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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

Recommendation ITU-T J.83

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Recommendation ITU-T J.83

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

Summary

Recommendation ITU-T J.83 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 (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 a number of digital cable television systems had been developed and provisionally implemented before this standardization effort was undertaken by 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.

Source

Recommendation ITU-T J.83 was approved on 14 December 2007 by ITU-T Study Group 9 (2005-2008) under Recommendation ITU-T A.8 procedure.

FOREWORD

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Introduction

The development of new digital technologies 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.

The second edition (1997) of this Recommendation incorporated Amendment 1 (10/1996), which brought the following changes with respect to the first edition of the Recommendation:

- a) Annex B includes 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). With the second edition, three other VSB modes are defined, i.e., 2-VSB, 4-VSB and 8-VSB.

The third edition (2007) of this Recommendation enhanced the modulation method by adding 128 QAM and 256 QAM in Annex A and 256 QAM in Annex C.

Recommendation ITU-T J.83

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

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 1 – 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 optimized 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.

NOTE 2 – The structure and content of this Recommendation have been organized for ease of use by those familiar with the original source material; as such, the usual style of ITU-T Recommendations has not been applied.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this 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; 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 the currently valid ITU-T Recommendations is published regularly. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

- [1] Recommendation ITU-R 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] Recommendation ITU-T H.222.0 (1995) | ISO/IEC 13818-1:1996, *Information technology – Generic coding of moving pictures and associated audio information: Systems*.
- [3] European Telecommunications Standards Institute (ETSI) EN 300 429 V1.2.1 (1998-04), *Digital Video Broadcasting (DVB); Framing structure, channel coding and modulation for cable 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
q	Number of bits: 2,3,4 for 16-QAM, 32-QAM, 64-QAM, respectively
R	Randomized sequence
r_m	In-band ripple (dB)
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
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 abbreviations:

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	Moving 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

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TS	Transport Stream
VLSI	Very Large Scale Integration
VSF	Vestigial SideBand
XOR	Exclusive OR
2-VSB	2 level VSB
4-VSB	4 level VSB
8-VSB	8 level VSB
16-VSB	16 level VSB

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 – 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 Clause 5 in [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 of 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 Clause 6 in [3], 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 Clause 7.1 in [3], 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.5.1.)	RS (204, 188) GF 256 (See Clause 7.2 in [3], 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 Clause 7.3 in [3], C.5.3.)	Convolutional interleaving, depth: I = 52. (See D.5.3.)	

Table 1 – Comparison of specifications in summary form indicating common features

Item		Annex B	Annex A	Annex C	Annex D
Modulation	Byte to symbol mapping	See B.5.5.	See Clause 8 in [3], C.6.1.		See D.6.1.
	Differential coding	See B.5.5.	See Clause 8 in [3], 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- 128-, 256-QAM (See Clause 9 in [3])	64-, 256-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 Clause 9 in [3])	13% See C.6.4	11.5% See D.6.3
	Baseband filter characteristics	Table B.2	See Annex A in [3]	Figure C.8	Figure D.11

Annex A

Digital multi-programme System A

(This annex forms an integral part of this Recommendation)

A.1 Introduction

This annex derives from work done by the Digital Video Broadcasting (DVB) Project, an industry-led consortium of over 260 broadcasters, manufacturers, network operators, software developers, regulatory bodies and others in over 35 countries around the world committed to designing global standards for the global delivery of digital television and data services. It has been adopted by the Joint Technical Committee (JTC) of the European Broadcasting Union (EBU), Comité Européen de Normalization ELECTrotechnique (CENELEC) and the European Telecommunications Standards Institute (ETSI) as European Norm (EN) 300 429 [3].

It describes the framing structure, channel coding and modulation (denoted "the System" for the purposes of this annex) for a digital multi-programme television distribution by cable. The System can be used transparently with the modulation/channel coding system used for digital multi-program television by satellite.

The System is based on MPEG-2 (see Reference [2]) as regards source coding and transport multiplexing with the addition of appropriate Forward Error Correction (FEC). It is based on Quadrature Amplitude Modulation (QAM). It allows for 16-, 32-, 64-, 128-, or 256-QAM constellations.

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 Specification

The text of ETSI EN 300 429 [3] is applied in this Annex A with the modifications as given below.

A.2.1 Un-numbered clause 'Foreword'

The introductory clause labelled 'Foreword' does not apply in the context of this annex.

6

Annex B

Digital multi-programme System B

(This annex forms an integral part of this Recommendation)

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 fibre, 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.

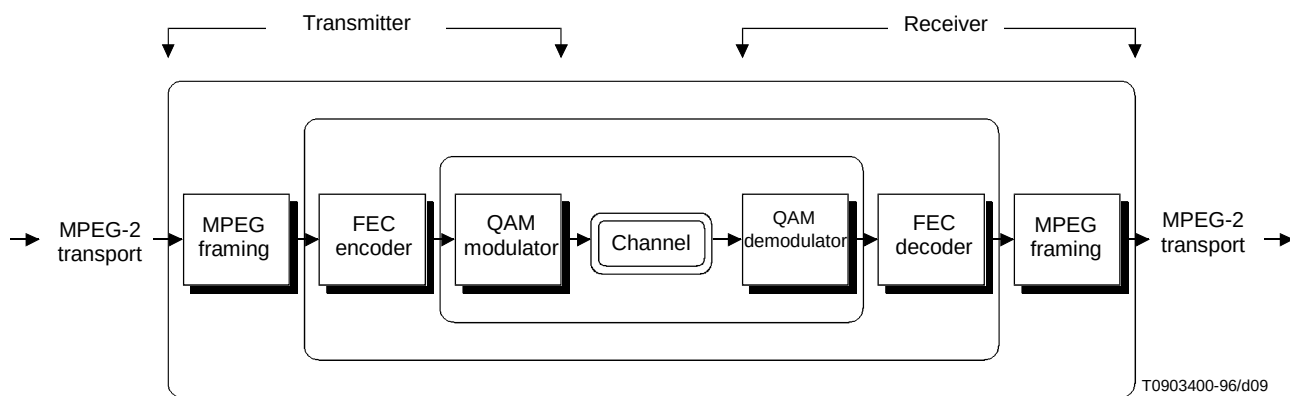


Figure B.1 – Cable transmission block diagram

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.

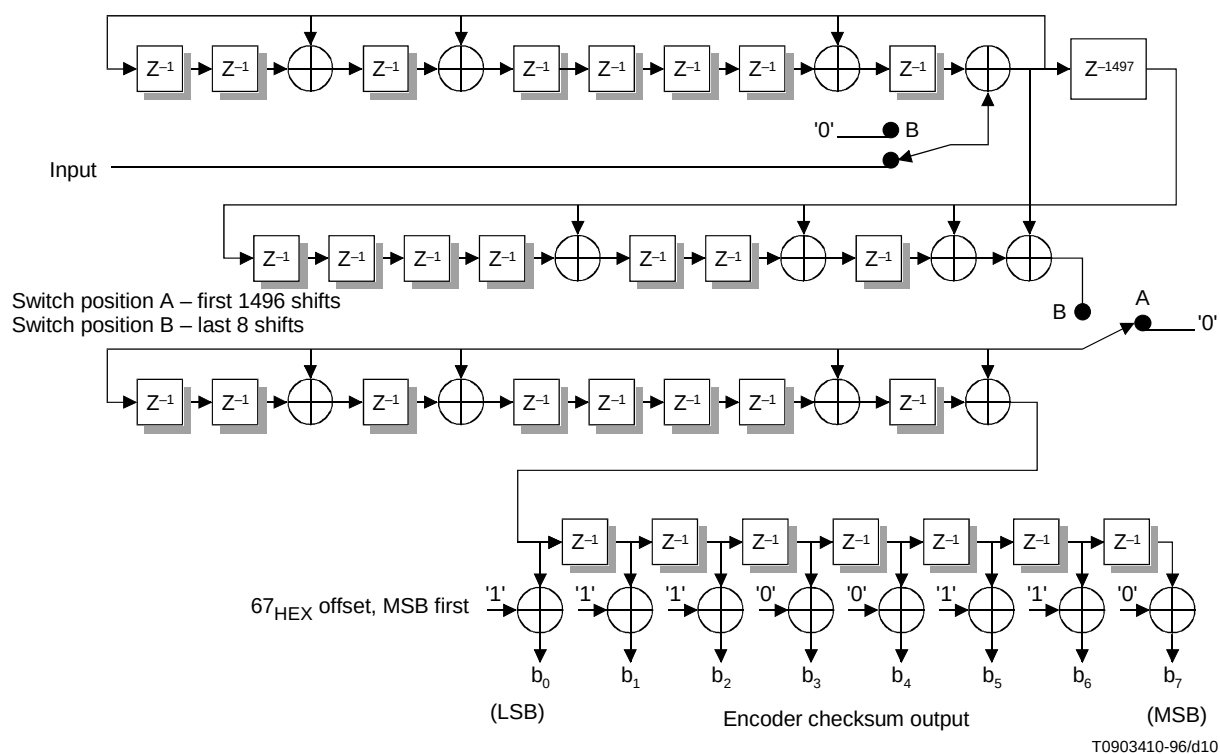


Figure B.2 – Checksum generator for the MPEG-2 sync byte encoder

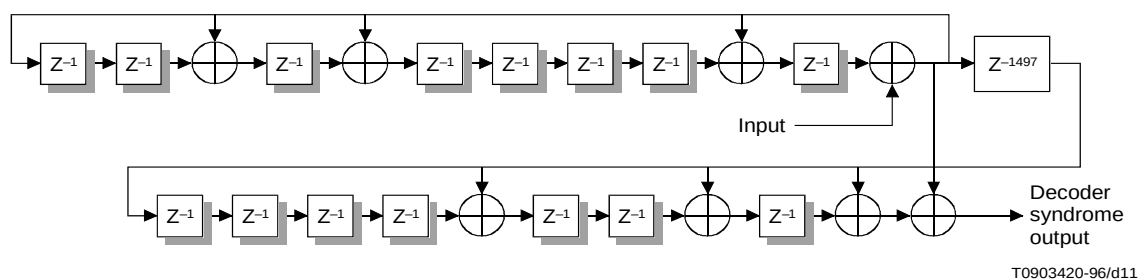


Figure B.3 – Syndrome generator for the MPEG-2 sync decoder

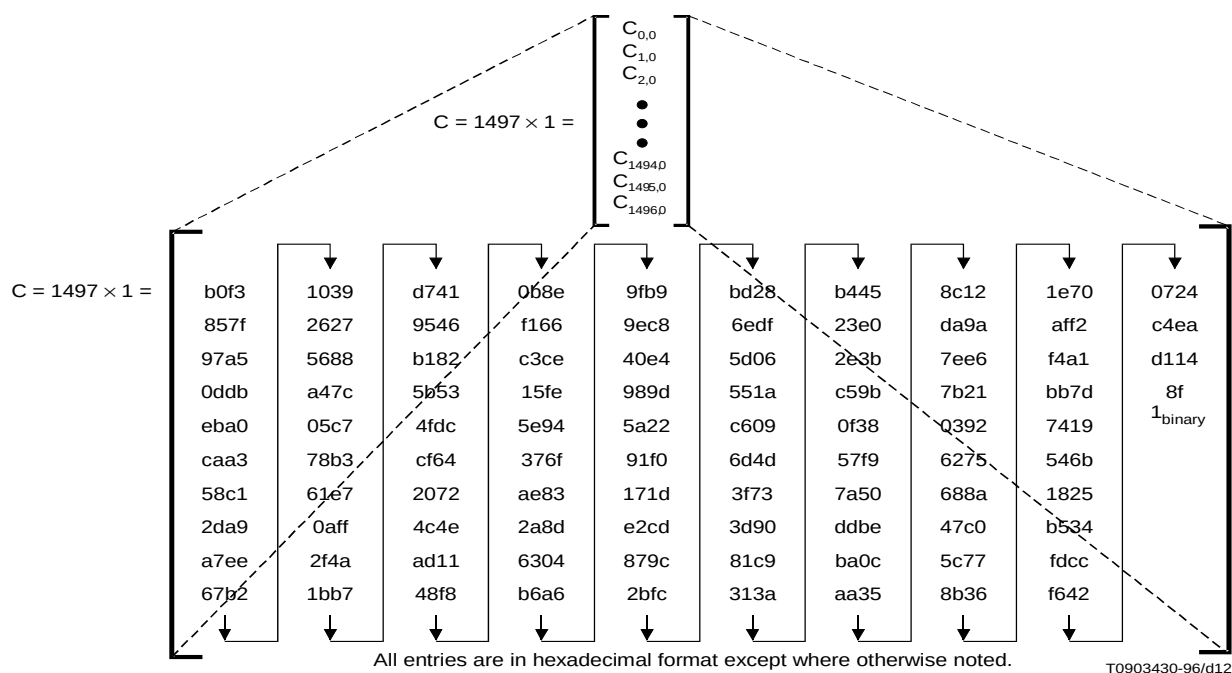


Figure B.4 – "C" column vector (replicated inside the parity check matrix)

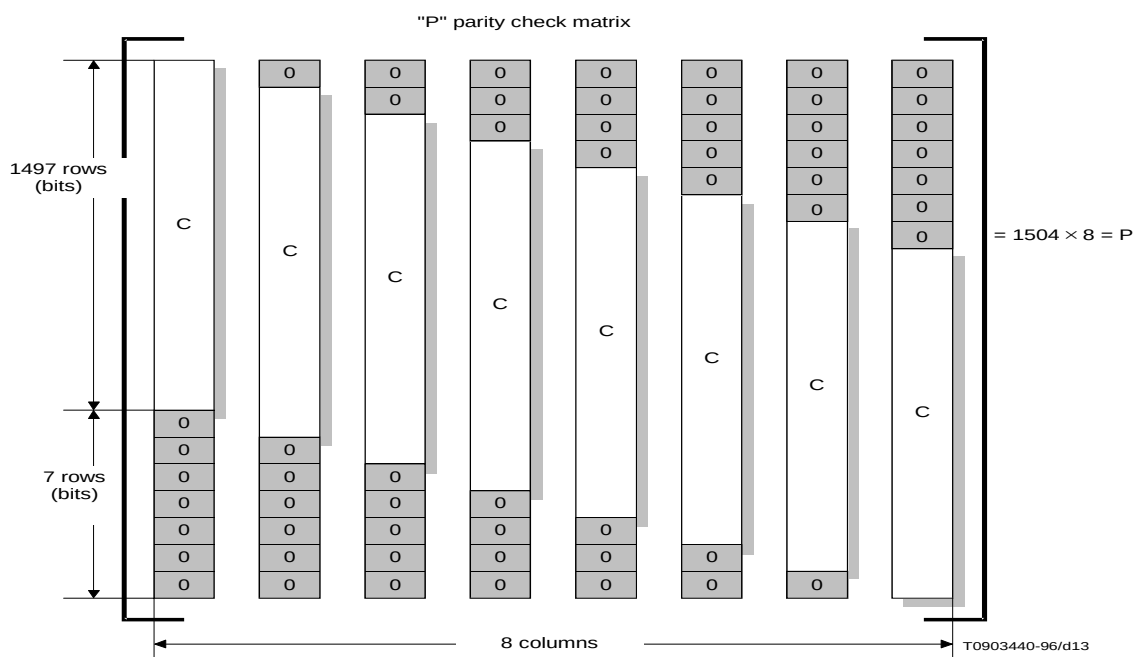


Figure B.5 – Structure of the parity check matrix "P"

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

S = [0100 0111] = 0 × 47

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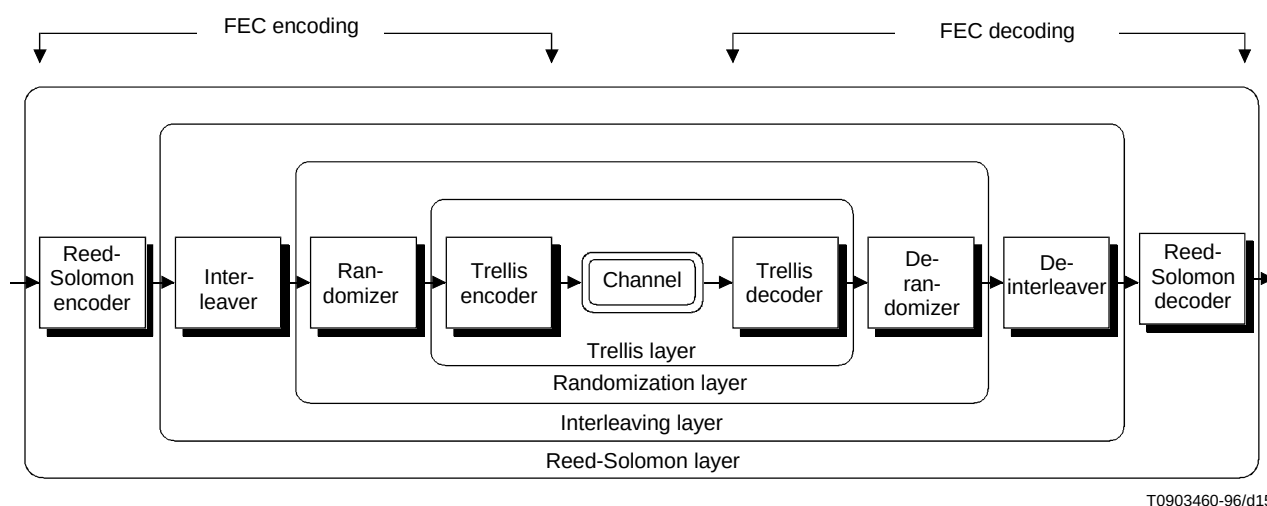
Figure B.6 – Received MPEG-2 vector and parity check matrix multiplication

A valid check sum 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.



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Figure B.7 – 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:

$$p(x) = 0$$

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:

$$m(x) = m_{121}x^{121} + m_{120}x^{120} + \dots + m_1x + m_0$$

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:

$$r(x) = r_4x^4 + r_3x^3 + r_2x^2 + r_1x + r_0$$

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:

$$c(x) = m_{121}x^{126} + m_{120}x^{125} + m_{119}x^{124} + \dots + r_4x^4 + r_3x^3 + r_2x^2 + r_1x + r_0$$

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:

$$\begin{aligned}\hat{c} &= xc(x) + c_- \\ &= m_{121}x^{127} + m_{120}x^{126} + \dots + m_1x^7 + m_0x^6 + r_4x^5 + r_3x^4 + r_2x^3 + r_1x^2 + r_0x + c_- \end{aligned}$$

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

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

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*J symbol periods of delay, and so on, upto the Ith path which has (I – 1)*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.

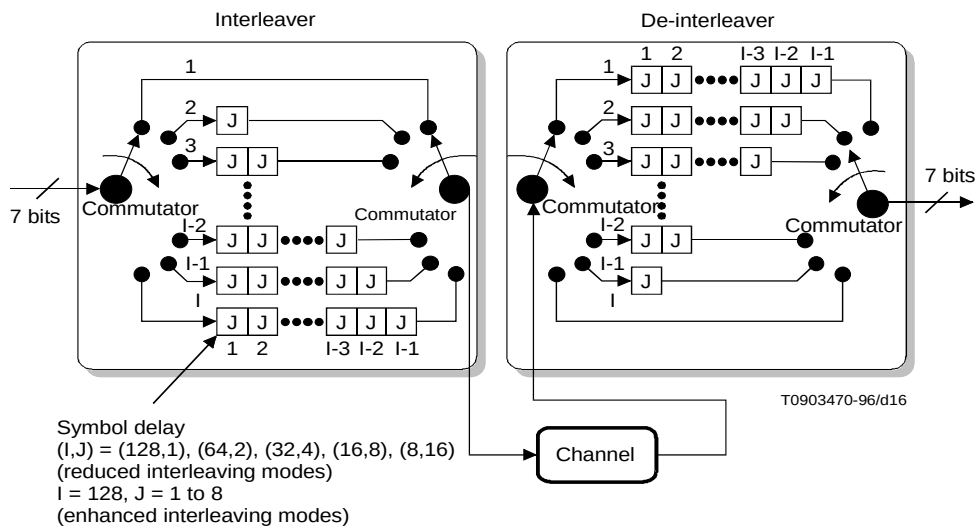


Figure B.8 – Interleaving functional block diagram

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