RECOMMENDATION ITU-R BO.1211*

Digital multi-programme emission systems for television, sound and data services for satellites operating in the 11/12 GHz frequency range**

(Question ITU-R 3/6)

(1995)

The ITU Radiocommunication Assembly,

considering

a) that digital source coding techniques have reached a level of maturity that can provide, at bit-rates suitable for efficient transmission, advantages in terms of vision and sound quality, in comparison to conventional analogue techniques;

b) that digital multiplexing techniques can provide greater flexibility to allocate, on a dynamic basis, the total data rate associated with each programme component (video, sound and data), to change the number of programmes contained in the same multiplex and to accommodate multimedia services;

c) that digital transmission techniques can offer better spectrum efficiency (data rates from about 25 to 50 Mbit/s, including suitable error protection, can for example be accommodated in a World Administrative Radio Conference for the Planning of the Broadcasting-Satellite Service (Geneva, 1977) (WARC BS-77) assignment) and power efficiency, in comparison to conventional analogue techniques, and can be flexibly configured to cope with the specific satellite bandwidth and power resources;

d) that the digital emissions may require less protection against interference than analogue emissions and thus, by virtue of this technique, spectrum efficiency may be improved;

e) that large scale integrated digital circuits can potentially offer lower equipment costs for mass production;

f) that new digital multi-programme TV systems can be designed to be used in the existing satellite channels in the 11/12 GHz bands;

* Radiocommunication Study Group 6 made editorial amendments to this Recommendation in 2001 in accordance with Resolution ITU-R 44.

** For fixed-satellite service (FSS) satellite applications, this Recommendation should be brought to the attention of Radiocommunication Study Group 4.

g) that extensive studies have been undertaken within ITU-R, with the objective of converging to a worldwide standard, on the following aspects:

– the requirements for digital television broadcasting systems by satellite,

– the integrated services digital broadcasting (ISDB) concept, including the transport of data services and service information,

– the common scrambling techniques for conditional access,

– the maximum possible commonality for different delivery media, such as satellite, satellite master antenna television systems (SMATV), terrestrial transmitters and cable,

– the possibility to implement the concept of a universal integrated television receiver, serving the interest of general public by offering affordably priced receivers throughout the world,

– the possibility for a future compatible evolution toward high definition television (HDTV),

– the ability of the various technical proposals to meet the above-mentioned requirements,

– computer simulations as well as laboratory and field trial evaluations of the various system approaches,

further considering

h) that digital multi-programme satellite services have already started in some countries and are planned to commence from 1995 onward in a number of other countries;

j) that satellite digital multi-programme TV systems are presently subject to developments in many parts of the world; Administrations outside Europe are still studying and considering a decision about adopting relevant standards and making submissions to ITU-R;

k) that a Memorandum of Understanding (MoU) for the development of harmonized Digital Video Broadcasting (DVB) services in Europe has been signed by more than 150 entities including equipment manufacturers, broadcasters, network operators and administrations;

l) that the DVB project has converged on a common system proposal for satellite emissions in the 11/12 GHz frequency range (referred to as DVB-S system) which is a European Telecommunication Standard (ETS);

m) that the DVB-S system comprises the following components, currently under development for the implementation of consumer receivers from 1995 onwards:

– image and audio coding, transport multiplexing, service information system according to draft International Standard ISO/IEC 13818;

– data service transport, common scrambling system, common interfaces to external equipment;

n) that the DVB-S system has the maximum commonality (including source coding, multiplexing and Reed-Solomon outer coding) with the DVB-C system proposed for distribution on cable, the DVB-CS system proposed for distribution on SMATV systems, and the DVB-T system being developed for terrestrial broadcasting, *recommends*

1 that the DVB-S framing structure, channel coding and modulation methods as specified in Annex 1 should be considered in converging to a worldwide standard for the introduction of digital multi-programme television services from satellites operating in the 11/12 GHz downlink frequency range.

NOTE 1 – In response to this Recommendation a Special Rapporteur has been appointed on convergence to a worldwide standard for digital multi-programme emission systems for television, sound and data services for satellites operating in the 11/12 GHz frequency range. This Special Rapporteur will study the DVB-S system along with other existing digital multi-programme satellite emission systems with the goal of converging to a worldwide standard.

ANNEX 1

Digital broadcasting systems for television, sound and data services – Framing structure, channel coding and modulation for 11/12 GHz satellite services

(ETS Standard – ETS 300 421)

CONTENTS

Page

1 Scope 4

2 Normative references 5

3 Symbols and abbreviations 5

3.1 Symbols 5

3.2 Abbreviations 6

4 Transmission system 7

4.1 System definition 7

4.2 Adaptation to satellite transponder characteristics 8

4.3 Interfacing 8

4.4 Channel coding 8

4.4.1 Transport multiplex adaptation and randomization for energy dispersal 8

4.4.2 Outer coding (RS), interleaving and framing 9

4.4.3 Inner coding (convolutional) 11

4.5 Baseband shaping and modulation 12

5 Error performance requirements 13

Page

Appendix 1 (normative) – Signal spectrum at the modulator output 14

Appendix 2 (informative) – Conceptual System description 16

Appendix 3 (informative) – Examples of bit rates versus transponder bandwidth 18

Appendix 4 (informative) – Examples of possible use of the System 21

1 Scope

This draft European Telecommunication Standard (ETS) describes the modulation and channel coding system (denoted the "System" for the purposes of this ETS) for satellite digital multi-programme television (TV)/high definition television (HDTV) services to be used for primary and secondary distribution in fixed-satellite service (FSS) and broadcasting-satellite service (BSS) bands. The System is intended to provide direct-to-home (DTH) services for consumer integrated receiver decoder (IRD), as well as collective antenna systems (satellite master antenna television (SMATV)) and cable television head-end stations, with a likelihood of remodulation (see [1]).

The System uses quaternary phase shift keying (QPSK) modulation and concatenated error protection strategy based on a convolutional code and a shortened Reed-Solomon (RS) code.

The System is suitable for use on different satellite transponder bandwidths.

Compatibility with Moving Pictures Experts Group-2 (MPEG-2) [2] coded TV services, with a transmission structure synchronous with the packet multiplex, is provided. Exploitation of the multiplex flexibility allows the use of the transmission capacity for a variety of TV service configurations, including sound and data services. All service components are time division multiplexed (TDM) on a single digital carrier.

The scope of this draft ETS is as follows:

– it gives a general description of the System for satellite digital TV transmission;

– it specifies the digitally modulated signal in order to allow compatibility between pieces of equipment developed by different manufacturers. This is achieved by describing in detail the signal processing principles at the modulator side, while the processing at the receive side is left open to different implementation solutions. However, it is necessary in this draft ETS to refer to certain aspects of reception;

– it identifies the global performance requirements and features of the System, in order to meet the service quality targets.

2 Normative references

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

[1] DTVB 1110/GT V4/MOD 252/DTVC 18, 7th revised version, January 1994: "Baseline modulation/channel coding system for digital multi-programme television by satellite" (Contribution from V4/MOD-B).

[2] [ISO/IEC DIS 13818-1 (June 1994): "Coding of moving pictures and associated audio".

[3] Forney, G.D. IEEE Trans. Comm. Tech., COM-19, pp. 772-781, October 1971: "Burst-correcting codes for the classic bursty channel".

[4] Intelsat Earth Station Standards (IESS) No. 308, revision 6 (26 October 1990): "Performance characteristics for Immediate Data Rate (IDR) digital carriers".

3 Symbols and abbreviations

3.1 Symbols

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

α: roll-off factor

C/N: signal-to-noise ratio

d_{free}: convolutional code free distance

E_b/N_0: ratio between the energy per useful bit and twice the noise power spectral density

f_N: Nyquist frequency

G_1, G_2: convolutional code generators

$g(x)$: RS code generator polynomial

I: interleaving depth (bytes)

I, Q: in-phase, quadrature phase components of the modulated signal

j: branch index of the interleaver

K: convolutional code constraint length

M: convolutional interleaver branch depth for $j = 1$, $M = N/I$

N: error protected frame length (bytes)

$p(x)$: RS field generator polynomial

r_m: in-band ripple (dB)

R_s: symbol rate corresponding to the bilateral Nyquist bandwidth of the modulated signal

R_u: useful bit rate after MPEG-2 [2] transport multiplexer

R'_u: bit rate after RS outer coder

T: number of bytes which can be corrected in RS error protected packet

T_s: symbol period

X, Y: di-bit stream after rate 1/2 convolutional coding

3.2 Abbreviations

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

AWGN: additive white Gaussian noise
BB: baseband
BER: bit error ratio
BSS: broadcasting-satellite service
BW: bandwidth
DTH: direct-to-home
EBU: European Broadcasting Union
ETS: European Telecommunication Standard
FDM: frequency division multiplex
FEC: forward error correction
FIFO: first-in, first-out shift register
FIR: finite impulse response
FSS: fixed-satellite service
hex: hexadecimal notation
HDTV: high definition television
IF: intermediate frequency
IMUX: input multiplexer – filter
IRD: integrated receiver decoder
ITU: International Telecommunication Union
ITU-T: Telecommunication Standardization Sector
MPEG: Moving Pictures Experts Group
MSB: most significant bit
MUX: multiplex
OBO: output back-off
oct: octal notation
OMUX: output multiplexer – filter
P: puncturing
PDH: plesiochronous digital hierarchy
PSK: phase-shift keying
PRBS: pseudo random binary sequence
QEF: quasi-error-free
QPSK: quaternary PSK
R: randomized sequence
RF: radio frequency
RS: Reed-Solomon
SMATV: satellite master antenna television
TBD: to be defined
TDM: time division multiplex
TV: television
TWTA: travelling wave tube amplifier

4 Transmission system

4.1 System definition

The System is defined as the functional block of equipment performing the adaptation of the baseband TV signals, from the output of the MPEG-2 [2] transport multiplexer, to the satellite channel characteristics. The following processes shall be applied to the data stream (see Fig. 1):

– transport multiplex adaptation and randomization for energy dispersal,

– outer coding (i.e. RS),

– convolutional interleaving,

– inner coding (i.e. punctured convolutional code),

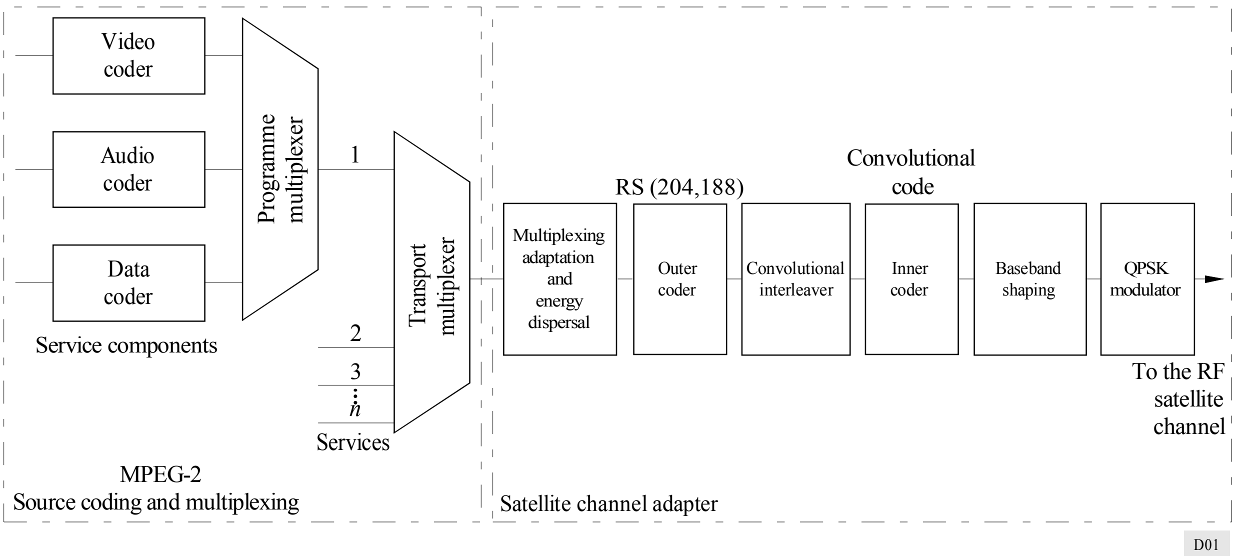
– baseband shaping for modulation,

– modulation.

The System functional description is given in Appendix 2.

FIGURE 1

Functional block diagram of the system



DTH services via satellite are particularly affected by power limitations, therefore, ruggedness against noise and interference, shall be the main design objective, rather than spectrum efficiency. To achieve a very high power efficiency without excessively penalizing the spectrum efficiency, the System shall use QPSK modulation and the concatenation of convolutional and RS codes. The convolutional code is able to be configured flexibly, allowing the optimization of the system performance for a given satellite transponder bandwidth (see Appendix 3).

Although the System is optimized for single carrier per transponder time division multiplex (TDM), it is able to be used for multi-carrier frequency division multiplex (FDM) type applications.

The System is directly compatible with MPEG-2 [2] coded TV signals. The modem transmission frame is synchronous with the MPEG-2 multiplex transport packets.

If the received signal is above *C/N* and *C/I* threshold, the forward error correction (FEC) techniques adopted in the System are designed to provide a "quasi-error-free" (QEF) quality target. The QEF means that less than one uncorrected error-event per transmission hour, corresponding to bit error ratio (BER) = 1×10^{-10} to 1×10^{-11} at the input of the MPEG-2 demultiplexer.

4.2 Adaptation to satellite transponder characteristics

Transmissions of digital multi-programme TV services will use satellites in both the FSS and the BSS bands. The choice of transponder bandwidth is a function of the satellite used and the data rates required by the service.

The symbol rate shall be matched to given transponder characteristics. Examples based on computer simulations for a hypothetical satellite chain, not including interference effects, are given in Appendix 3.

4.3 Interfacing

The System, as defined in this draft ETS, shall be delimited by the following interfaces given in Table 1.

TABLE 1

System interfaces

Location	Interface	Interface type	Connection
Transmit station	Input	MPEG-2 [2] transport multiplex	From MPEG-2 multiplexer
	Output	70/140 MHz IF	To RF devices
Receive installation	Output	MPEG-2 transport multiplex	To MPEG-2 demultiplexer
	Input	To be defined	From RF devices (indoor unit)

4.4 Channel coding

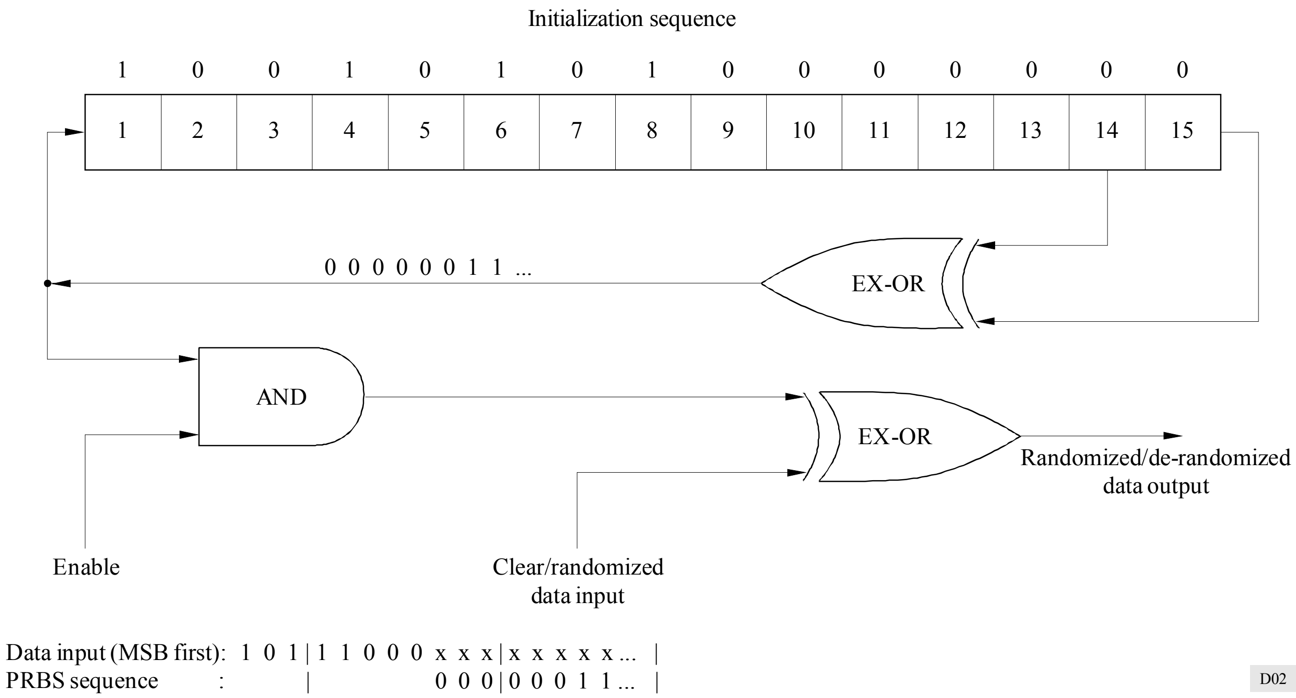
4.4.1 Transport multiplex adaptation and randomization for energy dispersal

The System input stream shall be organized in fixed length packets (see Fig. 3), following the MPEG-2 [2] transport multiplexer. The total packet length of the MPEG-2 transport Multiplex (MUX) packet is 188 bytes. This includes 1 sync-word byte (i.e. 47_{hex}). The processing order at the transmitting side shall always start from the MSB (i.e. 0) of the sync word-byte (i.e. 01000111).

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. 2.

FIGURE 2

Randomizer/de-randomizer schematic diagram



The polynomial for the pseudo random binary sequence (PRBS) generator shall be:

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

Loading of the sequence "100101010000000" into the PRBS registers, as indicated in Fig. 2, 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_{hex} to $B8_{hex}$. This process is referred as to 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. $B8_{hex}$). 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 sequence 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.

4.4.2 Outer coding (RS), interleaving and framing

The framing organization shall be based on the input packet structure (see Fig. 3a)).

Reed-Solomon RS (204,188, $T = 8$) shortened code, from the original RS (255,239, $T = 8$) code, shall be applied to each randomized transport packet (188 bytes) of Fig. 3b) to generate an error protected packet (see Fig. 3c)). Reed-Solomon coding shall also be applied to the packet sync byte, either non-inverted (i.e. 47_{hex}) or inverted (i.e. $B8_{hex}$).

Code generator polynomial: $g(x) = (x + \lambda^0)(x + \lambda^1)(x + \lambda^2) \dots (x + \lambda^{15})$, where $\lambda = 02_{hex}$.

Field generator polynomial: $p(x) = x^8 + x^4 + x^3 + x^2 + 1$.

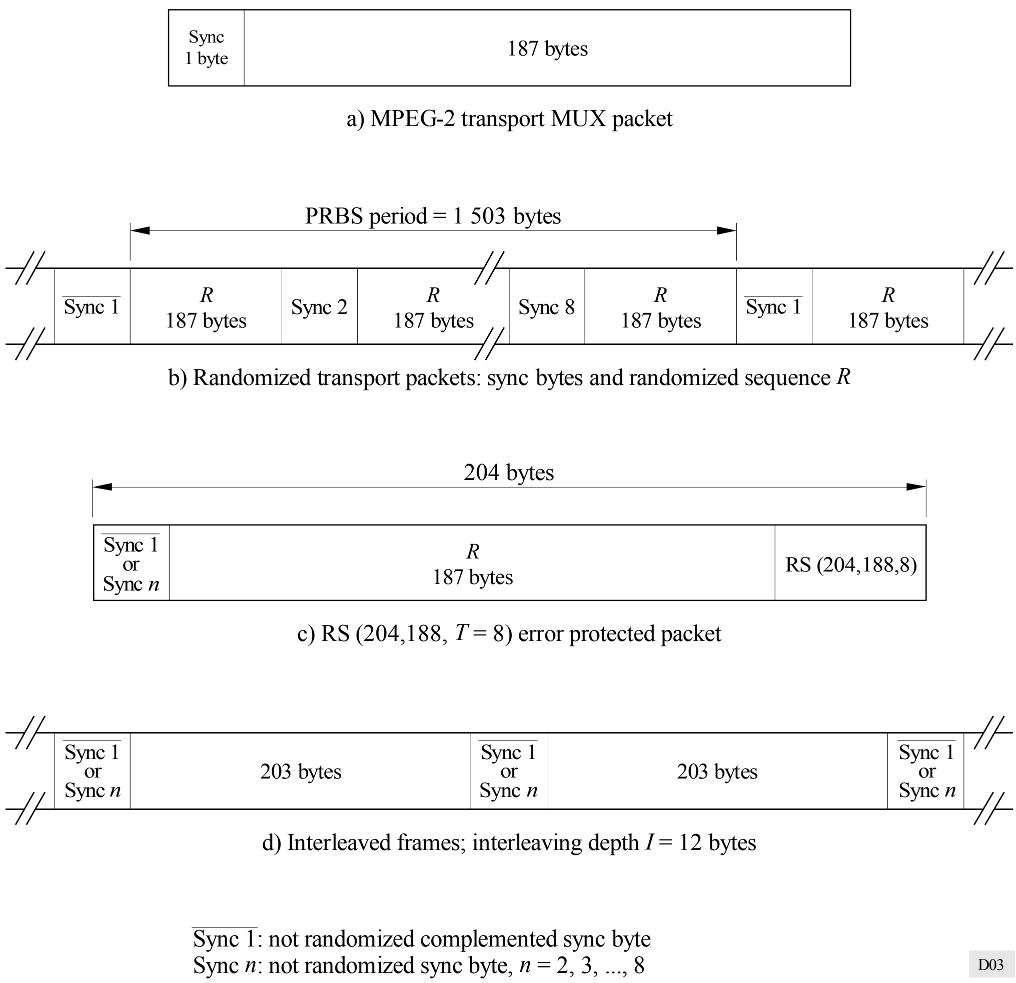
The shortened Reed-Solomon Code may be implemented by adding 51 bytes, all set to zero, before the information bytes at the input of a (255,239) encoder. After the RS coding procedure these null bytes shall be discarded.

Following the conceptual scheme of Fig. 4, convolutional interleaving with depth $I = 12$ shall be applied to the error protected packets (see Fig. 3c)). This results in an interleaved frame (see Fig. 3d)).

The convolutional interleaving process shall be based on the Forney approach (see [3]) which is compatible with the Ramsey type Ill 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 [2] sync bytes (preserving the periodicity of 204 bytes).

FIGURE 3

Framing structure



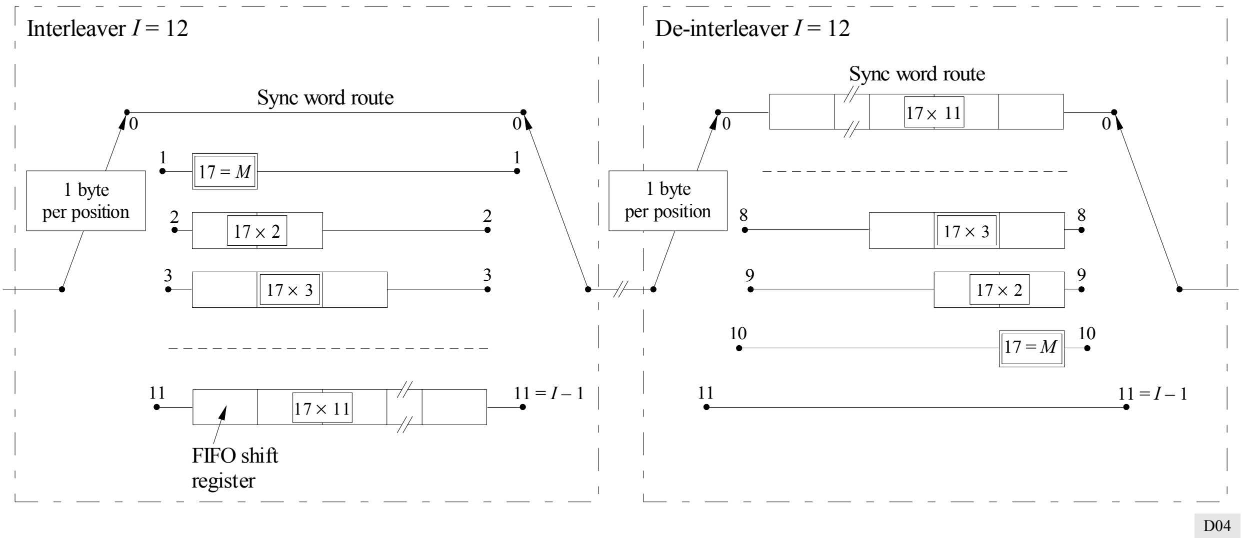
The interleaver may be composed of $I = 12$ branches, cyclically connected to the input byte-stream by the input switch. Each branch shall be a first-in, first-Out (FIFO) shift register, with depth (Mj) cells (where $M = 17 = N/I$, $N = 204$ = error protected frame length, $I = 12$ = interleaving depth, 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.

FIGURE 4

Conceptual diagram of the convolutional interleaver and de-interleaver



4.4.3 Inner coding (convolutional)

The System shall allow for a range of punctured convolutional codes, based on a rate 1/2 convolutional code with constraint length $K = 7$. This will allow selection of the most appropriate level of error correction for a given service or data rate. The System shall allow convolutional coding with code rates of 1/2, 2/3, 3/4, 5/6 and 7/8.

The punctured convolutional code shall be used as given in Table 2 (see also Fig. 5).

NOTE 1 – At the receiver, each of the code rates and puncturing configurations is in a position to be tried until lock is acquired. π phase ambiguity in the demodulator is able to be resolved by decoding the MPEG-2 [2] sync byte delimiting the interleaved frame.

TABLE 2

Punctured code definition

Original code			Code rates									
			1/2		2/3		3/4		5/6		7/8	
K	G_1 (X)	G_2 (Y)	P	d_{free}	P	d_{free}	P	d_{free}	P	d_{free}	P	d_{free}
7	171_{oct}	133_{oct}	$X = 1$ $Y = 1$ $I = X_1$ $Q = Y_1$	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

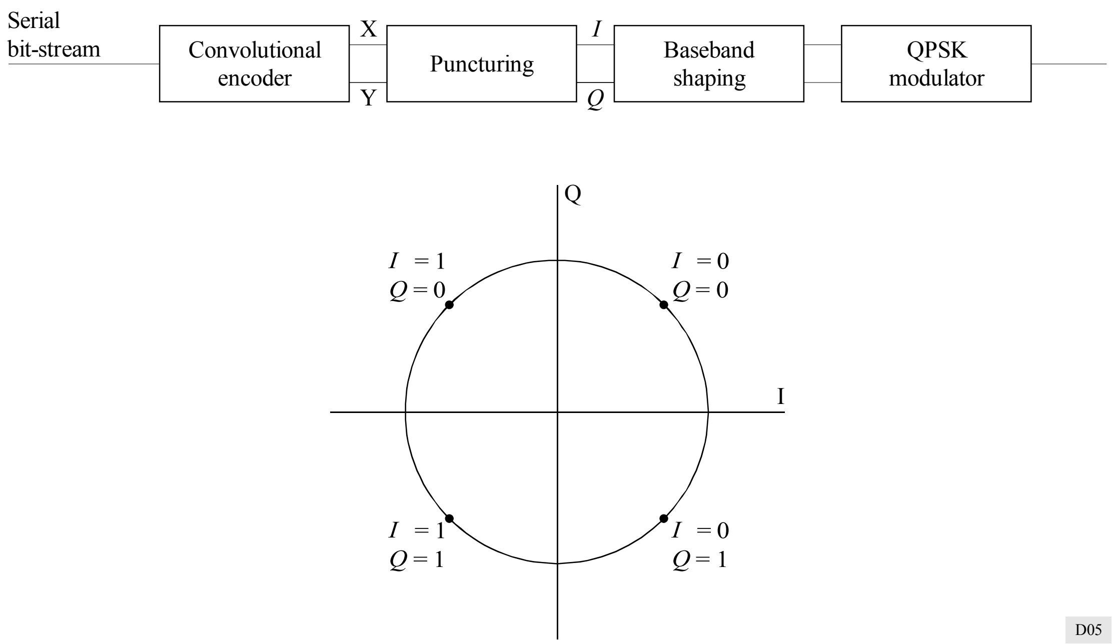
0: non transmitted bit

4.5 Baseband shaping and modulation

The System shall employ conventional Gray-coded QPSK modulation with absolute mapping (no differential coding). Bit mapping in the signal space as given on Fig. 5 shall be used.

FIGURE 5

QPSK constellation



Prior to modulation, the I and Q signals (mathematically represented by a succession of Dirac delta functions spaced by the symbol duration $T_S = 1/R_S$, with appropriate sign) shall be square-root raised cosine filtered. The roll-off factor shall be 0.35.

The baseband square-root raised cosine filter shall have a theoretical function defined by the following expression:

$$H(f) = 1 \qquad \text{for} \qquad |f| < f_N(1 - \mathrm{a})$$

$$H(f) = \left\{ \frac{1}{2} + \frac{1}{2} \sin \frac{\pi}{2 f_N} \left[\frac{f_N - |f|}{\alpha} \right] \right\}^{1/2} \qquad \text{for} \quad f_N(1-\alpha) \le |f| \le f_N(1+\alpha)$$

$$H(f) = 0 \qquad \text{for} \qquad |f| > f_N(1 + \mathrm{a})$$

where:

f_N: Nyquist frequency

$$= \frac{1}{2T_s} = \frac{R_s}{2}$$

α: roll-off factor

$= 0.35$

A template for the signal spectrum at the modulator output is given in Appendix 1.

5 Error performance requirements

The modem, connected in IF loop, shall meet the BER versus E_b/N_0 performance requirements given in Table 3.

TABLE 3

IF-loop performance of the System

Inner code rate	Required E_b/N_0 for BER = 2×10^{-4} after Viterbi QEF after Reed-Solomon
1/2	4.5
2/3	5.0
3/4	5.5
5/6	6.0
7/8	6.4

NOTE 1 – The figures of E_b/N_0 refer to the useful bit-rate before RS coding and include a modem implementation margin of 0.8 dB and the noise bandwidth increase due to the outer code (10 log 188/204 = 0.36 dB).

NOTE 2 – Quasi-error-free (QEF) means less than one uncorrected error event per hour, corresponding to BER = 1×10^{-10} to 1×10^{-11} at the input of the MPEG-2 demultiplexer.

Indicative figures of the System performance by satellite are given in Appendix 4.

APPENDIX 1

TO ANNEX 1
(normative)

Signal spectrum at the modulator output

Figure 6 gives a template for the signal spectrum at the modulator output.

Figure 6 also represents a possible mask for a hardware implementation of the Nyquist modulator filter as specified in § 4.5. The points A to S shown on Figs. 6 and 7 are defined in Table 4. 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 is to be applied on the filter response.

Figure 7 gives a mask for the group delay for the hardware implementation of the Nyquist modulator filter.

Figures 6 and 7 are based on [4], with slight modification due to different roll off.

FIGURE 6

Template for the signal spectrum mask at the modulator output

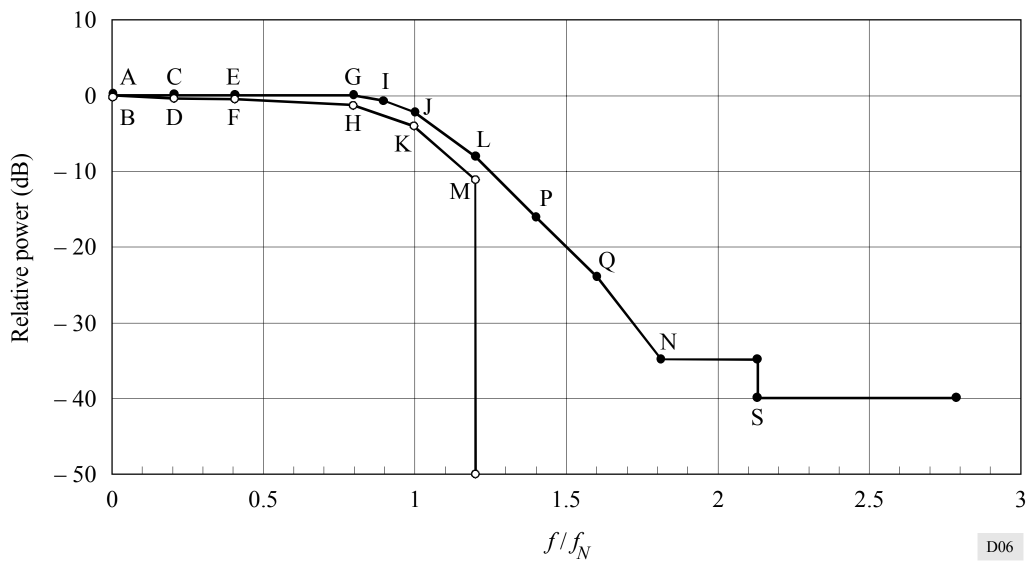


FIGURE 7

Template of the modulator filter group delay

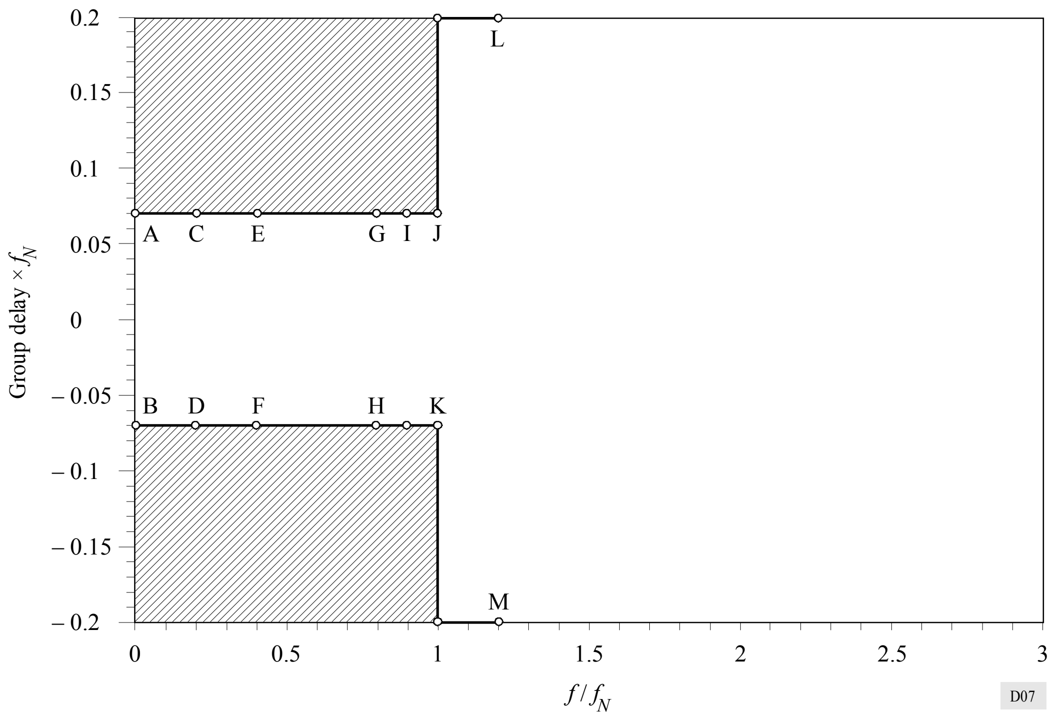


TABLE 4

Definition of points given in Fig. 6

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

APPENDIX 2

TO ANNEX
(informative)

Conceptual System description

The modulator and demodulator may perform the functions indicated in the block diagrams of Fig. 8.

Due to the similarity of the modulator and demodulator block diagrams, only the latter is described as follows:

– *IF interface and QPSK demodulator*: this unit performs the quadrature coherent demodulation function and the analogue to digital conversion, providing "soft decision" *I* and *Q* information to the inner decoder.

– *Matched filter*: this unit performs the complementary pulse shaping filtering of raised cosine 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.

– *Carrier/clock recovery unit*: this device recovers the demodulator synchronization. The probability of slips generation over the full *C*/*N* range of the demodulator should be very low.

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

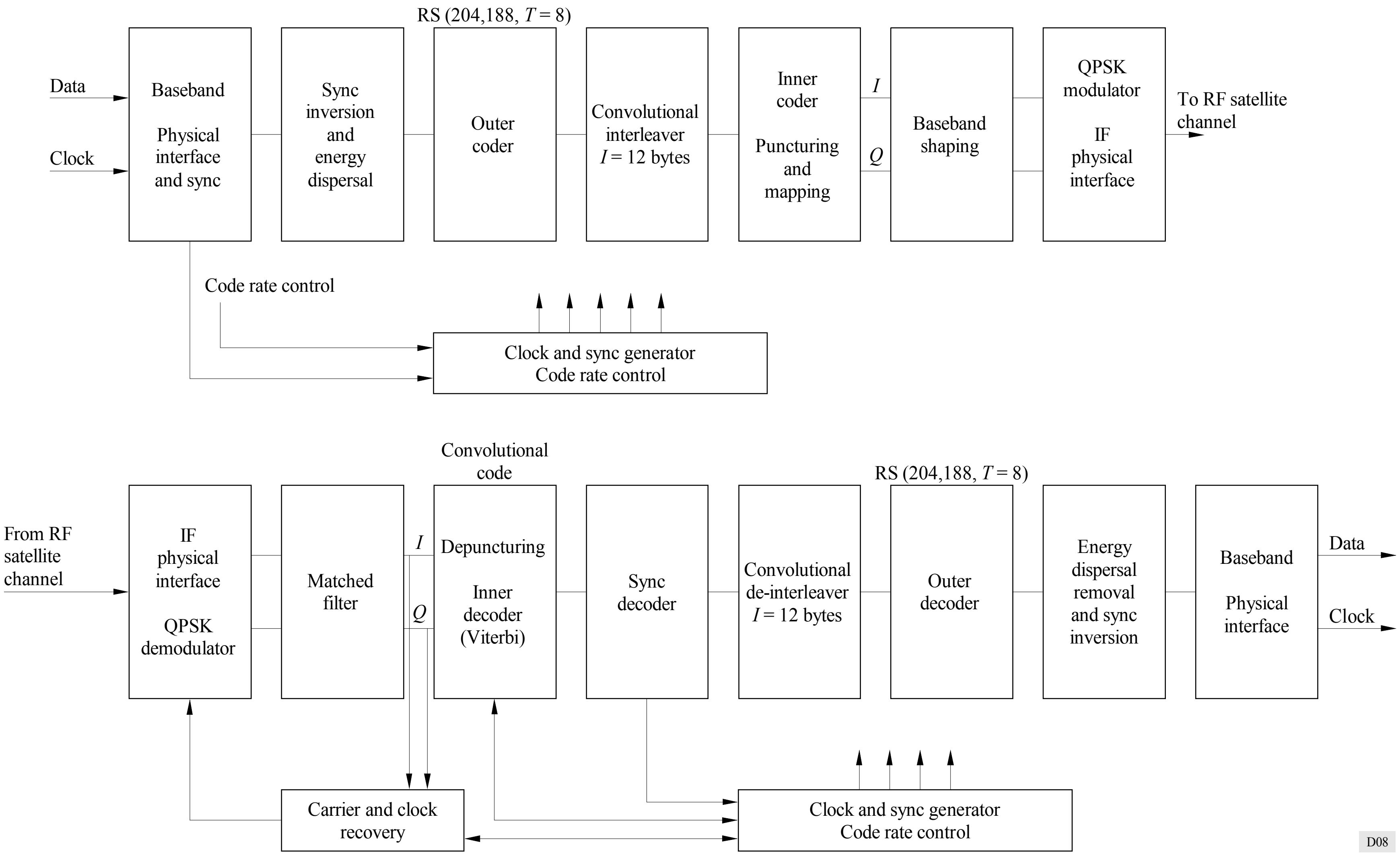
– *Sync byte decoder*: by decoding the MPEG-2 [2] sync bytes, this decoder provides synchronization information for the de-interleaving. It is also in a position to recover ambiguity of QPSK demodulator (not detectable by the Viterbi decoder).

– *Convolutional de-interleaver*: this device 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.

– *Outer decoder*: this unit provides second level error protection. It is in a position to provide QEF output (i.e. BER of about 1×10^{-10} to 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.

FIGURE 8

Conceptual block diagram of the System at the transmitting and receiving side



– *Energy dispersal removal*: this unit recovers the user data by removing the randomizing pattern used for energy dispersal purposes and changes the inverted sync byte to its normal MPEG-2 sync byte value.

– *Base-band physical interface*: this unit adapts the data structure to the format and protocol required by the external interface.

NOTE 1 – A possibility is provided by the MPEG-2 [2] system to set on the error flag bit in the packet header if the correction capability of the outer code is exceeded.

APPENDIX 3

TO ANNEX 1
(informative)

Examples of bit rates versus transponder bandwidth

The transmission symbol rate R_s, can be matched to given transponder characteristics, to achieve the maximum transmission capacity compatible with the acceptable signal degradation due to transponder bandwidth limitations. Table 5 gives examples of the useful bit rate capacity R_u achievable on a satellite transponder with bandwidth BW corresponding to $BW/R_s = 1.28$.

TABLE 5

Examples of bit rates versus transponder bandwidth

BW (at –3 dB) (MHz)	BW′ (at –1 dB) (MHz)	R_s (for BW/R_s = 1.28) (MBd)	R_u (for QPSK + 1/2 convolutional) (Mbit/s)	R_u (for QPSK + 2/3 convolutional) (Mbit/s)	R_u (for QPSK + 3/4 convolutional) (Mbit/s)	R_u (for QPSK + 5/6 convolutional) (Mbit/s)	R_u (for QPSK + 7/8 convolutional) (Mbit/s)
54	48.6	42.2	38.9	51.8	58.3	64.8	68.0
46	41.4	35.9	33.1	44.2	49.7	55.2	58.0
40	36.0	31.2	28.8	38.4	43.2	48.0	50.4
36	32.4	28.1	25.9	34.6	38.9	43.2	45.4
33	29.7	25.8	23.8	31.7	35.6	39.6	41.6
30	27.0	23.4	21.6	28.8	32.4	36.0	37.8
27	24.3	21.1	19.4	25.9	29.2	32.4	34.0
26	23.4	20.3	18.7	25.0	28.1	31.2	32.8

NOTE 1 – R_u stands for the useful bit rate after MPEG-2 MUX. R_s (symbol rate) corresponds to the –3 dB bandwidth of the modulated signal.

NOTE 2 – The figures of Table 5 correspond to an E_b/N_0 degradation of 1.0 dB (with respect to AWGN channel) for the case of 0.35 roll-off and 2/3 code rate, including the effects of IMUX, OMUX and TWTA.

Other BW/R_s values may be adopted for different service requirements, depending on the trade-off between transmission capacity and E_b/N_0 degradation.

Figures 9 and 10 show the IMUX and OMUX filter characteristics adopted in the computer simulations, with a 33 MHz (–3 dB) total bandwidth.

FIGURE 9

Hypothetical IMUX filter characteristic

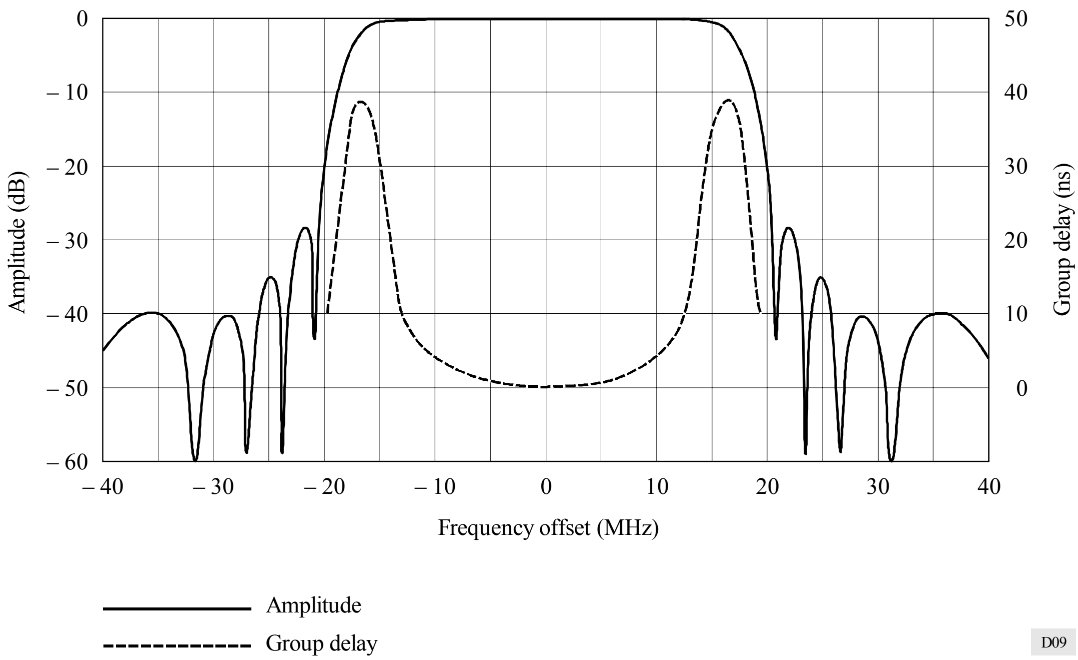


FIGURE 10

Hypothetical OMUX filter characteristic

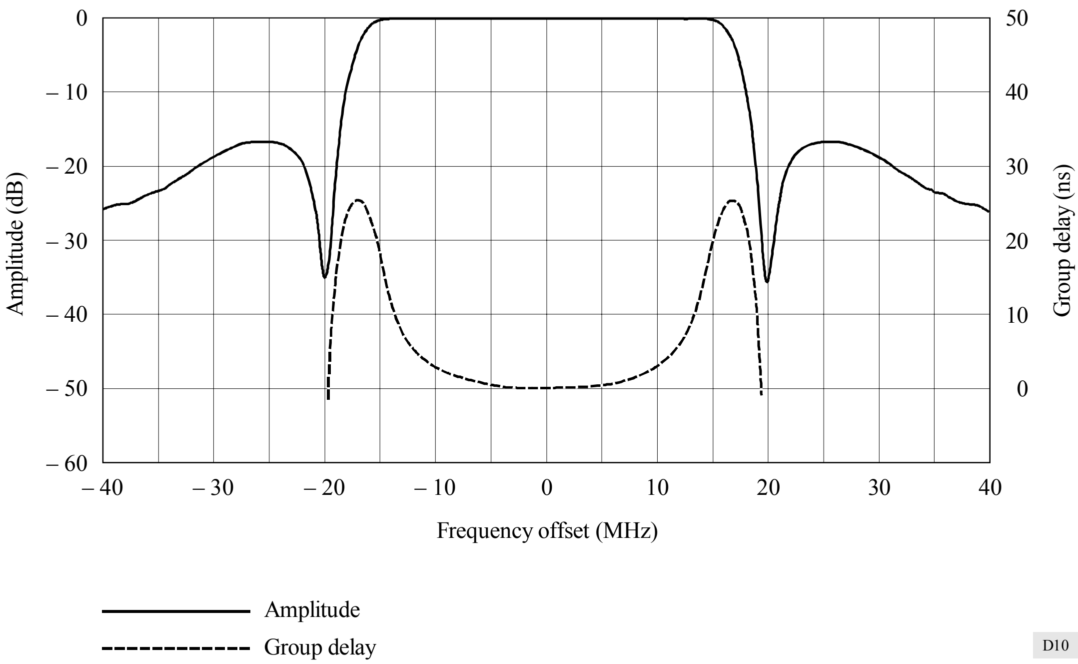
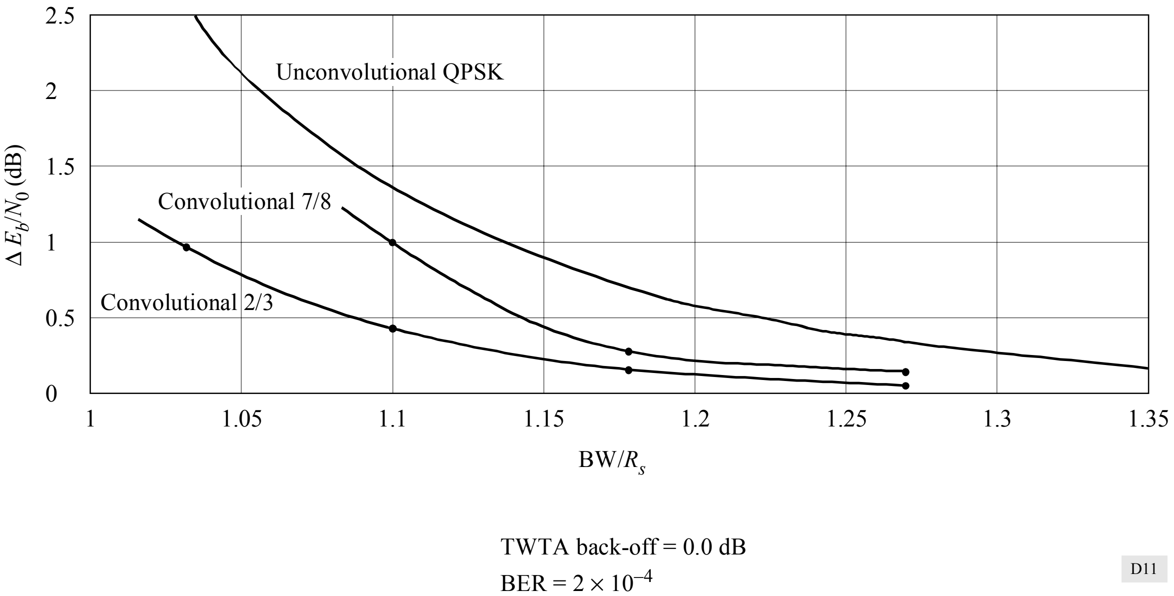


Figure 11 gives an example of the E_b/N_0 degradation on a computer simulated satellite transponder (travelling wave tube amplifier output back-off (TWTA OBO) = 0 dB) due to bandwidth limitations on IMUX and OMUX (see Figs. 9 and 10), for a ratio BW/R_s between 1 and 1.35. The reference 0 dB degradation refers to the case of a satellite transponder without bandwidth limitations (BW = ∞, TWTA OBO = 0 dB). The results are obtained by computer simulations, with inner code rates 2/3 and 7/8, at BER = 2×10^{-4}. Other results could be obtained for different transponder filter characteristics. When using the results of Fig. 11, suitable margins should be allowed to take into account thermal and ageing instabilities of the transponder characteristics.

FIGURE 11

Example degradation due to transponder bandwidth limitation



APPENDIX 4

TO ANNEX 1
(informative)

Examples of possible use of the System

Table 6 considers possible examples of use of the System for a nominal transponder bandwidth (−3dB) of 33 MHz. Different inner code rates are given with the relevant bit rates.

TABLE 6

Example of System performance over 33 MHz transponder

Bit rate R_u (after MUX) (Mbit/s)	Bit rate R'_u (after RS) (Mbit/s)	Symbol rate (MBd)	Convolutional inner code rate	RS outer code rate	C/N (33 MHz) (dB)
23.754	25.776	25.776	1/2	188/204	4.1
31.672	34.368	25.776	2/3	188/204	5.8
35.631	38.664	25.776	3/4	188/204	6.8
39.590	42.960	25.776	5/6	188/204	7.8
41.570	45.108	25.776	7/8	188/204	8.4

NOTE 1 – The figures in Table 6 refer to computer simulation results achieved on a hypothetical satellite chain, including IMUX, TWTA and OMUX (see Figs. 9 and 10), with modulation roll-off of 0.35. The C/N figures are based on the assumption of soft-decision Viterbi decoding in the receiver. The ratio $BW/R_s = 1.28$ has been adopted.

NOTE 2 – 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 “quasi-error-free” at the RS coder output. Degradation due to interference is not taken into account.

Figure 12 shows that the example highlighted in Table 6 with rate 2/3 inner code would be suitable for connection to a plesiochronous digital hierarchy (PDH) terrestrial network at 34.368 Mbit/s, including the same Reed-Solomon error protection used by satellite.

FIGURE 12

Example of connection of the System with the terrestrial PDH network