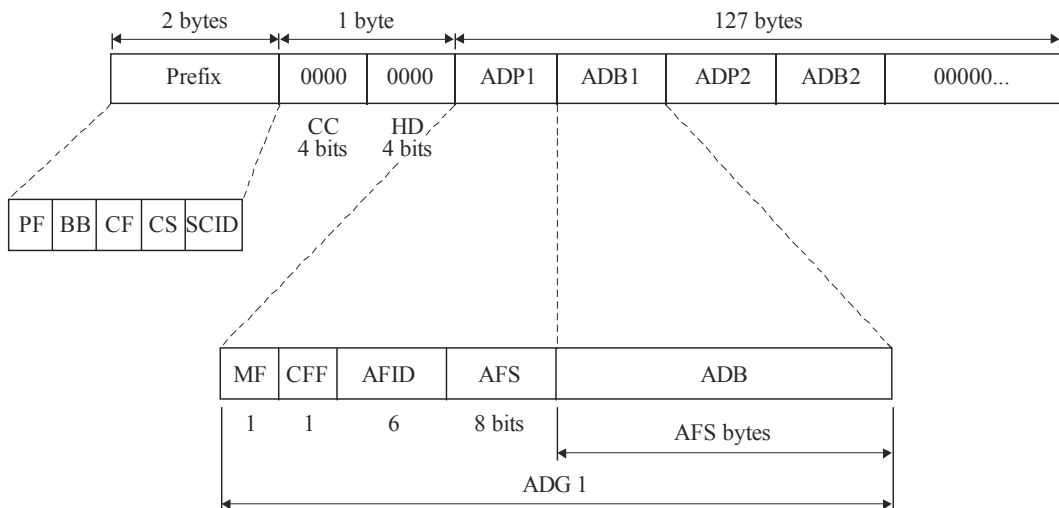
An example of auxiliary data packet structure with two ADG fields is illustrated in Fig. 35. The semantic definition of the (relevant) fields in the auxiliary data packet are given in Table 12.

FIGURE 34
Video application packet structures



BO.1516-34

FIGURE 35
Auxiliary data packet structure



BO.1516-35

TABLE 12

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 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 _b for Aux packets
HD	Header designator	HD = 0000 _b 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 _b Reference time stamp only 000001 _b Encryption control word packet (CWP) only 000011 _b Reference time stamp and CWP 000010 _b , and 000100 _b to 111111 _b : 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_b

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_b

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_b

AFS = 125 (0x7D)

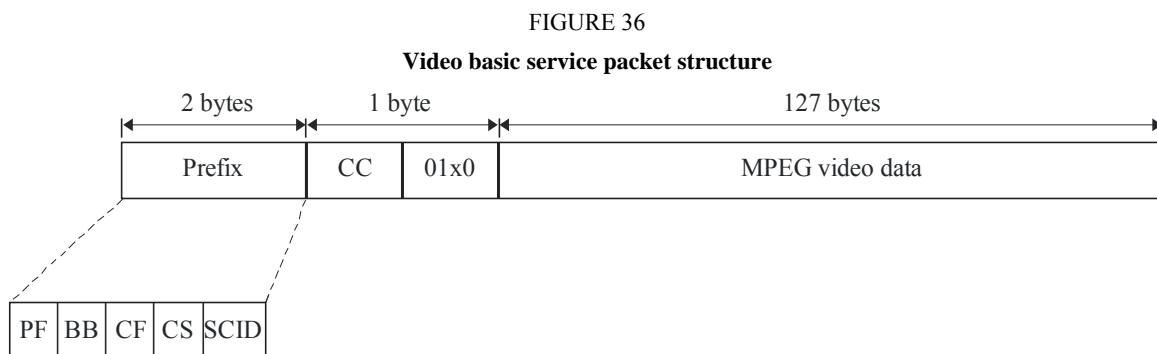
ADB = 5 byte time stamp followed by 120 bytes of CWP

NOTE 1 – For multi-service programmes, 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 programme. This is possible because:

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

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. 36. The semantic definition of the (relevant) fields in basic video service packet structure is given in Table 13.



BO.1516-36

TABLE 13

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

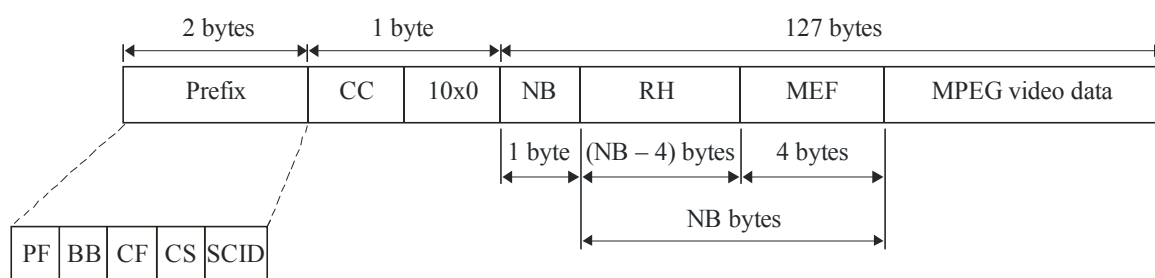
BB	Bundle boundary	BB bit is set to 1 in first basic video packet containing a redundant video sequence header, and 0 in all other packets. The decoder should ignore this bit
CF	Control flag	CF = 1: The transport block of this packet is not scrambled CF = 0: The transport block of this packet is scrambled
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 _b for basic video service packets The HD(1) bit, indicated by x in HD = 01x0 _b , 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 group of pictures (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. 37. The semantic definition of the (relevant) fields in the redundant data packet is given in Table 14.

FIGURE 37
Redundant data packet structure



BO.1516-37

TABLE 14

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 _b for redundant data packets The HD(1) bit, indicated by x in HD = 10x0 _b , reflects the toggle state of the HD of the last basic video service packet (x value in HD = 01x0 _b) 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. $5 \leq NB \leq 126$

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 back-up to operational mode.

The structure of a non-MPEG data packet is illustrated in Fig. 38. The semantic definition of the (relevant) fields in the non-MPEG video data packet is given in Table 15.

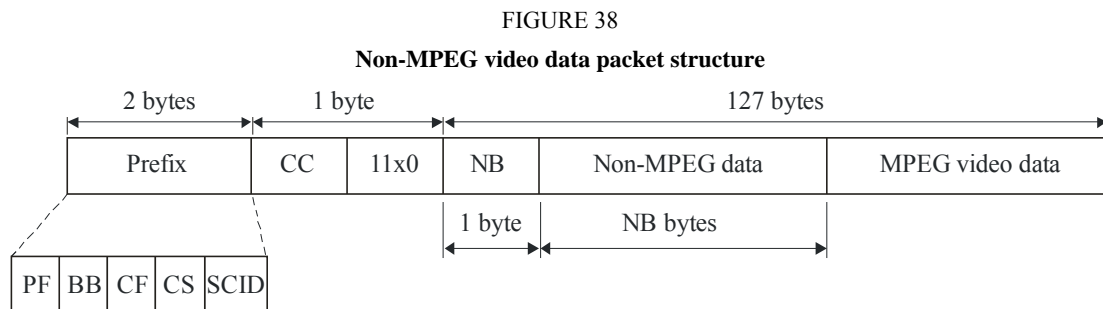


TABLE 15

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 _b for non-MPEG video data packets The HD(1) bit, indicated by x in HD = 11x0 _b , reflects the toggle state of the HD of the last basic video service packet (x value in HD = 01x0 _b) 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. $5 \leq \text{NB} \leq 126$
	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. 39. 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 16. 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 39

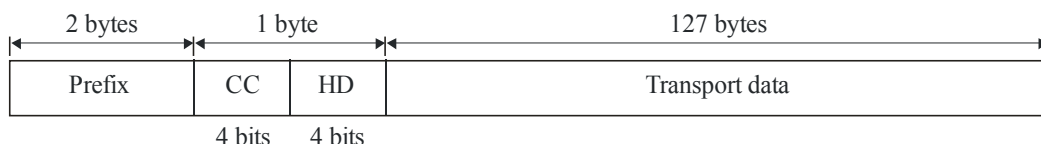
General audio application packet structure

TABLE 16

The semantic definition of elements in the CC HD byte

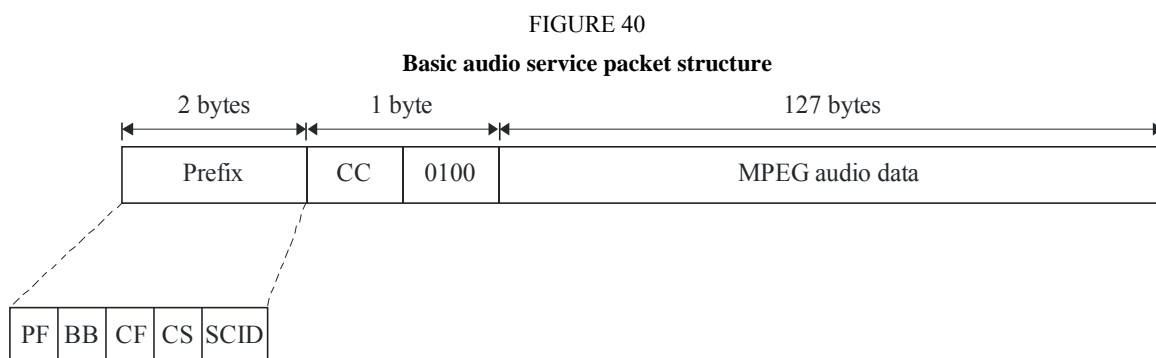
CC	Continuity counter	This 4 bit field (unsigned integer, MSB first) is incremented by one with each packet with the same SCID. After it reaches the maximum value of 15 (1111 _b), the continuity counter wraps around to 0. The continuity counter is set to 0 (0000 _b) and shall not be incremented when the HD field is equal to "0x00" (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 _b Auxiliary data packets 0100 _b Basic audio service packets 1100 _b 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_b carry basic audio service (i.e. MPEG audio bits) information. The structure of the basic audio service packet is illustrated in Fig. 40 and the semantic definition of the (relevant) fields is given in Table 17.



BO.1516-40

TABLE 17

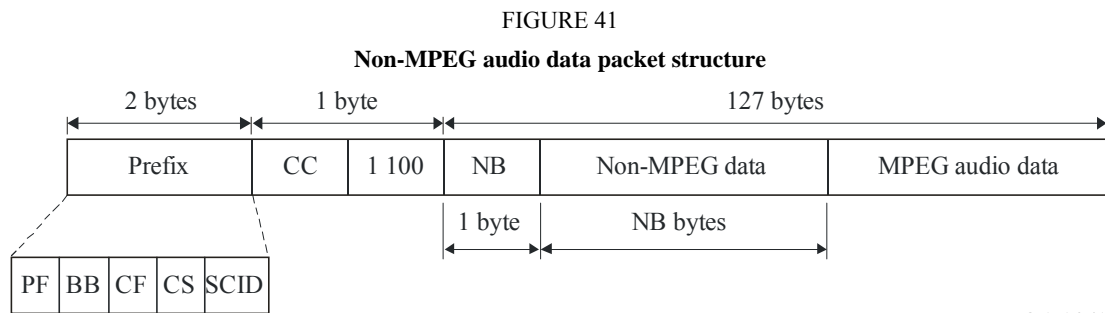
The semantic definition of the (relevant) fields in the basic audio service packet

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 _b 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 back-up to operational mode.

The structure of a non-MPEG audio data packet is illustrated in Fig. 41 and the semantic definition of the (relevant) fields is given in Table 18.



BO.1516-41

TABLE 18

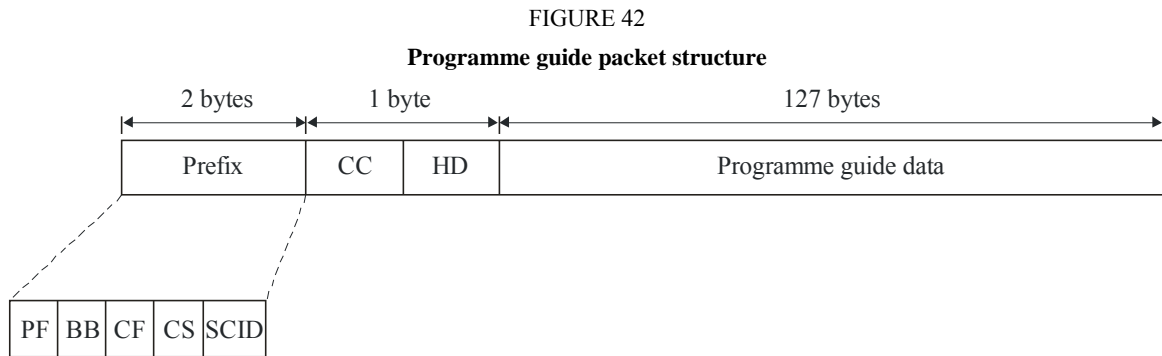
The semantic definition of the (relevant) fields in the non-MPEG audio data packet

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 _b 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 Programme guide packets

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

Master programme guide (MPG), special programme 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. 42. The CF bit in the prefix field is set to 1 for all these streams (i.e. not scrambled). The SCID of the master programme guide packets is always a fixed value that is predefined by the user.



BO.1516-42

TABLE 19

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

BB	Bundle boundary	BB = 0 for programme guide packets
CF	Control flag	CF = 1 for programme guide packets (not scrambled)
SCID	Service channel ID	SCID: this is a fixed value predefined by the user to identify master programme guide data; format is a 12-bit field (unsigned integer, MSB first). Typical value is 0x001
HD	Header designator	HD = 0100 _b for programme 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, CA, MPG, SPG, DIP, 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 bytes 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.

Appendix 2

to Annex 1

Control signal for System D

CONTENTS

- 1 Introduction
- 2 TMCC information encoding
 - 2.1 Order of change
 - 2.2 Modulation-code combination information
 - 2.3 TS identification
 - 2.4 Other information
- 3 Outer coding for TMCC information
- 4 Timing references
- 5 Channel coding for TMCC

1 Introduction

This Appendix defines the control signal of System D. System D uses TMCC signal for an appropriate demodulation/decoding at the receiver. TMCC signal carries the following information:

- modulation-code combination for each slot;
- MPEG-2 TS identification for each slot; and
- others (e.g. order of change, flag bit for emergency alert broadcasting).

TMCC information is transmitted in advance to the main signal because the main signal cannot be demodulated without the TMCC information. The minimum interval for TMCC information renewal is a duration of one super-frame. The receivers principally decode the TMCC information at every super-frame. The TMCC signal conveys timing references in addition to the information above.

2 TMCC information encoding

The information carried by the TMCC signal is formatted as shown in Fig. 43. Details for each item are described below.

FIGURE 43
TMCC information format

Order of change	Modulation code combination for each slot	Relative TS ID for each slot	Corresponding table between relative TS ID and MPEG-2 TS_ID	Other information
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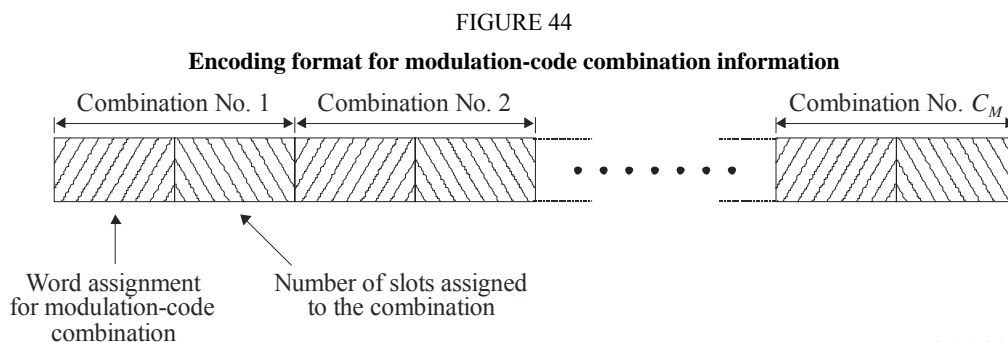
BO.1516-43

2.1 Order of change

The “order of change” is a 5-bit number that indicates renewal of the TMCC information. It is incremented each time the TMCC is renewed. The receiver may detect just the bits and may decode the TMCC information only when the bits change. The use of order of change is optionally defined by the system.

2.2 Modulation-code combination information

This represents combinations of the modulation scheme and the convolutional code rate for each slot. To reduce the transmission bits for this information, the information is encoded into the format shown in Fig. 44. The maximum number of modulation-code combinations, C_M , that are used simultaneously is defined by the system taking into account the service requirements. The word assignment for the modulation-code combination is that defined in Table 20. When the number of modulation-code combinations being used is less than the maximum number specified by the system, the word “1111” is applied to the rest of the combinations and the number of slots assigned is set to zero.



BO.1516-44

TABLE 20

Word assignment for modulation-code combination

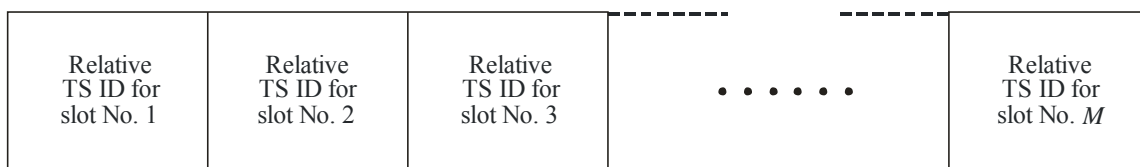
Word	Modulation-code combination
0000	Reserved
0001	BPSK($r = 1/2$)
0010	QPSK($r = 1/2$)
0011	QPSK($r = 2/3$)
0100	QPSK($r = 3/4$)
0101	QPSK($r = 5/6$)
0110	QPSK($r = 7/8$)
0111	TC8-PSK($r = 2/3$)
1000-1110	Reserved
1111	Dummy

2.3 TS identification

Instead of transmitting MPEG-2 TS_ID (16 bits) for each slot, a combination of “relative TS IDs” that identify only the TSs being transmitted and the corresponding table between these two kinds of IDs are employed. This results in reduced transmission bits. The relative TS IDs for each slot are transmitted sequentially from slot No. 1. The maximum number of TSs transmitted simultaneously, T_M , is defined by the system.

FIGURE 45

Data arrangement of relative TS ID information

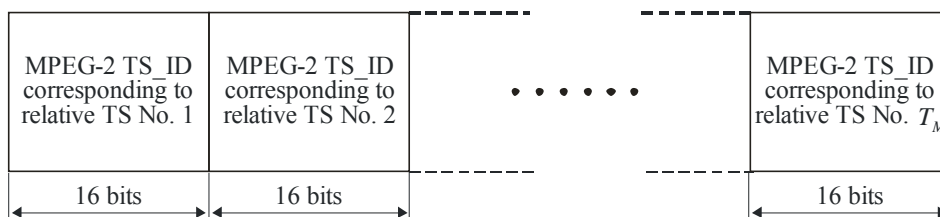


BO.1516-45

The corresponding table is composed of an array of numbers that are 16-bit numbers to represent each MPEG-2 TS_ID. The numbers are arranged from the relative TS ID number 0 to T_M .

FIGURE 46

Data arrangement of correspondance table



BO.1516-46

2.4 Other information

The encoding format for the other information is defined appropriately by the system.

3 Outer coding for TMCC information

Since TMCC information is indispensable for the demodulation at receivers, the TMCC signal should be protected with an FEC level higher than the FEC used for the main signal. For the same reason, it shall be transmitted with the modulation-code combination having the most robustness against transmission noise.

4 Timing references

Two kinds of timing references are contained, i.e. the frame sync word that indicates the start of each frame and the frame identification words that identify the first frame (frame No. 1). These words shall be transmitted by each frame.

After dividing the outer coded TMCC data into M blocks (where M is the number of frames in a super-frame), the sync words shall be inserted in each block, as shown in Fig. 47. The sync word W1 shall be inserted at the beginning of each block. The word W2 shall be inserted at the end of the block that is transmitted in the first frame, while the word W3 shall be inserted at the end of the remaining blocks. The words W1, W2, and W3 shall consist of 2 bytes. W1 shall be 1B95_h, W2 shall be A340_h, and W3 shall be 5CBF_h (W3 is obtained by inverting the bits of W2).

Note that the first 6 bits of the words will be changed by the payload information (contents of the main signal and/or TMCC signal) due to convolutional coding (constraint length of 7), which is applied to the TMCC signal at the succeeding process stage. In other words, the first 6 bits of the word are used as the termination bits of the convolutional code. Consequently, the unique bit pattern in the synchronizing word is 10 bits out of 16 bits of the original word.

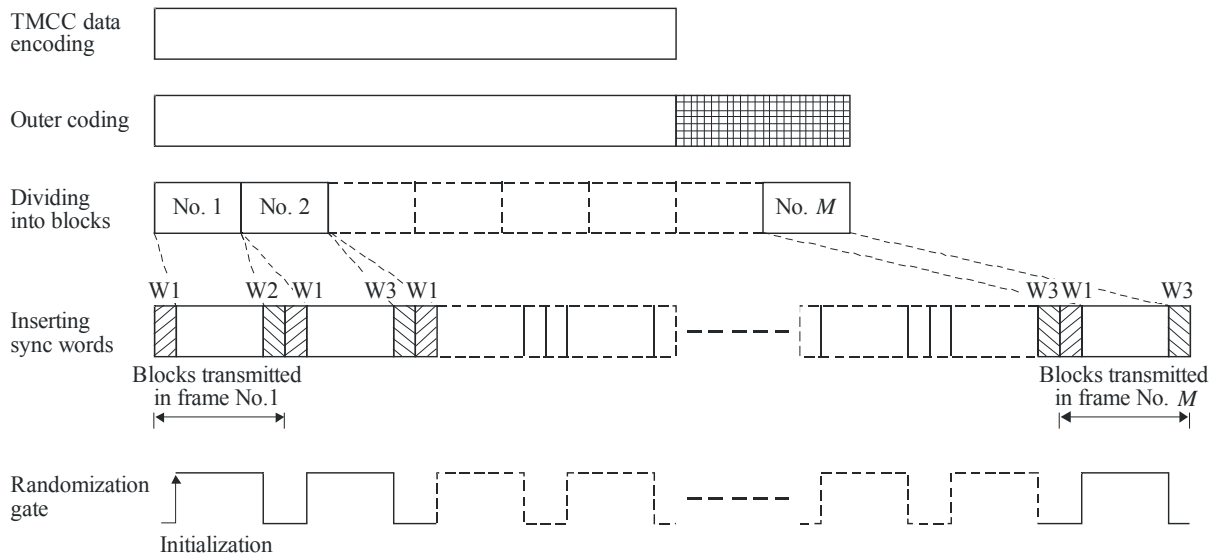
5 Channel coding for TMCC

The TMCC signal shall be randomized for energy dispersal. The polynomial for the pseudo-random binary sequence generator is the same as that for the main signal. The pseudo-random sequence is initiated at the third byte (just after the sync word) of the first block. The first bit of the output of the generator is applied to the first bit (i.e. MSB) of the third byte of the first block. The pseudo-random sequence is added to the data except to the timing reference words.

Interleaving processes may not be needed for TMCC signal consisting of a small amount of bits because the effect of interleaving is limited. An appropriate interleaving process should be specified, if necessary.

FIGURE 47

Generation of TMCC signal



BO.1516-47

Appendix 3

to Annex 1

Availability status of integrated circuits for common integrated receiver decoder

CONTENTS

- 1 Introduction
- 2 Analysis
- 3 Conclusion

1 Introduction

This Appendix describes the current state of integrated circuits (IC) development and availability. Several reputable IC manufacturers were contacted to review their current product offering, future plans, and evaluation of possibility to develop an IC supporting the four systems.

Several IC manufacturers already offer IC that supports Systems A, B and C and one supplier offers IC that supports Systems A and D. Furthermore, in the near future, all four systems are likely to be supported by several suppliers.

Report ITU-R BO.2008 – Digital multiprogramme broadcasting by satellite –was used as a basis to evaluate feasibility of IC supporting common elements of the four systems and associated cost impact.

2 Analysis

Recent evaluation has confirmed assumptions that have been identified in Report ITU-R BO.2008. Several manufacturers are offering IC for identified common IRD elements and, thus, making possible to develop an IRD supporting Systems A, B and C.

Required new functions of System D IRD were evaluated. It was determined that while all common elements of a universal IRD are required, link layer as depicted in Fig. 1 of Report ITU-R BO.2008 would require upgrade impacting modifications in decoder sections of the satellite tuner/decoder module as depicted in Figs 7 and 8. Typically two ICs are used to implement satellite tuner and its decoder modules. All four systems can use a common tuner chip (IC).

The satellite decoder chip includes demodulator function. System D requires, on chip, larger RAM to support block de-interleave function. Systems A, B and C use convolutional de-interleave function, which requires reduced RAM array. While there are additional functions to support control signalling required in this chip, it was determined that its impact would be negligible.

To evaluate decoder chip pricing, we assumed the same volume as typically used in estimating IRD costs. While the typical IRD cost splitting as listed in Report ITU-R BO.2008 estimates satellite demodulator + decoder function to cost USD 30, its present cost is estimated to be in the USD 4 range in typical volume. The upgraded satellite demodulator + decoder chip is estimated to cost in the USD 9 range within a year.

Report ITU-R BO.2008 indicates estimated cost of an IRD to be USD 300. We believe when compared with the estimated USD 5 (USD 9 – USD 4) increase cost to support System D, most IRD manufacturers will desire a common IRD design. While the price difference is estimated to be in the USD 5 range, over time this difference is expected to shrink. Current industry trends based on improvements in manufacturing processes projects 20% price reduction on a yearly basis.

3 Conclusions

Report ITU-R BO.2008 concluded that advances in IC manufacturing would make common element based IRD design possible. Several IC manufacturers are now supplying chips supporting Systems A, B and C. Based on evaluating Report ITU-R BO.2008 and present state of technologies, we conclude that a common element based IRD supporting the four systems will be feasible within a year with negligible cost impact to the total IRD costs.
