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SERIES J: TRANSMISSION OF TELEVISION, SOUND
PROGRAMME AND OTHER MULTIMEDIA SIGNALS

Digital transmission of television signals

**Digital multi-programme systems for television,
sound and data services for cable distribution**

ITU-T Recommendation J.83

(Previously CCITT Recommendation)

ITU-T J-SERIES RECOMMENDATIONS

TRANSMISSION OF TELEVISION, SOUND PROGRAMME AND OTHER MULTIMEDIA SIGNALS

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FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Geneva, October, 1997).

This second edition of ITU-T Recommendation J.83 was prepared by ITU-T Study Group 9 (1997-2000) and incorporates Amendment 1 and Amendment 2 approved under the WTSC Resolution No. 1 procedure on the 17th of October 1996 and the 22nd of April 1997 respectively.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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SUMMARY

This Recommendation "Digital multi-programme systems for television, sound and data services for cable distribution" covers the definition of the framing structure, channel coding and modulation for digital multi-programme signals for television, sound and data services distributed by cable networks.

This Recommendation has four Annexes (A, B, C and D), that provide the specifications for the four digital television cable systems submitted to the ITU-T. This reflects the fact that standardization of digital cable television systems is being addressed for the first time by the ITU-T and that a number of systems had been developed and provisionally implemented when this standardization effort was undertaken by the ITU.

This Recommendation recommends that those implementing new digital multi-programme services on existing and future cable networks should use one of the systems whose framing structure, channel coding and modulation are specified in Annexes A, B, C and D.

INTRODUCTION

The development of new digital technology is now reaching the point at which it is evident that they enable digital systems to offer significant advantages, in comparison with conventional analogue techniques, in terms of vision and sound quality, spectrum and power efficiency, service flexibility, multimedia convergence and potentially lower equipment costs. Moreover, the use of cable distribution for the delivery of video and audio signals to individual viewers and listeners is continually growing, and has already become the dominant form of distribution in many parts of the world. It is also evident that these potential benefits can best be achieved through the economies of scale resulting from the widespread use of digital systems designed to be easily implementable on existing infrastructure and which take advantage of the many possible synergies with related audiovisual systems.

Administrations and private operators planning the introduction of digital cable television services are encouraged to consider the use of one of the systems described in Annexes A, B, C and D, and to seek opportunities for further convergence, rather than developing a different system based on the same technologies.

This second edition of this Recommendation incorporates Amendment 1 and Amendment 2. These amendments brought the following changes with respect to the first edition of the Recommendation:

- a) In Annex B there is now a specification for 256-QAM;
- b) In Annex B, two distinct operating modes of interleaving capability are specified, called *level 1* and *level 2*. Level 1 is specified for 64-QAM transmission only and this mode already existed in the first edition of Annex B. Level 2 encompasses 64-QAM and 256-QAM transmission, and for both modulation schemes is capable of supporting variable interleaving.
- c) In the first edition of Annex D, 24 bits were identified which determined the VSB mode for the data in the frame and two such modes were defined: 16-VSB Cable and 8-VSB Terrestrial (trellis coded). In this second edition, three other VSB modes are defined, i.e. 2-VSB, 4-VSB and 8-VSB.

Table 1/J.83 has been updated to take account of these extensions. In addition, a new Appendix I containing a short Bibliography has been added.

DIGITAL MULTI-PROGRAMME SYSTEMS FOR TELEVISION, SOUND AND DATA SERVICES FOR CABLE DISTRIBUTION

(Geneva, 1995; revised in 1997)

1 Scope

The scope of this Recommendation is the definition of the framing structure, channel coding and modulation for digital multi-programme television, sound and data signals distributed by cable networks (e.g. CATV systems) possibly in frequency-division multiplex. A separate Recommendation defines the transmission characteristics for digital multi-programme signals distributed through SMATV networks.

NOTE – The system input is specified to be the MPEG-2 transport layer; this provides some ancillary data capacity in the forward channel, which can be used to accommodate the needs of interactive services (a description of the provision and characteristics of the return channel is outside the scope of this Recommendation).

Being highly flexible, the MPEG-2 transport layer can be configured to deliver any desired mix of television, sound and data signals (with sound either related or unrelated to the video signal content, and at various possible levels of quality). The transport layer can even be totally devoted to the delivery of sound programming, although it may not necessarily be optimised for this application.

The specific case of the delivery of a multiplex only containing sound signals may be addressed in a future Recommendation.

This Recommendation is intended to ensure that the designers and operators of cable distribution (e.g. CATV) networks carrying multi-programme signals will have the information they need to be able to establish and maintain fully satisfactory networks. It also provides the information needed by the designers and manufacturers of equipment (including receivers) for digital multi-programme signals distributed by cable networks.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in the text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of currently valid ITU-T Recommendations is regularly published.

- [1] ITU-R Recommendation BO.1211 (1995), *Digital multi-programme emission systems for television, sound and data services for satellites operating in the 11/12 GHz frequency range*.
- [2] ITU-T Recommendation H.222.0 (1995) | ISO/IEC 13818-1:1996, *Information technology – Generic coding of moving pictures and associated audio information: Systems*.

3 Terms and definitions

No unconventional terms or definitions are used in this Recommendation.

4 Symbols and abbreviations

4.1 Symbols

This Recommendation uses the following symbols.

α	Roll-off factor
A_k, B_k	Most Significant Bits at the output of the Byte to m-tuple converter
byte	Eight bits
f_0	Channel centre frequency
f_N	Nyquist frequency
$g(x)$	RS code generator polynomial
$G_{(256)}$	RS primitive field generator polynomial
$G_{(16)}$	Randomizer generator polynomial
I	Interleaving depth (bytes)
I, Q	In-phase, Quadrature phase components of the modulated signal
j	Branch index
k	Number of bytes mapped into n symbols
m	Power of 2^m -level QAM: 4,5,6 for 16-QAM, 32-QAM, 64-QAM, respectively
M	Convolutional interleaver branch depth for $j = 1$, $M = N/I$
ms	millisecond
n	Number of symbols mapped from k bytes
N	Error protected frame length (bytes)
$p(x)$	RS field generator polynomial
$PN(x)$	Pseudorandom sequence, identified by the number following the symbol
r_m	In-band ripple (dB)
R	Randomized sequence
R_s	Symbol rate corresponding to bilateral Nyquist bandwidth of modulated signal
R_u	Useful bit rate after MPEG-2 transport multiplexer
$R_{u'}$	Bit rate after RS outer coder
q	Number of bits: 2,3,4 for 16-QAM, 32-QAM, 64-QAM, respectively
T	Number of bytes which can be corrected in RS error-protected packet
T_s	Symbol period

4.2 Abbreviations

This Recommendation uses the following symbols.

ATM	Asynchronous Transfer Mode
BB	BaseBand
BER	Bit Error Ratio
bps	Bits per second
CATV	Community Antenna Television
C/N	Carrier to Noise ratio

DTVC	Digital Television by Cable
FEC	Forward Error Correction
FIFO	First In First Out
HEC	Header Error Control
HEX	Hexadecimal
IF	Intermediate Frequency
IRD	Integrated Receiver Decoder
LSB	Least Significant Bit
MMDS	Multichannel Multipoint Distribution System
MPEG	Motion Picture Experts Group
MSB	Most Significant Bit
MUX	Multiplex
P	Parity
PDH	Plesiochronous Digital Hierarchy
PN	Pseudorandom Noise
ppm	Parts per million
PRBS	PseudoRandom Binary Sequence
QAM	Quadrature Amplitude Modulation
QEF	Quasi Error Free
RF	Radio Frequency
RS	Reed-Solomon
SMATV	Satellite Master Antenna Television
SNR	Signal-to-Noise Ratio
sps	Symbols per second
Sync	Synchronizing signal
TBD	To Be Determined
TDM	Time Division Multiplex
TS	Transport Stream
VLSI	Very Large Scale Integration
VSF	Vestigial SideBand
XOR	Exclusive OR
2-VSF	2 level VSF
4-VSF	4 level VSF
8-VSF	8 level VSF
16-VSF	16 level VSF

5 Digital multi-programme systems for cable distribution

It is recommended that those implementing new digital multi-programme services on existing and future cable networks should use one of the systems whose framing structure, channel coding and modulation are specified in Annexes A, B, C and D. The specifications are compared in Table 1, indicating common features.

Table 1/J.83 – Comparison of specifications in summary form indicating common features

Item		Annex B	Annex A	Annex C	Annex D
Input signals		Modified MPEG-2 transport stream. A parity checksum is substituted for the sync byte, supplying improved packet delineation functionality, and error detection capability independent of the FEC layer. (See B.4.)	MPEG-2 transport Stream (See A.3, C.3, D.3.)		
Framing structure		An FEC frame consists of a 42- or 40-bit sync trailer following 60 or 88 RS blocks, with each block containing 128 symbols. An RS symbol consists of 7 bits. Thus, there is a total 53 802 or 78 888 bits in an FEC frame for 64- or 256-QAM respectively. (See B.5.3.)	The framing organization is based on the MPEG-2 transport packet structure. (See A.4, C.4, D.4.)		
Channel coding	Randomization	The 3-word polynomial for the PRS: $x^3 + x + \alpha^3$ over GF 128. (See B.5.4.)	The 15-bit polynomial for the PRBS: $1 + x^{14} + x^{15}$ (See A.5.1, C.5.1.)		The 16-bit polynomial for the PRBS: $1 + x + x^3 + x^6 + x^7 + x^{11} + x^{12} + x^{13} + x^{16}$. (See D.5.1.)
	FEC	Concatenated coding, RS (128, 122) GF 128 with convolutional coding. (See B.5.)	RS (204, 188) GF 256 (See A.5.2, C.5.2.)		RS (207, 187) GF 256 (See D.5.2.)
	Interleaving	Convolutional interleaving depth: I = 128, 64, 32, 16, 8 J = 1, 2, 3, 4, 5, 6, 7, 8, 16. (See B.5.2.)	Convolutional interleaving, depth: I = 12. (See A.5.3, C.5.3.)		Convolutional interleaving, depth: I = 52. (See D.5.3.)
Modulation	Byte to symbol mapping	See B.5.5.	See A.6, C.6.1.		See D.6.1.
	Differential coding	See B.5.5.	See A.6, C.6.2.		None
	Trellis coding	See B.5.5.	None		
	Bandwidth	6 MHz	8 MHz	6 MHz	
	Constellation	64- or 256-QAM Figure B.18 or B.19	16-, 32-, 64-QAM Figure A.7	64-QAM Figure C.7	2-, 4-, 8-, 16-VSB
	Roll-off factor	18% or 12% for 64- or 256-QAM respectively. See B.6.1.	15% See A.7	13% See C.6.4	11.5% See D.6.3
	Baseband filter characteristics	Table B.2	Figure A.8	Figure C.8	Figure D.11

Annex A

Digital multi-programme System A

A.1 Introduction

This Annex derives from work done on digital television satellite broadcasting in Europe; it describes the framing structure, channel coding and modulation (denoted "the System" for the purposes of this Annex) for digital multi-programme television distribution by cable. This System can be used transparently with the modulation/channel coding system used for digital multi-programme television by satellite (see Reference [1]). The System allows for further evolution as technology advances.

The System is based on MPEG-2 (see Reference [2] as regards source coding and transport multiplexing. It is based on Quadrature Amplitude Modulation (QAM). It allows for 16-, 32-, or 64-QAM constellations and permits future extension to higher constellations, such as 128-QAM and 256-QAM.

The System FEC is designed to improve the Bit Error Ratio (BER) from 10^{-4} to a range of 10^{-10} to 10^{-11} , ensuring "Quasi Error Free" (QEF) operation with approximately one uncorrected error event per transmission hour.

A.2 Cable system concept

The cable system shall be defined as the functional block of equipment performing the adaptation of the baseband TV signals to the cable channel characteristics (see Figure A.1). In the cable head-end, the following TV baseband signal sources can be considered:

- satellite signal(s);
- contribution link(s);
- local programme source(s).

The following processes shall be applied as shown in Figure A.1.

A.2.1 Baseband interfacing¹⁾ and sync

This unit shall adapt the data structure to the format of the signal source. The framing structure shall be in accordance with MPEG-2 transport layer including sync bytes.

A.2.2 Sync 1 inversion and randomization

This unit shall invert the Sync 1 byte according to the MPEG-2 framing structure, and randomizes the data stream for spectrum shaping purposes.

A.2.3 Reed-Solomon (RS) coder

This unit shall apply a shortened Reed-Solomon (RS) code to each randomized transport packet to generate an error-protected packet. This code shall also be applied to the Sync byte itself.

A.2.4 Convolutional interleaver

This unit shall perform a depth $I = 12$ convolutional interleaving of the error-protected packets. The periodicity of the sync bytes shall remain unchanged.

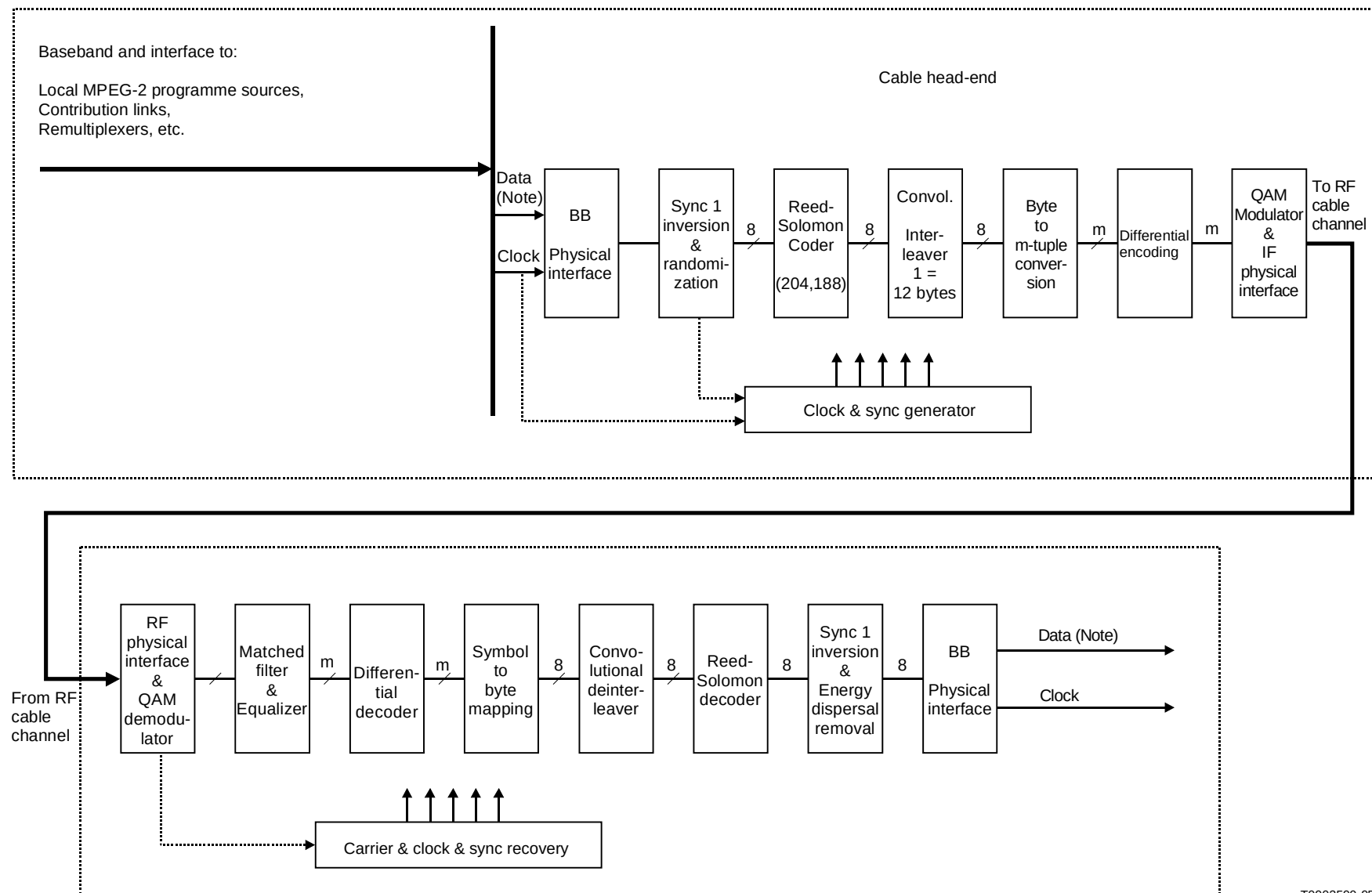
A.2.5 Byte to m-tuple conversion

This unit shall perform a conversion of the bytes generated by the interleaver into QAM symbols.

A.2.6 Differential encoding

In order to get a rotation-invariant constellation, this unit shall apply a differential encoding of the two Most Significant Bits (MSBs) of each symbol.

¹⁾ Interfaces are not part of this Recommendation.



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NOTE – MPEG-2 transport MUX packets.

Figure A.1/J.83 – Conceptual block diagram of elements at the cable head-end and receiving site

A.2.7 QAM modulation and physical interface

This unit performs a square-root raised cosine filtering of the I and Q signals prior to QAM modulation. This is followed by interfacing the QAM modulated signal to the Radio Frequency (RF) cable channel.

A.2.8 Cable receiver

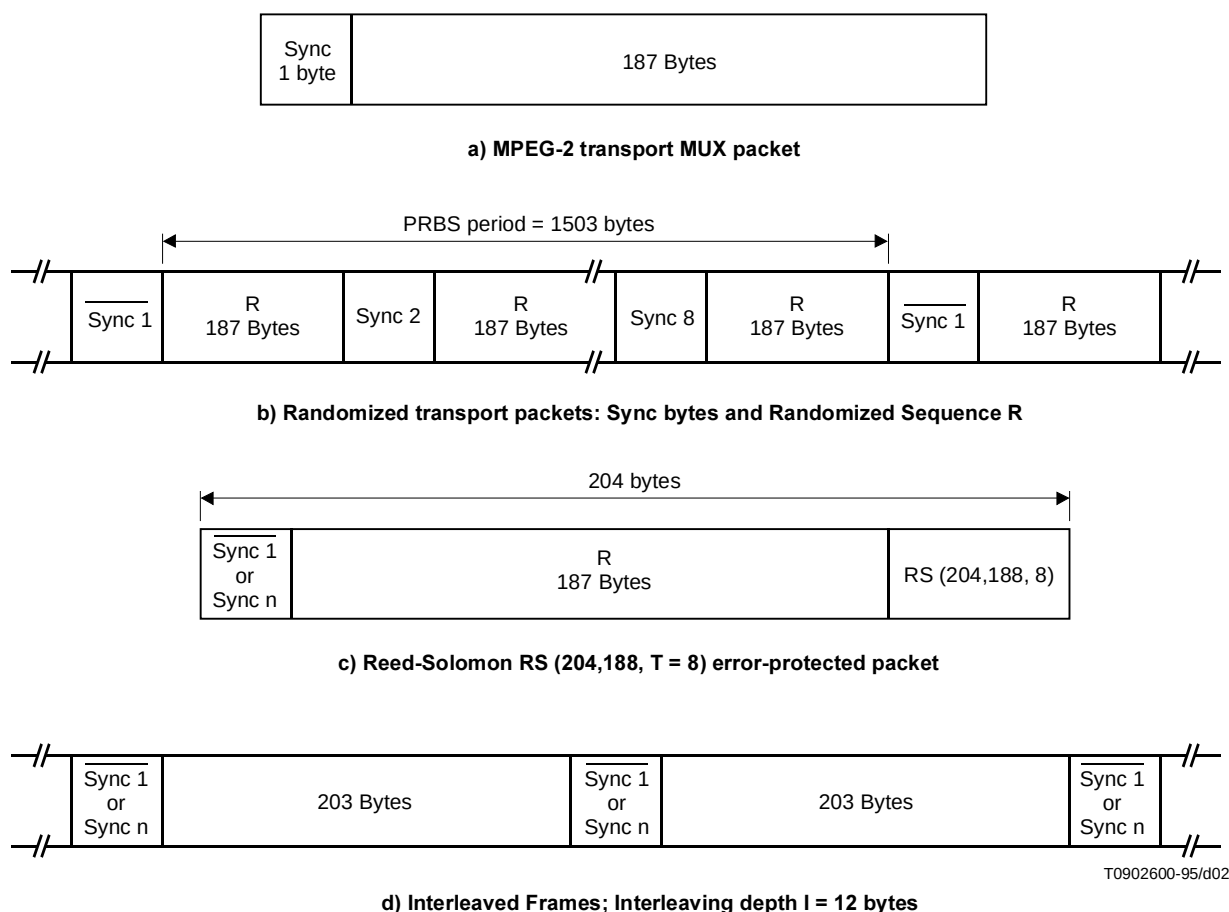
A System receiver shall perform the inverse signal processing, as described for the modulation process above, in order to recover the baseband signal.

A.3 MPEG-2 transport layer

The MPEG-2 transport layer is defined in Reference [2]. The transport layer for MPEG-2 data is comprised of packets having 188 bytes, with one byte for synchronization purposes, three bytes of header containing service identification, scrambling and control information, followed by 184 bytes of MPEG-2 or auxiliary data.

A.4 Framing structure

The framing organization shall be based on the MPEG-2 transport packet structure. The System framing structure is shown in Figure A.2.



Sync 1 Not randomized complemented sync byte.
Sync n Not randomized sync byte, n = 2, 3, ..., 8.

Figure A.2/J.83 – Framing structure

A.5 Channel coding

To achieve the appropriate level of error protection required for cable transmission of digital data, an FEC based on Reed-Solomon encoding shall be used. In contrast to the Baseline System for satellite described in Reference [1], no convolutional coding shall be applied to cable transmission. Protection against burst errors shall be achieved by the use of byte interleaving.

A.5.1 Randomization for spectrum shaping

The System input stream shall be organized in fixed length packets (see Figure A.2), following the MPEG-2 transport multiplexer. The total packet length of the MPEG-2 transport 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 the System for satellite (see Reference [1]) and to ensure adequate binary transitions for clock recovery, the data at the output of the MPEG-2 transport multiplex shall be randomized in accordance with the configuration depicted in Figure A.3.

The polynomial for the PseudoRandom Binary Sequence (PRBS) generator shall be:

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

Loading of the sequence "100101010000000" into the PRBS registers, as indicated in Figure A.3, 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 shall be bit wise inverted from 47_{HEX} to B8_{HEX}.

The first bit at the output of the PRBS generator shall be applied to the first bit 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 continues, but its output shall be disabled, leaving these bytes unrandomized. The period of the PRBS sequence shall therefore be 1503 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.

A.5.2 Reed-Solomon coding

Following the energy dispersal randomization process, systematic shortened Reed-Solomon encoding shall be performed on each randomized MPEG-2 transport packet, with $T = 8$. This means that 8 erroneous bytes per transport packet can be corrected. This process adds 16 parity bytes to the MPEG-2 transport packet to give a codeword (204, 188).

NOTE – RS 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:

$$\lambda = 02_{\text{HEX}}$$

Field Generator Polynomial: $p(x) = x^8 + x^4 + x^3 + x^2 + 1$

The shortened Reed-Solomon code shall be implemented by appending 51 bytes, all set to zero, before the information bytes at the input of a (255, 239) encoder; after the coding procedure these bytes are discarded.

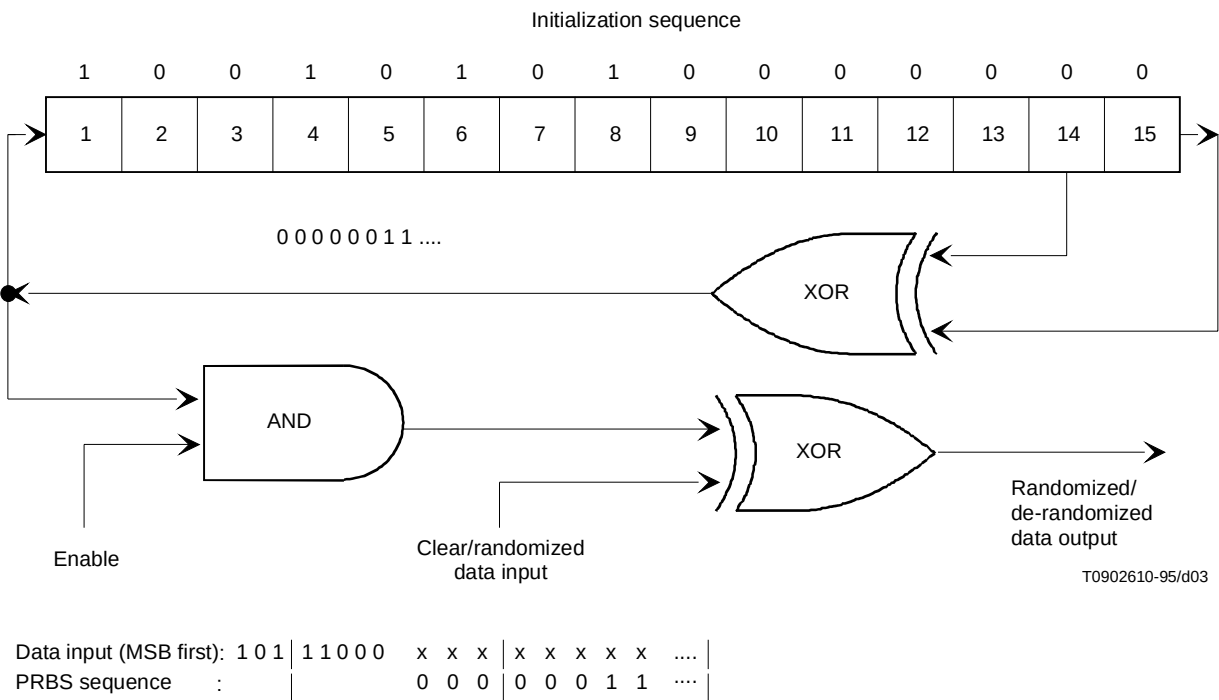


Figure A.3/J.83 – Scrambler/descrambler schematic diagram

A.5.3 Convolutional interleaving

Following the scheme of Figure A.4, convolutional interleaving with depth $I = 12$ shall be applied to the error-protected packets [see Figure A.2 c)]. This results in an interleaved frame [see Figure A.2 d)].

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 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 – 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.

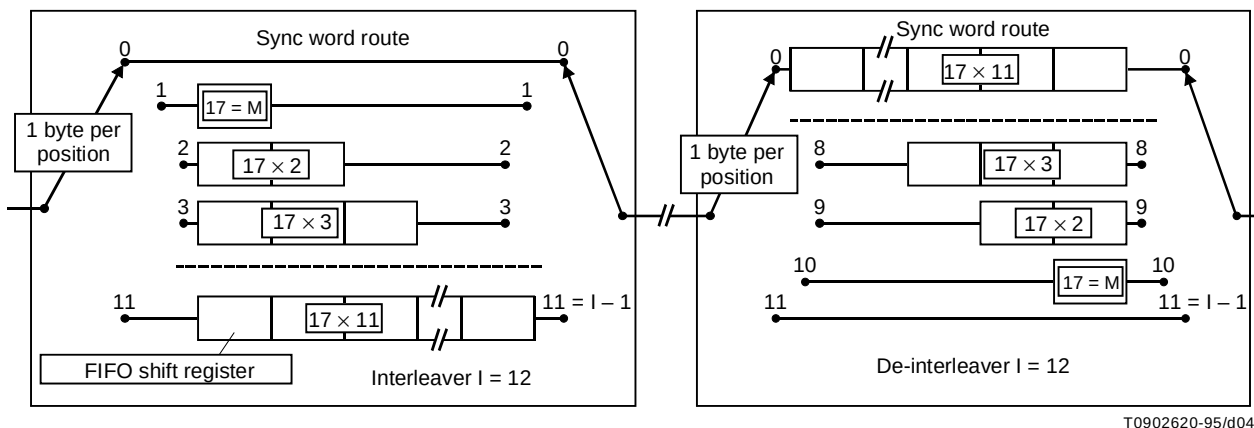


Figure A.4/J.83 – Conceptual diagram of the convolutional interleaver and de-interleaver

A.6 Byte to symbol mapping

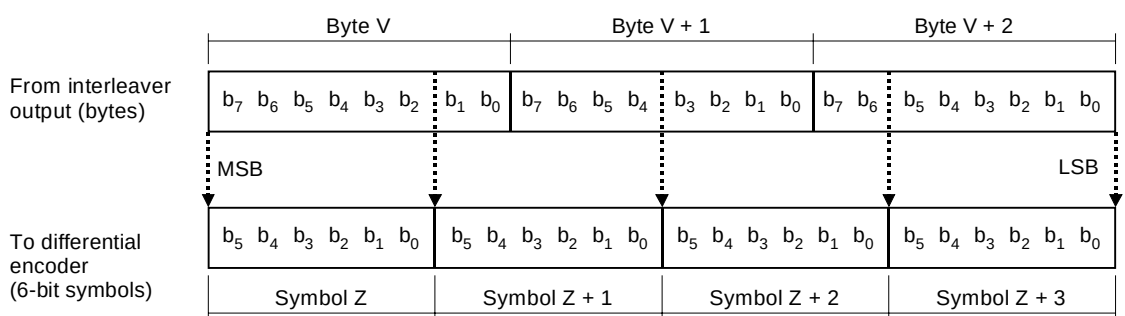
After convolutional interleaving, an exact mapping of bytes into symbols shall be performed. The mapping shall rely on the use of byte boundaries in the modulation system.

In each case, the MSB of symbol Z shall be taken from the MSB of byte V.

Correspondingly, the next significant bit of the symbol shall be taken from the next significant bit of the byte. For the case of 2^m -QAM modulation, the process shall map k bytes into n symbols, such that:

$$8k = n \cdot m$$

The process is illustrated for the case of 64-QAM (where $m = 6$, $k = 3$ and $n = 4$) in Figure A.5:



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NOTE 1– b_0 shall be understood as being the Least Significant Bit (LSB) of each byte or m-tuple.

NOTE 2 – In this conversion, each byte results in more than one m-tuple, labelled Z, Z + 1, etc. with Z being transmitted before Z + 1.

Figure A.5/J.83 – Byte to m-tuple conversion for 64-QAM

The two most significant bits of each symbol shall then be differentially coded in order to obtain a $\pi/2$ rotation-invariant QAM constellation. The differential encoding of the two MSBs shall be given by the following expression:

$$I_k = \overline{(A_k \oplus B_k)} \cdot (A_k \oplus I_{k-1}) + (A_k \oplus B_k) \cdot (A_k \oplus Q_{k-1})$$

$$Q_k = \overline{(A_k \oplus B_k)} \cdot (B_k \oplus Q_{k-1}) + (A_k \oplus B_k) \cdot (B_k \oplus I_{k-1})$$

Figure A.6 gives an example of implementation of byte to symbol conversion.

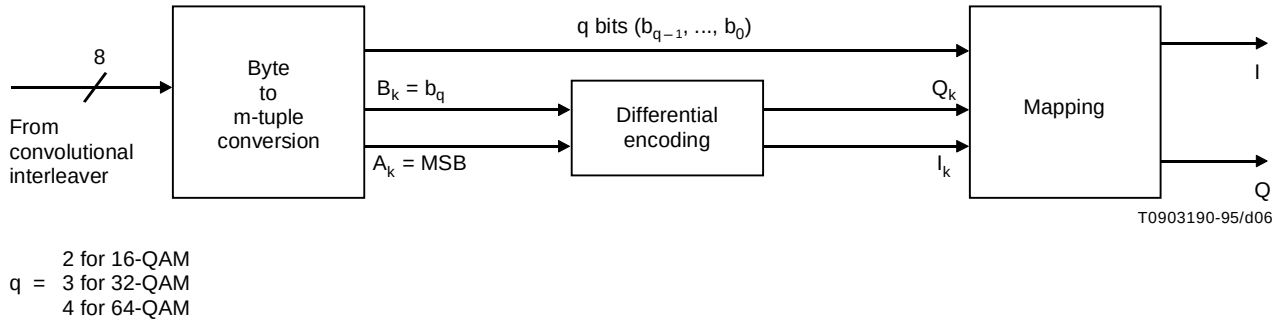


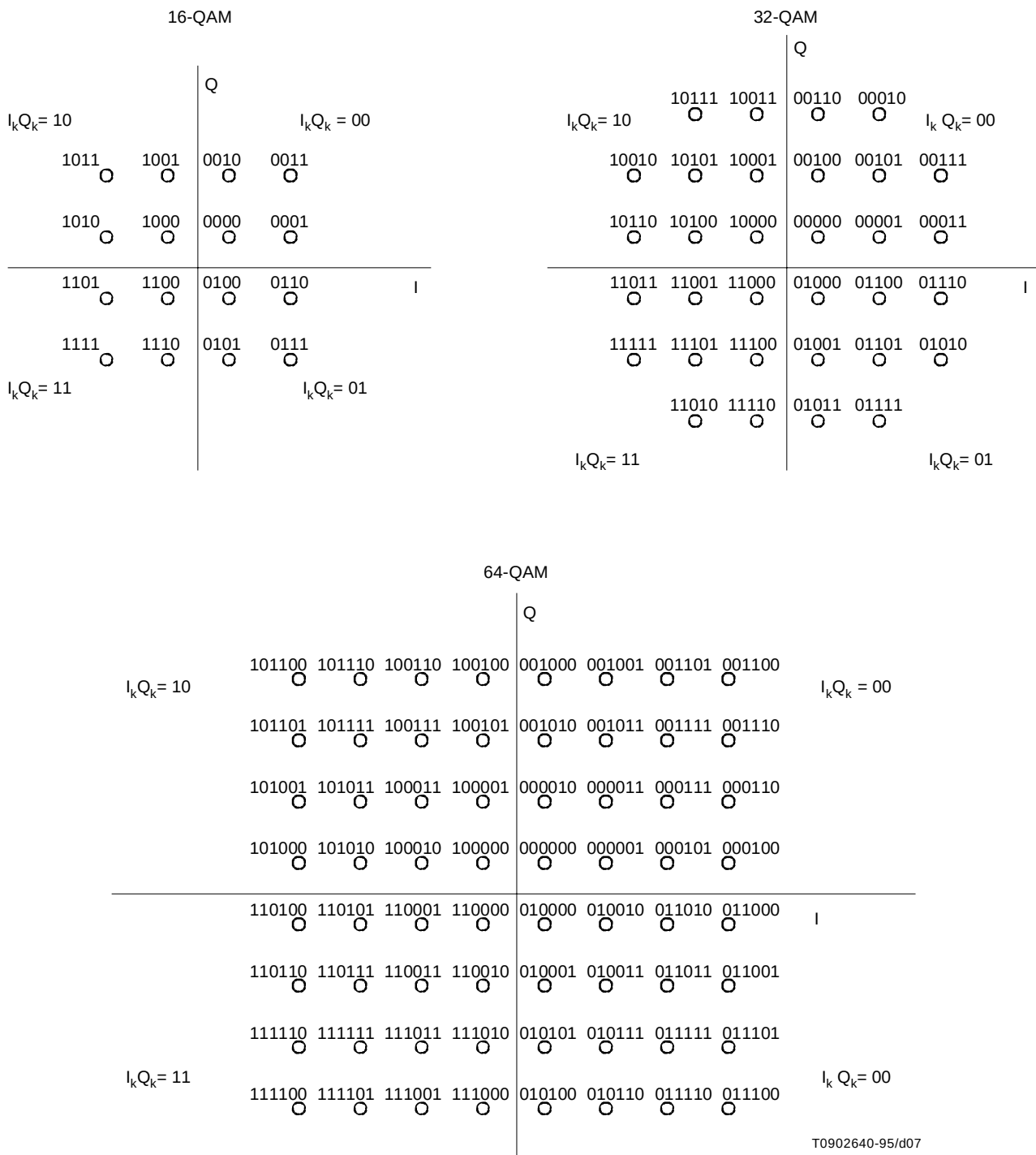
Figure A.6/J.83 – Example of an implementation of the byte to m-tuple conversion and the differential encoding of the two MSBs

A.7 Modulation

The modulation of the System shall be Quadrature Amplitude Modulation (QAM) with 16, 32, or 64 points in the constellation diagram.

The System constellation diagrams for 16-QAM, 32-QAM and 64-QAM are given in Figure A.7.

As shown in Figure A.7, the constellation points in Quadrant 1 shall be converted to Quadrants 2, 3 and 4 by changing the two MSB (i.e. I_k and Q_k) and by rotating the q LSBs according to the rule given in Table A.1.



NOTE – $I_k Q_k$ are the two MSBs in each quadrant.

Figure A.7/J.83 – Constellation diagrams for 16-QAM, 32-QAM and 64-QAM

Table A.1/J.83 – Conversion of constellation points of quadrant 1 to other quadrants of the constellation diagram given in Figure A.7

Quadrant	MSBs	LSBs rotation
1	00	
2	10	$+\pi/2$
3	11	$+\pi$
4	01	$+3\pi/2$
NOTE – Receivers shall support at least 64-QAM modulation.		

Prior to modulation, the I and Q signals shall be square-root raised-cosine filtered. The roll-off factor shall be 0.15.

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

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

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

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

where:

$$f_N = \frac{1}{2 T_s} = \frac{R_s}{2} \text{ is the Nyquist frequency and roll-off factor } \alpha = 0.15.$$

The transmitter filter characteristic is given in A.8.

A.8 Baseband filter characteristics

The template given in Figure A.8 shall be used as a minimum requirement for hardware implementation of the Nyquist filter. This template takes into account not only the design limitations of the digital filter, but also the artefacts coming from the analogue processing components of the System (e.g. D/A conversion, analogue filtering, etc).

The value of in-band ripple r_m in the pass-band up to $(1 - \alpha) f_N$ as well as at the Nyquist frequency f_N shall be lower than 0.4 dB. The out-of-band rejection shall be greater than 43 dB.

The filter shall be phase-linear with the group delay ripple $\leq 0.1 T_s$ (ns) up to f_N ,

where:

$$T_s = \frac{1}{R_s} \text{ is the symbol period.}$$

NOTE – The values for in-band ripple and out-of-band rejection given in this Annex are subject to further study.

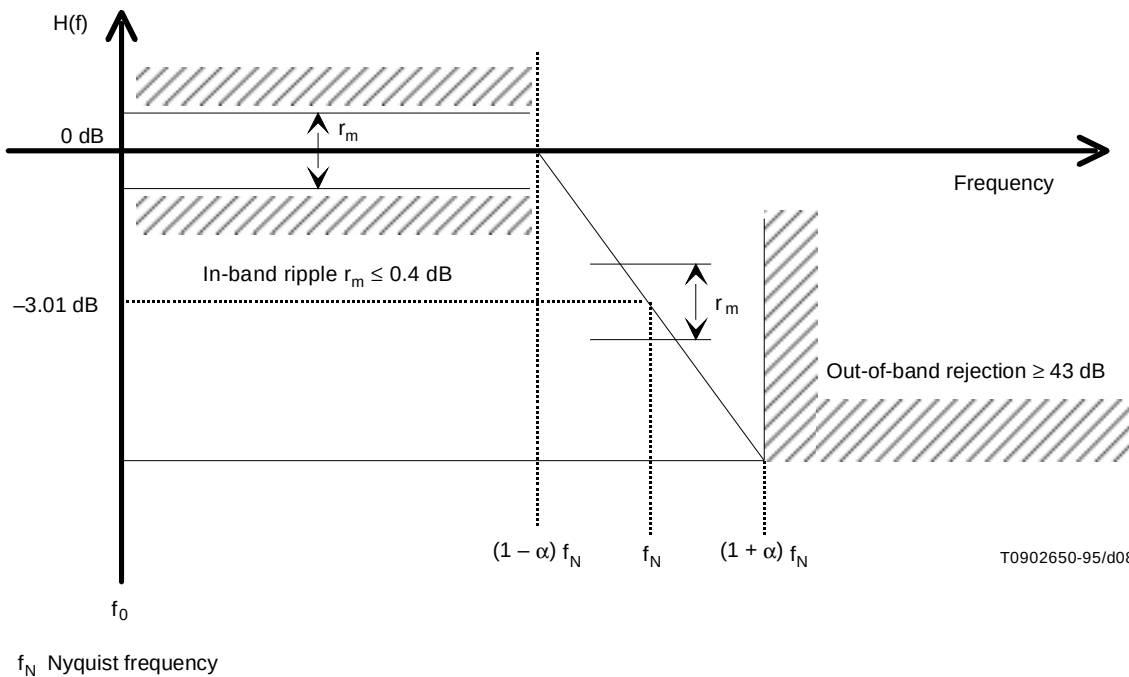


Figure A.8/J.83 – Half-Nyquist baseband filter amplitude characteristics

Annex B

Digital multi-programme System B

B.1 Introduction

This Annex describes the framing structure, channel coding, and channel modulation for a digital multi-service television distribution system that is specific to a cable channel. The system can be used transparently with the distribution from a satellite channel, as many cable systems are fed directly from satellite links. The specification covers both 64- and 256-QAM. Most features of both modulation schemes are the same. Where there are differences, the specific details for each modulation scheme will be covered.

The design of the modulation, interleaving and coding is based upon testing and characterization of cable systems in North America. The modulation is Quadrature Amplitude Modulation with a 64-point signal constellation (64-QAM) and with a 256-point signal constellation (256-QAM), transmitter selectable. The Forward Error Correction (FEC) is based on a concatenated coding approach that produces high coding gain at moderate complexity and overhead. Concatenated coding offers improved performance over a block code, at a similar overall complexity. The system FEC is optimized for quasi error free operation at a threshold output error event rate of one error event per 15 minutes.

The data format input to the modulation and coding is assumed to be MPEG-2 transport. However, the method used for MPEG synchronization is decoupled from FEC synchronization. For example, this enables the system to carry ATM packets easily without interfering with ATM synchronization. In fact, ATM synchronization may be performed by defined ATM synchronization mechanisms.

There are two modes supported: Mode 1 has a symbol rate of 5.057 Msymbols/s and Mode 2 has a symbol rate of 5.361 Msymbols/s. Typically, Mode 1 will be used for 64-QAM and Mode 2 will be used for 256-QAM. The system will be compatible with future implementations of higher data rate schemes employing higher order QAM extensions.

B.2 Cable system concept

Channel coding and transmission are specific to a particular medium or communication channel. The expected channel error statistics and distortion characteristics are critical in determining the appropriate error correction and demodulation. The cable channel, including optical fiber, is primarily regarded as a bandwidth-limited linear channel, with a balanced combination of white noise, interference, and multi-path distortion. The Quadrature Amplitude Modulation (QAM) technique used, together with adaptive equalization and concatenated coding is well suited to this application and channel.

The basic layered block diagram of cable transmission processing is shown in Figure B.1. The following subclauses define these layers from the "outside" in, and from the perspective of the transmit side.

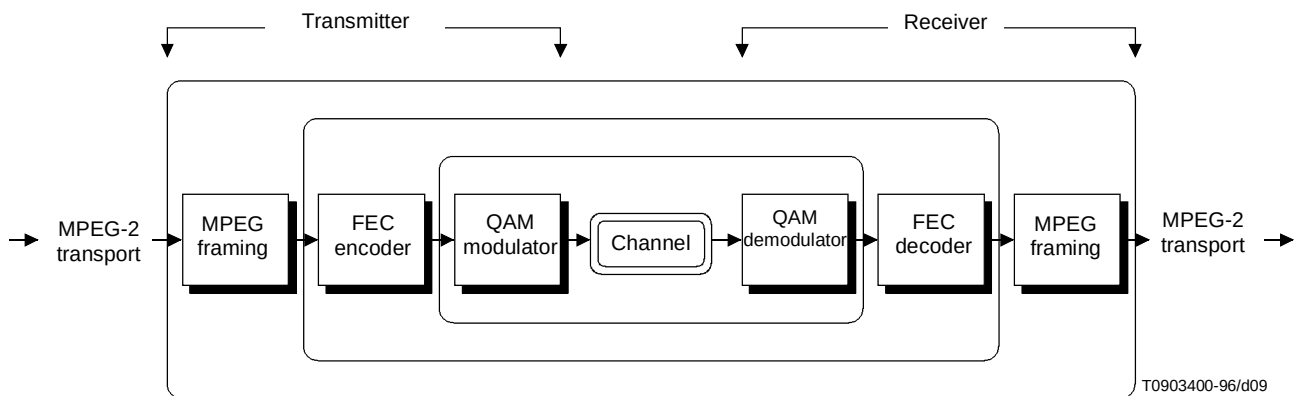


Figure B.1/J.83 – Cable transmission block diagram

B.3 MPEG-2 transport layer

The MPEG-2 transport layer is defined in Reference [2]. The transport layer for MPEG-2 data is comprised of packets having 188 bytes, with one byte for synchronization purposes, three bytes of header containing service identification, scrambling and control information, followed by 184 bytes of MPEG-2 or auxiliary data.

B.4 MPEG-2 transport framing

The MPEG transport framing is the outermost layer of processing. It is provided as a robust means of delivering MPEG packet synchronization to the receiver output. This processing block receives an MPEG-2 transport data stream consisting of a continuous stream of fixed length 188 byte packets. This data stream is transmitted in serial fashion, MSB first. The first byte of a packet is specified to be a sync byte having a value of 47_{HEX}.

The sync byte is intended for the purpose of packet delineation. The cable transmission system has incorporated an additional layer of processing to provide an additional functionality by utilizing the information bearing capacity of this sync byte. A parity checksum which is a coset of an FIR parity check linear block code is substituted for this sync byte, supplying improved packet delineation functionality, and error detection capability independent of the FEC layer.

The parity checksum is computed over the adjacent 187 bytes, which constitute the immediately preceding MPEG-2 packet contents (minus sync byte). It is then possible to support simultaneous packet synchronization and error detection. The decoder computes a sliding checksum on the serial data stream, using the detection of a valid code word to detect the start of packet. Once a locked alignment condition is established, the absence of a valid code word at the expected location will indicate a packet error. The error flag of the previous packet may optionally be set as the data is passed out of the decoder. The normal sync word must be re-inserted in place of the checksum to provide a standard MPEG-2 data stream as an output.

The syndrome is computed by passing the 1496 payload bits through a Linear Feedback Shift Register (LFSR) as described by the following equation:

$$f(x) = [1 + x^{1497}b(x)]/g(x)$$

where:

$$g(x) = 1 + x + x^5 + x^6 + x^8; \text{ and}$$

$$b(x) = 1 + x + x^3 + x^7.$$

This computational structure is illustrated in Figures B.2 and B.3. All addition operations are assumed to be modulo 2. For an encode operation, the LFSR is first initialized so that all memory elements contain zero value. The 1496 bits which constitute the MPEG-2 transport stream packet payload are then shifted into the LFSR. The encoder input is set to zero after the 1496 data bits are received, and eight additional shifts are required to sequentially output the eight computed syndrome bits. This 8-bit result must then be passed through an additional FIR filtering function $g(x)$ (initialized to an all-zeros state prior to introduction of the 8 syndrome bits) to generate an encoder checksum. An offset of 67_{HEX} is added to this checksum result for improved autocorrelation properties, and causes a 47_{HEX} result to be produced during a syndrome decode operation when a valid code word is present. The final 8-bit checksum with added offset is transmitted MSB first following the 1496 payload bits to implement a systematic encoder.

A parity check matrix may be used by the decoder to identify a valid checksum. A syndrome generator, as shown in Figure B.3, may also be employed for this purpose. The code has been designed such that when the appropriate 188 bytes of the modified MPEG-2 transport stream packet (which includes the associated checksum) are multiplied with the parity check matrix, a valid code word is indicated when the calculated product produces a 47_{HEX} result. Each of the 8 columns of the parity check matrix "P" includes a 1497 bit vector, hereafter referred to as "C". This vector is defined in Figure B.4.

Proceeding from the leftmost column of the matrix "P", the 1497-bit column "C" is duplicated in subsequent columns of the matrix "P", shifted down by one bit position. The bit positions unoccupied by the column data are filled with zeros, as illustrated in Figure B.5.

Note that the checksum is calculated based on the previous 187 bytes and not the 187 bytes yet to be received by the MPEG-2 sync decoder. This is in contrast to the conventional notion of an MPEG packet structure, in that the sync byte is usually described as the first byte of a received packet.

The received vector "R" is the MPEG-2 data consisting of 187 bytes followed by the checksum byte, yielding a total of 1504 bits. This "R" vector is multiplied (modulo 2) by the parity check "P" matrix, yielding an "S" vector whose length is 8-bits, as illustrated in Figure B.6.

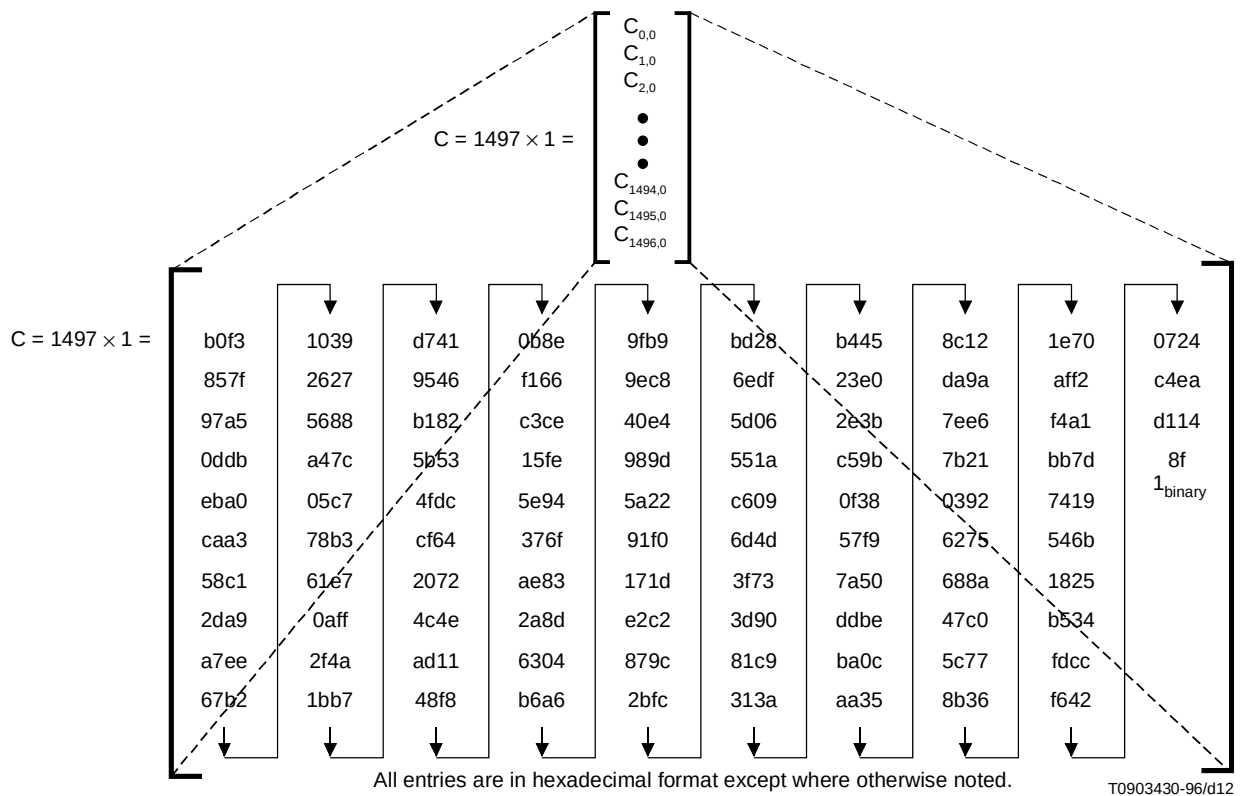


Figure B.4/J.83 – "C" column vector (replicated inside the parity check matrix)

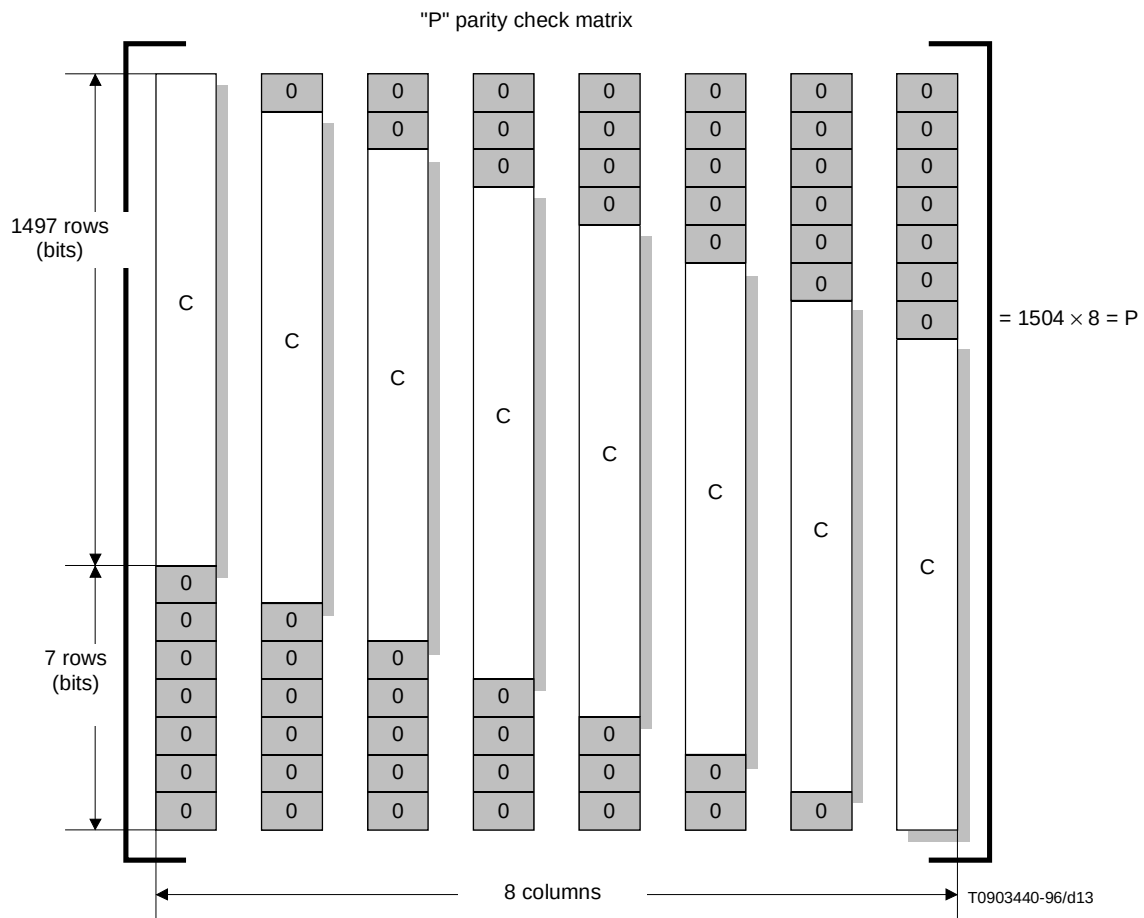


Figure B.5/J.83 – Structure of the parity check matrix "P"

$$\begin{bmatrix} \text{R "Vector"} \\ \text{(Alignment window)} \\ 1 \times 1504 \end{bmatrix} \times \begin{bmatrix} \text{P "Matrix"} \\ \text{(Parity check)} \\ 1504 \times 8 \end{bmatrix} = \begin{bmatrix} \text{S "Vector"} \\ \text{(Received checksum)} \\ 1 \times 8 \end{bmatrix}$$

$S = [0100 \ 0111] = 0 \times 47$

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Figure B.6/J.83 – Received MPEG-2 vector and parity check matrix multiplication

A valid checksum is indicated when $S = [0100, 0111] = 47_{\text{HEX}}$.

For carriage of transport protocols other than MPEG-2 Transport, e.g. ATM, this outer layer is removed or bypassed. The FEC layer accepts and delivers data without any constraints on protocol. The framing section could be replaced with one appropriate to the alternative transport protocol if required by an application. All other portions of this specification (modulation, coding, interleaving) are implemented as described below. For the case of ATM, no framing layer is required. The ATM HEC typically provides adequate packet framing and error detection. Isochronous ATM streams are therefore carried transparently without overhead for MPEG or quasi-MPEG packet encapsulation.

B.5 Forward error correction

The Forward Error Correction (FEC) definition is composed of four processing layers, as illustrated in Figure B.7. There are no dependencies on input data protocol in any of the FEC layers. FEC synchronization is fully internal and transparent. Any data sequence will be delivered from the encoder input to decoder output.

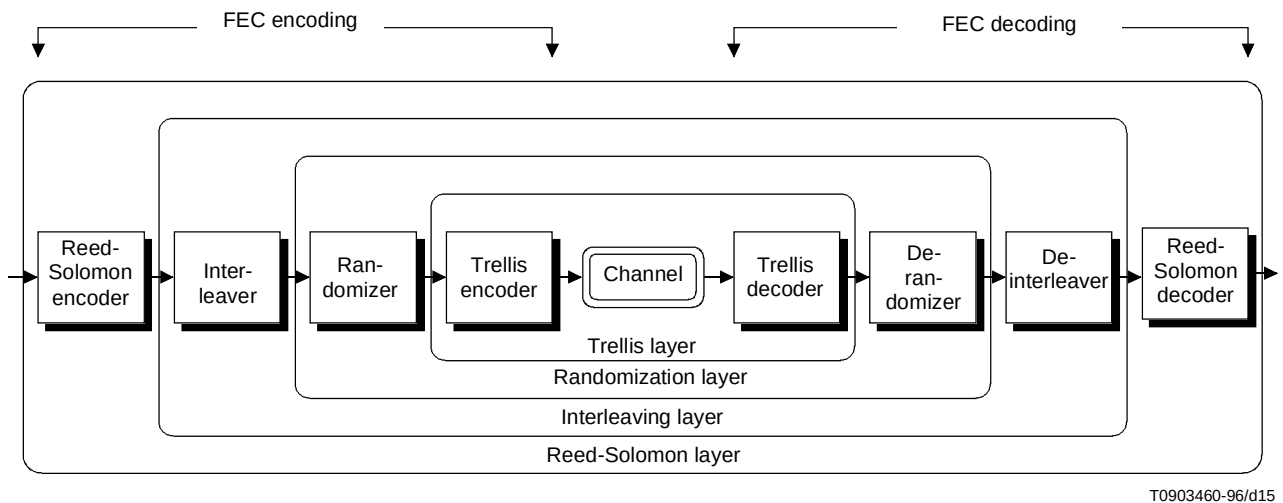


Figure B.7/J.83 – Layers of processing in the FEC

The FEC section uses various types of error correcting algorithms and interleaving techniques to transport data reliably over the cable channel.

- Reed-Solomon (RS) Coding – Provides block encoding and decoding to correct up to three symbols within an RS block.
- Interleaving – Evenly disperses the symbols, protecting against a burst of symbol errors from being sent to the RS decoder.
- Randomization – Randomizes the data on the channel to allow effective QAM demodulator synchronization.
- Trellis Coding – Provides convolutional encoding and with the possibility of using soft decision trellis decoding of random channel errors.

The following subclauses define these 4 layers.

B.5.1 Reed-Solomon coding

The MPEG-2 transport stream is Reed-Solomon (RS) encoded using a (128, 122) code over GF(128). This code has the capability of correcting up to $t = 3$ symbol errors per RS block. The same RS code is used for both 64-QAM and 256-QAM. However, the FEC frame format is different for each modulation type, as described in a later subclause.

The Reed-Solomon encoder implementation is described in this subclause. A systematic encoder is utilized to implement a $t = 3$, (128,122) extended Reed-Solomon code over GF(128). The primitive polynomial used to form the field over GF(128) is:

$$p(x) = x^7 + x^3 + 1$$

where:

$$p(\alpha) = 0.$$

The generator polynomial used by the encoder is:

$$\begin{aligned} g(x) &= (x + \alpha)(x + \alpha^2)(x + \alpha^3)(x + \alpha^4)(x + \alpha^5) \\ &= x^5 + \alpha^{52} x^4 + \alpha^{116} x^3 + \alpha^{119} x^2 + \alpha^{61} x + \alpha^{15} \end{aligned}$$

The message polynomial input to the encoder consists of 122, 7-bit symbols, and is described below:

$$m(x) = m_{121} x^{121} + m_{120} x^{120} + \dots + m_1 x + m_0$$

This message polynomial is first multiplied by x^5 , then divided by the generator polynomial $g(x)$ to form a remainder, described by the following:

$$r(x) = r_4 x^4 + r_3 x^3 + r_2 x^2 + r_1 x + r_0$$

This remainder constitutes five parity symbols which are then added to the message polynomial to form a 127-symbol code word that is an even multiple of the generator polynomial.

The generated code word is now described by the following polynomial:

$$c(x) = m_{121} x^{126} + m_{120} x^{125} + m_{119} x^{124} + \dots + r_4 x^4 + r_3 x^3 + r_2 x^2 + r_1 x + r_0$$

A valid code word will have roots at the first through fifth powers of α .

An extended parity symbol (c_-) is generated by evaluating the code word at the sixth power of α .

$$c_- = c(\alpha^6)$$

This extended symbol is used to form the last symbol of a transmitted Reed-Solomon block. The extended code word then appears as follows:

$$\begin{aligned} \hat{c} &= xc(x) + c_- \\ &= m_{121} x^{127} + m_{120} x^{126} + \dots + m_1 x^7 + m_0 x^6 + r_4 x^5 + r_3 x^4 + r_2 x^3 + r_1 x^2 + r_0 x + c_- \end{aligned}$$

The structure of a Reed-Solomon block which illustrates the order of transmitted symbols output from the RS encoder is shown below:

$$m_{121}m_{120}m_{119}\dots m_1m_0r_4r_3r_2r_1r_0c_- \quad (\text{order sent is left to right})$$

B.5.2 Interleaving

Interleaving is included in the modem between the RS block coding and the randomizer to enable the correction of burst noise induced errors. In both 64-QAM and 256-QAM a convolutional interleaver is employed.

Convolutional interleaving is illustrated in Figure B.8. At the start of an FEC frame defined in a subsequent subclause, the interleaving commutator position is initialized to the top-most branch and increments at the RS symbol frequency, with a single symbol output from each position. With a convolutional interleaver, the RS code symbols are sequentially shifted into the bank of I registers (the width of each register is 7 bits which matches the RS symbol size). Each successive register has J symbols more storage than the preceding register. The first interleaver path has zero delay, the second has a J symbol period of delay, the third $2 \cdot J$ symbol periods of delay, and so on, upto the I^{th} path which has $(I - 1) \cdot J$ symbol periods of delay. This is reversed for the de-interleaver in the Cable Decoder such that the net delay of each RS symbol is the same through the interleaver and de-interleaver. Burst noise in the channel causes a series of bad symbols. These are spread over many RS blocks by the de-interleaver such that the resultant symbol errors per block are within the range of the RS decoder correction capability.

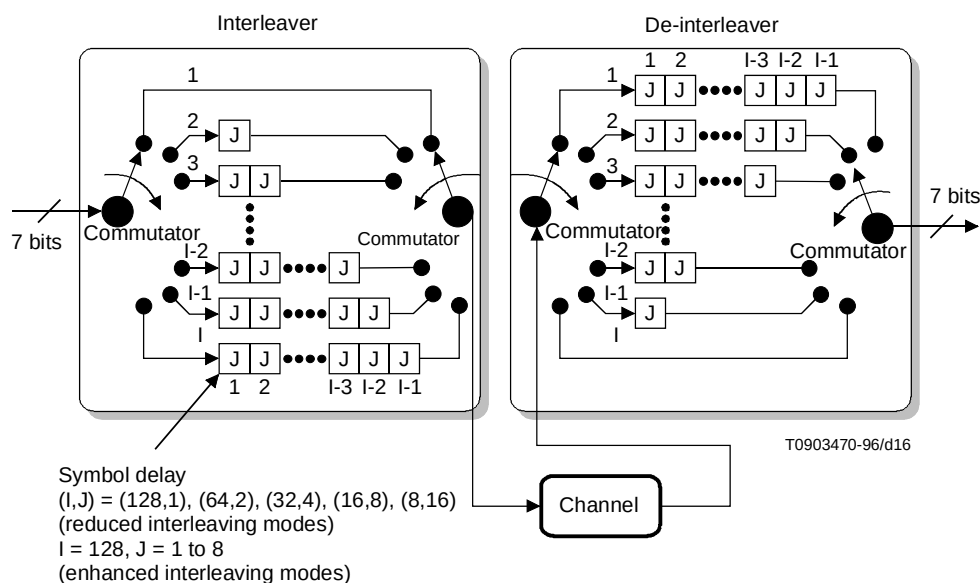


Figure B.8/J.83 – Interleaving functional block diagram

With regard to interleaving capability, two distinct operating modes are specified, hereafter referred to as *level 1* and *level 2*.

Level 1 is specified for 64-QAM transmission only. This mode accommodates the installed base of legacy 64-QAM-only digital set tops. While operating in level 1, a single interleaving depth will be supported; namely $I = 128, J = 1$.

Level 2 shall encompass 64-QAM and 256-QAM transmission, and will for both modulation schemes be capable of supporting variable interleaving. This will include both enlarged and reduced interleaving depths relative to the nominal 64-QAM (level 1) configuration. Four data bits are transmitted in-band during the FEC frame sync interval to convey the interleaving parameters to the receiver for a given channel.

Table B.1 describes the interleaver parameters for level 1 operation, with associated latency and burst protection. Table B.2 describes the decoding of the 4-bit in-band control word into the I and J interleaving parameters for level 2 operation, also with associated burst protection and latency.

Table B.1/J.83 – Level 1 interleaving

Control word (4 bits)	I (# of taps)	J (increment)	Burst protection	Latency
xxxx	128	1	95 μ s	4.0 ms

Table B.2/J.83 – Level 2 interleaving

Control word (4 bits)	I (# of taps)	J (increment)	Burst protection 64-QAM/256-QAM	Latency 64-QAM/256-QAM
0001	128	1	95 μ s /66 μ s	4.0 ms/2.8 ms
0011	64	2	47 μ s /33 μ s	2.0 ms/1.4 ms
0101	32	4	24 μ s /16 μ s	0.98 ms/0.68 ms
0111	16	8	12 μ s /8.2 μ s	0.48 ms/0.33 ms
1001	8	16	5.9 μ s /4.1 μ s	0.22 ms/0.15 ms
1011	Reserved			
1101	Reserved			
1111	Reserved			
0000	128	1	95 μ s /66 μ s	4.0 ms/2.8 ms
0010	128	2	190 μ s /132 μ s	8.0 ms/5.6 ms
0100	128	3	285 μ s /198 μ s	12 ms/8.4 ms
0110	128	4	379 μ s /264 μ s	16 ms/11 ms
1000	128	5	474 μ s /330 μ s	20 ms/14 ms
1010	128	6	569 μ s /396 μ s	24 ms/17 ms
1100	128	7	664 μ s /462 μ s	28 ms/19 ms
1110	128	8	759 μ s /528 μ s	32 ms/22 ms

B.5.3 Frame synchronization sequence

The frame synchronization sequence trailer delineates the FEC frame, providing synchronized RS coding, interleaving, and randomization. Additionally, trellis groups for 256-QAM only are aligned with the FEC frame. The FEC framing does not perform MPEG packet of trellis decoder synchronization. The RS block and 7-bit symbol structures are aligned with the end of the frame for both 64- and 256-QAM.

For 64-QAM, an FEC frame consists of a 42-bit sync trailer which is appended to the end of 60 RS blocks, with each RS block containing 128 symbols. Each RS symbol consists of 7 bits. Thus, there is a total of 53 760 data bits and 42 frame sync trailer bits in this FEC frame. The first 4 7-bit symbols of the frame sync trailer contain the 28-bit "unique" synchronization pattern (1110101 0101100 0001101 1101100) or (75 2C 0D 6C)_{HEX}. The remaining 2 symbols (14 bits) are utilized as follows: first 4 bits for interleaver mode control, and 10 bits are reserved and set to zero. The frame sync trailer is inserted by the encoder and detected at the decoder. The decoder circuits search for this pattern and determine the location of the frame boundary and interleaver depth mode when found. The FEC frame for 64-QAM is shown in Figure B.9.

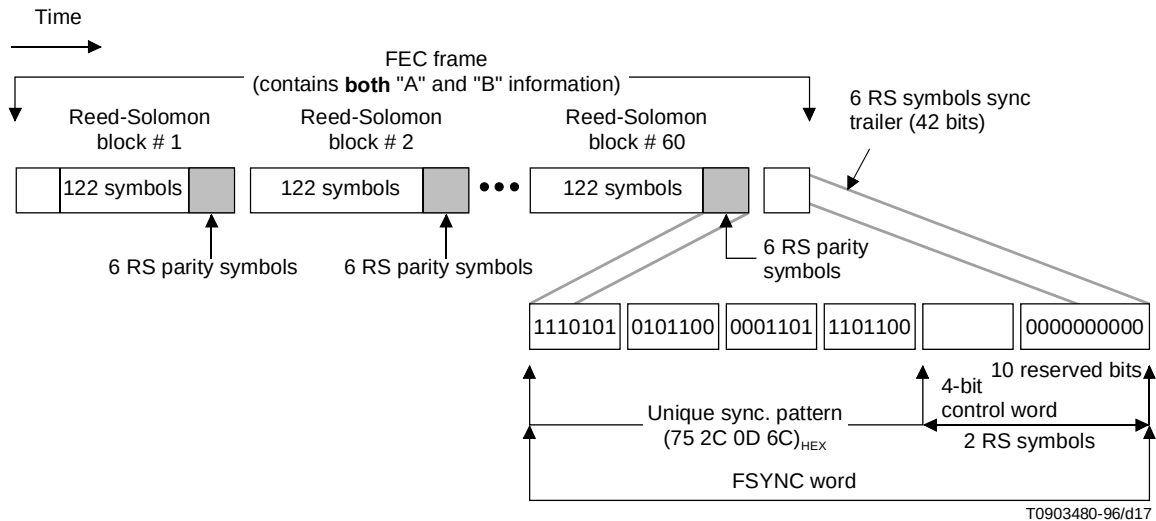


Figure B.9/J.83 – Frame packet format for 64-QAM

For 256-QAM, an FEC frame consists of a 40-bit sync trailer which is appended to the end of 88 RS blocks, with each RS block containing 128 symbols. Each RS symbol consists of 7 bits. Thus, there is a total of 78 848 data bits and 40 frame sync trailer bits in this FEC frame. The 40-bit frame sync trailer is divided as follows: 32 bits are the "unique" synchronization pattern (0111 0001 1110 1000 0100 1101 1101 0100) or (71 E8 4D D4)_{HEX}, 4 bits are a control word which determine the size of the interleaver employed, and 4 bits are a reserved word which is set to zero. The FEC frame for 256-QAM is shown in Figure B.10.

Note that there is no synchronization relationship between the transmitted RS block and transport data packets. Thus, MPEG-2 transport stream packet synchronization is obtained independently from RS frame synchronization. This keeps the FEC and transport layers de-coupled and independent.

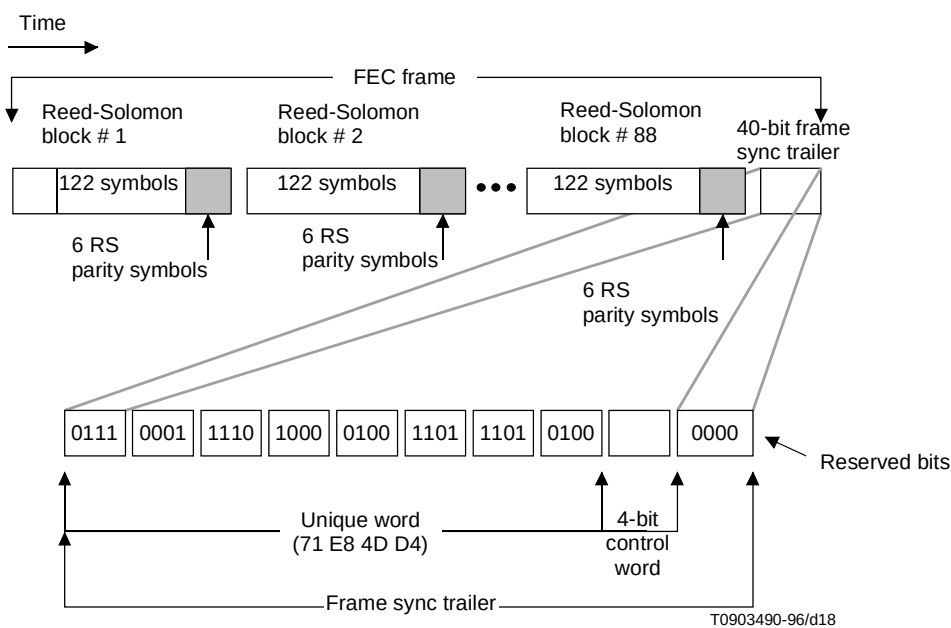


Figure B.10/J.83 – Frame packet format for 256-QAM

B.5.4 Randomization

The randomizer shown in Figure B.11 is the third layer of processing in the FEC block diagram. The randomizer provides for even distribution of the symbols in the constellation, which enables the demodulator to maintain proper lock. The randomizer adds a Pseudorandom Noise (PN) sequence of 7-bit symbols over GF(128) (i.e. bit-wise exclusive-OR) to the symbols within the FEC frame to assure a random transmitted sequence.

For both 64- and 256-QAM, the randomizer is initialized during the FEC frame trailer, and is enabled at the first symbol after the trailer. Thus the trailer itself is not randomized, and the initialized output value randomizes the first data symbol.

Initialization is defined as pre-loading to the 'all ones' state for the randomizer structure shown in Figure B.11. The randomizer uses a linear feedback shift register specified by a GF(128) polynomial defined as follows:

$$f(x) = x^3 + x + \alpha^3$$

where:

$$\alpha^7 + \alpha^3 + 1 = 0.$$

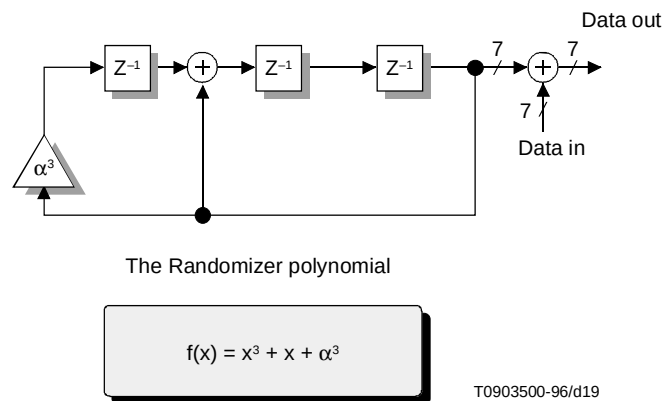


Figure B.11/J.83 – Randomizer (7-bit symbol scrambler)

B.5.5 Trellis coded modulation

As part of the concatenated coding scheme, trellis coding is employed for the inner code. It allows the introduction of redundancy to improve the threshold Signal-to-Noise Ratio (SNR) by increasing the symbol constellation without increasing the symbol rate. As such, it is more properly termed "trellis coded modulation."

B.5.5.1 64-QAM modulation mode

For 64-QAM, the input to the trellis coded modulator is a 28-bit sequence of four, 7-bit RS symbols, which are labelled in pairs of 'A' symbols, and 'B' symbols. A block diagram of a 64-QAM trellis coded modulator is shown in Figure B.12. All 28 bits are assigned to a trellis group, where each trellis group forms 5-QAM symbols, as shown in Figure B.13.

Of the 28 input bits that form a trellis group, each of two groups of 4 bits of the differentially pre-coded bit streams in a trellis group are separately encoded by a Binary Convolutional Code (BCC). Each BCC produces 5 coded bits, as shown in Figure B.12. The remaining bits are sent to the mapper uncoded. This will produce an overall output of 30 bits. Thus, the overall code rate for 64-QAM trellis coded modulation is 14/15.

The trellis group is formed from RS symbols as follows: For the "A" symbols, the RS symbols are read, from MSB to LSB, $A_{10}, A_8, A_7, A_5, A_4, A_2, A_1$ and $A_9, A_6, A_3, A_0, A_{13}, A_{12}, A_{11}$. The four MSBs of the second symbol are input to the BCC, one bit at a time, LSB first. The remaining bits of the second symbol and all the bits of the first symbol are input to the mapper, uncoded, LSB first one bit at a time. The four bits sent to the BCC will produce 5 coded bits labelled, U_1, U_2, U_3, U_4, U_5 . The same process is done for the "B" bits. The process can be seen in Figure B.12. With 64-QAM, 4 RS symbols conveniently fit into one trellis group, and in this case the sync word may occupy every bit position within a trellis group.

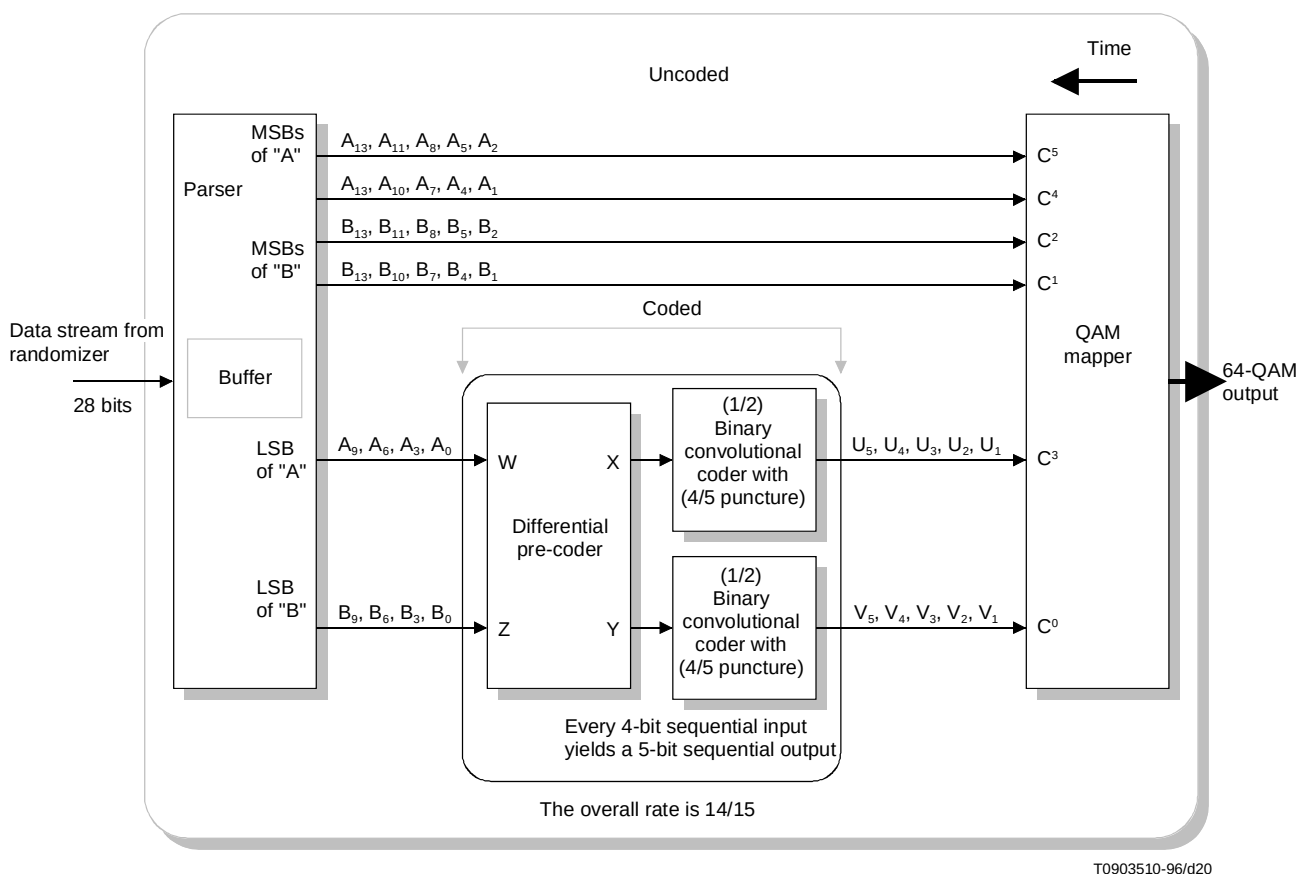


Figure B.12/J.83 – 64-QAM trellis coded modulator block diagram

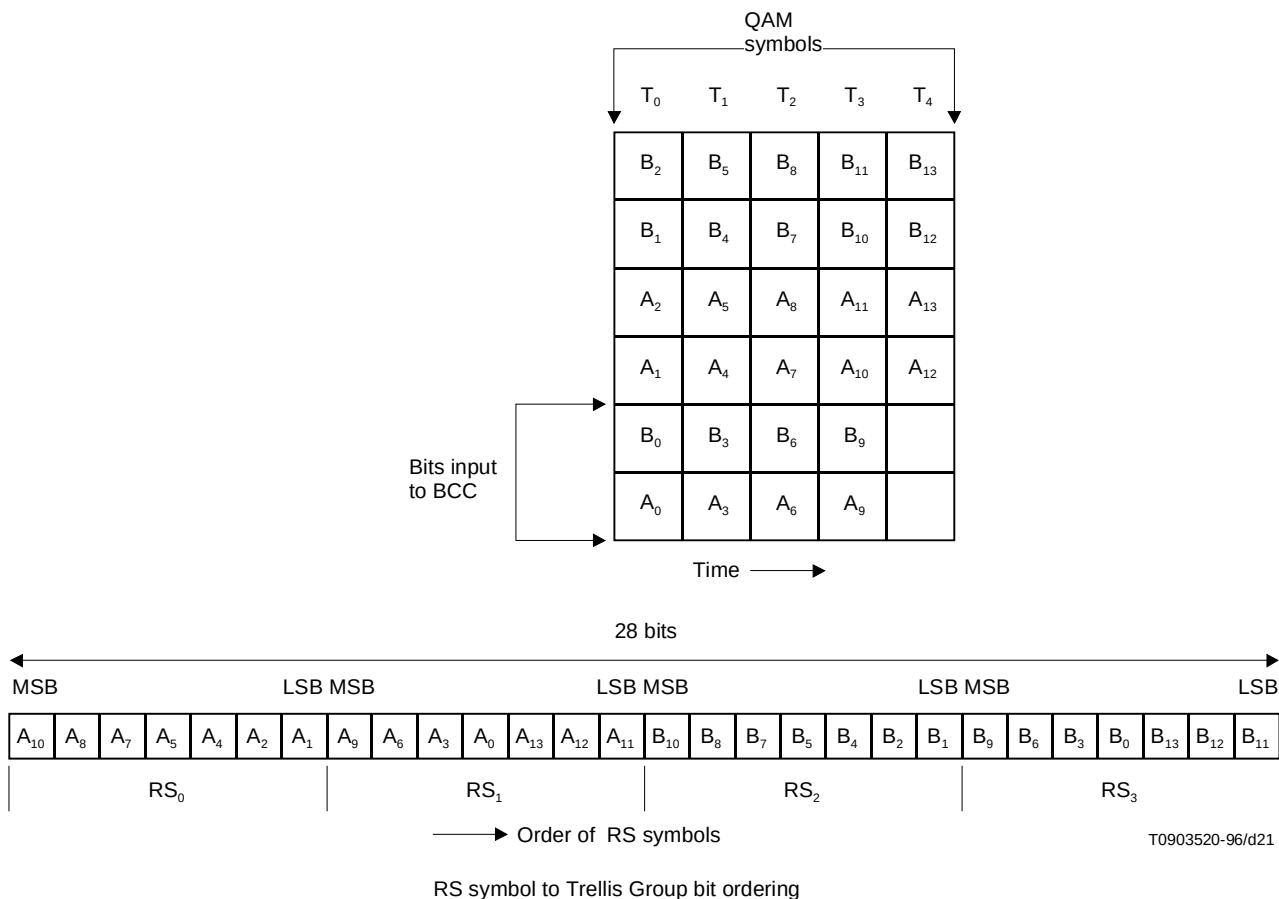


Figure B.13/J.83 – 64-QAM trellis group

B.5.5.2 256-QAM modulation mode

For 256-QAM, an analogous trellis coding is employed using the same BCC as 64-QAM, with the same rate 1/2 generator and the same 4/5 puncture matrix. The 256-QAM trellis coded modulator is shown in Figure B.14. In this case all the FEC frame sync information is embedded only in the trellis group convolutionally encoded bit positions of a trellis group as shown in Figure B.15.

There are two distinct types of trellis groups in 256-QAM: hereafter referred to as a *non-sync* group and a *sync* group. Each trellis group generates 5-QAM symbols at the modulator, the non-sync group contains 38 data bits while the sync group contains 30 data bits and 8 sync bits. Figure B.15 shows both a non-sync trellis group and a sync trellis group. Since there are 88 RS blocks plus 40 frame sync bits per FEC frame, there will be a total of 2076 trellis groups per frame. Of these trellis groups, 2071 are non-sync trellis groups and 5 are sync trellis groups. The 5 sync trellis groups come at the end of the frame. The frame sync trailer is aligned to the trellis groups. In the encoder, the trellis group is further divided into two groups: one uncoded bit stream and one coded bit stream. The MSB of the first RS symbol in the FEC frame is assigned to the first bit in the first non-sync trellis group, as shown in the ordering in Figure B.15. The output from each BCC is the five parity bits labelled U_1 through U_5 and V_1 through V_5 , respectively, as shown in Figure B.14.

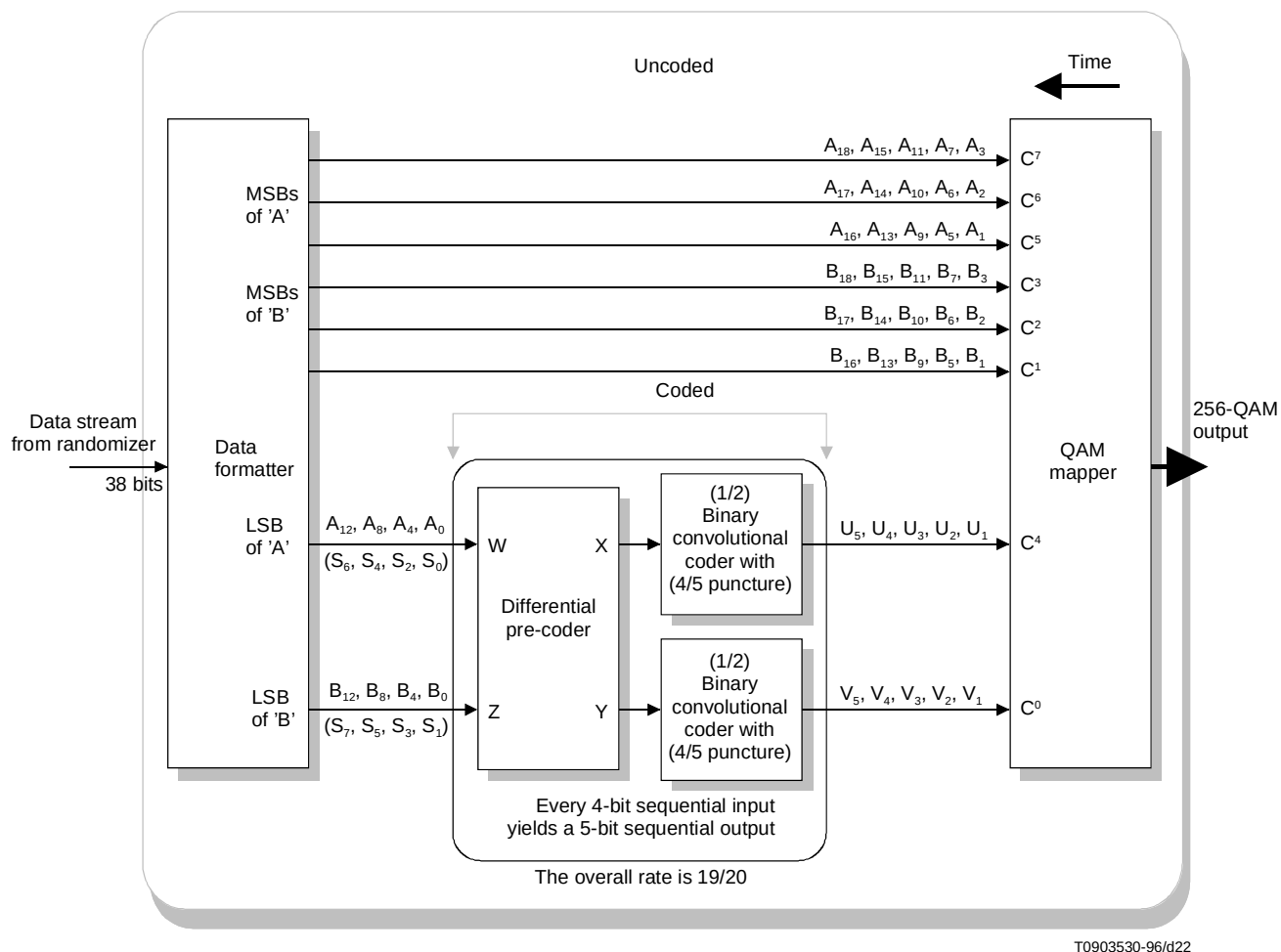


Figure B.14/J.83 – 256-QAM trellis coded modulator block diagram

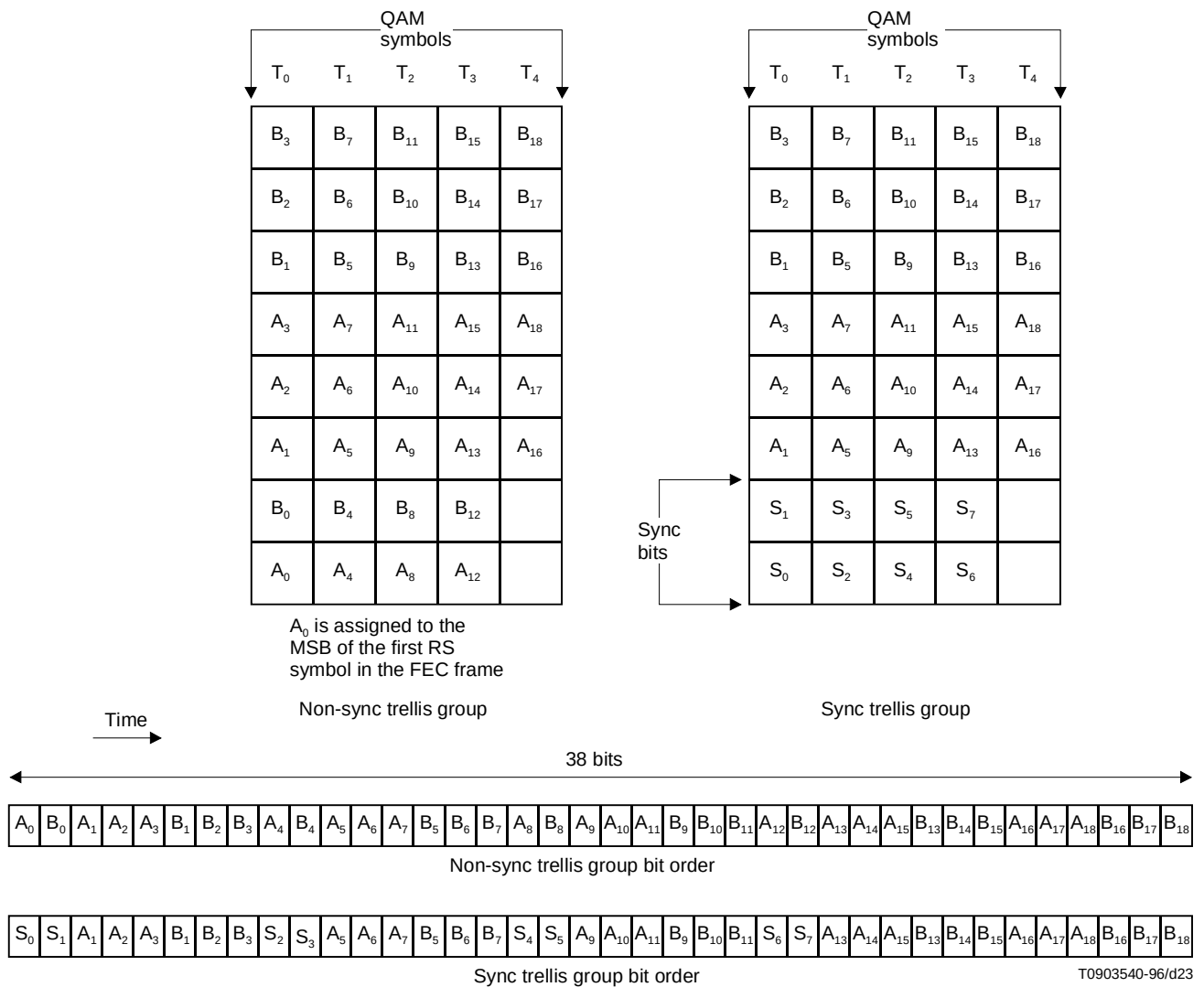


Figure B.15/J.83 – 256-QAM sync and non-sync trellis groups

To form trellis groups from RS code words, the RS code words are serialized beginning with the MSB of the first symbol of the first RS code word following the frame sync trailer. Bits are placed into trellis group locations from RS symbols in the order: A₀ B₀ A₁ ... B₃ A₄ B₄ ... B₁₆ B₁₇ B₁₈ as shown in Figure B.15. For sync trellis groups, the bits from serialized RS symbols are placed beginning at location A₁ instead of A₀. The last five trellis groups in an FEC frame each contain 8 of the 40 sync bits, S₀ S₁ ... S₇ in the frame sync trailer shown in Figure B.10.

Of the 38 input bits that form a trellis group, each of two groups of 4 bits of type differentially pre-coded bit streams in a trellis group are separately encoded by a Binary Convolutional Coder (BCC). Each BCC produces 5 coded bits, as shown in Figure B.14. The remaining bits are sent to the QAM mapper uncoded. This produces a total output of 40 bits per trellis group. Thus, the overall code rate for 256-QAM trellis coded modulation is 19/20.

B.5.5.3 Rotationally invariant pre-coding

The differential pre-coder shown in Figure B.16 performs the 90° rotationally invariant trellis coding. Rotationally invariant coding is employed for both 64- and 256-QAM modulation. The key for robust modem design is to have very fast recovery from carrier phase slips. Non-rotationally invariant coding requires resynchronization of the FEC when the carrier phase tracking changes quadrant alignment, leading to a burst of errors at the FEC output.

The differential pre-coder allows the information to be carried by the change in phase, rather than by the absolute phase. For 64-QAM, the 3rd and the 6th bits of the 6-bit symbols are differentially encoded, and for 256-QAM, the 4th and 8th bits are differentially encoded. If you mask out the 3rd and the 6th bits in 64-QAM as in Figure B.18 (labelled C^3 and C^0) and the 4th and 8th bits in 256-QAM as in Figure B.19 (labelled C^4 and C^0), the 90° rotational invariance of the remaining bits is inherent in the labelling of the symbol constellation.

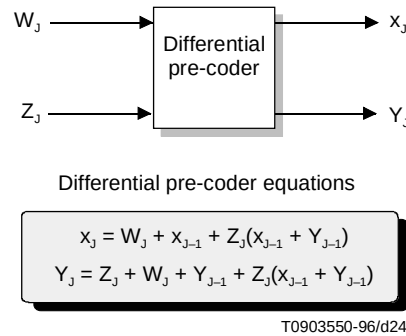


Figure B.16/J.83 – Differential pre-coder

B.5.5.4 Binary Convolutional Coder

The trellis coded modulator includes a punctured rate 1/2 binary convolutional encoder that is used to introduce the redundancy into the LSBs of the trellis group. The convolutional encoder is a 16-state non-systematic rate 1/2 encoder with the generator: $G1 = 010\ 101$, $G2 = 011\ 111$ (25,37_{octal}), or equivalently the generator matrix $[1 \oplus D^2 \oplus D^4, 1 \oplus D \oplus D^2 \oplus D^3 \oplus D^4]$. At the beginning of a trellis group, the BCC commutator is initially in the G1 position. For each input bit presented to the tapped delay line, two bits (G1, followed by G2) are subsequently produced at the output in accordance with the associated set of generator coefficients. For each trellis group, 4 input bits produce 8 convolutionally encoded bits. This time output of the encoder is selected according to a puncture matrix: $[P1, P2] = [0001; 1111]$ ("0" denotes NO transmission, "1" denotes transmission), which produces a single serial bit stream. The puncture matrix essentially converts the rate 1/2 encoder to rate 4/5, since only 5 of the 8 encoded bits are retained after puncturing. The internal structure of the punctured encoder is illustrated in Figure B.17.

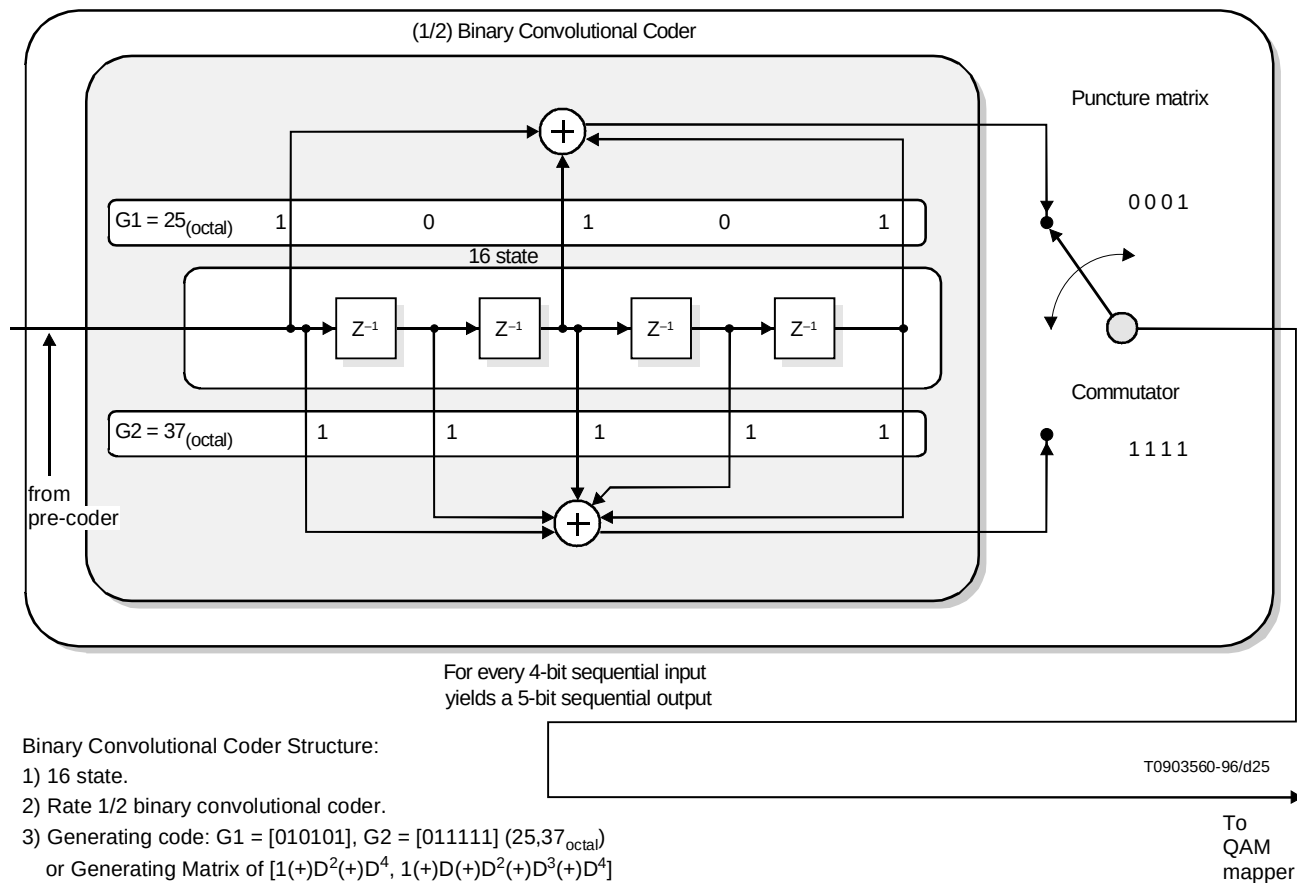
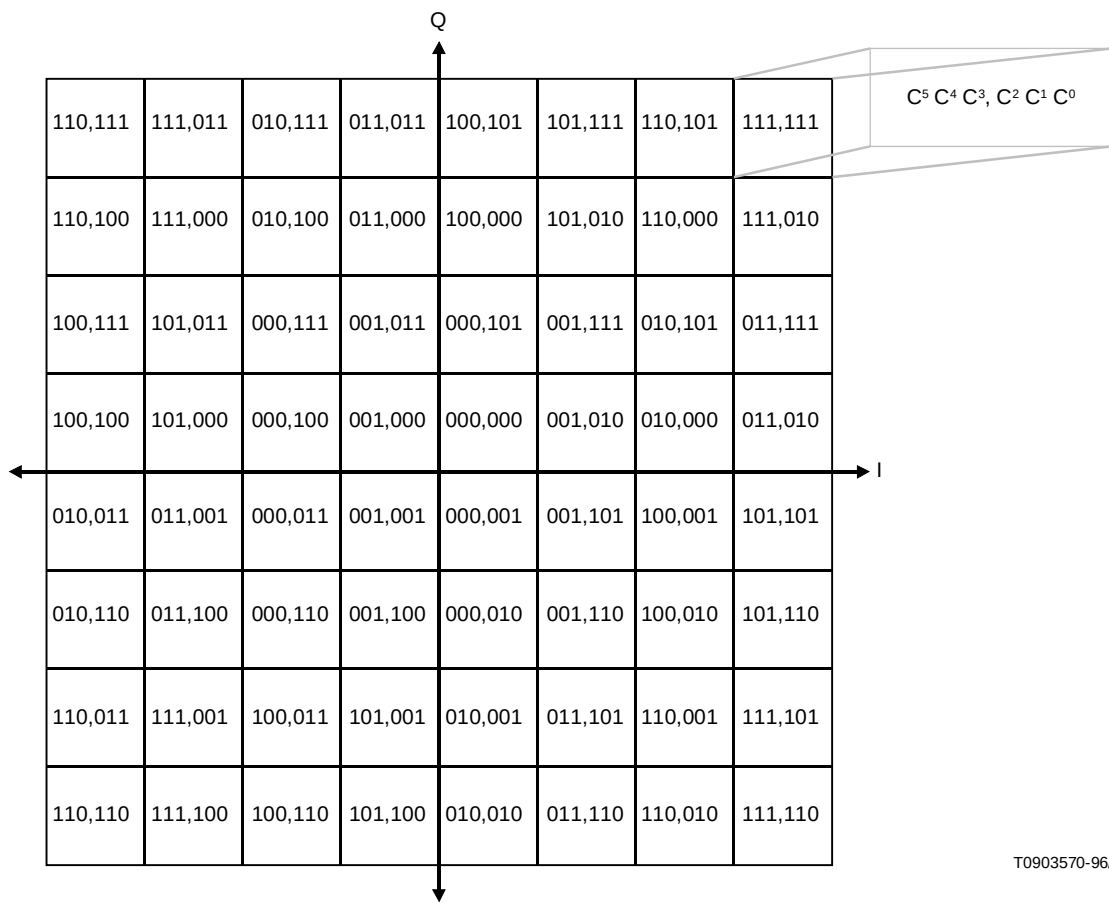


Figure B.17/J.83 – Punctured Binary Convolutional Coder

B.5.5.5 QAM constellation mapping

For 64-QAM, the QAM mapper receives the coded and uncoded 3-bit 'A' and 'B' data from the trellis coded modulator. It uses these bits to address a look-up table which produces the 6-bit constellation symbol. The 6-bit constellation symbol is then sent to the 64-QAM modulator where the signal constellation illustrated in Figure B.18 is generated.

For 256-QAM, the QAM mapper receives the coded and uncoded 4-bit 'A' and 'B' data from the trellis coded modulator. It uses these bits to address a look-up table which produces the 8-bit constellation symbol. The 8-bit constellation symbol is then sent to the 256-QAM modulator where the signal constellation illustrated in Figure B.19 is generated.



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Figure B.18/J.83 – 64-QAM constellation

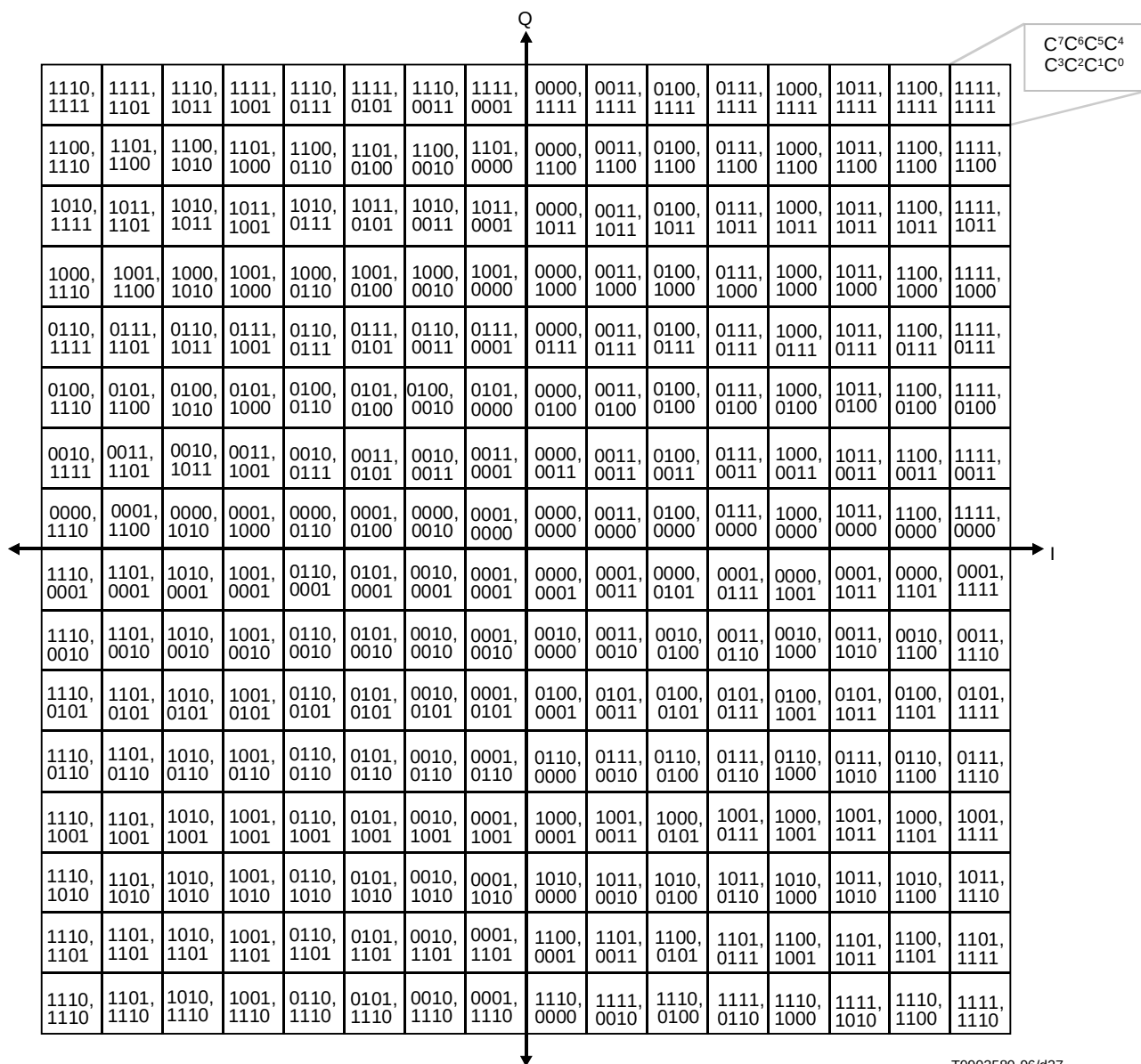


Figure B.19/J.83 – 256-QAM constellation

B.6 Modulation and demodulation

B.6.1 QAM characteristics

The cable transmission format is summarized in Table B.3 for 64-QAM and 256-QAM. Table B.4 contains a summary of the pertinent characteristics of the variable interleaving modes.

Table B.3/J.83 – Cable transmission format

Parameter	64-QAM format	256-QAM format
Modulation	64-QAM, rotationally invariant coding	256-QAM, rotationally invariant coding
Symbol size	3 bits for "I" and 3 bits for "Q" dimensions	4 bits for "I" and 4 bits for "Q" dimensions
Transmission band	54 to 860 MHz (Note)	54 to 860 MHz (Note)
Channel spacing	6 MHz (Note)	6 MHz (Note)
Symbol rate	5.056941 Msps $\pm$ 5 ppm (Note)	5.360537 Msps $\pm$ 5 ppm (Note)
Information bit rate	26.97035 Mbps $\pm$ 5 ppm (Note)	38.81070 Mbps $\pm$ 5 ppm (Note)
Frequency response	Square root raised cosine filter (Roll-off $\approx$ 0.18)	Square root raised cosine filter (Roll-off $\approx$ 0.12)
FEC framing	42-bit sync trailer following 60 RS blocks (see B.5.3)	40-bit sync trailer following 88 RS blocks (see B.5.3)
QAM constellation mapping	6 bits per symbol (see B.5.5)	8 bits per symbol (see B.5.5)
NOTE – These values are specific to 6 MHz channel spacing. Additional sets of values for differing channel spacing are under study.		

Table B.4/J.83 – Variable interleaving modes

	Level 1	Level 2
QAM format	64-QAM (see Table B.3)	64- or 256-QAM (see Table B.3)
Interleaving	Fixed interleaving (see B.5.2) I = 128 J = 1	Variable interleaving (see B.5.2) I = 128,64,32,16,8 J = 1,2,3,4,5,6,7,8,16

B.6.2 QAM modulator RF output

The 64-QAM and 256-QAM modulator RF output specifications for a 75 Ω cable impedance are shown in Table B.5.

Table B.5/J.83 – QAM modulator RF output

Parameter	Specification
Input power level range	–9 to +16 dBmV
I/Q Phase offset	< 1.0°
I/Q Crosstalk	$\geq$ –50 dB
I/Q Amplitude imbalance	0.05 dB max.
I/Q Timing skew	< 3.0 nsec.

Annex C

Digital multi-programme System C

C.1 Introduction

This Annex describes the framing structure, channel coding and modulation of digital multi-programme system for cable distribution.

The system employs the transport multiplexing based on MPEG-2 (see Reference [2]), guaranteeing interoperability with other media such as digital broadcasting, ISDN networks or packaged media. The framing structure and the channel coding are the same as in Annex A. The modulation is 64-QAM, and the QAM symbol rate and the roll-off factor are optimized for the 6 MHz channel plan.

The field experiment using a 64-QAM receiver with an equalizer was carried out in Japan. As the results of the experiment, quasi-error-free operation was confirmed.

The system also allows for further evolution to higher order QAM constellations, and the appropriate modifications to its channel coding and symbol mapping are currently under study.

C.2 Cable system concept

The cable system shall be defined as the functional block of equipment performing the adaptation of the baseband TV signals to the cable channel characteristics.

In the cable head-end, the TV baseband signal may come from broadcasting, second distribution links, contribution links and local programme sources.

The following process shall be applied as shown in Figure C.1: System configuration.

C.2.1 Baseband interfacing and sync

This unit shall adapt the data structure to the format of the signal source. The framing structure shall be in accordance with MPEG-2 transport layer including sync bytes.

C.2.2 Sync 1 inversion and randomization

This unit shall invert the MPEG-2 Sync byte (Sync 1) every eight packets, according to the MPEG-2 framing structure, and shall randomize the data stream for spectrum shaping purposes.

C.2.3 Reed-Solomon (RS) coder

This unit shall apply a shortened Reed-Solomon (RS) code to each randomized transport packet to generate an error-protected packet. This code shall also be applied to the Sync byte itself.

C.2.4 Convolutional interleaver

This unit shall perform a depth $I = 12$ convolutional interleaving of the error-protected packets. The periodicity of the sync bytes shall remain unchanged.

C.2.5 Byte to m-tuple conversion

This unit shall perform a conversion of the bytes generated by the interleaver into QAM symbols.

C.2.6 Differential encoding

In order to get a rotation-invariant constellation, this unit shall apply a differential encoding of the two Most Significant Bits (MSBs) of each symbol.

C.2.7 QAM modulation and physical interface

This unit performs a square-root raised cosine filtering of the I and Q signals prior to QAM modulation. This is followed by interfacing the QAM modulated signal to the Radio Frequency (RF) cable channel.

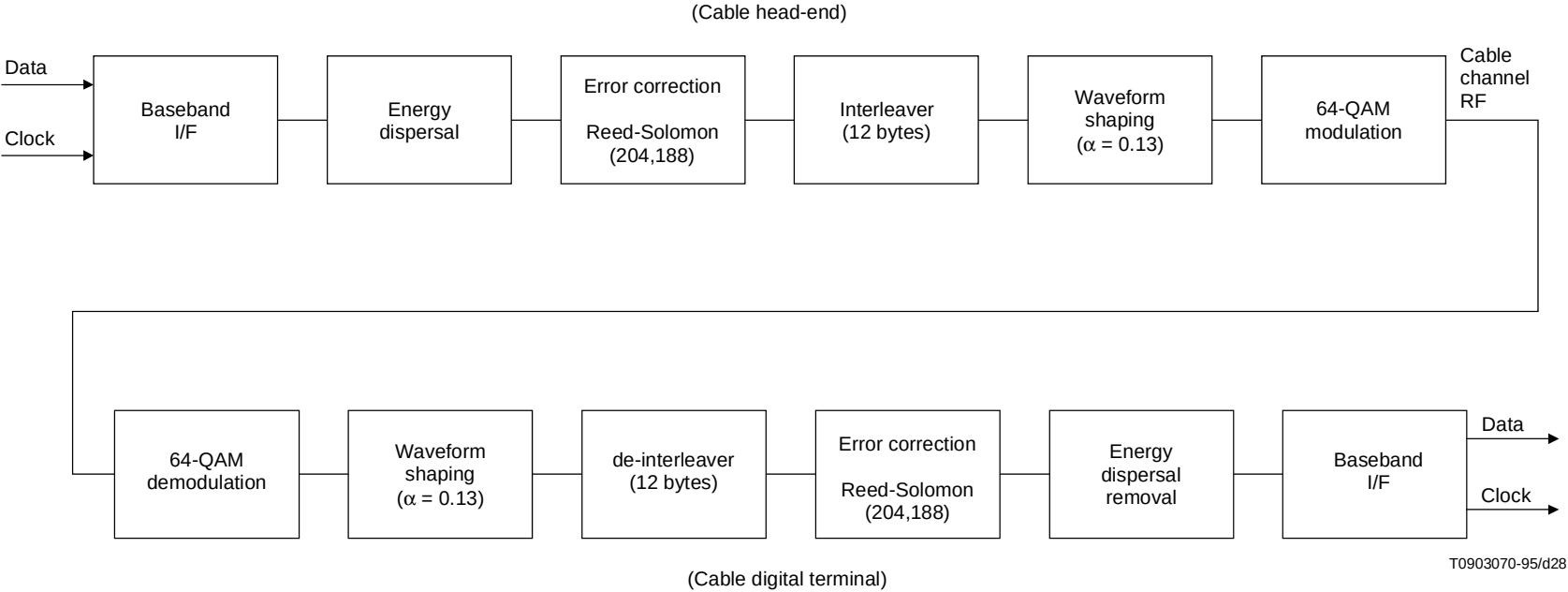


Figure C.1/J.83 – System configuration

C.2.8 Cable receiver

A System receiver shall perform the inverse signal processing, as described for the modulation process above, in order to recover the baseband signal.

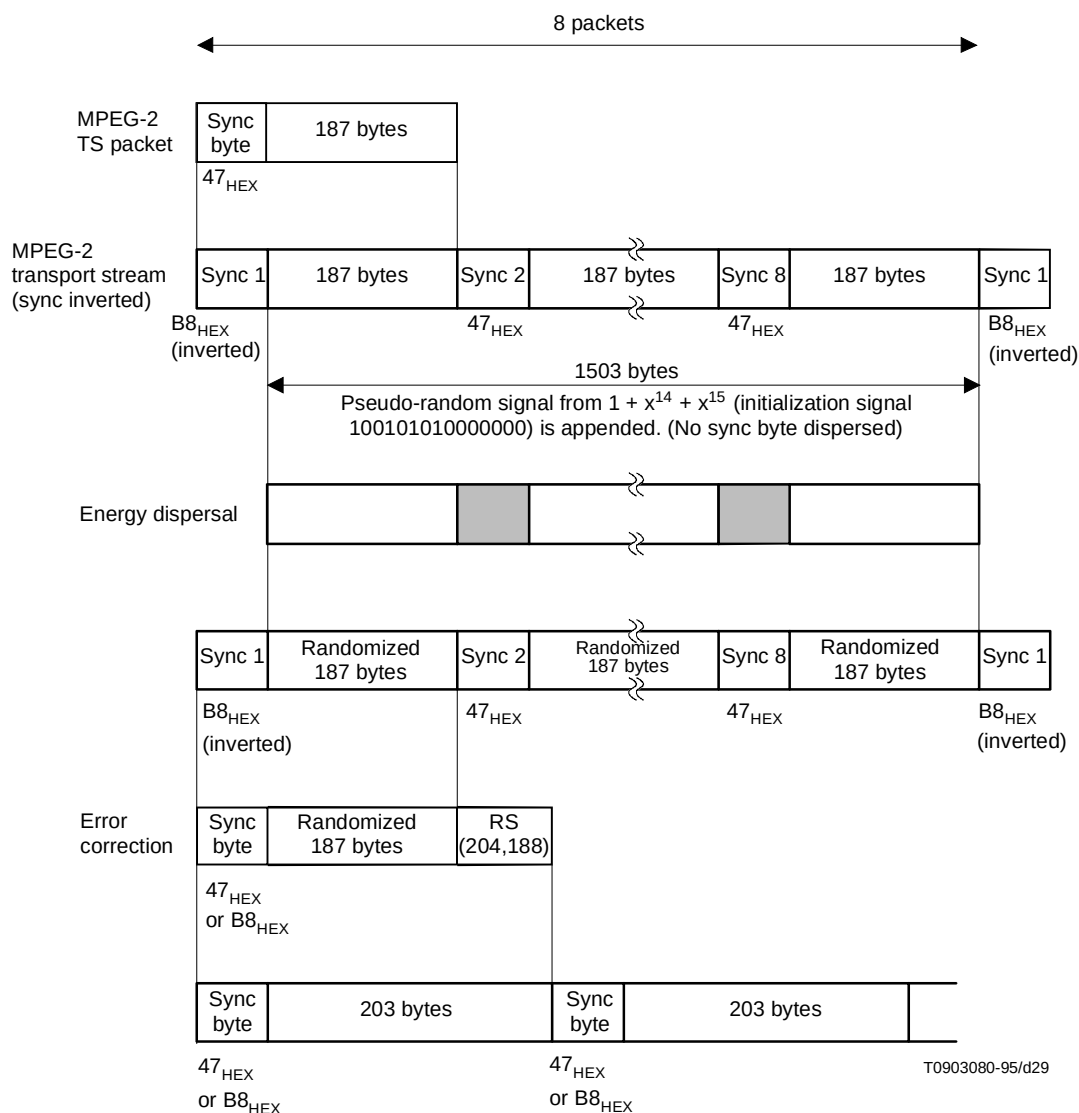
In addition, each cable receiver should install an equalizer to prevent increase of the bit-error caused by the reflection in the cable system.

C.3 MPEG-2 transport layer

The transport layer for the digital multi-programme system is based on MPEG-2 (see Reference [2]). The transport multiplexing is performed in Transport Stream-Packet having 188 bytes, in conformance with MPEG-2.

C.4 Framing structure

The framing organization shall be based on the MPEG-2 transport packet structure. The System framing structure is shown in Figure C.2: Transmission signal configuration.



Interleaving: Convolutional interleaving (by byte unit).
No delay in sync byte.

Figure C.2/J.83 – Transmission signal configuration

Code Generator Polynomial:

$$g(x) = (x + \lambda^0)(x + \lambda^1)(x + \lambda^2) \dots (x + \lambda^{15});$$

where:

$$\lambda = 02_{\text{HEX}}$$

Field Generator Polynomial:

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

C.5.3 Convolutional interleaving

Following the scheme of Figure C.4, convolutional interleaving with depth $I = 12$ shall be applied to the error-protected packets.

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

C.6 Modulation

C.6.1 Byte to symbol mapping

After convolutional interleaving, an exact mapping of bytes into symbols shall be performed. The mapping shall rely upon the use of byte boundaries in the modulation system.

In each case, the MSB of symbol Z shall be taken from the MSB of byte V . Correspondingly, the next significant bit of the symbol shall be taken from the next significant bit of the byte. For the case of 2^m -QAM modulation, the process shall map k bytes into n symbols, such that;

$$8k = n \cdot m$$

The process is illustrated for the case of 64-QAM (when $m = 6$, $k = 3$ and $n = 4$) in Figure C.5.

C.6.2 Differential encoding

The two MSBs of each symbol shall then be differentially coded in order to obtain a $\pi/2$ rotation-invariant QAM constellation. The differential encoding of the two MSBs shall be given by the following expression:

$$\begin{aligned} I_k &= \overline{(A_k \oplus B_k)} \cdot (A_k \oplus I_{k-1}) + (A_k \oplus B_k) \cdot (A_k \oplus Q_{k-1}) \\ Q_k &= \overline{(A_k \oplus B_k)} \cdot (B_k \oplus Q_{k-1}) + (A_k \oplus B_k) \cdot (B_k \oplus I_{k-1}) \end{aligned}$$

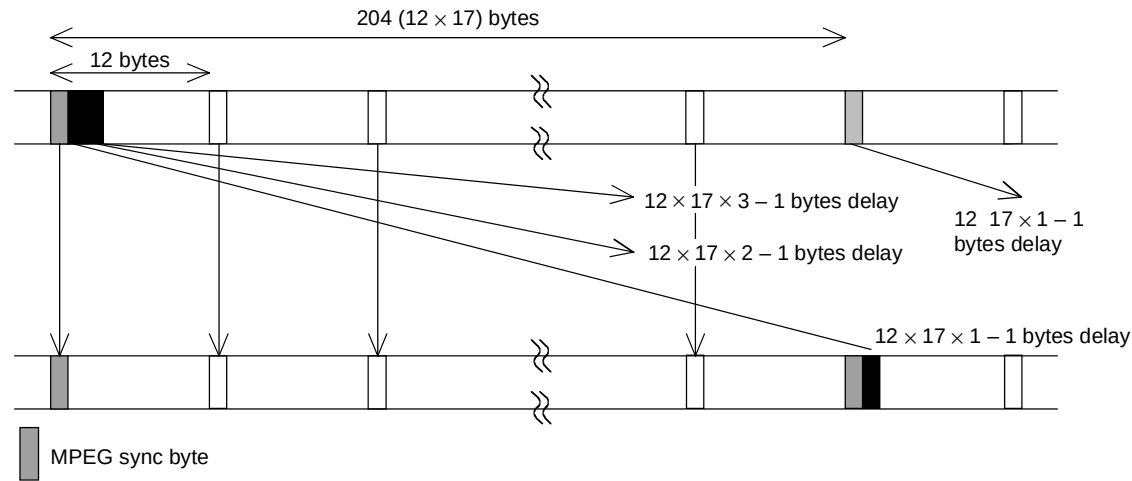
Figure C.6 gives an example of implementation of byte to symbol conversion.

C.6.3 64-QAM constellation

The system can be adapted to 6 MHz channel spacing. The byte to modulation scheme described in this subclause is directly related to the byte to symbol mapping method given in C.6.1.

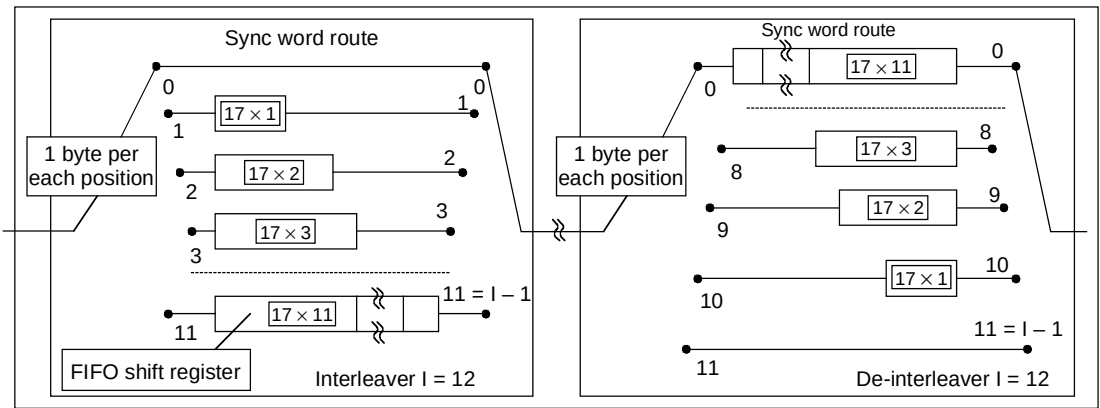
The modulation of the system shall be Quadrature Amplitude Modulation (QAM) with 64 points in the constellation chart.

The System constellation chart for 64-QAM is given in Figure C.7.



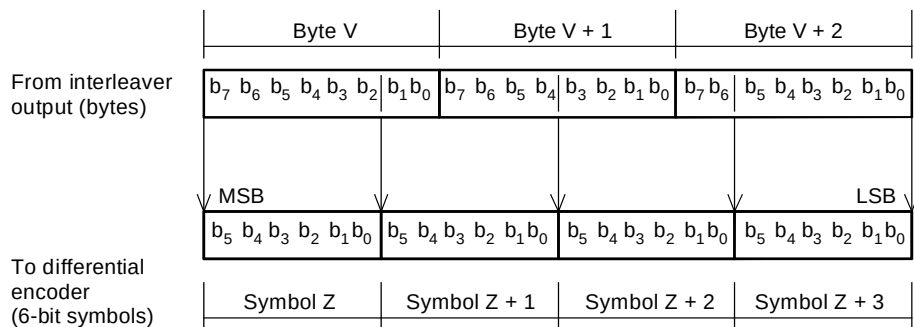
Interleaving depth	$I (= 12)$
Synchronization	Routing SW (Cycle I)
Required memory capacity	$1/2 \times I \times L$

$L (= 17 \times 11)$



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Figure C.4/J.83 – Interleaving configuration

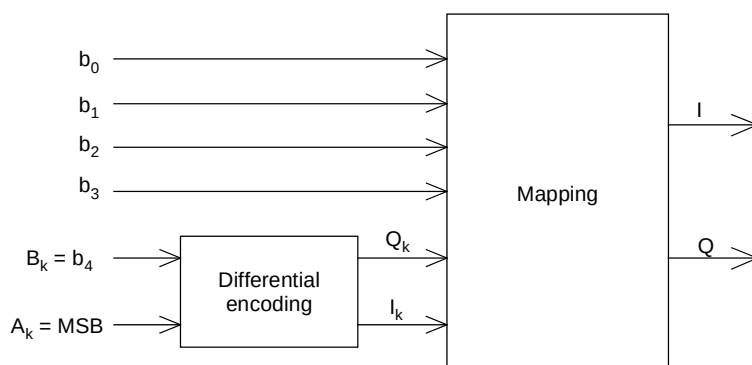


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NOTE 1 – b_0 shall be understood as being the Least Significant Bit (LSB) of each byte or m-tuple.

NOTE 2 – In this conversion, each byte results in more than one m-tuple, labelled Z, Z + 1, etc., with Z being transmitted before Z + 1.

Figure C.5/J.83 – Byte to m-tuple conversion for 64-QAM



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Figure C.6/J.83 – Example of implementation of byte to symbol conversion and the differential encoding of the two MSBs

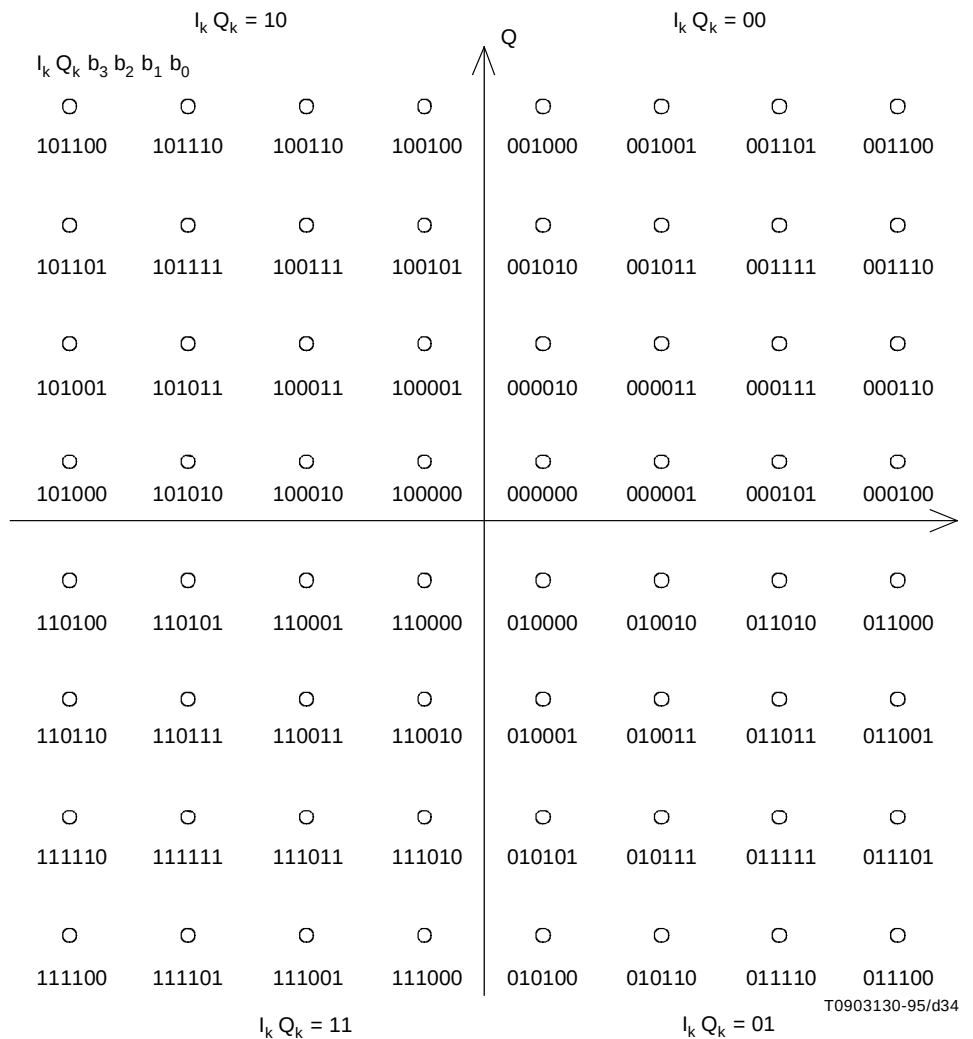


Figure C.7/J.83 – Constellation chart for 64-QAM

C.6.4 Roll-off factor

Prior to modulation, the I and Q signals shall be square-root raised cosine filtered. The roll-off factor shall be 0.13.

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

$$H(f) = 1 \text{ for } |f| \leq f_N(1 - \alpha)$$

$$H(f) = \left[\frac{1}{2} + \frac{1}{2} \sin \frac{\pi}{2f_N} \left[\frac{f_N - |f|}{\alpha} \right] \right]^{1/2} \text{ for } f_N(1 - \alpha) \leq |f| \leq f_N(1 + \alpha), H(f) = 0 \text{ for } |f| > f_N(1 + \alpha)$$

where:

$$f_N = \frac{1}{2 T_s} = \frac{R_s}{2} \text{ is the Nyquist frequency and roll-off factor } \alpha = 0.13.$$

NOTE – Transmission filter characteristics are given in the following subclause. The roll-off factor applies under the condition with adjacent channel signals interference (i.e. from TV signal, etc.) and with the specified baseband filter characteristics.

C.6.5 Baseband filter characteristics

The template given in Figure C.8 shall be used a minimum requirement for hardware implementation of the Nyquist filter. This template takes into account not only the design limitations of the digital filter, but also the artifacts coming from the analogue processing components of the system (e.g. D/A conversion, analogue filtering, etc.).

The value of in-band ripple r_m in the pass-band up to $(1 - \alpha)f_N$ shall be lower than 0.4 dB. The out-of-band rejection shall be greater than 43 dB. The ripple r_N at the Nyquist frequency f_N shall be lower than 1.0 dB.

The filter shall be phase-linear with the group delay ripple $\leq 1.0 T_s$ (ns) in the pass-band up to $(1 - \alpha)f_N$ and $\leq 2.0 T_s$ (ns) at f_N ,

where:

$$T_s = \frac{1}{R_s} \text{ is the symbol period.}$$

NOTE – The values for in-band ripple and out-of-band rejection given in this Annex are subject to the operation condition of the cable systems and may require further study

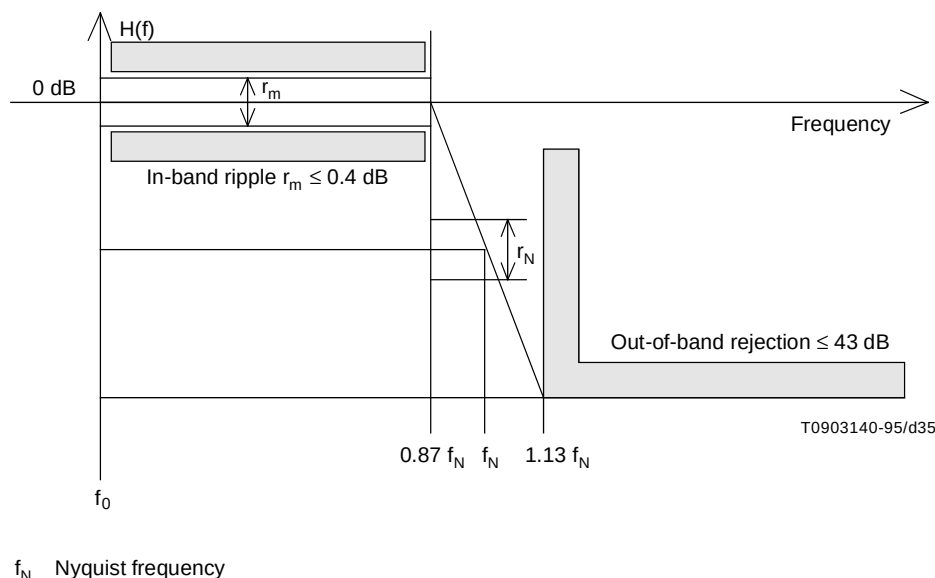


Figure C.8/J.83 – Half-Nyquist baseband filter amplitude characteristics

Annex D

Digital multi-programme System D

D.1 Introduction

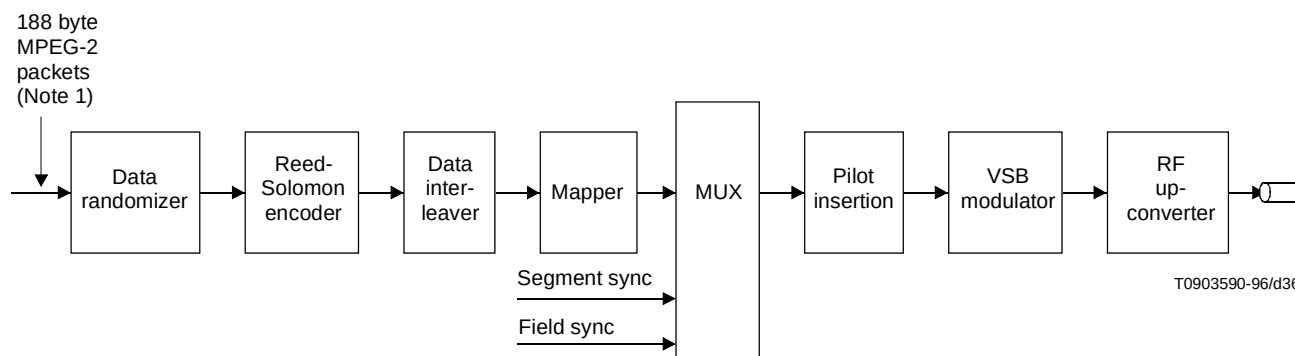
This Annex derives from work done on digital television terrestrial broadcasting in North America; it describes the framing structure, channel coding and modulation for digital multi-programme television distribution by cable, based on MPEG-2 transport multiplexing, and on 16-VSB (Vestigial SideBand) digital transmission.

D.2 Cable system concept

The 16-VSB system will support a nominal payload data rate of 38.78 Mbit/s in a 6 MHz channel². A functional block diagram of a representative 16-VSB cable transmitter is shown in Figure D.1. The input to the transmission subsystem from the transport subsystem is equivalent to a nominal 38.78 Mbit/s serial data stream comprised of 188-byte MPEG-compatible data packets, see Reference [2] (including a sync byte and 187 bytes of data)².

² Parameter value for 6 MHz channel bandwidth; value can be adjusted to match other channel bandwidths.

The incoming data is randomized and then processed for Forward Error Correction (FEC) in the form of Reed-Solomon (RS) coding (20 RS parity bytes are added to each packet), and 1/12 data field interleaving. The randomization and FEC processes are not applied to the sync byte of the transport packet, which is represented in transmission by a Data Segment Sync signal as described below. Following randomization and forward error correction processing, convolutional byte interleaving is performed and then the data packets are formatted into Data Frames for transmission and Data Segment Sync and Data Field Sync are added.



NOTE 1 – Provided by terrestrial broadcasts, satellite, or local origination.

NOTE 2 – Includes private cable [hotels, apartment buildings, condominiums, and schools, – wired, and MMDS (Multichannel Multipoint Distribution System) – wireless microwave].

Figure D.1/J.83 – 16-VSB transmitter (cable or SMATV head-end – Note 2)

D.3 MPEG-2 transport layer

The MPEG-2 transport layer is defined in Reference [2]. The transport layer for MPEG-2 data is comprised of packets having 188 bytes, with one byte for synchronization purposes, three bytes of header containing service identification, scrambling and control information, followed by 184 bytes of MPEG-2 or auxiliary data.

D.4 Framing structure

Figure D.2 shows how the data are organized for transmission. Each Data Frame consists of two Data Fields, each containing 313 Data Segments. The first Data Segment of each Data Field is a unique synchronizing signal (Data Field Sync) and includes the training sequence used by the equalizer in the receiver. The remaining 312 Data Segments each carry the equivalent of two 188-byte transport packets plus its associated FEC overhead. The actual data in each Data Segment comes from several transport packets because of the data interleaving. Each Data Segment consists of 832 symbols. The first 4 symbols are transmitted in binary form and provide segment synchronization. This Data Segment Sync signal also represents the sync byte for each of the two 188-byte MPEG-compatible transport packets. The remaining 828 symbols of each Data Segment carry data representing two groups of 187 data bytes each followed by 20 Reed-Solomon bytes. These 828 symbols are transmitted as 16-level signals and therefore carry four bits per symbol. Thus, $828 \times 4 = 3312$ bits of data are carried in each Data Segment, which exactly matches the requirement to send two protected transport packets:

$$187 \text{ data bytes} + 20 \text{ RS parity bytes} = 207 \text{ bytes}$$

$$2 \times 207 \text{ bytes} \times 8 \text{ bits/byte} = 3312$$

The exact symbol rate is given by the equation below:

$$S_r \text{ (MHz)} = 4.5/286 \times 684 = 10.76 \dots \text{ MHz}^3$$

The 16-level symbols combined with the binary Data Segment Sync and Data Field Sync signals are used to modulate a single carrier in suppressed-carrier mode. Before transmission, however, most of the lower sideband is removed. The resulting spectrum is flat, except for the band edges where a nominal square-root raised-cosine response results in 620 kHz transition regions. The nominal VSB transmission spectrum is shown in Figure D.3³.

At the suppressed-carrier frequency, 310 kHz from the lower band edge, a small pilot is added to the signal.

The cable system may also carry standard television signals on other channels as shown in Figure D.3. The nominal average VSB signal power is 6 dB below peak sync power of standard television signals carried in adjacent channels.

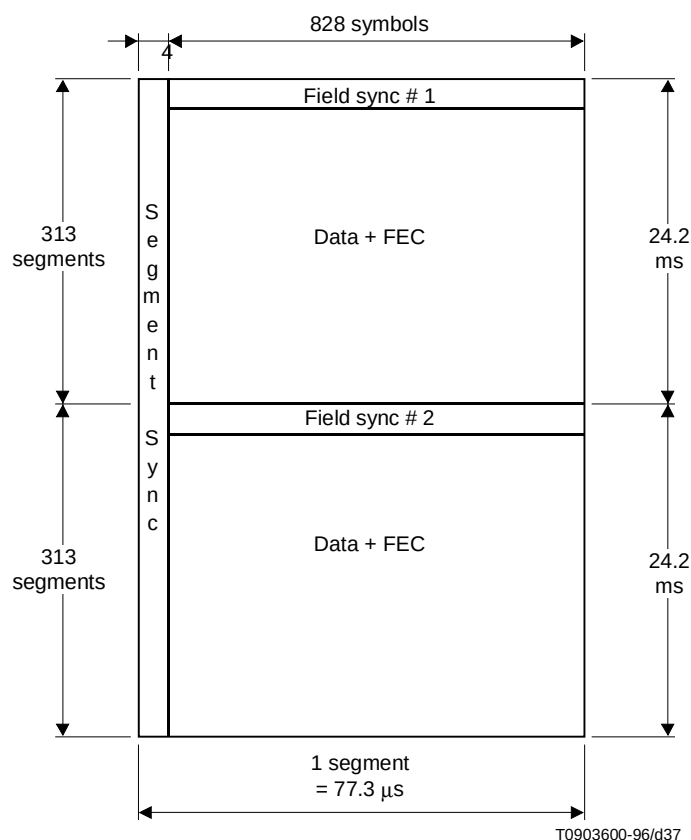


Figure D.2/J.83 – VSB data frame

³ Parameter value for 6 MHz channel bandwidth; value can be adjusted to match other channel bandwidths.

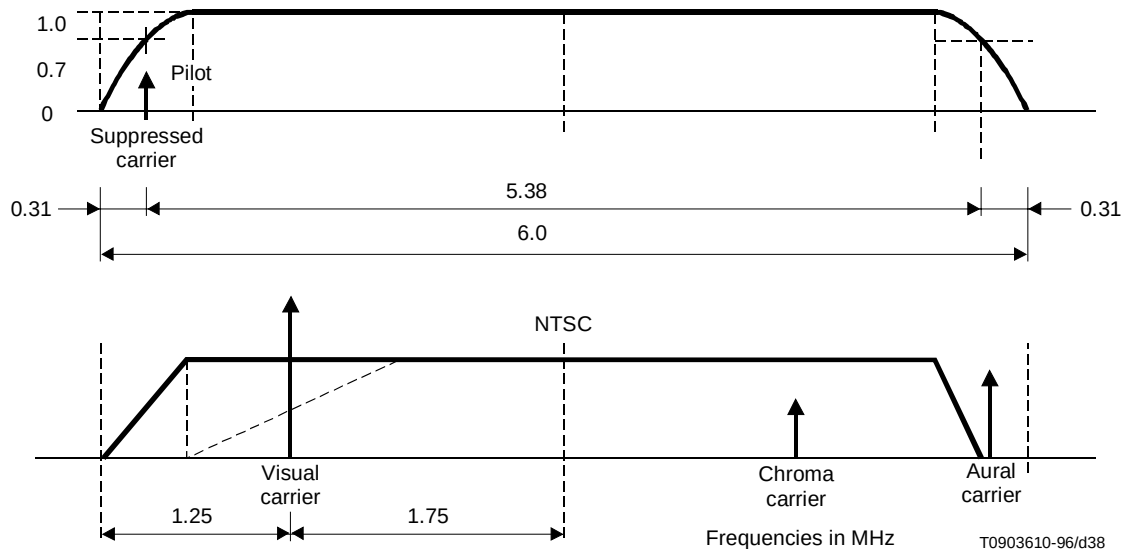


Figure D.3/J.83 – VSB and NTSC channel occupancy

D.5 Channel coding

D.5.1 Data randomizer

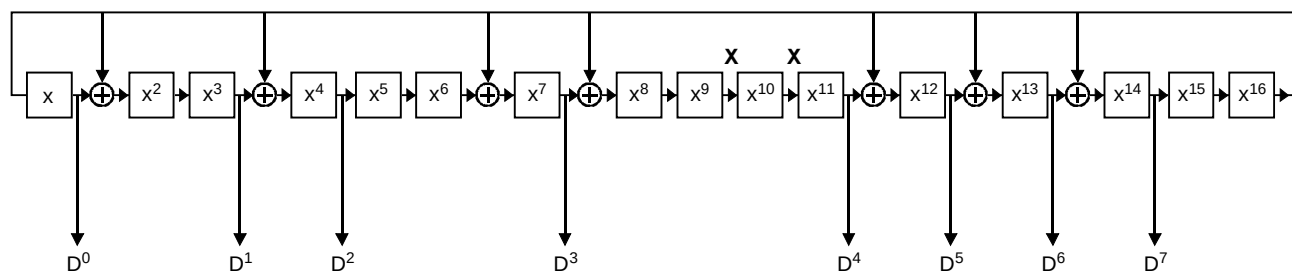
A data randomizer is used on all input data to randomize the data payload (not including Data Field Sync or Data Segment Sync, or RS parity bytes). The data randomizer XOR-s all the incoming data bytes with a 16-bit maximum length PseudoRandom Binary Sequence (PRBS) which is initialized at the beginning of the Data Field. The PRBS is generated in a 16-bit shift register that has 9 feedback taps. Eight of the shift register outputs are selected as the fixed randomizing byte, where each bit from this byte is used to individually XOR the corresponding input data bit. The data bits are XOR-ed MSB to MSB ... LSB to LSB.

The initialization (pre-load) to F180 hex (load to 1) occurs during the Data Segment Sync interval prior to the first Data Segment.

The randomizer generator polynomial and initialization are shown in Figure D.4.

Generator polynomial $G_{(16)} = x^{16} + x^{13} + x^{12} + x^{11} + x^7 + x^6 + x^3 + x + 1$
The initialization (pre-load) occurs during the field sync interval

Initialization to F180 hex (Load to 1)
 $x^{16} x^{15} x^{14} x^{13} x^9 x^8$



The generator is shifted with the Byte Clock and one 8-bit Byte of data is extracted per cycle

Figure D.4/J.83 – Randomizer polynomial

D.5.2 Reed-Solomon encoder

The RS code used in the VSB transmission subsystem is $t = 10$ (207, 187) code. The RS data block size is 187 bytes, with 20 RS parity bytes added for error correction. Two RS blocks of 207 bytes are transmitted per Data Segment.

The 20 RS parity bytes are sent at the end of each respective group of 187 bytes. The parity generator polynomial and the primitive field generator polynomial are shown in Figure D.5.

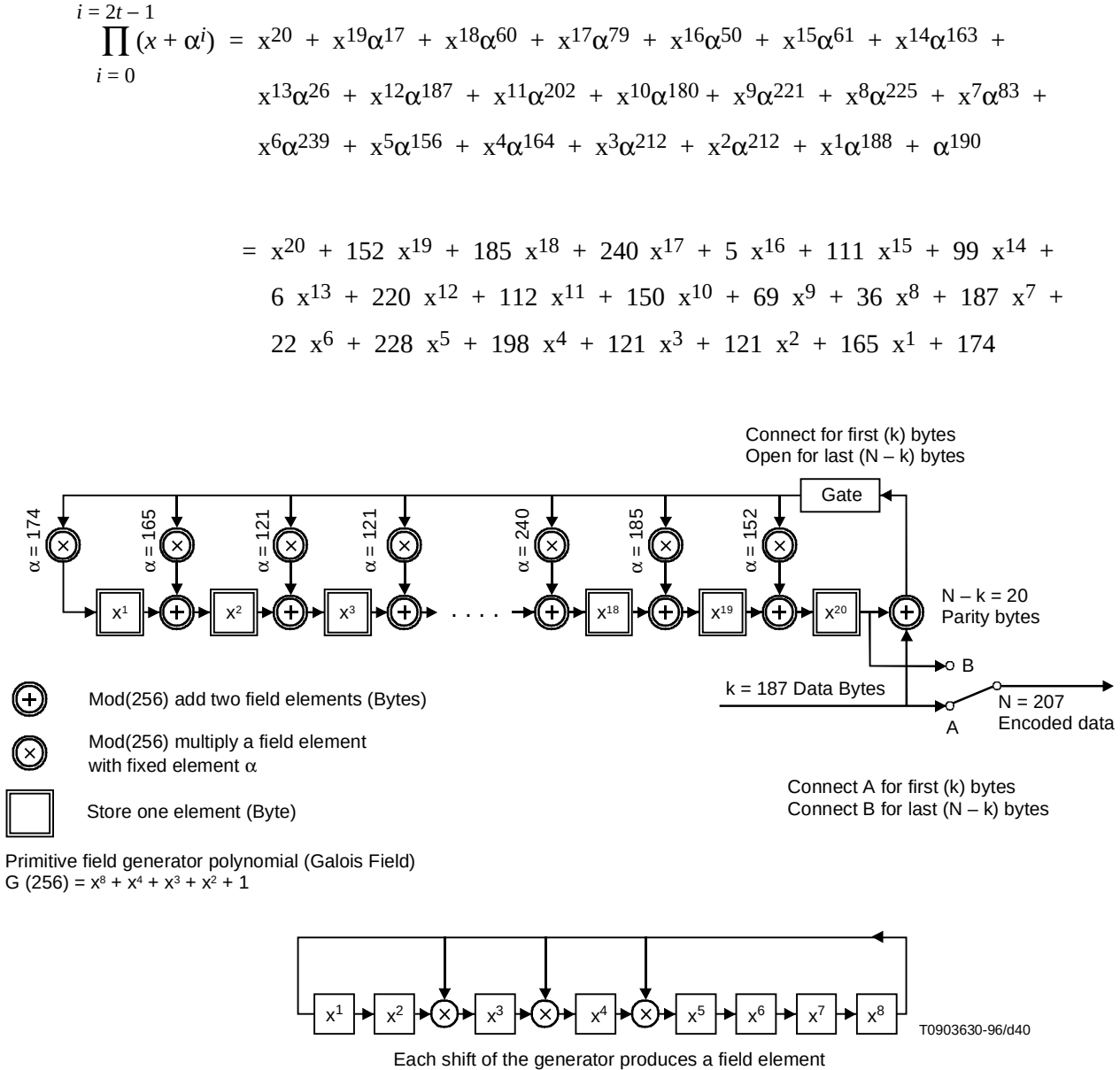


Figure D.5/J.83 – Parity generator polynomial for Reed-Solomon (207, 187) with $t = 10$

D.5.3 Interleaving

The interleaver employed in the VSB transmission system is a 26 data segment (intersegment) convolutional byte interleaver. Interleaving is provided to a depth of about 1/12 of a data field (2 ms deep). Only data bytes are interleaved. The interleaver is synchronized to the first data byte of the data field. The convolutional interleaver is shown in Figure D.6.

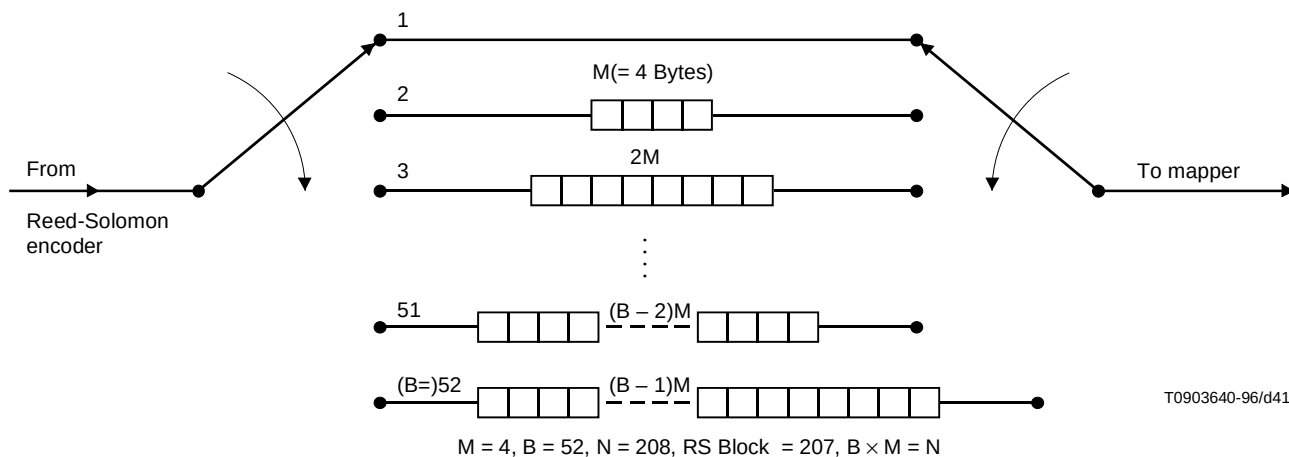


Figure D.6/J.83 – Convolutional interleaver

D.5.4 Data segment sync

The multi-level data is passed through a multiplexer that inserts the various synchronization signals (Data Segment Sync and Data Field Sync).

A two-level (binary) 4-symbol Data Segment Sync is inserted into the 16-level digital data stream at the beginning of each Data Segment. (The MPEG sync byte is replaced by Data Segment Sync.) The Data Segment Sync embedded in random data is shown in Figure D.7.

A complete segment consists of 832 symbols: 4 symbols for Data Segment Sync, and 828 data plus parity symbols. The Data Segment Sync is binary (2-level). The same sync pattern occurs regularly at 77.3 μ s intervals, and is the only signal repeating at this rate. Unlike the data, the four symbols for Data Segment Sync are not Reed-Solomon encoded, nor are they interleaved. The Data Segment Sync pattern is a 1001 pattern, as shown in Figure D.7.

D.5.5 Data field sync

The data are not only divided into Data Segments, but also into Data Fields, each consisting of 313 segments. Each Data Field (24.2 ms) starts with one complete Data Segment of Data Field Sync, as shown in Figure D.8. Each symbol represents one bit of data (2-level). The 832 symbols in this segment are defined below. See Figure D.8.

D.5.5.1 Sync

This corresponds to Data Segment Sync and is defined as 1001.

D.5.5.2 PN511

This pseudo-random sequence is defined as $x^9 + x^7 + x^6 + x^4 + x^3 + x + 1$ with a pre-load value of 010000000.

The generator for the PN511 sequence is shown in Figure D.9.

D.5.5.3 PN63

This pseudo-random sequence is repeated three times. It is defined as $x^6 + x + 1$ with a pre-load value of 100111. The middle PN63 is inverted on every other Data Field Sync.

The generator for the PN63 sequence is shown in Figure D.9.

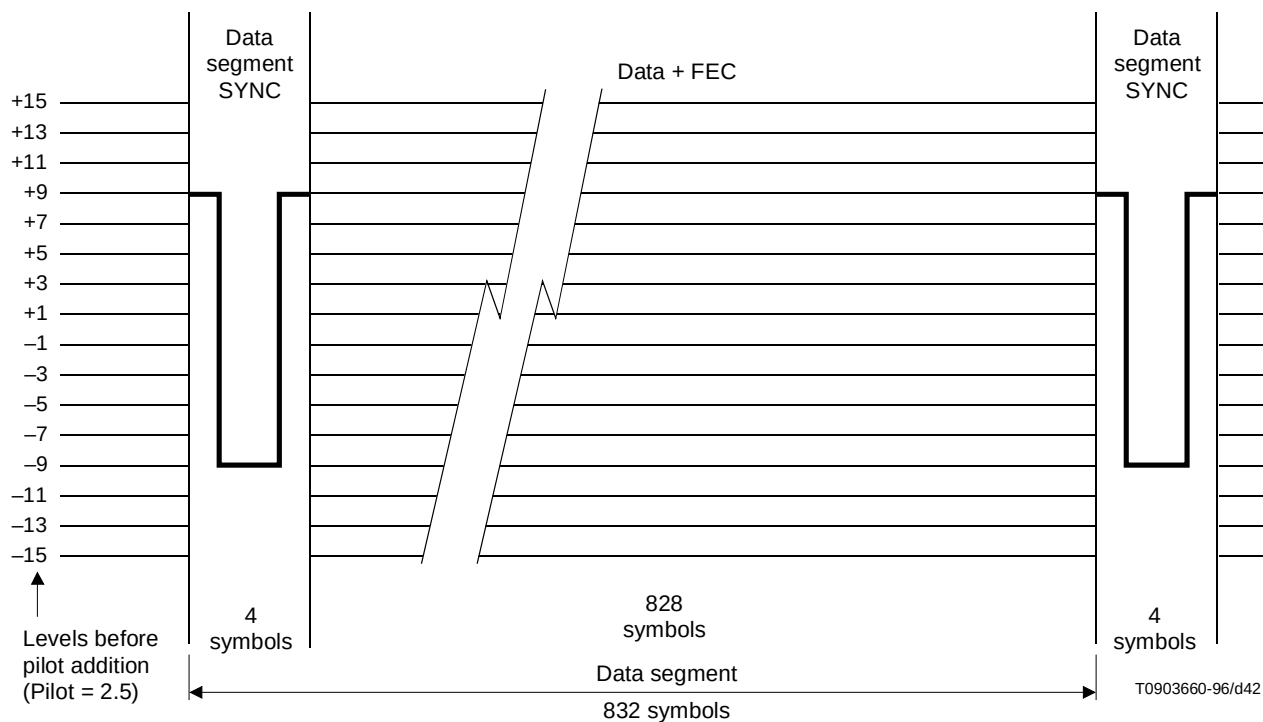


Figure D.7/J.83 – 16-VSB data segment

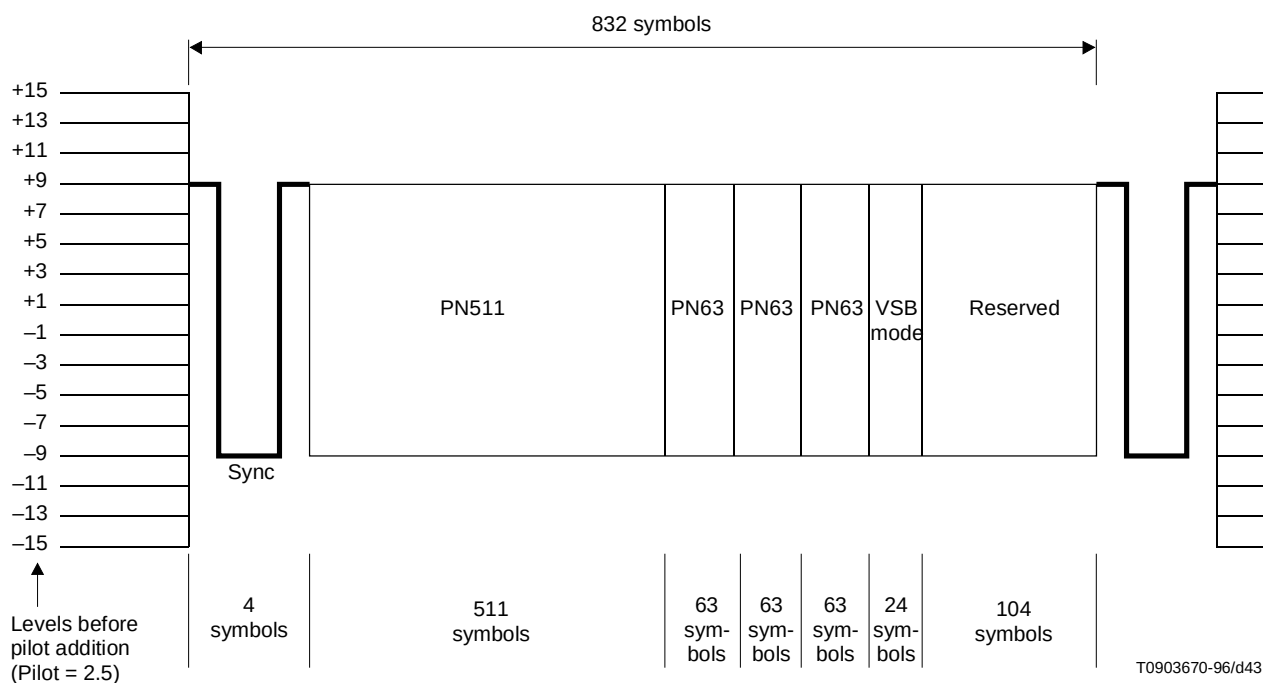


Figure D.8/J.83 – 16-VSB data field sync

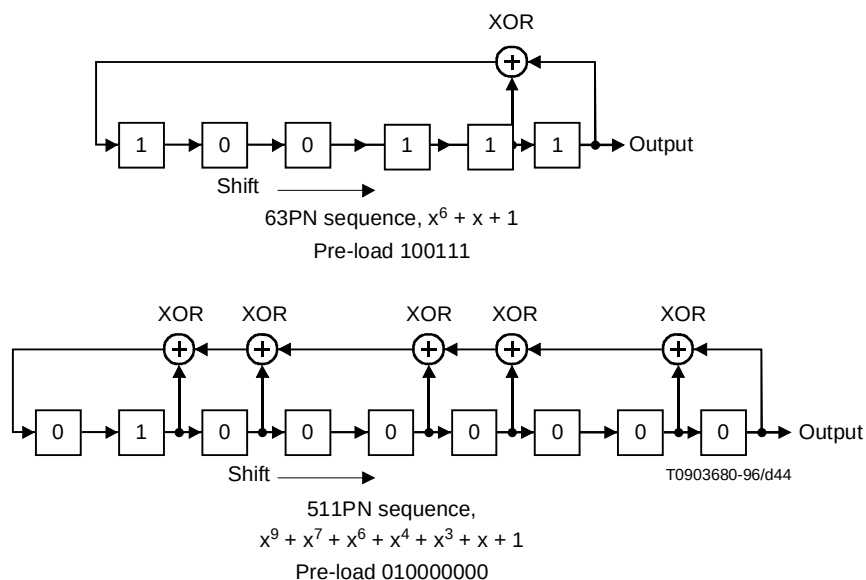


Figure D.9/J.83 – Field sync PN sequence generators

D.5.5.4 VSB mode

These 24 bits determine the VSB mode for the data in the frame. The first two bytes are reserved. The suggested fill pattern is 0000111100001111. The next byte is defined as:

P A B C $\overline{\text{P A B C}}$

where P is the even parity bit, the MSB of the byte, and A, B, C are the actual mode bits.

P A B C

0 0 0 0 2 VSB

1 0 0 1 4 VSB

1 0 1 0 8 VSB

0 0 1 1 Reserved

1 1 0 0 16-VSB Cable

0 1 0 1 8-VSB Terrestrial (Note) (trellis coded)

0 1 1 0 Reserved

1 1 1 1 Reserved

NOTE – In the 8-VSB terrestrial mode, the preceding bits are defined as:

0 0 0 0 $\overline{\text{P A B C}}$ P A B C 1 1 1 1

D.5.5.5 Reserved

The last 104 bits is reserved space. It is suggested that this be filled with a continuation of the PN63 sequence.

All sequences are pre-loaded before the beginning of the Data Field Sync.

Like the Data Segment Sync, the Data Field Sync is not Reed-Solomon encoded, nor is it interleaved.

D.6 Modulation

D.6.1 Bit-to-symbol mapping

Figure D.10 shows the mapping of the outputs of the interleaver to the nominal signal levels of $(\pm 1, \pm 3, \pm 5, \pm 7, \pm 9, \pm 11, \pm 13, \pm 15)$. As shown in Figures D.7 and D.8, the nominal levels of Data Segment Sync and Data Field Sync are -9 and $+9$.

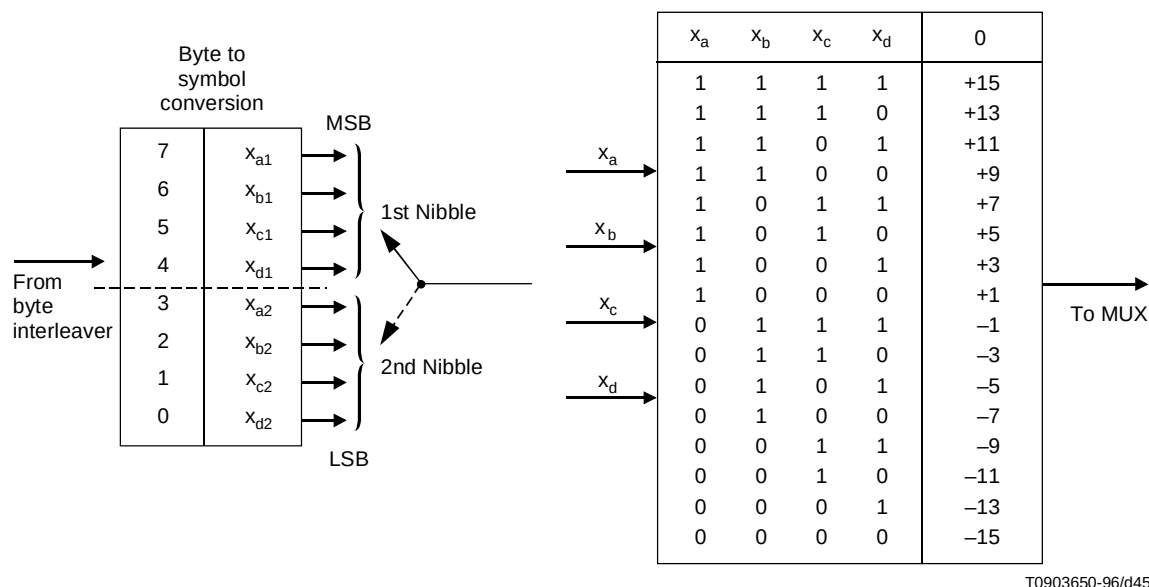


Figure D.10/J.83 – 16-VSB mapper

D.6.2 Pilot addition

A small in-phase pilot is added to the data signal. The frequency of the pilot is the same as the suppressed-carrier frequency as shown in Figure D.3. This may be generated in the following manner. A small (digital) DC level (2.5) is added to every symbol (data and syncs) of the digital baseband data plus sync signal $(\pm 1, \pm 3, \pm 5, \pm 7, \pm 9, \pm 11, \pm 13, \pm 15)$. The power of the pilot is 11.3 dB below the average data signal power.

D.6.3 16-VSB modulation method

The VSB modulator receives the 10.76 Msymbols/s, 16-level composite data signal (pilot and syncs added). The digital multi-programme system performance is based on a linear-phase raised-cosine Nyquist filter response in the concatenated transmitter and receiver, as shown in Figure D.11⁴. The system filter response is essentially flat across the entire band, except for the transition regions at each end of the band. Nominally, the roll-off in the transmitter has the response of a linear-phase square-root raised-cosine filter. Tolerances, both in-band and out-of-band, are under study.

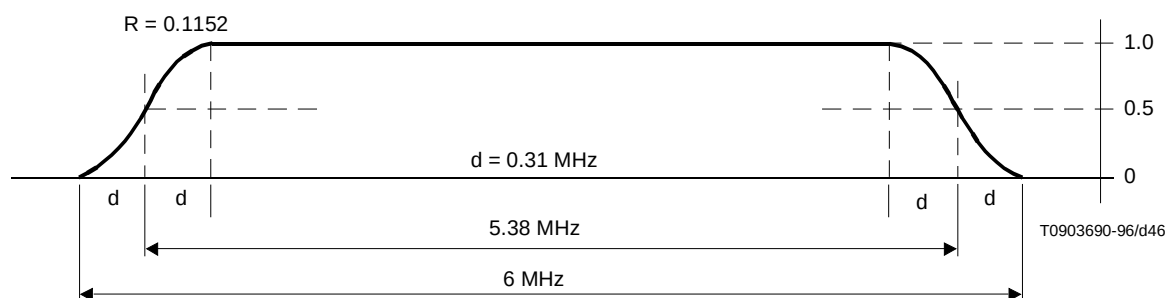


Figure D.11/J.83 – Nominal VSB system channel response (linear-phase raised-cosine Nyquist filter)

⁴ Parameter value for 6 MHz channel bandwidth; value can be adjusted to match other channel bandwidths.

D.6.4 Up-conversion

The modulation process is usually accomplished at an IF frequency. The modulated IF is then up-converted to the final frequency carried by the cable system.

D.7 16-VSB cable receiver

The 16-VSB cable receiver is shown in Figure D.12. All of the inverse functions of the transmitter are performed in the receiver: down conversion (tuner), detection, sync and timing recovery, de-interleaving, Reed-Solomon forward error correction, and data de-randomization.

In addition, an equalizer removes intersymbol interference making use of the data field sync as a training reference signal, and a phase tracker reduces the effect of phase-noise of the local oscillator of the tuner. Following the phase tracker is the slicer to recover the data from the multi-level symbols.

The demodulating carrier is recovered from the pilot and the sync and clock are recovered from the segment sync.

The receiver may be fed with cable mode signals from the cable distribution system or, if the receiver is a terrestrial broadcast receiver, may also be fed from 8-VSB trellis-coded terrestrial broadcasts, or from private cable sources (SMATV or MMDS, or other).

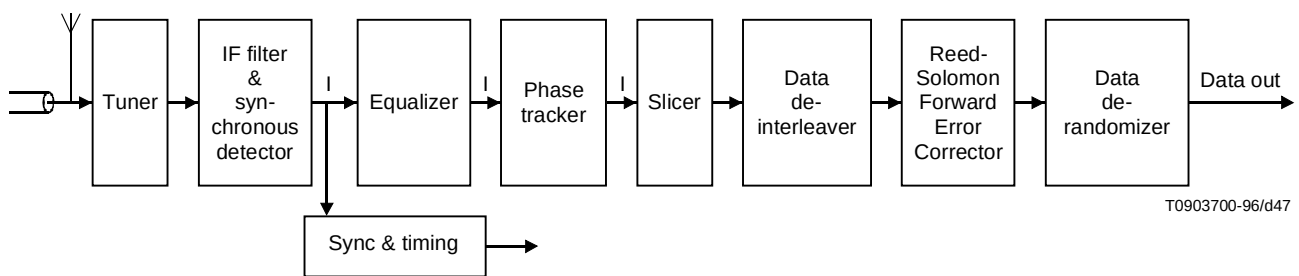


Figure D.12/J.83 – 16-VSB receiver

D.8 Other VSB modes

D.8.1 Introduction

Subclause D.5.5.4 above identified the VSB mode bits in the VSB data field sync interval for VSB modes 2-VSB, 4-VSB, 8-VSB, 16-VSB, and 8-VSB (trellis coded). This entire Annex up to this point has described the 16-VSB mode processes including the 188-byte MPEG input packets, data frame structure, data randomizer, Reed-Solomon Forward Error Correction (FEC), byte interleaving, mapper, data segment sync, data field sync, VSB modulation, and the 16-VSB receiver.

D.8.2 Overview of VSB modes

The group of VSB modes form a family with a high degree of commonality of processes. Each mode of the family uses the same symbol rate synchronizing, spectrum shaping, and channel bandwidth. The difference between the modes is due to the variation of bits per symbol and the consequent payload data rate that can be accommodated. Table D.1 shows the relationships.

Table D.1/J.83 – VSB mode comparison table

VSB mode	Transport data rate	Noise burst tolerance	C/N threshold
2-VSB	9.7 Mbit/s	387 μ sec	10 dB
4-VSB	19.39 Mbit/s	193 μ sec	16 dB
8-VSB	29.09 Mbit/s	129 μ sec	22 dB
16-VSB	38.79 Mbit/s	97 μ sec	28 dB
8-VSB (Trellis)	19.39 Mbit/s	193 μ sec	15 dB

Each of the VSB modes are treated in the following subclauses.

D.8.3 MPEG-2 transport layer

As in D.3 above, the transport layer provides 188-byte packets.

D.8.4 Framing structure

Figure D.2 shows the data symbols, data segments, and data field organization which is common to all VSB modes. Table D.2 shows the relationship between Reed-Solomon FEC protected MPEG-2 packets and data segments and VSB modes.

Table D.2/J.83 – Comparison table of data frame – MPEG packet-VSB mode

VSB mode	Bits/symbol	Bits/segment	FEC protected MPEG packets/segment
2	1	828	1/2
4	2	1656	1
8	3	2484	1 1/2
16	4	3312	2
8T	2	1656	1

D.8.5 Channel coding

D.8.5.1 Data randomizer

Data randomizing for 2-VSB, 4-VSB, 8-VSB and 8-VSB (trellis) is accomplished as in D.5.1.

D.8.5.2 Reed-Solomon encoder

The RS FEC process for 2-VSB, 4-VSB, 8-VSB, and 8-VSB (trellis coded) is as described in D.5.2.

D.8.5.3 Interleaving

Interleaving for 2-VSB, 4-VSB, 8-VSB, and 8-VSB (trellis coded) is as described in D.5.3. Because bit rates are different for the various VSB modes, the performance with noise bursts varies as shown in Table D.1.

D.8.5.4 Data segment sync

The data segment sync is common to all VSB modes. The particulars of the data segment sync are described in D.5.4. However, the embedding of the sync in the multi-level symbol data varies. This is shown for 2-VSB, 4-VSB, 8-VSB and 8-VSB (trellis coded) in Figures D.13, D.14, D.15, and D.16 respectively.

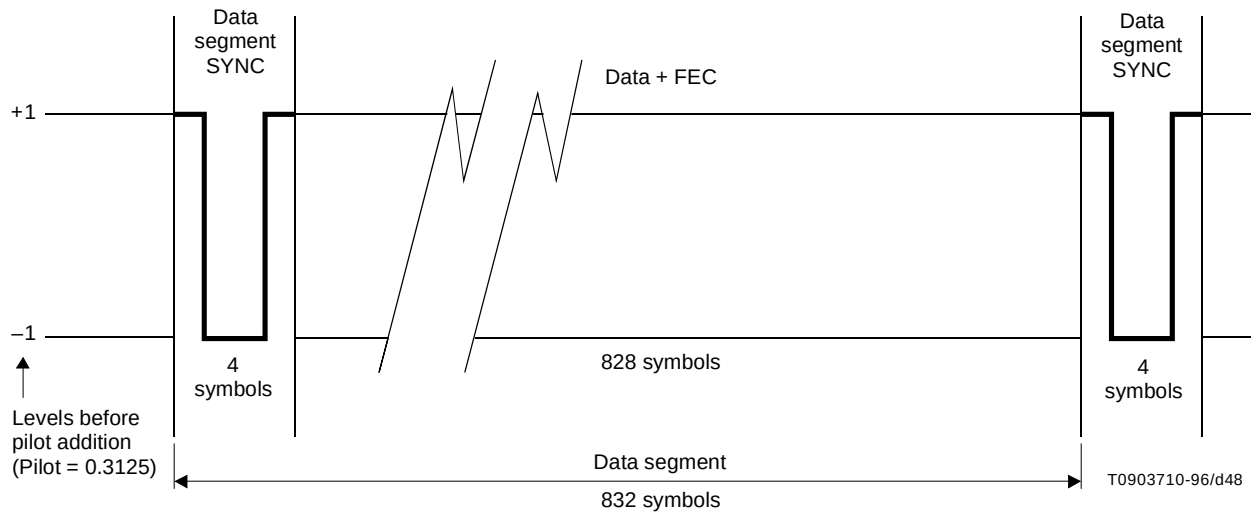


Figure D.13/J.83 – 2-VSB data segment

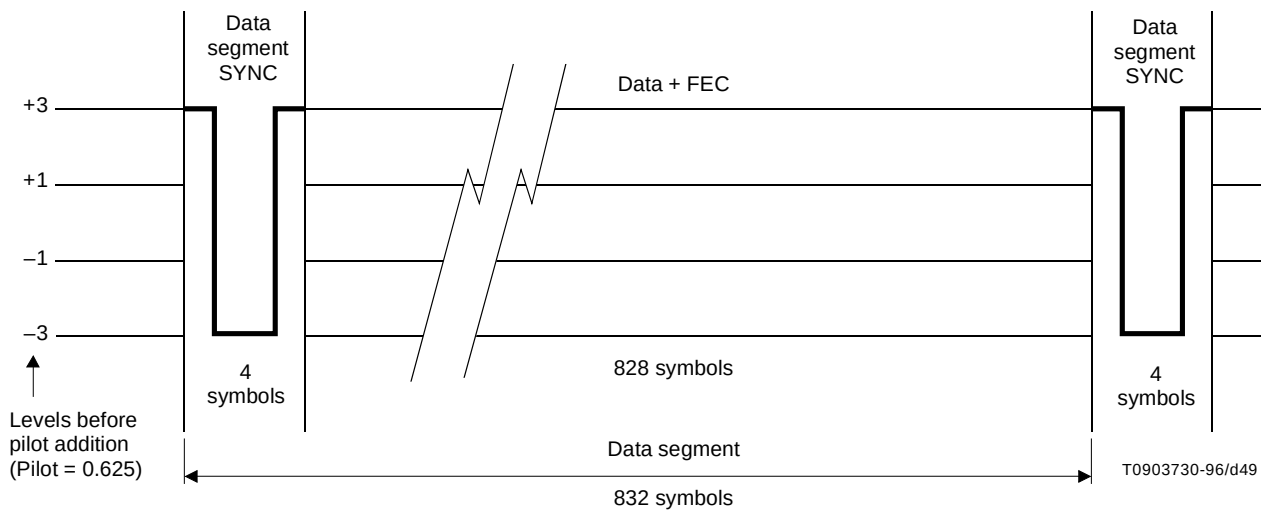


Figure D.14/J.83 – 2-VSB data segment

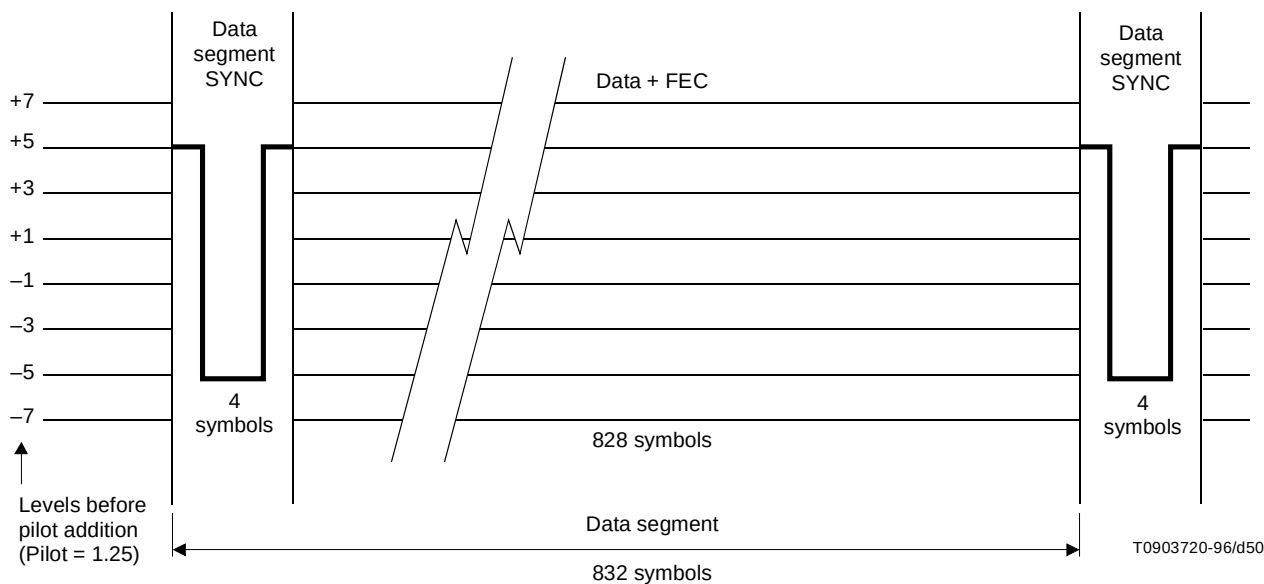


Figure D.15/J.83 – 8-VSB data segment

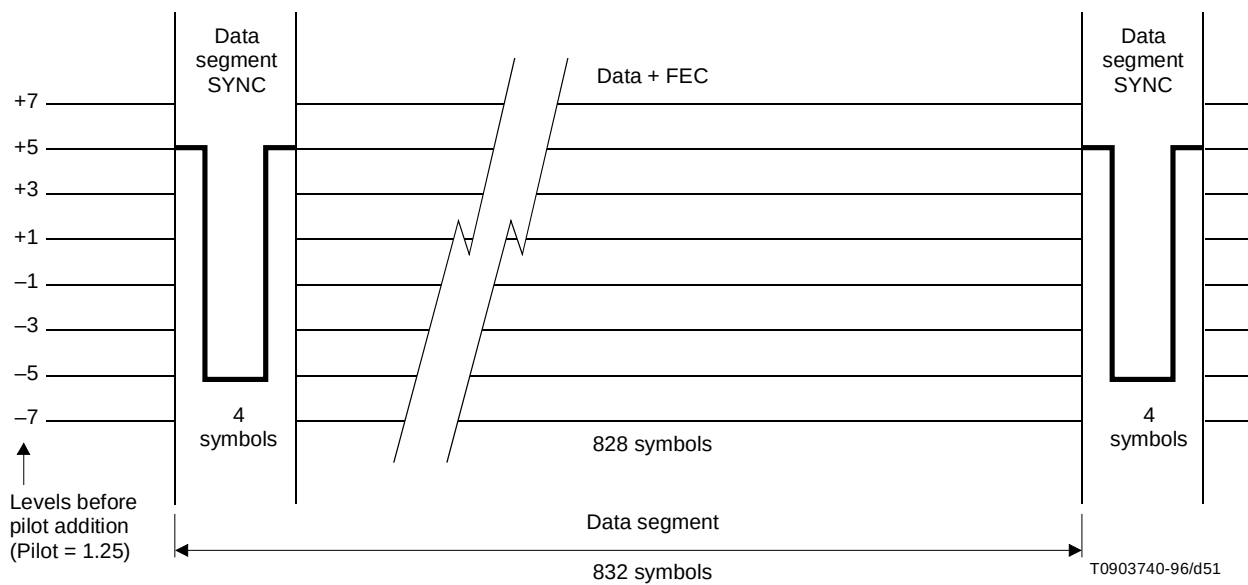


Figure D.16/J.83 – 8-VSB data segment (trellis coded)

D.8.5.5 Data field sync

The data field sync is common to all VSB modes. The particulars of the data field sync are described in D.5.5 data field sync. However, as in the case of the segment sync embedding of the data field sync in the multi-level symbol data varies. This is shown for 2-VSB, 4-VSB, 8-VSB, and 8-VSB (trellis coded) in Figures D.17, D.18, D.19, and D.20, respectively.

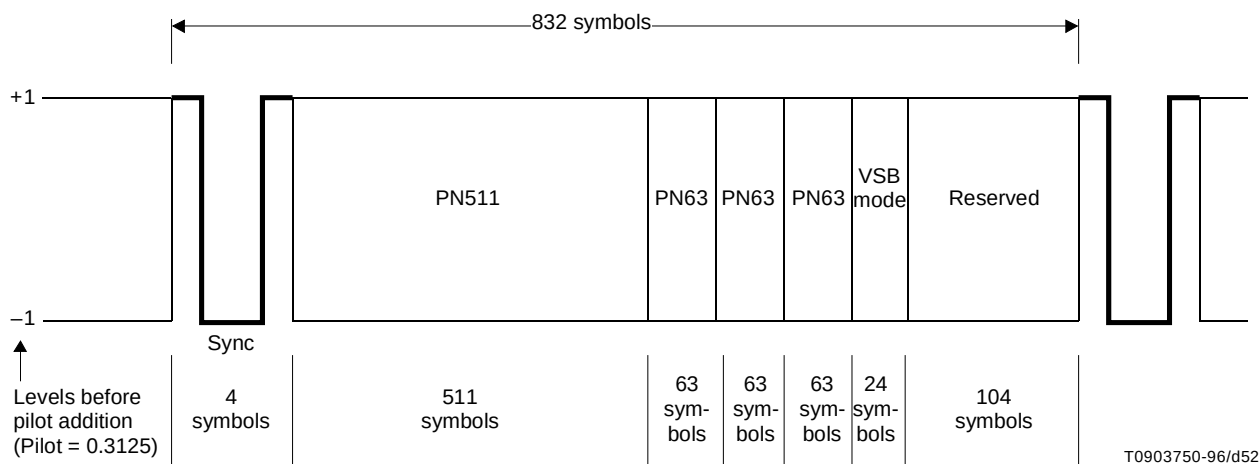


Figure D.17/J.83 – 2-VSB data field sync

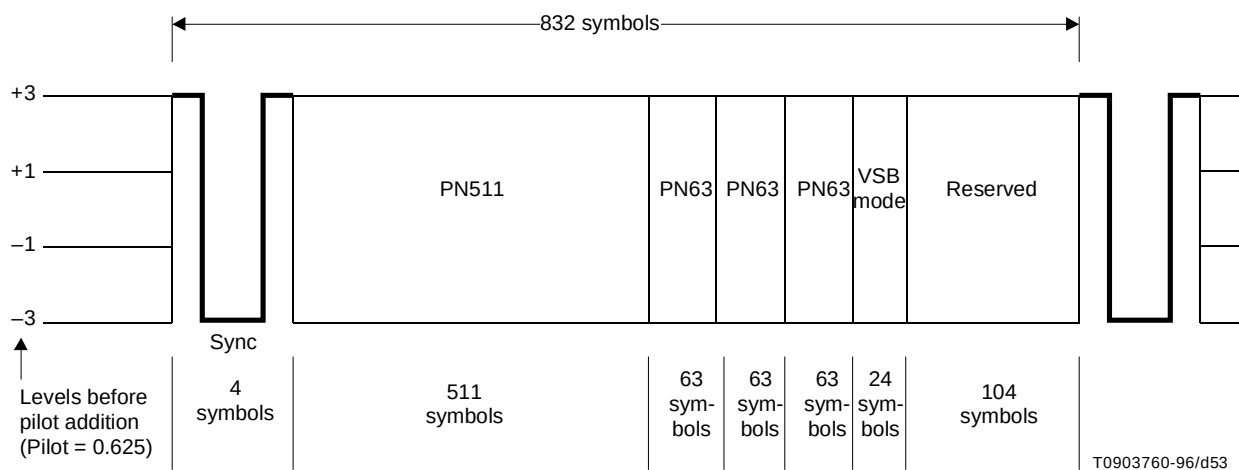


Figure D.18/J.83 – 4-VSB data field sync

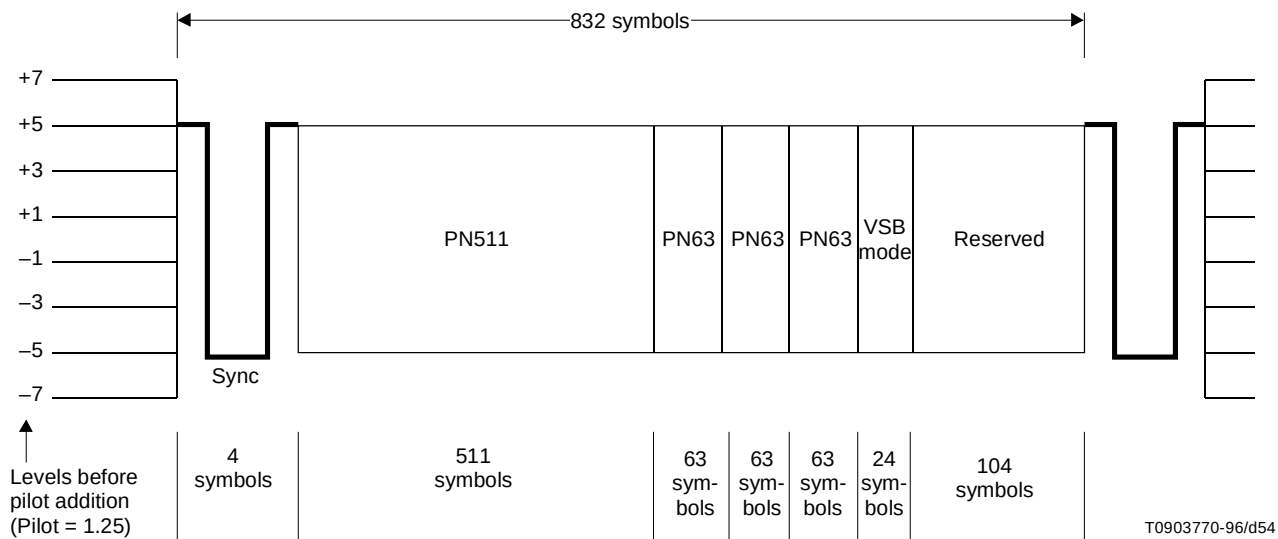
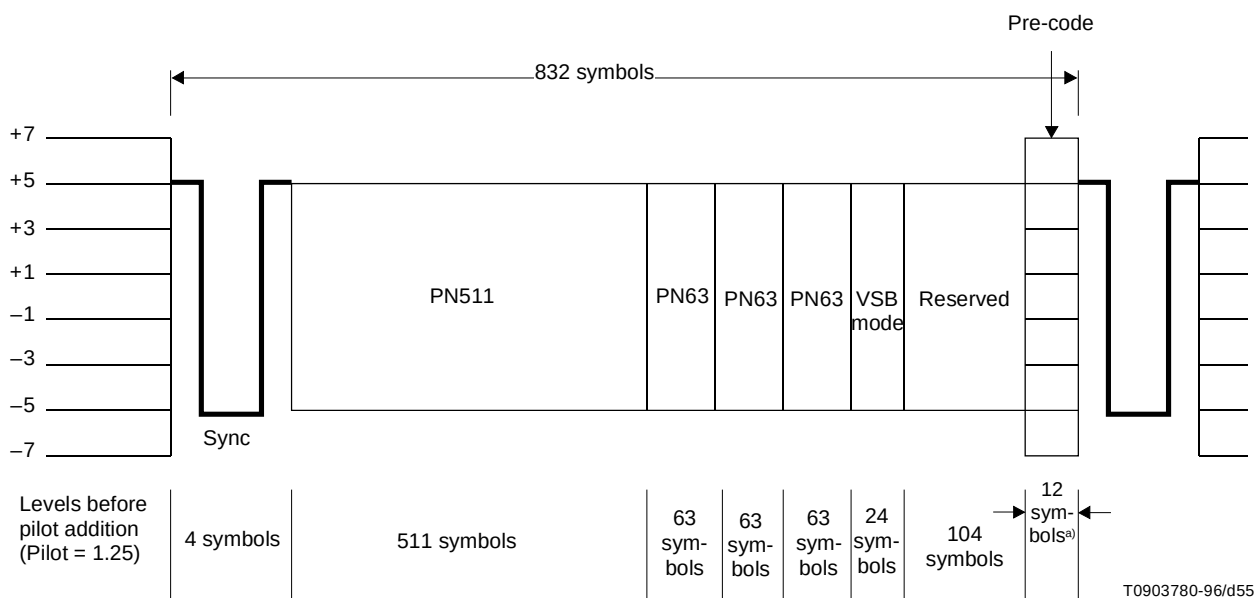


Figure D.19/J.83 – 8-VSB data field sync



a) For trellis coded 8-VSB, the last 12 symbols of the previous segment are duplicated in the last 12 reserved symbols of the field sync.

Figure D.20/J.83 – 8-VSB (trellis coded) data field sync

D.8.6 Modulation

D.8.6.1 Bit to symbol mapping

The bit to symbol mapping for 2-VSB, 4-VSB, 8-VSB, and 8-VSB (trellis coded) is shown in Figures D.21, D.22, D.23, and D.24, respectively. The data symbol levels are ± 1 for 2-VSB, $\pm 1, \pm 3$ for 4-VSB, and $\pm 1, \pm 3, \pm 5, \pm 7$ for 8-VSB and 8-VSB (trellis coded)⁵.

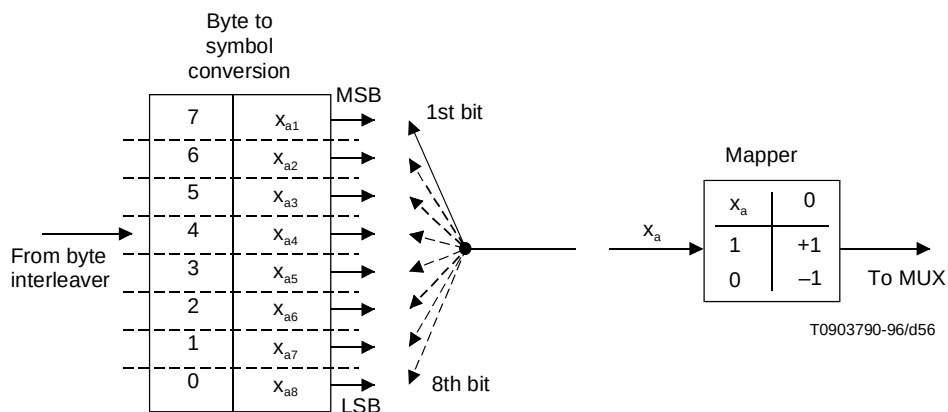


Figure D.21/J.83 – 2-VSB mapper

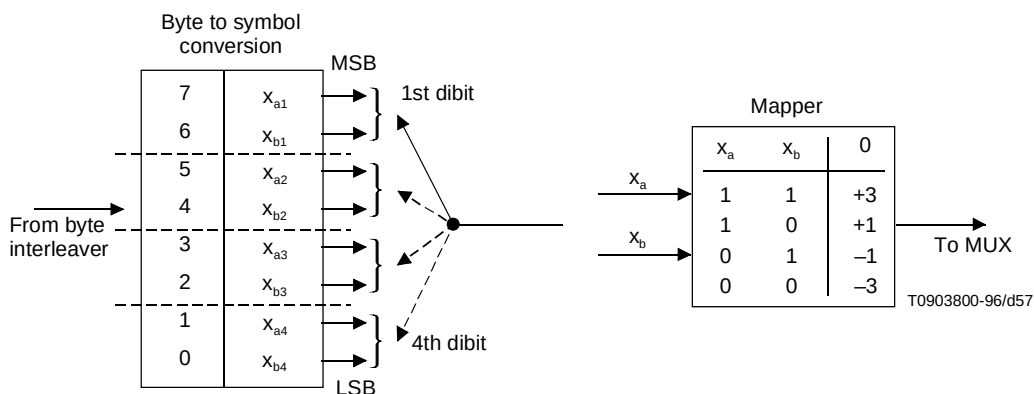


Figure D.22/J.83 – 4-VSB mapper

⁵ The 8-VSB (trellis coded) mode is described in detail in [1] of the Bibliography in Appendix I.

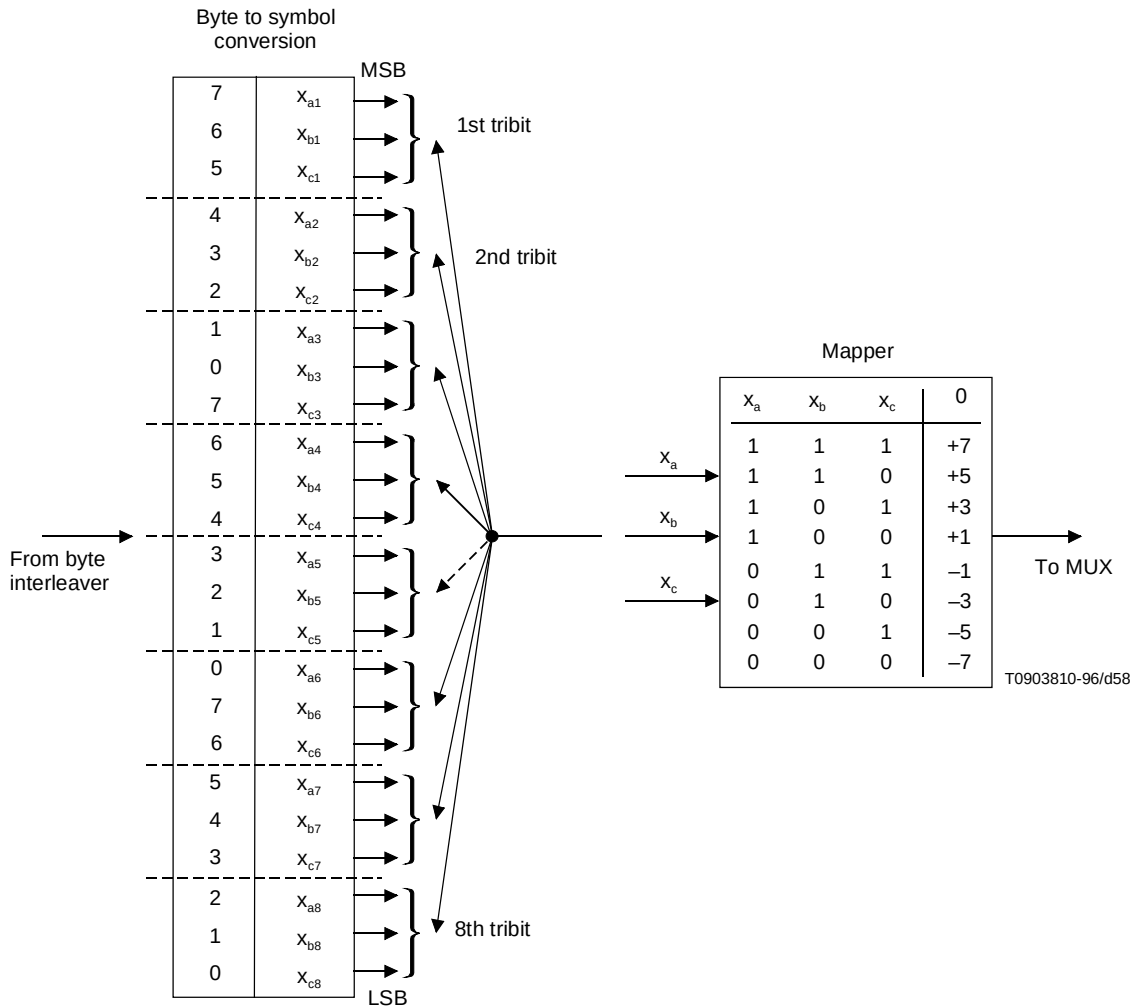


Figure D.23/J.83 – 8-VSB mapper

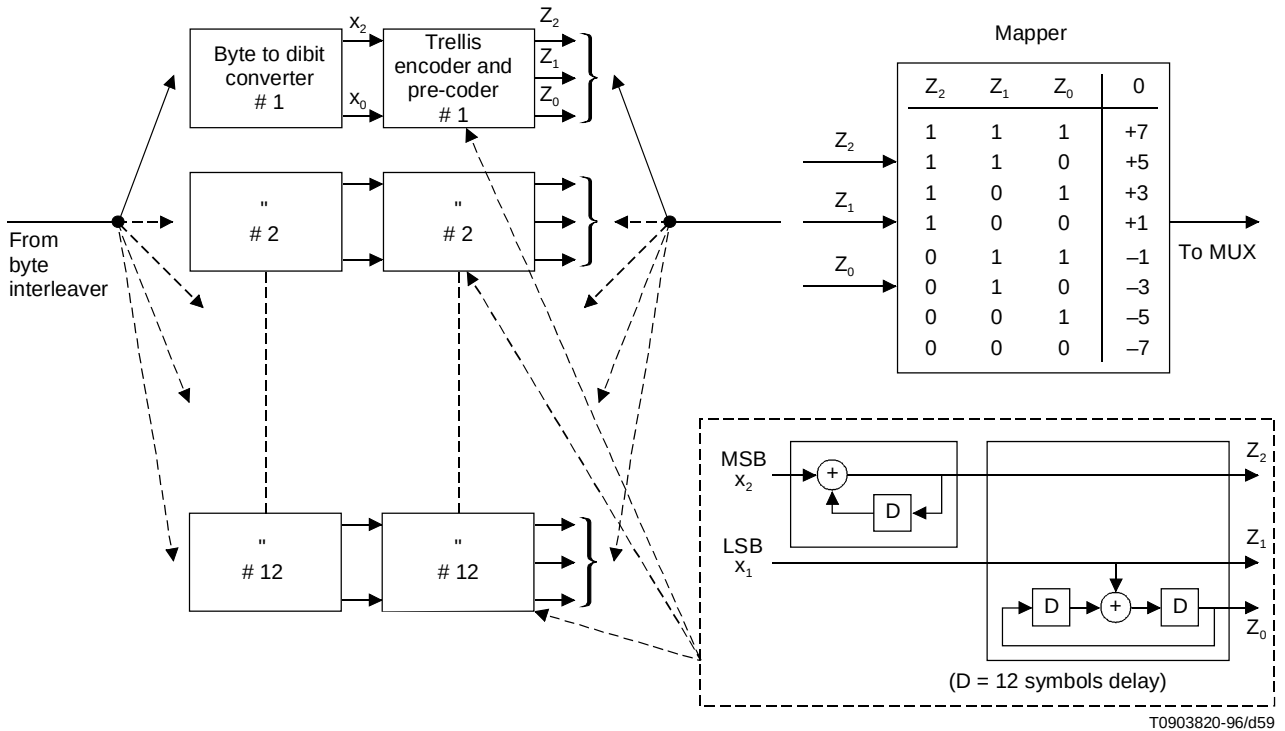


Figure D.24/J.83 – 8-VSB (pre-coder, trellis encoder) mapper
rate = 2/3

D.8.6.2 Pilot addition

The pilot addition may be done in the same manner as in D.6.2 by adding a small DC level to the baseband symbols (data and sync). The value of the DC level is 0.3125 for 2-VSB, 0.625 for 4-VSB, 1.25 for 8-VSB and 8-VSB (trellis coded).

This is shown in Figures D.13, D.14, D.15 and D.16 for 2-VSB, 4-VSB, 8-VSB, and 8-VSB (trellis coded), respectively.

D.8.6.3 VSB modulation method

The modulation method for all VSB systems is accomplished in the same manner as in D.6.3. The overall channel frequency response is shaped as in Figure D.11.

D.8.6.4 Up-conversion

The modulation method is usually accomplished at an IF frequency. The modulated IF is then up-converted to the final channel frequency of the RF transmission system.

D.8.7 VSB receiver

The VSB receiver has all the inverse functions of the transmitter. The VSB mode of transmission is indicated to the receiver in the manner described in D.5.5.4. The receiver functions are then adjusted to accommodate to the various required data slice levels and to the differing incoming bit rates⁶.

Appendix I

Bibliography

- [1] Advanced Television Systems Committee (ATSC): Digital television standard, Washington D.C., September 1995.
- [2] Advanced Television Systems Committee (ATSC): Guide to the use of the ATSC digital standard, Washington D.C., October 1995.

⁶ The 8-VSB (trellis coded) receiving process is described in detail in [2] of the Bibliography in Appendix I.

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