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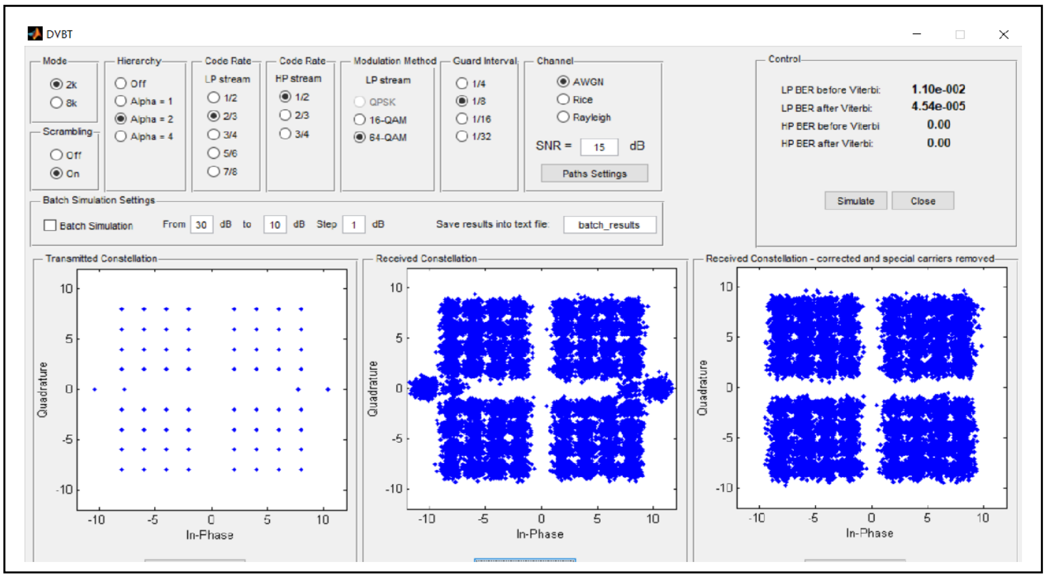
Πρόγραμμα Μεταπτυχιακών Σπουδών
Τεχνολογίες Ήχου, Βίντεο και Μετάδοσης

Master of Science in
Audio, Video and Broadcasting Engineering

# ΜΕΤΑΠΤΥΧΙΑΚΗ ΔΙΠΛΩΜΑΤΙΚΗ ΕΡΓΑΣΙΑ

## Μετάδοση Ψηφιακής Τηλεόρασης Σύμφωνα με την Προδιαγραφή DVB-T και Προσομοίωση στο MATLAB v.2023

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# MSc Thesis

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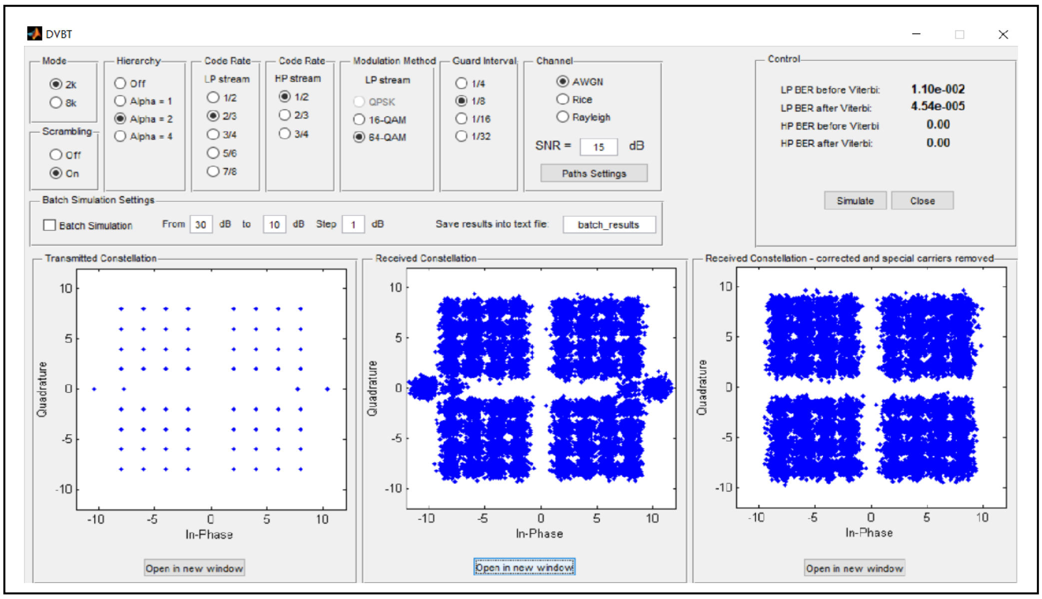


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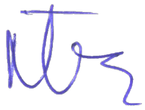

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Ο κάτωθι υπογεγραμμένος Δαλαμήτρας Βασίλης του Γεωργίου, με αριθμό μητρώου MSCAVB-004 φοιτητή του Προγράμματος Μεταπτυχιακών Σπουδών «Τεχνολογίες Ήχου, Βίντεο και Μετάδοσης» του Τμήματος Ηλεκτρολόγων και Ηλεκτρονικών Μηχανικών της Σχολής Μηχανικών του Πανεπιστημίου Δυτικής Αττικής, δηλώνω ότι:

«Είμαι συγγραφέας αυτής της μεταπτυχιακής διπλωματικής εργασίας και ότι κάθε βοήθεια την οποία είχα για την προετοιμασία της, είναι πλήρως αναγνωρισμένη και αναφέρεται στην εργασία. Επίσης, οι όποιες πηγές από τις οποίες έκανα χρήση δεδομένων, ιδεών ή λέξεων, είτε ακριβώς είτε παραφρασμένες, αναφέρονται στο σύνολό τους, με πλήρη αναφορά στους συγγραφείς, τον εκδοτικό οίκο ή το περιοδικό, συμπεριλαμβανομένων και των πηγών που ενδεχομένως χρησιμοποιήθηκαν από το διαδίκτυο. Επίσης, βεβαιώνω ότι αυτή η εργασία έχει συγγραφεί από μένα αποκλειστικά και αποτελεί προϊόν πνευματικής ιδιοκτησίας τόσο δικής μου, όσο και του Ιδρύματος.

Παράβαση της ανωτέρω ακαδημαϊκής μου ευθύνης αποτελεί ουσιώδη λόγο για την ανάκληση του πτυχίου μου».

Ο Δηλών

Δαλαμήτρας Βασίλειος του Γεωργίου

# ΠΕΡΙΛΗΨΗ

Η παρούσα Διπλωματική Εργασία παρουσιάζει το σύστημα DVB-T (Digital Video Broadcasting Terrestrial), το οποίο είναι το ευρωπαϊκό πρότυπο του συνόλου DVB (που τυποποιήθηκε από το ETSI), για τη μετάδοση ψηφιακής επίγειας τηλεόρασης. Επιπλέον, παρουσιάζεται ένας αναπτυγμένος κώδικας για προσομοιωμένη μετάδοση σύμφωνα με την προδιαγραφή DVB-T, μέσω της γλώσσας προγραμματισμού MATLAB v.2023. Εξετάζεται ο λόγος σφάλματος (BER) πριν και μετά την αποκωδικοποίηση Viterbi, καθώς και μετά από όλες τις διορθώσεις σφαλμάτων κατά τη διάρκεια των προσομοιωμένων μεταδόσεων. Αυτό φωτίζει τον σκοπό των λειτουργικών μπλοκ και την επίδραση των ρυθμίσεων στα προκύπτοντα σφάλματα στη μετάδοση. Πραγματοποιείται η προσομοίωση της επίδρασης της τεχνικής διαμόρφωσης που χρησιμοποιείται (QPSK, 16-QAM και 64-QAM - όλες μη ιεραρχικές), η προσομοίωση της επίδρασης του λόγου κωδικοποίησης του συνελικτικού κωδικοποιητή, η προσομοίωση της ιεραρχικής διαμόρφωσης και η προσομοίωση της επίδρασης του καναλιού μετάδοσης. Παρουσιάζονται τα αποτελέσματα με διαγράμματα για το BER και τα συμπεράσματα για κάθε προσομοίωση. Τέλος, παρουσιάζονται γενικά συμπεράσματα για την προσομοίωση του συστήματος DVBT.



ΛΕΞΕΙΣ-ΚΛΕΙΔΙΑ: AWGN Channel, BER, DVBT, HDTV, MATLAB, MPEG, NTSC, OFDM, PAL, QAM, QPSK, REYLEIGH Channel, RICEAN Channel, SDI, SDTV, SECAM, SNR

# ABSTRACT

This Msc Thesis presents the DVB-T (Digital Video Broadcasting Terrestrial) system, which is the European standard of the DVB consortium (standardized by ETSI) for the broadcast transmission of digital terrestrial television. Additionally, a developed code for simulated transmission according to the DVB-T specification is presented, using the programming language MATLAB v.2023. The bit error rate (BER) is examined before and after Viterbi decoding, as well as after all error corrections during simulated transmissions. This illuminates the purpose of functional blocks and the influence of settings on the resulting errors in the transmission. Simulations are carried out on the impact of the modulation technique used (QPSK, 16-QAM, and 64-QAM - all non-hierarchical), the effect of the convolutional encoder's code rate, the simulation of hierarchical modulation, and the influence of the transmission channel. The results are presented with diagrams for the BER and conclusions for each simulation. Finally, general conclusions are presented for the simulation of the DVBT system.



KEYWORDS: AWGN Channel, BER, DVBT, HDTV, MATLAB, MPEG, NTSC, OFDM, PAL, QAM, QPSK, REYLEIGH Channel, RICEAN Channel, SDI, SDTV, SECAM, SNR

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# INTRODUCTION:

In the next chapters, we present the analog TV modulation and transmission used in the past, the transition to the digital video creation, the compression of the digital TV signal, and the MPEG format for a DVB-T transmission. A simulated transmission according to DVB-T Specification, via the MATLAB v.2023 programming language was developed in order, to obtain information regarding the block structure of the encoder/decoder and modulator/demodulator in the DVB-T system. Bit error rate (BER) before and after Viterbi decoding, as well as after all error corrections is examined during simulated transmissions. This illuminates the purpose of functional blocks and the effect of the DVBT parameters settings on resulting errors in the transmission. Conclusions for different cases of simulations are presented to determine the role of the DVB-T blocks in the transmission.

# CHAPTER 1:
# Analog TV Broadcasting

## 1.1 Analog TV

Television sets are the leading entertainment gadgets globally. In 2021, statistics from the worldwide TV industry indicated that around 1.72 billion households owned a television. Projections suggest that this number is set to exceed 1.8 billion by 2026[1]. Radio and television play crucial roles in updating the public with news and events that impact their daily lives while also entertaining them. They also hold educational value, as they can disseminate courses and informative content. As of now, in regions where internet access is limited, TV and radio are especially vital communication tools. [2].

Early television iterations primarily relied on electromechanical processes. However, by around 1950, after numerous advancements, the majority of broadcasting systems transitioned to fully electronic mechanisms. This transition was also observed in the USA, where, by 1942, the Federal Communications Commission (FCC) embraced the guidelines set by the National Television System Committee (NTSC). [3]. Subsequently, several proposals for color TV systems emerged. In Europe, the Sequential Couleur A Memoire (SECAM) system, which translates to "sequential color with memory," was introduced. This was followed by the Phase Alternating Line (PAL) system, which, with minor modifications, was implemented not only in Europe but globally. Meanwhile, in the United States, the NTSC proposed a television system to the FCC. This NTSC standard received approval and was also adopted by numerous other nations.[4, 5, 6].

## 1.2 Color Management systems

Primarily two significant analog television standards existed: the 625-line system operating at a 50 Hz frame rate and the 525-line system at a 60 Hz frame rate. Consequently, the composite color video-and-blanking signal (CVBS, CCVS) of these two systems was channeled using these three distinct color transmission standards:

- *PAL (Phase Alternating Line)*
- *NTSC (National Television System Committee)*
- *SECAM (Séquentiel Couleur a Mémoire)*

All three major systems, NTSC, PAL, and SECAM, utilize the luminance signal, labeled as Y. This signal, Y, can be immediately applied to monochrome TVs, preserving its compatibility with older television models. Given that many users had invested in black and white TVs, ensuring a smooth transition was pivotal. Hence, when integrating color information into the CVBS signal, it was paramount not to interfere adversely with the legacy equipment. As such, the color data is transmitted parallel to the brightness signal. There's a distinction in the bandwidths assigned to the luminance and color signals. For example, in the PAL system prevalent in Europe, the Y signal's bandwidth hovers around 5.5 MHz, whereas the U and V signals are approximately 1.3 MHz. The NTSC system in the US shows a comparable distinction in bandwidths, particularly with its luminance and its I and Q color differential signals. The distinct factor in these color TV systems is their approach to generating these color signals, with NTSC using I and Q.

$$I = -0.27\,(B-Y) + 0.74\,(R-Y)$$

$$Q = 0.41\,(B-Y) + 0.48\,(R-Y) \qquad (1)$$

For the PAL system, the color indicators are termed U and V, and they're derived from specific subsequent mathematical operations:

$$U = 0.49\,(B-Y)$$

$$V = 0{,}88\,(R-Y) \qquad (2)$$

In the SECAM system, the color designations are referred to as $D_R$ and $D_B$, and their values are determined through certain following mathematical processes:

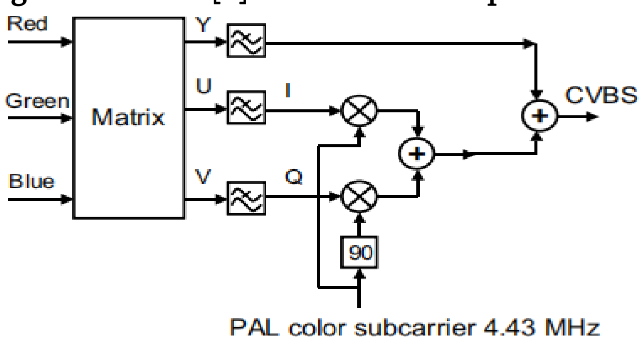
$$D_R = -1.9\,(R-Y)$$

$$D_B = 1.5\,(R-Y) \qquad (3)$$

We notice that in all three cases the color differences Cb and Cr are used. The color signals are modulated by an IQ modulator (NTSC and PAL) or by modulation frequency (SECAM). In the case of PAL the modulation IQ frequency is :

$$f_{sc} = 283.75\,f_h + 25\ \text{Hz} = 4.43\ \text{MHz} \qquad (4)$$

**In the SECAM framework, the chrominance signal undergoes frequency modulation using two distinct subcarriers on a per-line basis. This configuration gives SECAM an edge over PAL, as it's less sensitive to phase variations line-by-line, ensuring consistent color without shifts caused by phase differences during transmission.[7]. The structural representation of**




**the PAL modulator can be seen in**

Figure 1.

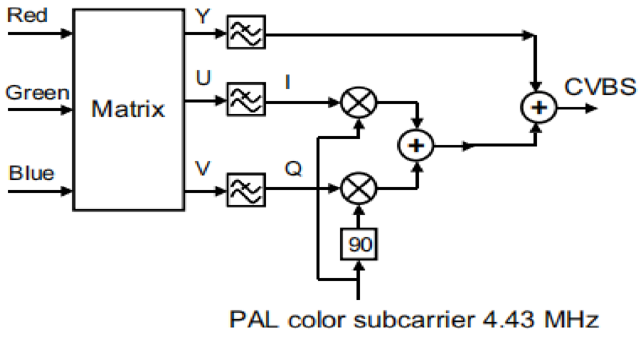



**Figure 1:** Block diagram of a PAL modulator

The composite video signal for PAL, NTSC, or SECAM (Figure 2) is produced by combining the monochrome signal, synchronization details, and the chrominance data, now termed as the CCVS (Composite Color, Video, and Sync) signal. The CCVS signal representation of a color bar signal is depicted in Figure 2. The color burst is distinctly visible. This color burst serves the purpose of transmitting the reference phase for the color subcarrier to the receiving device, enabling its color oscillator to synchronize with it. [8].

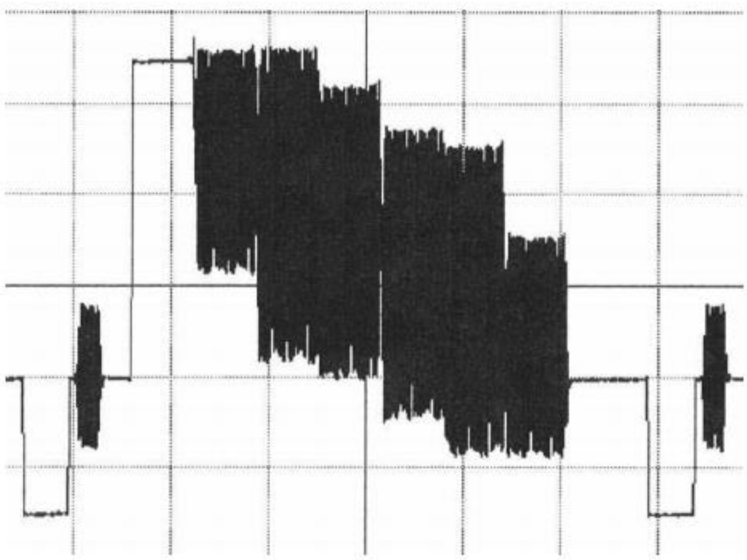


**Figure 2:** Oscillogram of a CVBS, CCVS (composite color video and sync) signal.

# CHAPTER 2:
# Analog TV Signal Modulation and Transmission

## *2.1 Analog Modulation Techniques:*

Essentially, there are three primary forms of analog modulation. These are amplitude modulation, frequency modulation, and phase modulation. Each of these has numerous subcategories, derivatives, or variations as mentioned in Table 1.

**Table 1:** Classification of Analog Modulation Technique.

| Sr. No. | MODULATION TECHNIQUES | REPRESENT ATION | TYPE |
|---|---|---|---|
| 1 | Amplitude Modulation Double-Sideband Suppressed Carrier | AM DSB-SC | Linear |
| 2 | Amplitude Modulation Double-Sideband With Full Carrier | AM DSB-FC | Linear |
| 3 | Amplitude Modulation Single-Sideband Suppressed Carrier | AM SSB-SC | Linear |
| 4 | Amplitude Modulation Single-Sideband With Full Carrier | AM SSB-FC | Linear |
| 5 | Amplitude Modulation Vestigial-Sideband | AM VSB | Linear |
| 6 | Narrow-Band Frequency Modulation | NBFM | Non-Linear |
| 7 | Wide-Band Frequency Modulation | WBFM | Non-Linear |
| 8 | Phase Modulation | PM | Non-Linear |

Within the realm of Amplitude Modulation, various derivatives exist. As evident from Figure 4, Single Side Band Suppressed Carrier (SSB-SC) showcases reduced bandwidth and power demands compared to Double Side Band Suppressed Carrier (DSB SC), Double Side Band Full Carrier (DSB FC), and Single Side Band Full Carrier (SSB FC). However, to detect the SSB-SC signal, a sharp cutoff Low Pass Filter (LPF) is essential, which isn't feasible in practice. By adopting the Vestigial Side Band (VSB) method over SSB-SC, one can utilize a low pass filter with a more gradual cutoff. Yet, this method consumes more bandwidth and power than SSB-SC but less than both DSBSC and DSB-FC. Theoretically, SSB-SC outperforms other AM techniques, but in practical scenarios, VSB emerges as a superior choice among amplitude modulation techniques. It's worth noting that amplitude-modulated signals need nonlinear amplifiers, which produce unwanted spectral components challenging to filter out. Compared to amplitude and phase modulation, Frequency Modulation stands out as superior. To address the issues in communication systems, the derivative of frequency modulation, known as narrow band FM (NBFM), is commonly used.

Table 2 displays properties such as representation, bandwidth needs, and power demands of different analog modulation methods. FM's significant advantage over AM is its ability to counteract noise effects, although it requires more bandwidth. A significant drawback of analog modulation systems, like AM, FM, and PM, for extended channel communication is that once noise infiltrates any part of the channel, it persists throughout. In stark contrast, when a digital signal is modulated and sent, the signal received is much less affected by the noise at the receiver's end [9].

**Table 2:** Performance Analysis of Analog Modulation Scheme.

| Sr. No. | TYPE OF ANALOG MODULATION | BANDWIDTH (B. W.) | % POWER SAVING | POWER REQUIREMENT |
|---|---|---|---|---|
| 1 | AM-DSB-FC | $2\omega_m$ | Standard | 3/2 Pc |
| 2 | AM-DSB-SC | $2\omega_m$ | 66.67% | 5/4 Pc |
| 3 | AM-SSB-FC | $\omega_m$ | 16.67% | 1/2 Pc |
| 4 | AM-SSB-SC | $\omega_m$ | 83.33% | 1/4 Pc |
| 5 | AM-VSB | $> \omega_m$ | >SSB-SC | Greater than SSB-SC |
| 6 | NBFM | $2\omega_m$ | Same as DSB-SC | Same as DSB-SC |
| 7 | WBFM | $2\omega_m M_F$ | More than NBFM | More than NBFM |
| | $\omega_m$ = modulating frequency | $M_F$ = modulation index in FM | $M_P$ = modulation in PM | Pc = carrier power |

## 2.2 Transmission Methods

Analog television broadcasting can be channeled through three main routes: Terrestrial, satellite, and cable. The preferred mode of transmission often hinges on the specific countries or regions in focus. For terrestrial and cable transmission of analog TV signals, amplitude modulation is predominantly adopted, with negative modulation being the common choice. The exception is the French Standard L, which uses positive modulation. The accompanying sound sub-carriers are usually frequency modulated. To economize on bandwidth, the vision carrier utilizes VSB-AM (vestigial side band amplitude modulation), meaning part of the spectrum gets eliminated through band-pass filtering. The fundamental blueprint of this can be seen in Figure 3 and Figure 4.

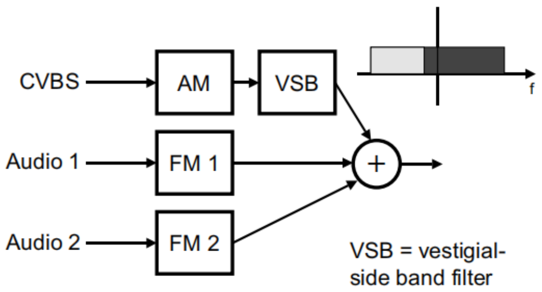



**Figure 3:** Principle of a TV modulator for analog terrestrial TV.

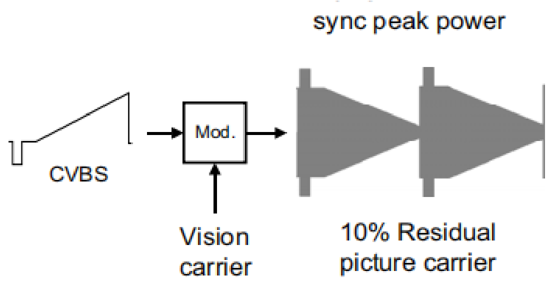



**Figure 4:** Vision modulator

## 2.2.1 Analog transmission

In analog television, the signal basically consists of three distinct signals (of which one has two components):

- Luminance signal (Y) which was used in black and white TV and continued to be used in color analog, initially for compatibility reasons, but also so that it can be managed differently (e.g. regarding sampling) because of its greater importance than that of color for image quality.
- The color signal, which consists of two components namely the color differences. Depending on the color system used, the signals that concerning color arise in another way. In the case of NTSC we have I and Q, in PAL the U and V are used, while in the case of SECAM the are used DR and DB. In all cases these components result directly from color differences with scaling.
- The audio signal, which is modulated and transmitted separately, within the specific channel.

These signals are summed and modulated. The overall modulated signal is integrated in the bandwidth of each channel. The bandwidth in case the channel belongs to the band of VHF is 7 MHz while in the case of UHF it has a bandwidth of 8 MHz. Regarding the transmission of the colour signal, Greece had initially adopted SECAM, however in its beginnings in the 1990s there was a transition to PAL B/G systems. A corresponding transition was made for important number of countries that had originally adopted SECAM. PAL B corresponds to the VHF channel (ie with 7 MHz bandwidth) and PAL G on a UHF channel (i.e. 8 MHz bandwidth) [9, 10].

#### 2.2.1.1 Black/White TV

**a) Luminance signal modulation**

The luminance signal (Y) has a bandwidth of about 5 MHz (calculated to a slightly larger value, however, because at the edges of the screen the detail is smaller, we consider the spectrum of the signal approaches this value). The brightness signal is amplitude modulated (Amplitude Modulation) to transmit. Because the channel has a bandwidth of 7 (VHF) or 8 (UHF) MHz it Double Side Band (DSB) configuration cannot be used since both laterals cannot be transmitted at the same time through the specific channel (in such a case it would be necessary spectrum of at least 10 MHz for brightness only). In essence, however, transmission is not required of both sides since they carry the same information. As an alternative to the bilateral AM configuration, the unilateral one could be applied modulation (Single Side Band, SSB) in which only one side is transmitted. It is saved with him the way significant bandwidth and the channel we have is sufficient to transmit the signal. The disadvantage in this case is that particularly complex filters are required cut off one side without affecting the other. Moreover, brightness has a spectrum that starts from very low (practically zero) frequencies (the zero frequency corresponds to the average brightness of the signal). The combination of the above two observations created the need to use a compromise solution: the use of residual side-band modulation (Vestigial Side Band Modulation, VSB) which reconciles the requirements between SSB and DSB. In particular, VSB provides for the transmission of the entire one of the sides and one section from the other side. This reduces the requirements for the band-pass filter (since it is not required to be very sharp in separating the channel sidebands), as well as the bandwidth requirements from the channel (it can be placed both on a channel VHF or UHF). Also, signals with zero frequency are allowed to pass through (that is, they have DC component). The following Figure 5, shows the total spectrum of the luminance signal and the mode which is divided into the part that is transmitted and the part that is not taken into account.

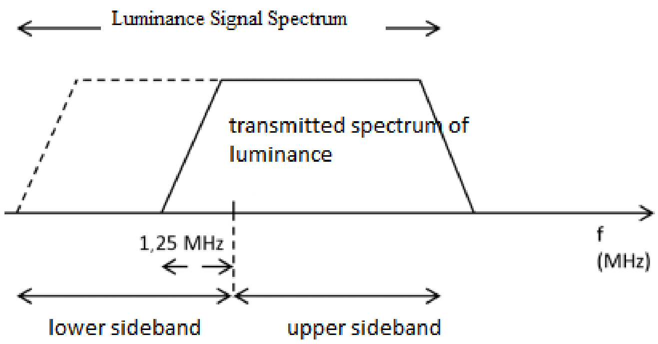



**Figure 5:** Total and transmitted luminance signal spectrum.

In particular, we observe that the entire upper lateral is transmitted from the lower lateral as well a rate corresponding to 0.75 MHz of its range is transmitted. Of course, there is also a loss in the spectrum due to the fact that inevitably the filters used have some slope, that is, they differ from the ideal whose cut-off curve is vertical. This loss, the deviation, that is, from the ideal case, must be considered during its design equipment.

### b) Audio Signal Modulation

The audio signal in the case of analog television is transmitted by frequency modulation (Frequency Modulation, FM). FM modulation is more efficient in transmitting sound regarding to face the noise. The sound carrier frequency is placed at a certain distance from the brightness carrier frequency. Specifically, in the case of the PAL B/G standards, this distance is 5.5 MHz. Thus, if the luminance carrier is 1.25 MHz away (of which 0.75 MHz corresponds to the lower side) from the beginning of the channel bandwidth, the audio carrier will be 6.75 MHz from the beginning of the channel.

Specifically:

$$f_{bc} = f_n + 1{,}25 \text{ MHz} \quad (5)$$

$$f_{sc} = f_{bc} + 5{,}5 \text{ MHz} \quad (6)$$

Where, $f_n$ is the frequency at which the channel spectrum starts, $f_{bc}$ the brightness signal carrier and $f_{sc}$ the sound signal carrier.

The distance between the two carriers can be varied in other standards (which are also PAL), e.g. in PAL D/K it is 6 MHz. The following Table 3 shows the characteristics that concern the various versions of the PAL system [11].

**Table 3:** Characteristics of PAL sub-classes.

| | PAL B | PAL G, H | PAL I | PAL D, K | PAL M | PAL N |
|---|---|---|---|---|---|---|
| frequency band | VHF | UHF | VHF/UHF | VHF/UHF | VHF/UHF | VHF/UHF |
| fields/sec | 50 | 50 | 50 | 50 | 60 | 60 |
| lines | 625 | 625 | 625 | 625 | 525 | 525 |
| Active Lines | 576 | 576 | 576 | 576 | 480 | 480 |
| Channel Bandwitdh | 7 MHz | 8MHz | 8MHz | 8MHz | 6MHz | 6MHz |
| Video Signal Bandwitdh | 5,0 MHz | 5,0 MHz | 5,5MHz | 6,0 MHz | 4,2MHz | 4,2MHz |
| colour carrier frequency | 4,43MHz | 4,43 MHz | 4,43 MHz | 4,43 MHz | 3,58 MHz | 3,58 MHz |
| Gap of Luminance carrier frequency and Sound carrier frequency | 5,5 MHz | 5,5 MHz | 6,0 MHz | 6,5 MHz | 4,5 MHz | 4,5 MHz |

#### 2.2.1.2 Black and White Television Signal Spectrum

The following Figure 6 shows the spectrum of the black and white television signal (i.e. the signal consisting of the luminance signal and the audio signal) on a VHF channel, 7 MHz bandwidth. In the case of a UHF channel (8 MHz) the shape of the spectrum is corresponding. This happens after so much the distance of the light and audio carrier frequencies as well as the bandwidth of the signal luminance remain the same[12]. In VHF, PAL B is used (in the case of Greece) while in the case of UHF PAL G is used.

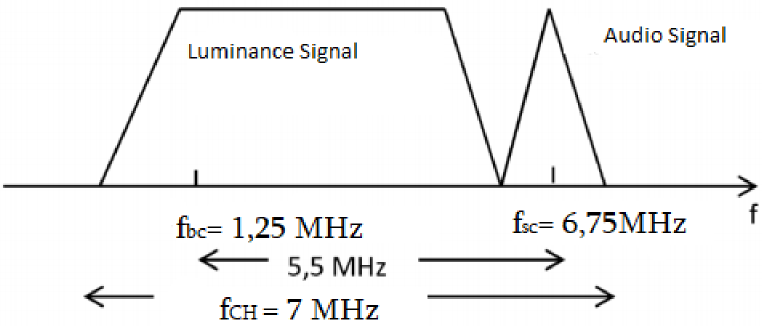



**Figure 6:** Spectrum of black and white television signal.

We notice that the channel bandwidth is 7 or 8 MHz (VHF, UHF respectively) and in this all signals must be included. Specifically:

- The luminance signal occupies the largest percentage of the bandwidth (from
- beginning of the channel up to 6.5 MHz).
- The audio signal after modulation occupies from 6.5 MHz up to 7 MHz,
- about 0.5 MHz.
- Luminance and audio carrier frequencies are 1.25 MHz and 6.75 MHz apart respectively from the beginning of the channel. The distance between them is 5.5 MHz.
- Due to the Vestigial Side Band (VSB) configuration, its bearing luminance includes a percentage of the lower side, around 0.75 MHz.

The next Table 4, gives the frequencies for the carriers of the luminance signal and the audio signal for some indicative channels in UHF. Specifically, for channels 21 to 24 for the PAL G system (625 lines, 8 MHz channel).

**Table 4:** Carrier frequencies for channels 21 through 24.

| CHANNEL | LUMINANCE CARRIER FREQUENCY(MHz) | Sound Carrier Frequency |
|---|---|---|
| 21 | 471,25 | 476,75 |
| 22 | 479,25 | 484,75 |
| 23 | 487,25 | 492,75 |
| 24 | 495,25 | 500,75 |

The used channels belong to the following areas:

- I (VHF) channels 2 to 4, frequency range 47 to 68 MHz.
- III (VHF) channels 2 to 4, frequency range 174 to 230 MHz.
- IV (UHF) channels 38 to 69, frequency range 606 έως 862 MHz.
- V (UHF) channels 21 to 37, frequency range 470 to 606 MHz.

The remaining frequencies, concerning the intermediate intervals, have been assigned to other uses.

### 2.2.1.3 Coloured TV Signal Modulation

As it mentioned in previous paragraph, at the three colored transmission standards are given (1), (2), (3):

a) ***In the case of NTSC*** the colour signals are called I and Q:

$$I = -0.27\ (B-Y) + 0.74\ (R-Y)$$

$$Q = 0.41\ (B - Y) + 0.48\ (R - Y)$$

b) ***In the case of PAL*** the colour signals are called U and V:

$$U = 0.49\ (B-Y)$$

$$V = 0{,}88\ (R-Y)$$

c) ***In the case of SECAM*** the colour signals are called DR and DB:

$$D_r = -1.9\ (R-Y)$$

$$D_b = 1.5\ (R-Y)$$

It is reminded that, although the colour information depends on three independent variables, namely the primary colours R, G, and B, is chosen to send the luminance and the two colour differences (we mentioned above because the values of these color differences are systematic larger of the three-color differences). Due to the independence of these quantities, the information is equivalent (in each case, we have a system of three independent variables) [13].

Color signals have less information (thus bandwidth) than brightness signal. Amplitude Modulation (AM) is used for transmission and both bands of the signal are sent, with the carrier suppressed. Specifically, the Double Side Band Suppressed Carrier (DSB-SC) mechanism is used. To perform AM DSB-SC modulation, the multiplication of the carrier by the information signal is required [14].

Let us consider an example to understand its mechanism modulation.

Let c(t) be the color information signal and let the carrier be a sine signal at $f_{CC}$ frequency (carrier color frequency). The AM DSB SC configuration is given by the following relationship:

$$c(t)\cos(2\pi f_{cc}t + \varphi) \qquad (7)$$

Here comes the following issue: The chrominance signal consists of two signals (derived from the Cb and Cr color differences), which are independent of each other, so a mechanism is required to simultaneously modulate these two signals with a carrier. This will enable more efficient use of bandwidth. For this reason, the orthogonal amplitude modulation QAM (Quadrature Amplitude Modulation) is used. QAM allows both signals to be transmitted simultaneously using the same carrier and improving (doubling) the performance of the DSB, in terms of the used bandwidth and with the same power consumption.

With QAM we can modulate a carrier based on two different voltages. The basics characteristics of QAM modulation, for the case of color television, are: Color information signals are derived based on the system used, e.g.

- For the case of PAL they are U and V and the resulting signal is called chroma signals.
- Two subcarriers are used with the same frequency but with a phase difference of π/2.

The range of a color video signal is limited, not going beyond roughly 1.5 MHz. As a result, the bandwidth needed to transmit color data is significantly less than 5 MHz. This characteristic permits the combination of the narrow band chrominance (color) and wide-band luminance (brightness) signals within the standard 7 MHz TV channel. This combination is realized by modulating the color signal on a carrier frequency that's within the channel's regular bandwidth. This specific frequency, known as the color sub-carrier frequency, is positioned near the top end of video frequencies to prevent clashing with the black and white signal. The bandwidth for color signals is confined to approximately ± 1.2 MHz around this sub-carrier. Below, Figure 7 illustrates the placement of monochrome (image), color, and audio signal spectrum, all accommodated within the unified channel bandwidth of 7 MHz.

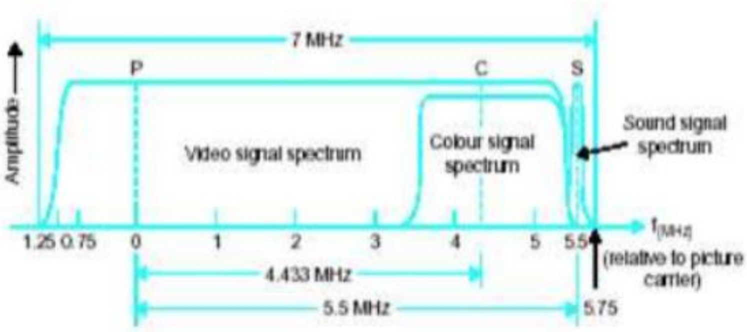



**Figure 7:** Channel Bandwidth for PAL colored TV signal.

Of significance is the American television system, which utilizes a 6 MHz channel bandwidth. In this system, as depicted in Figure 8 , the color sub-carrier is positioned 3.58 MHz apart from the picture carrier.

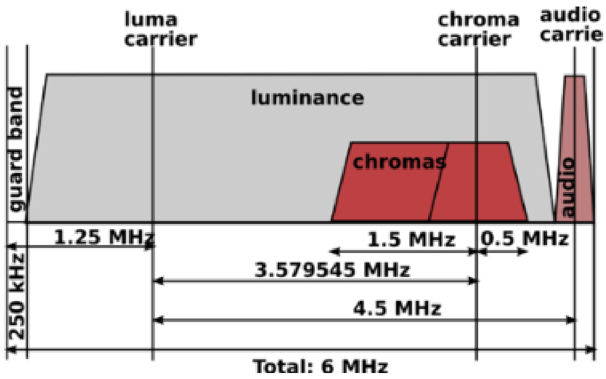



**Figure 8:** Channel Bandwidth for NTSC colored TV signal.

# CHAPTER 3:
# Serial Digital TV Signal

## *3.1 Digital Video Creation*

The mechanism of creation a Digital Video Signal is based on standard ITU-BT R601 (Studio Encoding Parameters of Digital Television for Standard 4:3 and wide screen 16:9 aspect ratio) [15]. This signal is generated and exploited basically in the studio, and it is not suitable (mainly because of the high transmission rate that will be required) for transmission.

The first version of the standard was created in the early 1980s and, since then, it was managed to bring together different directions that had begun to be created in different countries in relation to the shift from analog to digital TV broadcasting. Specifically, the ability to download and play videos in interlaced format, with sequential sending of the fields, using the color representation mode and sampling the luminance signal with frequency 13.5 MHz.

As an extension of this, BT.656 describes a digital video protocol for the transmission of non-standard definition PAL or NTSC compressed signal (625 and 525 lines respectively). It is based on in 4:2:2 type encoding with the parameters defined in the ITU-RBT.601 standard [16].

The first step in creating the digital signal is to get the values of the fundamentals of red, green and blue color components (Red, Green, Blue) as three distinct signals that we denote by the next three symbols (R, G, B). Initially, the signals are corrected with Gamma correction. This process involves matching luminance to voltage through an exponential relationship according to the specifications of CCIR Rec.709 (17). This particular standard deals with parameters related to 16:9 HD TV video.

In the projection system (television receiver) gamma correction takes place so that the voltage relationship among video signal and luminance to become linear again. Gamma correction is done with a function in independent variable whose exponent is the inverse of the exponent used in inverse gamma correction, i.e. (1/0.45=2.2).

These three signals are then processed to give the luminance signal Y and the chrominance signals, i.e., the Cb and Cr chrominance differences.

$$Y = 0{,}299\ R + 0{,}587\ G + 0{,}114\ B \tag{8}$$

$$Cb = R\text{-}Y = 0{,}701R - 0{,}587G - 0{,}114B \tag{9}$$

$$Cr = B\text{-}Y = -0{,}299T - 0{,}587G + 0{,}886B \tag{10}$$

The values of the variable Y range from 0 to 1.0, while the color difference Cr from -0.701 to +0.701 and Cb from -0.886 to +0.886. To map the value ranges to the unit, it is done scaling values and color differences range from -0.5 to 0.5.

$$Cb=0,56*(B-Y) \quad (11)$$

$$Cr=0,71*(R-Y) \quad (12)$$

At this point we consider that we have the original form of the digital signal, which includes brightness and color differences.

## *3.2 Filtering and Sampling*

Then the signal is filtered using a low-pass filter. Specifically, the luminance signal is filtered at 5.75 MHz, while the chrominance signals at 2.75 MHz with use of low-pass filters. Similar filtering happens in the color signal on analog TV (NTSC, PAL, SECAM as well, reduced to 1,3Mhz.

Then, the Y, Cb, Cr signals are converted to discrete-time signals, with the use of sampling. Sampling frequencies for signals should exceed twice than the maximum frequency component (according to the filtering they have undergone) to satisfy Shannon's theorem and appear in the next Table 5. We notice that the luminance sampling frequency is twice than the sampling of color differences and as result of that, the number of luminance samples being equal to number of samples concerning color differences.

**Table 5:** Luminance and chrominance sampling rates.

| Signal | Sampling Frequency (MHz) |
|---|---|
| Y | 13,5 |
| Cb | 6,75 |
| Cr | 6,75 |

The procedure followed is shown in the following Figure 9. The block diagrams of the system is the mechanism for generating the R, G, B signals, the transformation matrix to Y, Cb, Cr, the Low Pass Filters (LFFs) and the sampling system (analog to digital) which performs sampling and then quantization with 8 or 10 bits.

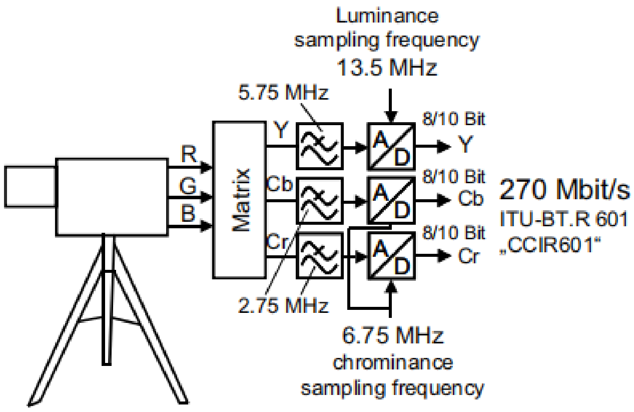



**Figure 9:** Digitization of luminance and chrominance.

We notice that the sampling gives a rate which is estimated at 270 Mbps (at case of using 10 bits per sample). Using 8 bits gives a corresponding rate of 216 Mbps.

## *3.3 Quantization*

The next step in signal processing is its quantization, which is performed with the mapping each sample to a sequence of bits. The quantization of the luminance signals and the chrominance are performed using a minimum of 8 bits for luminance and 4 bits for color differences. Common values are 8 or 10 bits for all three sizes. In case we have uniform quantization using 8 and 10 bits, we will have 28 and 210 quantization levels. The number of values are respectively 256 and 1.024. The number of bits we choose for quantization is related to the quality of quantization and specifically the signal to quantization noise ratio (Signal to Quantization Noise Ratio, SQNR). The SQNR is related to the approximation (rounding) that performed and depends from the number of the available levels . It is about the difference between its actual price of the analog signal and the nearest quantized value. This rounding is the error quantization (or, otherwise, quantization noise). Since the quantization error is at most half of the quantization interval, we can calculate it by using the number of bits let N. Specifically, let the maximum value is $2^{N-1} - 1$ ($\tilde{}2^{N-1}$) and in the negative (absolute value): $-2^{N-1}$. The value of SQNR can expressed as follows:

$$\mathrm{SQNR} = 20 \times \log \frac{2^{N-1}}{\frac{1}{2}} = 20 \times N \times \log 2 = 6.02N(\mathrm{dB}) \qquad (13)$$

The transmission rate for sampling rates of 13.5 MHz and 6.75 MHz respectively and quantization (α) with using 8 bits for all three resulting components and (b) 10 bits for the three components is resulting from the next relationships.

In the first case we have quantization using 8 bits:

Y: 13,5 MHz * 8 bits / sample = 108 Mbps

Cb: 6,75 MHz * 8 bits / sample = 54 Mbps

Cr: 6,75 MHz * 8 bits / sample = 54 Mbps (14)

** The individual rates add up to 216 Mbps.*

In the second case where quantization using 10 bits we have the following individual rates:

Y: 13,5 MHz * 10 bits / sample = 135 Mbps

Cb: 6,75 MHz * 10 bits / sample = 67,5 Mbps

Cr: 6,75 MHz * 10 bits / sample = 67,5 Mbps (15)

** The individual rates add up to 270 Mbps.*

After the quantization, the luminance and chrominance samples are multiplexed, as described in the next section.

### 3.3.1 Color Sub sampling

The luminance and chrominance sampling in the video signal is as follows: Y Cb Y Cr Y … This sequence is called 4:2:2 and, basically, it is a technique that refers to color sub-sampling, which exploits some of the human characteristics vision system. Specifically, due to the increased sensitivity of the human eye to luminance, in relation to its color sensitivity and given the need to compress it signal (therefore lowering the necessary bandwidth for its transmission) we can we get fewer samples of color information than the samples that concern the brightness. This mechanism called color sub-sampling. In this case, when creating the signal, we have alternating luminance and chrominance samples based on the presented standard above (i.e Y, Cb, Y, Cr, Y, …). However, this is not the only case where we can have variation in the sampling rate of the luminance and chrominance signal. There may be many cases and combinations. For this reason, a way of describing the sub-sampling is needed. For the description, it is enough to give (analog) the rates at which the brightness and color sampling. The following notation is commonly used:

$$J:a:b \quad (16)$$

This description consists of three integers, e.g. 4:2:0, which are interpreted as follows: The first integer *J* gives us how many (luminance) samples we take horizontally as a reference, usually 4. The second integer *a* tells us how many chrominance samples we have in the first J (horizontal) samples, while the third integer, *b*, tells us how many chrominance samples we have in the next J (horizontal, of the next line) samples. Color sub-sampling is done starting from the format 4:4:4 (which has equal samples of luminance and each chrominance component).

The following Figure 10 shows three color sub-sampling scenarios, starting each time from the full 4:4:4.

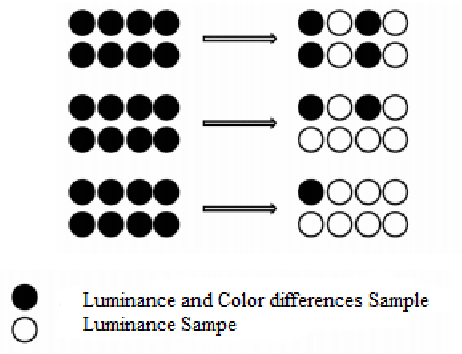



**Figure 10:** Color sub sampling from 4:4:4 to 4:2:2, 4:2:0 and 4:1:0.

As can be understood, color sub-sampling reduces the requirements regarding the transmission rate. Let's calculate the rate again assuming that we have applied color sub-sampling of the 4:2:0 format. The quantization is done using 8 bits for the three components (luminance and color differences).

Y: 13,5 MHz * 8 bits / sample = 108Mbps

Cb: (1/4)*6,75 MHz * 8 bits / sample = 13,5Mbps

Cr: (1/4)*6,75 MHz * 8 bits / sample = 13,5Mbp (17)

The individual streams add up to a total of 135 Mbps. A decrease relative to the original rate is noticed (14).

## *3.4 Video Structure*

In the data flow, specific code words, SAV (start of active video) and EAV (end of active video), denote the commencement and conclusion of the active video signal, as illustrated in Figure 11. The span between EAV and SAV is recognized as the horizontal blanking interval, devoid of any video-related information; that is, the sync pulse isn't present in the digital signal. This interval can be leveraged to relay additional data, like audio signals, commonly termed as '*embedded audio*'.

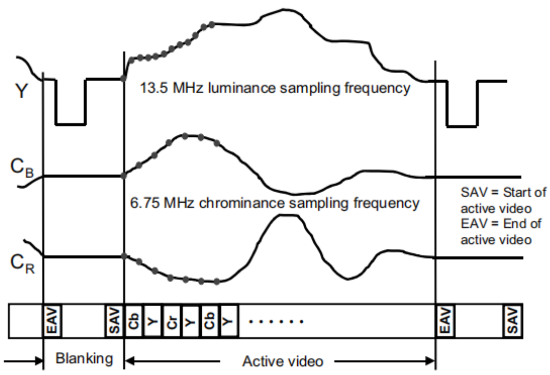



**Figure 11:** Component sampling as the standards set by ITU-BT.R601.

Both SAV and EAV code words, illustrated in Figure 12, are composed of either four 8-bit or 10-bit code words. Both SAV and EAV initiate with a code word where every bit is marked as one, succeeded by two code words with all bits marked as zero. The final, fourth code word holds details about the corresponding field or the vertical blanking interval. This code word plays a pivotal role in identifying the commencement of a frame, field, and the active picture region vertically. The fourth code word's topmost bit always stands at 1. The succeeding bit, whether bit 8 in 10-bit or bit 6 in 8-bit transmission, denotes the field. A zero indicates the initial field's line, while a one signifies the second field's line. The subsequent bit, either bit 7 in 10-bit or bit 5 in 8-bit transmission, highlights the vertical direction's active video section. A zero shows the visually active video segment, while otherwise, it indicates the vertical blanking interval. Bit 6 (for 10-bit) or bit 4 (for 8-bit) clarifies if the current code word is an SAV or EAV – SAV for a zero and EAV otherwise. The bits ranging from 5 to 2 in a 10-bit or 3 to 0 in an 8-bit serve the purpose of error protection for SAV and EAV code words. The fourth code word in the timing reference sequence (TRS) incorporates the subsequent details:

- F = Represents the Field. A '0' indicates the first field, while a '1' signifies the second field
- V = Stands for Vertical Blanking. A '1' here means the vertical blanking interval is currently active.
- H = Distinguishes between SAV and EAV. A '0' is for SAV, and a '1' identifies EAV
- P0, P1, P2, P3 = These are protective bits, designed according to the Hamming code

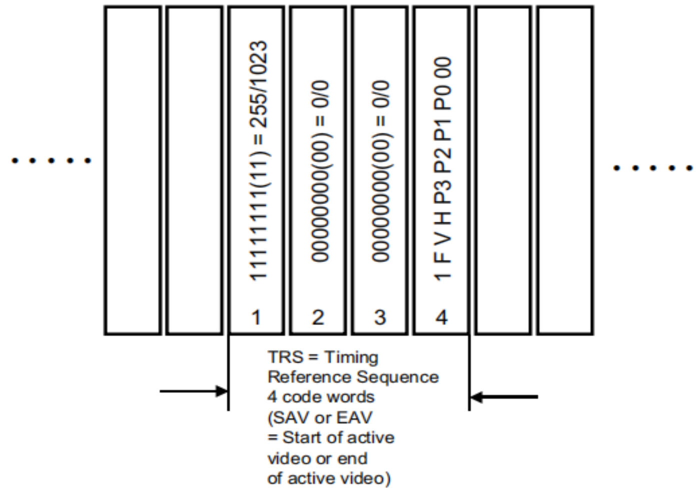



**Figure 12:** SAV and EAV code words in the ITU-BT.R601 signal.

The brightness signal, represented as (Y), along with the color disparity signals, labeled as (Cb, Cr), don't exploit the complete dynamic range they have at their disposal (Figure 13).

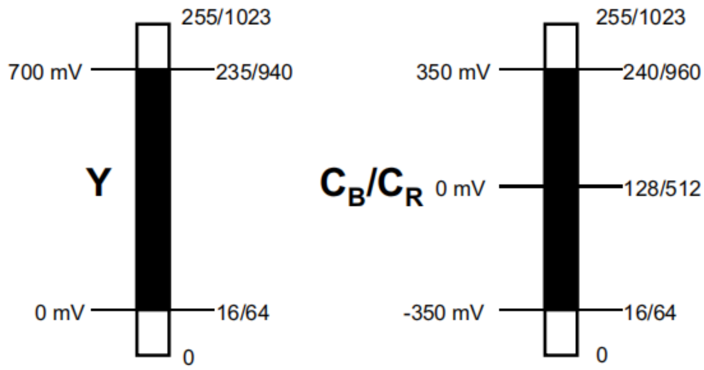



**Figure 13:** Level diagram.

The luminance signal (Y) and the color difference signals (Cb, Cr) don't utilize their entire potential dynamic range. Instead, there's a restricted zone kept as a buffer. This not only offers headroom but also ensures the easy recognition of SAV and EAV. For an 8-bit system, the Y signal lies between 16 and 240 decimal values, whereas for a 10-bit system, it spans from 64 to 960 decimal values. Similarly, for both Cb and Cr, the range is 16 to 240 for 8 bits and 64 to 960 for 10 bits. The space beyond this designated range serves as a buffer and for synchronization recognition. Such a video signal, in alignment with ITU-BT.R601 and typically accessible as an SDI (Serial Digital Interface) signal, is the foundational input for an MPEG encoder [18].

# CHAPTER 4:
# Serial Digital High Definition TV Signal

## *4.1 High Definition Television – HDTV*

The Standard Definition Television (SDTV) signal is still used by many of broadcasting stations in various countries, even though it was created in the 1950s. As we present before, the basic systems they use 625 and 525 lines with 25 and 50 Hz scanning respectively and 4:3 aspect ratios. On the other hand, the 20 century editions of cameras and television receivers can create and reproduce a much higher definition signal. In addition, it is able to use the 16:9 ratio which is also used in modern systems normal definition. The first thoughts about HDTV were doubling the lines and pixels per line. Doing so would increase its size signal, as shown in the Figure 14.

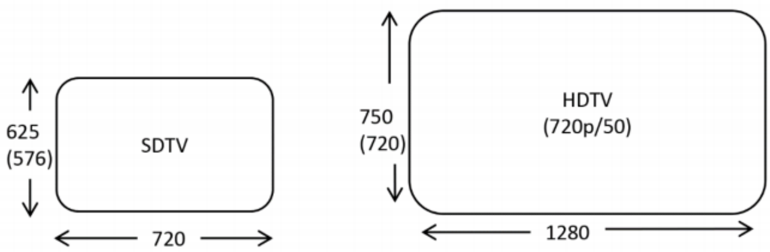



**Figure 14:** Dimension Compare between SD vs HD.

The HDTV requires sophisticated data reduction techniques (MPEG- 4, H.264), which is 2 to 3 times more efficient than its corresponding compression standard MPEG-2 which was commonly used in standard definition digital television. Next, we will consider some features of the high signal generation process clarity, according to the respective standard, namely ITU-R BT.709 and ITU-R BT.1120.

This standard deals with ultra-high-definition television signal issues and specifically with screen resolution, frame rate, color sub sampling, quantization levels and the representation of colors. Due to ITU 1,125 lines with frequencies of 50 Hz and 60 Hz of which 1,080 are active lines and each line has 1,920 pixels on both systems [19],[20]. The EBU – TECH 3299 standard: High Definition (HD), Image Formats for Television Production, (21) refers to 4 systems in Europe as shows Table 6.

**Table 6:** HDTV Systems 1 to 4.

| EBU System | Nomenclature [1] of abbreviations [activeLines$Scanning /frame-rate] | Luma or R'G'B' Samples per active line (S/AL) | Active lines per frame (picture) (AL/F) | Frame rate, Hz | Luma or R'G'B' sampling [2] frequency (fs), MHz | Luma sample periods per total line (S/TL) | Total lines per frame | Net image Bit Rate (4:2:2, 10 bit) [GBit/s] | Corresponding SMPTE system nomenclature |
|---|---|---|---|---|---|---|---|---|---|
| S1 | 720p/50 | 1280 | 720 | 50 | 74.25 | 1980 | 750 | 0.92 | SMPTE 296M System 3 |
| S2 | 1080i/25 | 1920 | 1080 | 25 (50 Hz field rate) | 74.25 | 2640 | 1125 | 1.04 | SMPTE 274 System 6 |
| S3 | 1080p/25 | 1920 | 1080 | 25 | 74.25 | 2640 | 1125 | 1.04 | SMPTE 274 System 9 |
| S4 | 1080p/50 | 1920 | 1080 | 50 | 148.5 | 2640 | 1125 | 2.08 | SMPTE 274 System 3 |

With reference to sampling rates, for the luminance signal the frequency sampling frequency has the value of 74.25 MHz. The color is sampled in half sampling frequency, specifically at 0.5 * 74.25 MHz = 37.125 MHz. The ITU-R BT.709 standard gives a sampling rate of 72 MHz for the luminance and 36 MHz for the color signal. For the formation of the signal Y:Cb:Cr the used mechanism is still 4:2:2. In order to avoid the folding effect, the bandwidth of the luminance signal is limited to 30 MHz and that of the chrominance signals at 15 MHz through low-pass filters.

For the 1125/60 setup (refer to Figure 15 ) and when employing a 10-bit resolution, the aggregate transmission speed amounts to:

Y: 74,25 x 10 Mbit/s = 742.5 Mbit/s

Cb: 0.5 x 74,25 x 10 Mbit/s = 371.25 Mbit/s

Cr: 0.5 x 74,25 x 10 Mbit/s = 371.25 Mbit/s (18)

----------------

1.485 Gbit/s

Total data transfer speed (1125/60)

Due to the marginally reduced sampling rates in the 1250/50 system (as seen in Figure 15), the overall data transfer speed, when using a 10-bit resolution, becomes:

Y: 72 x 10 Mbit/s = 720 Mbit/s

Cb: 0.5 x 72 x 10 Mbit/s = 360 Mbit/s

Cr: 0.5 x 7,2 x 10 Mbit/s = 360 Mbit/ (19)

----------------

1.44 Gbit/s

Total data transfer speed (1250/50)

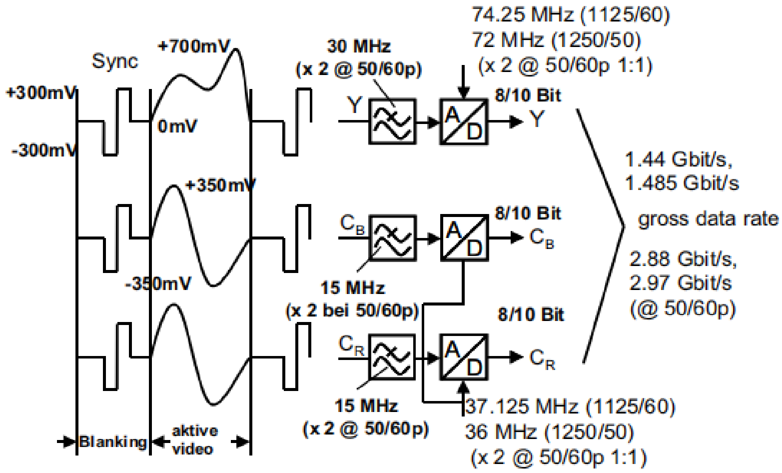



**Figure 15:** Sampling of an HDTV signal according to ITU-R.BT709.

Support is available for both interlaced and progressive scanning methods. Due to their inherent technology, plasma and LCD screens lean towards progressive scanning, as interlaced methods can result in undesirable visual distortions. For 50/60 progressively scanned frames, the sampling rates for the luminance signal rise to 148.5 and 144 MHz, respectively. Simultaneously, the chrominance signals' rates are set at 74.25 and 72 MHz, respectively. This causes the overall data speeds to reach 2.97 Gbit/s and 2.88 Gbit/s, respectively. The configuration of the raw digital HDTV data signal mirrors the ITU-R BT.601 standard. Both parallel and serial interfaces (specifically, HD-SDI) are determined. The HD signal uses the HD-SDI protocol, which stands for High-Definition Serial Digital Interface [22],[23]. DVI, or Digital Visual Interface, is familiar in the PC realm and is slated to succeed the conventional VGA interface. It supports data transfer speeds up to 1.65 Gbit/s. HDMI, representing High-Definition Multimedia Interface, can accommodate speeds as high as 5 Gbit/s, conveying both video and audio data. HDCP, or High Bandwidth Digital Content Protection, is implemented to shield HD digital content on DVI and HDMI interfaces from unauthorized duplication, a stipulation from the movie industry. Unlike the "HD Ready" label, "Full HD" offers the complete physical pixel resolution of 1920 x 1080 [24].

# CHAPTER 5:
# MPEG Video Compression

## *5.1 Compression Mechanisms*

As it mentioned before, the demands on the video-rated are too high to be supported by current transmission technologies. In the case of SD (Standard Definition, SD) the rate corresponds to 270 Mbps, and in the case of HD the rate corresponds to more than 1 Gbps. They must be reduced to values up to 2-7 Mbps considering current transmission technologies; thus, a compression ratio from 38:1 in the simplest case, up to 137:1 in the most complex is required. For the case of HD signal transmission, the compressed signal is about 15 to 20 Mbps. Data compression [29] can be achieved by eliminating extraneous or nonessential information from the data stream. "Redundant" refers to information that's superfluous, while "irrelevant" pertains to what's unnecessary. Data is considered superfluous if it repetitively appears in the stream, lacks informative value, or can be seamlessly and losslessly reconstructed using algorithms at the receiver's side. One method to reduce redundancy is through variable-length coding. For instance, rather than sending ten consecutive zeroes, a more concise special code indicating 'ten consecutive zeroes' can be transmitted.

Most of the digital video coding and compression standards rely on exploitation of various types of redundancy aiming to reduce it, thus also reducing the volume of data.

The types of redundancy are as follows:

- ✓ Spatial redundancy, which is due to the similarities between the elements that they build a frame.
- ✓ Temporal redundancy, which is due to similarities between successive images of a video.
- ✓ Statistical redundancy, which is due to the statistics of the appearance of various symbols to encode.
- ✓ Psycho optical redundancy, which is due to the mechanism of the human visual system and in the characteristics of the response of the vision system to the various stimuli.

In the case of digital television, the signal to be compressed is the one produced by the Camera man (camera). As it presented before, the components of the primary colors, which are red, green, and blue (R, G, B) are combined to give the Y and luminance signal of blue and red color differences (Cb and Cr). Color resolution is half that of luminance resolution, in the case of a 4:2:2 signal. In the case where we have a quantization of 10 bits, the rate can reach up to 270 Mbps according to the ITU-BT.R601 standard.

A method for video data reduction is MPEG (Motion Pictures Expert Group) [25],[26]. The MPEG-2 has been used and still is used in TV, however the MPEG-4 (AVC/H264) standard is also used, which ensures more efficient compression [27].

The MPEG video compression method consists of the following steps (Figure 16):

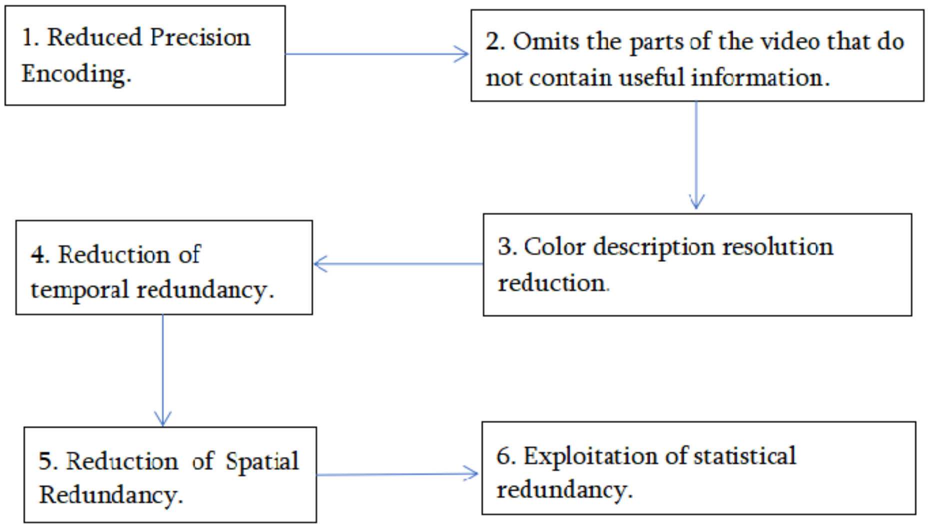



**Figure 16:** Basic steps in a typical video compression process.

1. Reducing the accuracy of the values of brightness and color differences, that is achieved by reducing the number of quantization levels and that, have as result, the reduction of bits used for the representation of the values, from 10 to 8.
2. Excludes sections of the video that lack pertinent details, specifically those segments which aren't representative of a visual cue. This involves disregarding the horizontal and vertical blanking periods, which, as per ITU BT.R601, don't hold any significant data, including teletext.
3. Further, color description resolution reduction beyond the standard 4:2:2 which is used when generating the signal. That it is possible, to use of different patterns (i.e 4:2:0) on the signal lines to exploit the relative limited sensitivity of the human eye to color information.
4. Taking advantage of the similarity that typically occurs between temporally successive video frames. The idea is to use the description of an existing frame to assist with the description of one of the following frames. They are used reference pictures and are used only the differences between consecutive images to be encoded. These differences have lower values in time periods when there is no high movement in the video.

5. Taking advantage of the similarity that may exist between the areas that make up one frame. This similarity is quite typical in many cases where we have almost uniform areas, such as the sky or other surfaces.
6. Taking advantage of the fact that some of the symbols appear more often than others. The content can have stable statistical properties with respect to the frequency of the symbol's appearance.

## *5.2 Video Signal Preparation*

### 5.2.1 Video Accuracy Downscaling (Decreasing the Quantization Depth from 10 Bits to 8 Bits)

As previously discussed in Chapter 3.2 on Quantization, when provided the right input, an 8-bit resolution A/D converter's quantization noise is already substantially below the threshold, making a 10-bit resolution in Y, CB, and CR superfluous outside of studio environments. Within studios, however, a 10-bit resolution is preferred as it facilitates post-processing and yields superior outcomes. When comparing to ITU BT.R601, dropping the data rate from 10 bits to 8 bits leads to a 20% reduction ((10-8)/10 = 20%). However, this constitutes a loss of data, and the original signal can't be fully restored during decoding. Based on rule (13), there's now a 12 dB increase in quantization noise. In terms of transmission speed, this signifies a 20% cutback, as, based on the given relationship, only 80% of the initial content is retained due to the shift from 10-bit to 8-bit quantization precision.

### 5.2.2 Excluding the Horizontal and Vertical Null Periods

In alignment with ITU BT.R601 (Fig. 17) the horizontal and vertical blanking periods of a digital video signal don't house essential details, including teletext. Though these sections can accommodate additional data like audio signals, such data must undergo separate MPEG encoding. As a result, MPEG entirely disregards these blanking periods. It's feasible to reconstruct the horizontal and vertical blanking intervals and any associated signals at the receiver side without any issues. In a PAL signal, out of 625 lines, only 575 are visible. The 50-line disparity, when evaluated against the total 625 lines, translates to an 8% data rate conservation once the vertical blanking is excluded. A single line spans 64 μs, yet the actual video content occupies just 52 μs. Compared to the 64 μs, this results in an additional 19% data rate savings. Accounting for the overlap between these two reductions, the overall data rate saving, achieved by eliminating this redundancy, rounds up to approximately 25%.

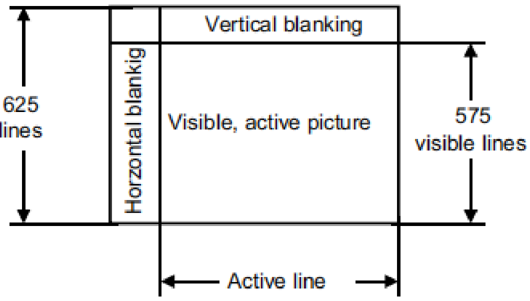



**Figure 17:** Horizontal and Vertical Blanking.

### 5.2.3 Decreasing the Vertical Chromatic Resolution (4:2:0)

As discussed in the chapter addressing video signal generation, right from the signal's inception, the components related to colors, particularly the color differentials Cb and Cr, are derived at a reduced sampling rate compared to luminance (Y). To be specific, this sampling rate is halved horizontally, with every 4 luminance samples corresponding to 2 color difference samples. Furthermore, the bandwidth for Cb and Cr is limited to 2.75 MHz, contrasting with the 5.75 MHz bandwidth allocated for luminance, denoting a 4:2:2 signal (Figure 18). Yet, for this 4:2:2 signal, the color resolution diminishes solely along the horizontal axis. In contrast, the vertical color resolution maintains its integrity, matching the complete resolution derived from the total line count in a TV frame.

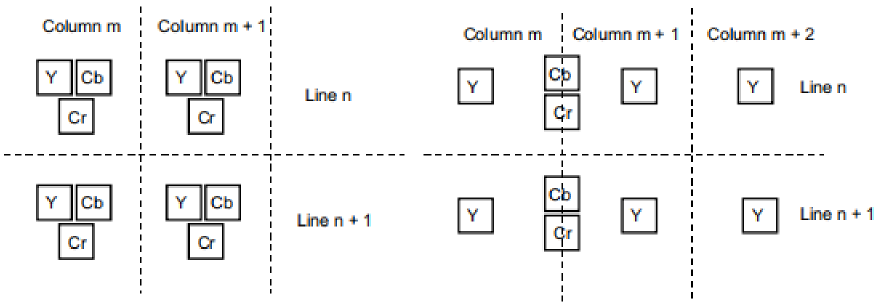



**Figure 18:** 4:4:4 resolution and 4:2:2 resolution.

Interestingly, the human eye doesn't differentiate between horizontal and vertical orientations when it comes to color resolution. Consequently, it's feasible to halve the color resolution vertically without any noticeable impact. Typically, MPEG-2 implements this reduction in its initial stages, converting the signal to a 4:2:0 format (Figure 19).

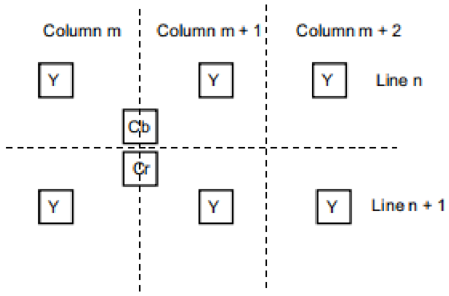



**Figure 19:** 4:2:0 resolution.

Now, every four Y pixels correspond to just one CB and one CR value respectively. This form of data omission leads to a precise data rate reduction of 25%.

The data minimization implemented thus far has achieved the subsequent outcome: From an initial data speed of 270 Mbit/s, by leveraging the steps outlined (as depicted in Figure 20), the ITU BT.R601 signal's rate is now condensed to 124.5 Mbit/s. This is less than half its original rate. Here's a breakdown (Figure 20):

- Starting at ITU BT.R601 = 270 Mbit/s.
- Transitioning to 8 bits from 10 results in a 20% decrease, bringing it to 216 Mbit/s.
- Omission of both horizontal and vertical blanking approximately leads to a 25% reduction, settling at 166 Mbit/s.
- The 4:2:0 step accounts for a further 25% reduction, culminating in 124.5 Mbit/s.

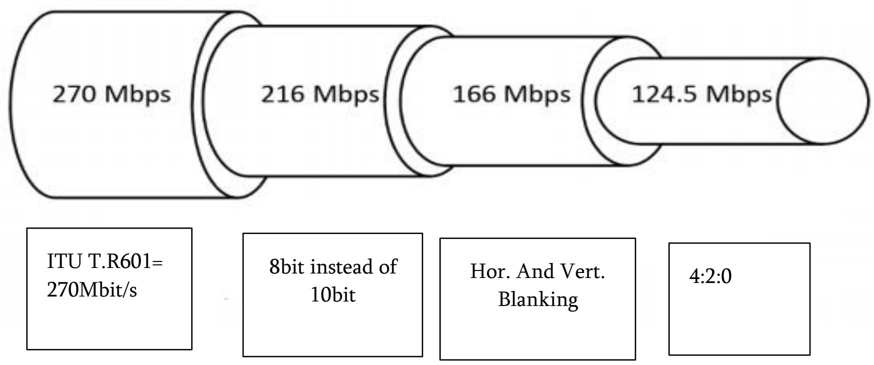



**Figure 20:** Data Reduction Steps.

The next steps are to further limit the rate to 2-4 Mbps. The achieving this goal is done by more complex mechanisms.

### 5.2.4 Temporal redundancy exploitation

The video consists of consecutive images the which are taken several times in one second. These pictures, in most cases, visually appear like each other. So, the differences between them, exist at the dynamic parts and are presented when movement takes place. In these cases, the static parts remain consistent from picture to picture. But even in the dynamic section, the changes may occur from moving objects between frames. Sending the entire description of a picture (a frame) leads to the repetition of the same information that could be recreated at the receiver. The idea behind limiting the time redundancy is to send only the necessary information, i.e. the information that could not be recreated at the receiver. This information relates to the areas where there are differences compared to the previous frames, so it is it is necessary to send this difference. The mechanism involves splitting the image into sub regions and the coding of images by I, P and B. When segmenting the pictures into regions, the static and dynamic regions are searched. To do the search for these areas manageable the picture is divided into equal parts of sub regions. Specifically, in blocks of 16x16 pixels. In these sub regions there are 16x16 brightness values and according to the 4:2:0 standard, the values corresponding to colors are in number 2/8=1/4 of their luminance. They correspond, that is, to an 8x8 area for each of the color differences Cb and Cr while the set of these values corresponds to a macro-block (Figure 21).

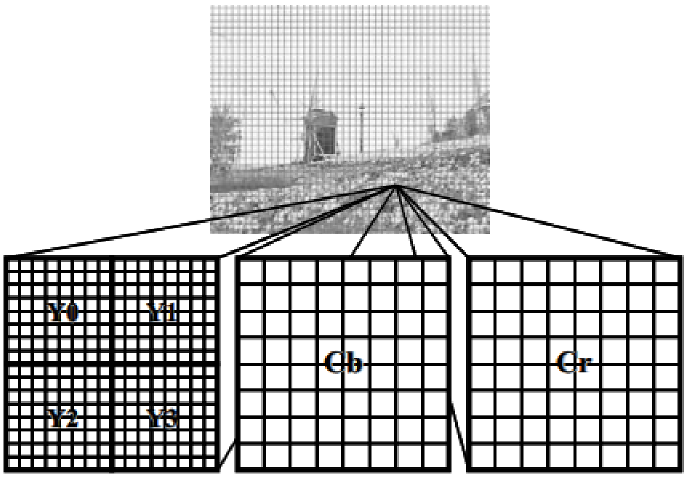



**Figure 21:** Macro block Structure with 4:2:0.

The mechanism is simple and works as following:

- Some of the frames are encoded independently without regard information from other contexts. These frames are called (in coding terms) type I (I-frames, *intracoded frames*). Such frames are coded and sent periodically.
- Some frames are encoded considering previous frames. The coding it is done by exploiting the similarities that may exist between the current context and of some (specific) precedent. Similarities are searched and found in macro-block level, i.e. in sections of size 16x16. These kinds of frameworks they are called P-frames (*predicted*

*frames*). The detection of any similarity in a frame of type P, is done using the previous type I or P frame (whichever is closer in time). The encoding of P-frames requires less data than the coding of I frames.

- There are frames which are encoded using similarity detection with previous and next frames. They depend, that is, on a previous (in time) frame of type I or P and a next (in time) frame similarly of type I or P. The frames of this type are called type B (B-frames, bidirectional frames). The encoding of B frames requires less data than a coding of type P frames (and of course type I frames).

The order of the frames and the way they alternate is predefined and invariant in time. Specifically, the group of pictures, Group of Pictures (GoP), is defined, which consists of a sequence of different types of frames, I, P and B.

The GoP (Figure 22) have some characteristics:

✓ A GoP is opened with an I frame, since its encoding must be self-contained and

not to have dependencies on previous and generally neighboring GoPs.

✓ A GoP can be closed with a frame of type P or I. It cannot be closed with a frame of type B since this will imply a dependency on a subsequent context.

✓ If a GoP is closed by an I frame, then we say that the GoP is *closed*.

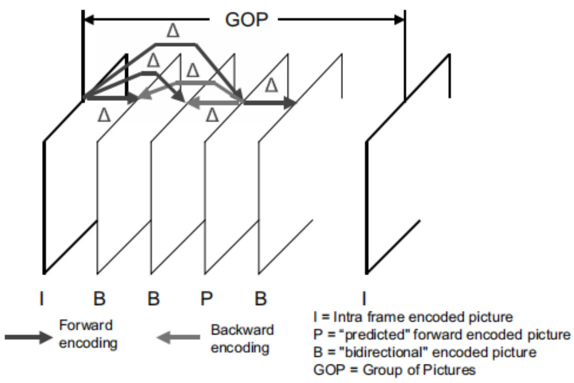



**Figure 22:** Closed GoP Structure.

The method for acquiring motion vectors through motion estimation is detailed as such: Beginning with a designated delta frame set for encoding, the system examines the prior frame (labeled as forward prediction P) and potentially the succeeding frame (dubbed bidirectional prediction B) for relevant macro block data surrounding the macro block set for encoding. This assessment leverages the block matching principle within a specific search vicinity around the macro block (Figure 23).

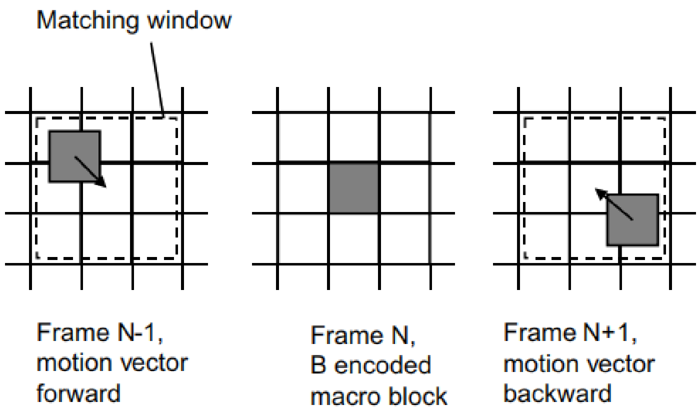



**Figure 23:** Motion vectors.

The process identifies a congruent block both ahead (and potentially behind, in the context of bidirectional coding). Once identified, both forward and backward motion vectors are derived and transmitted. If necessary, any further block differences for both forward and backward motion can also be conveyed. Yet, this block difference undergoes separate DCT coding coupled with quantization, which will be elaborated in the succeeding chapter. This method notably conserves storage space. A Group of Pictures (GOP) is typically composed of a specific count and structure of B and P pictures placed between two I pictures. Usually spanning around 12 frames, a GOP's sequence tends to follow I, B, B, P, B, B, P, ... with B pictures nestled between I and P ones. To decode a B picture at the receiver, it's imperative to first have the prior I and P pictures and the succeeding I or P picture's information. Nonetheless, as per MPEG [28], GOP structures can be flexible. To ensure the receiver doesn't demand excessive storage, GOP structures might need adjustments during transmission. This ensures that backward prediction data precedes the actual B pictures. Therefore, transmitted frames might not align with their original sequence. Rather than following the sequence I0, B1, B2, P3, B4, B5, P6, B7, B8, P9, the transmission order shifts to I0, B-2, B-1, P3, B1, B2, P6, B4, B5, P9, etc. (as illustrated in Figure 24). Essentially, the succeeding P or I picture are received and decoded at the destination before the corresponding B pictures arrive. This method ensures that the storage allocation at the receiving end remains predictable and restricted. To regain the authentic sequence, the frame identifiers need to be encoded and transmitted. For this task, among other tools, the DTS (decoding time stamp) values located in the PES header are utilized. Figure 25 provides an understanding of the advantages yielded by capitalizing on temporal redundancy. An I-coded frame typically occupies more space than P and B frames. The size of a P-type frame can be as much as half of an I frame, while B frames can be reduced to just 25%. Nevertheless, these figures are approximations, as the exact proportions can usually be adjusted.

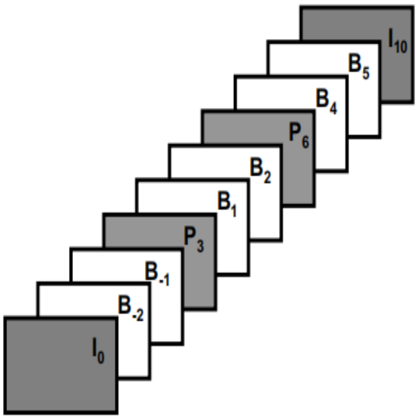



**Figure 24:** Order of picture transmission.

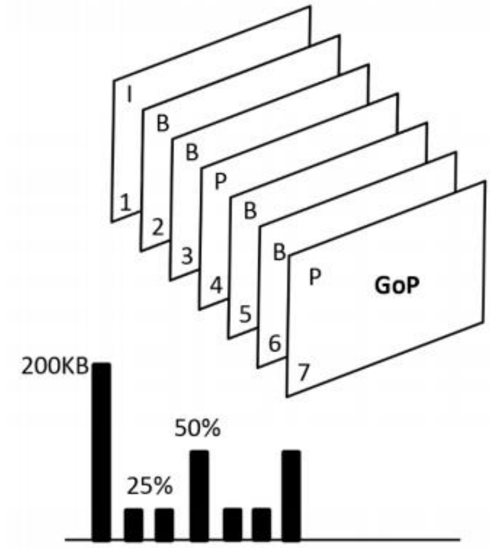



**Figure 25:** Indicative frame sizes depending on the type of encoding

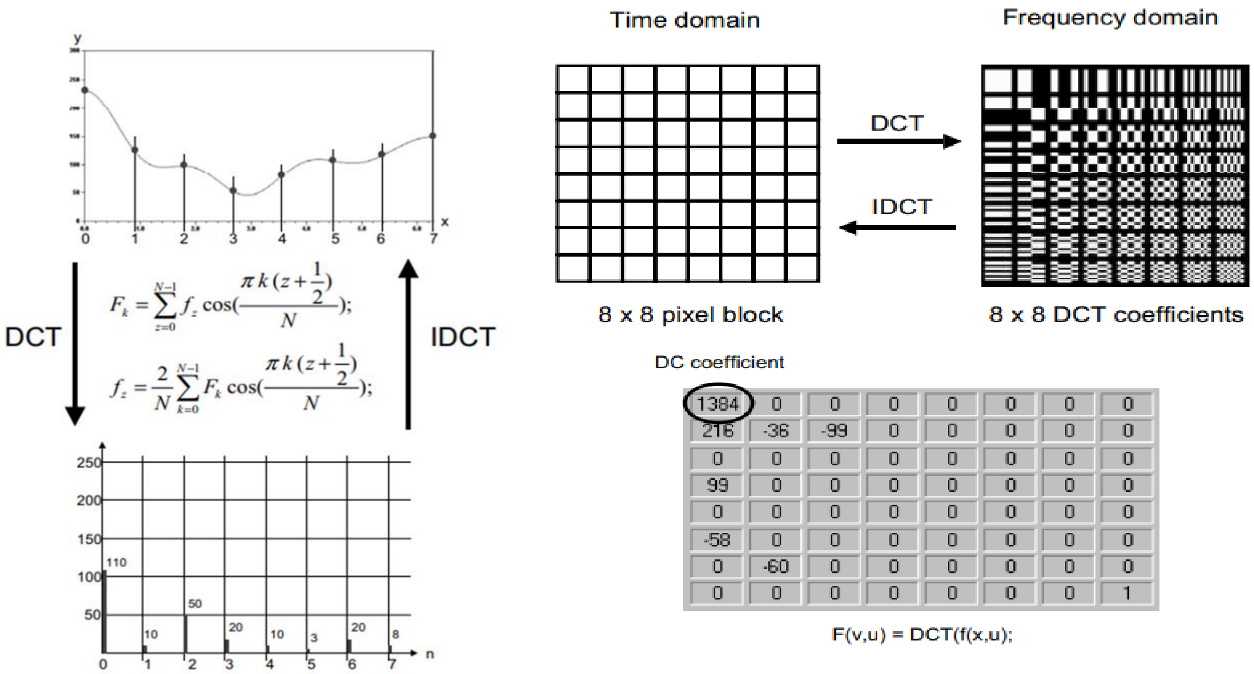
### 5.2.5 Spatial Redundancy Exploitation

Spatial redundancy comes from the similarities in different regions within a picture. This phenomenon is quite frequent and exploits the spatial redundancy with the standards still-frame compression. The most common method is JPEG (Joint Pictures Expert Group). The foundational mechanism underlying JPEG is the Discrete Cosine Transform, often abbreviated as DCT, as visualized in (Figure 26). This very DCT serves as the cornerstone for the MPEG video encoding technique. The mathematical expression representing the two-dimensional DCT is as follows:

$$F(u,v) = \frac{2}{N} C(u) C(v) \sum_{x=0}^{N-1} \sum_{y=0}^{N-1} f(x,y) \cos\frac{(2x+1)u\pi}{2N} \cos\frac{(2y+1)v\pi}{2N}$$

$$C(u), C(v) = \begin{cases} \frac{1}{\sqrt{2}} & \text{for } u,v=0 \\ 1 & \text{otherwise} \end{cases} \qquad (20)$$

This compression technique leverages the understanding that human vision isn't particularly adept at detecting distinct variations (especially those occurring over a limited pixel range, equating to elevated spatial frequencies) within an image. Essentially, our eyes aren't highly attuned to intense spatial frequencies. This observation also applies to analog video where more noise can be allowed at high frequencies compared to low ones. Based on the above, the lowest (spatial) frequencies need to be coded with greater detail (In terms of quantization), while the higher frequencies are not required to be coded in detail. This reduction in detail can cause significant savings in required transmission rate. The DCT is the algorithm that distinguishes the information concerning high (spatial) frequencies from them with low frequency. Initially, there's a shift from the video signal's time domain to its frequency domain. The Discrete Cosine Transform acts as a specific instance of either the discrete Fourier Transform or, equivalently, the Fast Fourier Transform.




a. b.

**Figure 26: a.**One-dimensional Discrete Cosine Transform **b.** Two-dimensional DCT

### 5.2.6 Discrete Cosine Transformation Method

The steps followed in the case of JPEG and MPEG include slicing of the picture in blocks, usually 8x8 pixels in size, subtracting the mean value (of 128 in the case that the quantization will be done with the help of 8 bits) and then the transformation by Discrete Cosine Transform. Then follows the adjustment of the coefficients based on human sensitivity eye, but also of the desired level of quality. Then the coefficients are scanned with in such a way that the largest and smallest values are efficiently grouped together. At the end of the process, quantization is performed (Figure 27).

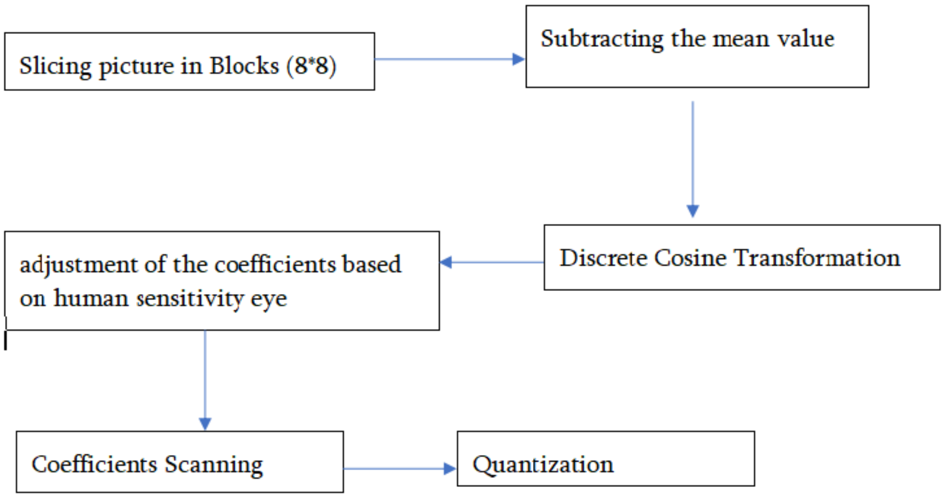



**Figure 27:** DCT Steps transform in JPEG and MPEG.

### 5.2.7 Statistical Redundancy Exploitation

Exploiting statistical redundancy is a lossless data compression method. It was developed in 1952 by David Huffman. The basis of this method is exploiting the fact that in any alphabet some of the symbols occur more frequently than others. This means that if the most frequently appearing symbols are coded more efficiently with shorter in length code words, (where the length translates to the number of bits used to encode the word), we expect to have a benefit compared to uniform encoding, which encodes all symbols uniformly (with words of the same length) without considering the statistical properties of the symbols' appearance. In Huffman coding, each symbol corresponds to a unique encoding and any symbol's encoding is not prefixed to another's encoding symbol. Huffman coding is the last stage of data encoding / compression (Figure 28).

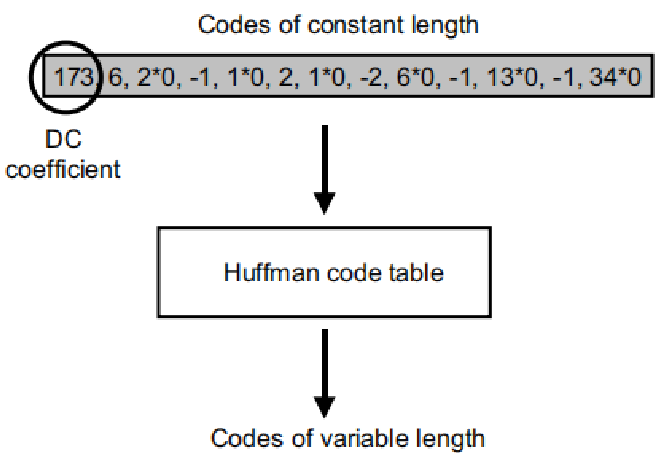



**Figure 28:** Huffman coding (variable length coding - VLC).

## 5.3 Profiles and Levels

By using the methods we mentioned to reduce redundancy, as well as some information, the omission of which is not so important in terms of fidelity, the video bitrate is possible to drop from 270 Mbps (with 4:2:2 color sub sampling) to 2 Mbps up to 6 Mbps. The most important step in the compression process is to limit the time redundancy combined with spatial redundancy and the implementation of the DCT, as well as the further color sub sampling (using the 4:2:0 mechanism). So, the compression level is not constant, but it depends on parameters that one can choose and change. For this purpose, the MPEG has created a series of levels and profiles by which it groups them these options (Figure 29). Here are some of the most common [29]:

- For standard definition utilizing the (4:2:0) format, it's termed as Main Profile at the Main Level.
- Standard definition leveraging (4:2:2) is referred to as High Profile at the Main Level.
- High definition with the 4:2:0 format aligns with the Main Profile at the High Level.
- High definition that uses the 4:2:2 signal falls under the High Profile at the High Level.

The rate in the video stream can be dynamically varied over time or remain constant. Changes in rate depend on parameter selections quantization. Typical options that define profiles and levels include image encoding types, the level of color sub sampling, the ratio of the picture, scaling and accuracy with respect to DCT.

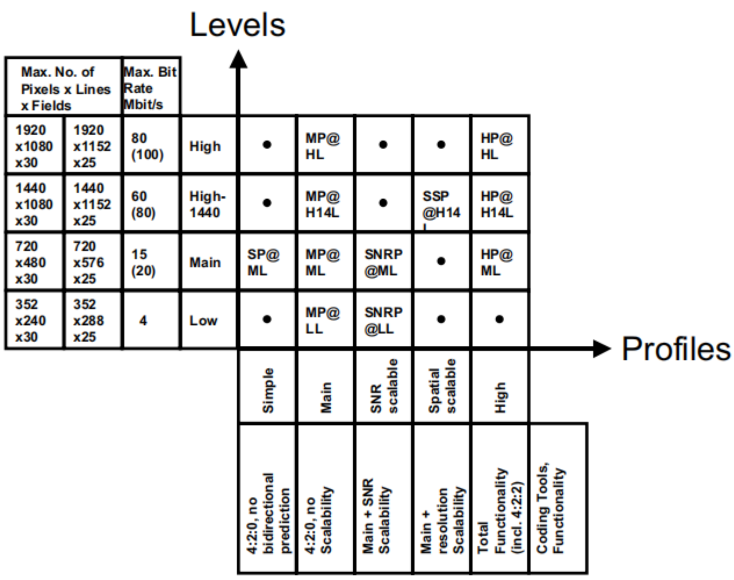



**Figure 29:** Some of MPEG-2 profiles and levels.

The most common profile and level combinations include the following: MP@ML, 720x480, 30, 4:2:0, 4 Mbps standard definition digital TV, MP@ML, 720x576, 25, 4:2:0, 4 Mbps standard definition digital TV, MP@HL, 1920x1080, 30, 4:2:0, 80 Mbps, digital TV high definition and MP@HL, 1920x720, 60, 4:2:0, 80 Mbps, digital high definition TV.

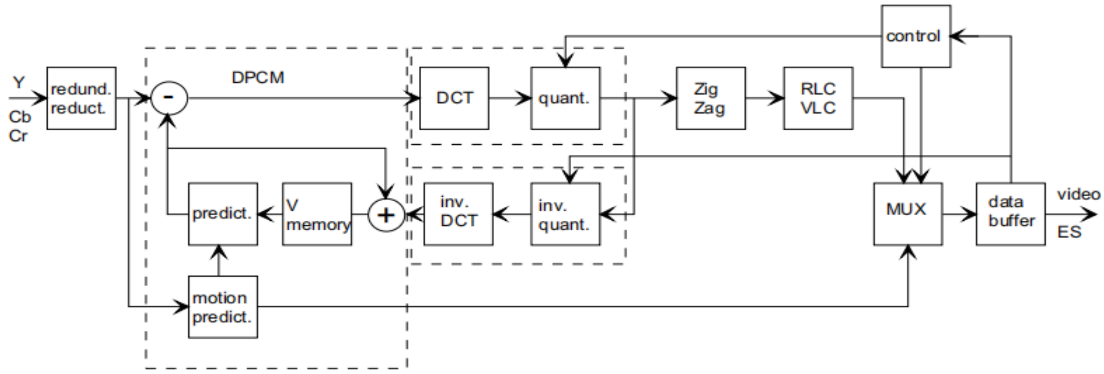



**Figure 30:** MPEG-2 encoder.

A 6 Mbit/s SDTV signal in the 4:2:0 format typically offers a quality comparable to traditional analog TV signals. However, in real-world scenarios, data rates can oscillate between 2 to 7 Mbit/s, which directly influences the image quality. Notably, high-paced content such as sports broadcasts may necessitate higher data rates. The data rate for the foundational video stream can either remain consistent or fluctuate based on the prevailing image content. This rate is regulated by adjusting the quantization factors, contingent upon the output buffer level of the MPEG encoder (Figure 30).

On the previous chapters, it presented and explained the basic flow and the idea behind the methods for digitization and compression of the TV signal, these procedures are the main parts of video and audio encoding. Next, a brief explanation of the DVB-MPEG Standard mechanism for creating packets that forms the Transport Streams take place. Through the Transport streams the video and audio data are ready to function as input on broadcasting channels, where channel coding take place, for the transmission to the receiver. This procedure, after the Transport stream building, up to transmission and reception at terrestrial TV receivers is standardized from DVB-T Protocol.

# CHAPTER 6:
# MPEG Data Stream

## *6.1 Elementary Streams and Multiplexing*

### 6.1.1 Elementary Streams and Packets Fragmentation

After video and audio have been encoded, the resulting data streams are called Elementary Streams (ES) [30]. These streams are the output of the encoders and are a continuous stream of bits (bitstream). The rate of ES may be constant or variable. This depends on the properties and operation of the encoder. The ES (Figure 31) is not suitable for transmission since it is not divided into packets, it does not contain its information program and cannot be multiplexed with other streams (in its current form). The DVB standard considers that a program consists of ES with a common timing base. For Transport Streams, where multiple streams are multiplexed to produce varied programs, each program possesses a distinct timing basis that is independent the another's. This timing is set by specific timing fields, known as the Program Clock Reference (PCR). The uninterrupted data from the elementary streams gets divided into packets, leading to the creation of Packetized Elementary Streams (PES). Typically, the division starts at frame edges, with each frame equating to a single PES packet. This suggests that PES packets don't maintain consistent lengths. These packets can encapsulate multiple compressed frames for video signals or several compressed segments for audio signals. Each Packetized Elementary Stream (PES) is derived from a singular Elementary Stream.

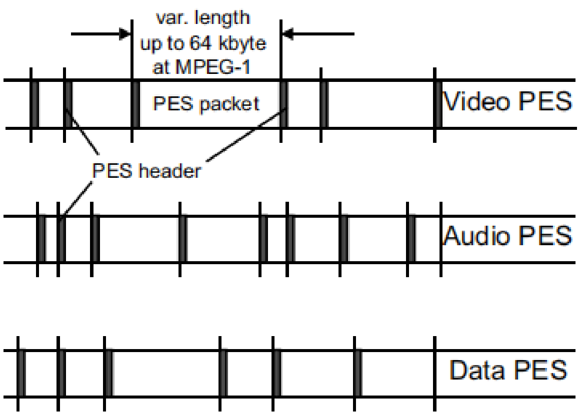



**Figure 31:** MPEG Elementary Streams.

## *6.2 The Packetized Elementary Stream (PES)*

A PES is divided into a header and its associated data. These PES packets can vary in size, typically extending up to 64 Kbytes. The header of a PES is initiated with a 24-bit start code, carrying the value "0x00 00 01," which signals the commencement of the PES packet. The byte succeeding this start code represents the stream's code (stream ID). This code denotes the original stream or its category (be it video, audio, or data) from which the PES packet stems. A 2-Byte length field succeeds this, designating the length of the packet information segment. If this field reads zero (both bytes indicating zero), the payload's length might surpass 64 Kbytes, necessitating the decoder to employ alternative means, like the start code, for packet segregation. Following the PES packet header is an optional header encompassing fields pertinent to the data segment. This potentially included PES packet header might encapsulate decoding and presentation timing data (like Decoding Time Stamps, DTS, and Presentation Time Stamps, PTS) along with other optional fields. (Figure 32).

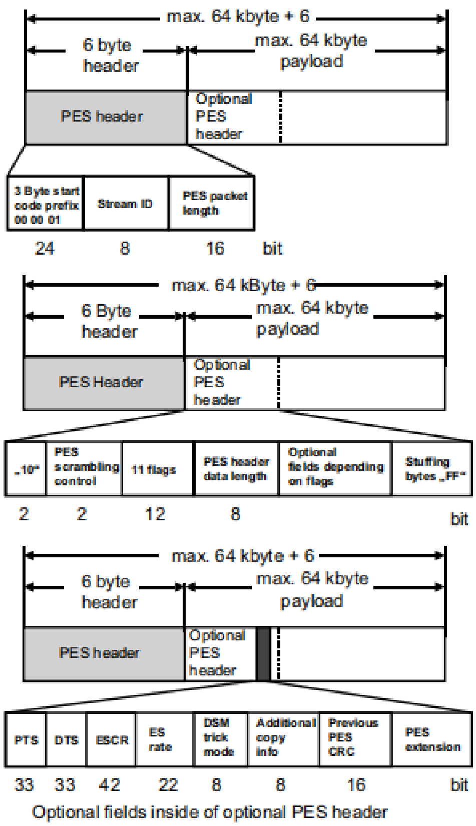



**Figure 32:** The PES packet.

The "Packetized Elementary Stream" (PES), with its extended packet configurations, isn't apt for transmission, especially when broadcasting multiple programs within a single multiplexed data signal. In MPEG-2, the goal is to combine anywhere from 6 to 20 distinct TV or radio broadcasts into a unified multiplexed MPEG-2 data signal. To achieve this, these packets are fragmented into smaller packets of a fixed length. PES packets are integrated into Transport Stream (TS) packets. From the PES, segments of 184 bytes are extracted, and a 4-byte header is appended (as illustrated in Figure 33), resulting in 188-byte "transport stream packets" that are then multiplexed.

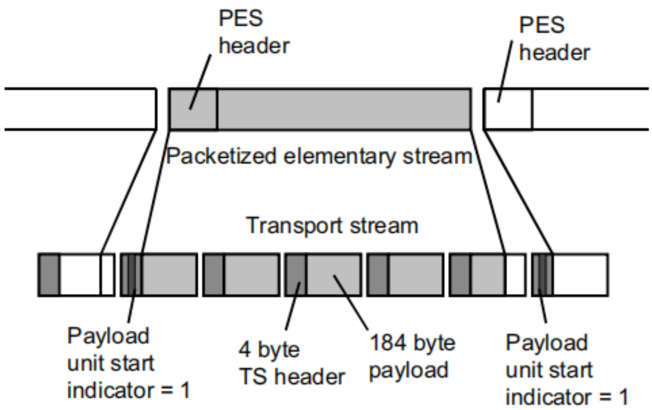



**Figure 33:** Creating MPEG-2 Transport Stream Packets.

Packetized Elementary Streams (PES) are multiplexed with each other as well as with system related information to provide a Transport Stream (TS).

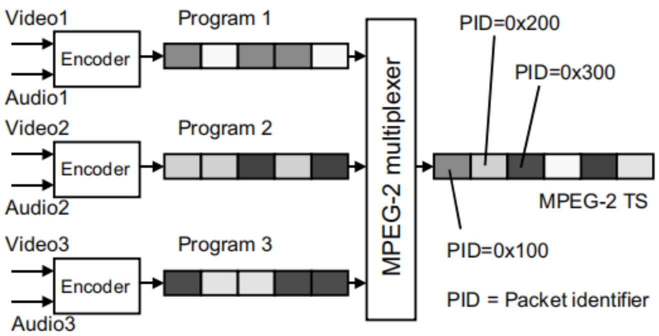



**Figure 34:** Multiplexed MPEG-2 transport stream packets.

Each program employs a distinct MPEG encoder. This encoder processes all elementary streams, establishes a PES framework, and subsequently arranges these PES packets into TS packets. While individual programs typically have a data rate ranging between approximately 2 to 7 Mbit/s, the combined data rate, encompassing video, audio, and data, can either be stable or fluctuate based on the content being showcased, a scenario termed "statistical multiplex". The transport streams from all programs converge into a unified multiplexed MPEG-2 data stream, forming an overarching transport stream (as visualized in Figure 34. This consolidated stream can reach data speeds close to 40 Mbit/s. It's common to find anywhere from 6 to 20 programs within a single transport stream. Though individual data rates might oscillate, the cumulative rate must remain unchanged. A program might offer video and audio, solely audio (as in radio broadcasts), or just data. This structure is adaptable and might undergo changes during broadcasting. To ascertain the structure's composition during decoding, the transport stream also embeds descriptive lists or "tables".

## *6.3 The MPEG-2 Transport Stream Packet*

An MPEG-2 transport stream encompasses transport stream packets from all programs, each being 188 bytes in length, carrying video, audio, and data signals. Depending on the respective data rates, packets from different elementary streams might appear with varying frequencies within the MPEG-2 transport stream. Each packet maintains a consistent length of 188 bytes, split between a 4-byte header and a 184-byte payload. This payload is tasked with carrying video, audio, or other forms of data. The header is packed with various critical transmission elements. Foremost among these is the "sync byte" found at the beginning of the header. This byte consistently registers a value of 47hex (or 0x47 in C/C++ notation) and is consistently spaced at 188-byte intervals in the transport stream. It's worth noting that the occurrence of a byte valued at 0x47 elsewhere in the packet, though possible, isn't deemed irregular.

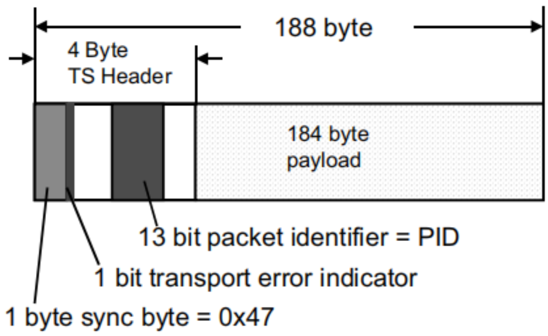



**Figure 35:** MPEG-2 transport stream packet.

The sync byte's primary function is to synchronize the packet with the transport stream. This synchronization is achieved through the consistent value of the sync byte coupled with its fixed spacing. As per MPEG standards, decoder synchronization takes place after the receipt of five transport stream packets. Another crucial feature of the transport stream is the 13-bit "packet identifier", commonly referred to as PID. The PID provides insights into the content housed within the packet's payload. Using a hexadecimal 13-bit number and associated tables present within the transport stream, one can identify the specific elementary stream or content in question. Directly succeeding the sync bit is another essential component: the "transport error indicator" bit. (Figure 35). This bit identifies transport stream packets that encountered errors during transmission. It's activated by demodulators at the transmission link's conclusion when excessive errors occur, especially if they can't be rectified by existing error-correction methods. In Digital Video Broadcasting (DVB), for instance, the primary protective measure against errors is the Reed Solomon error correction code, which operates as an initial phase within the DVB-T modulator. In the following Table 7 the fields that make up the header of a Transport packet Stream, as well as the number of bits of which they consist of, are described [31].

**Table 7:** TS packet header fields.

| | Name | Number of bits | Description |
|---|---|---|---|
| 1 | Sync byte | 8 | Synchronization field with fixed value '0100 0111' (0x47):It is 188 bytes away (at the beginning of each packet) of the Transport Stream. The constant value and distance of the field synchronization help in synchronization, which receives country every 5 TS packets. Note, that there may be one or more Bytes within the packet, which have the same value. For the confirmation that this is the sync Byte, the receiver checks for 5 packets the Bytes which are separated by n*188 (where n from 1 to 5). If these also have the value 0x47 then it considers it as a synchronization byte. We consider that we have loss of synchronization after loss 3 packages. |
| 2 | Transport_error_ indicator | 1 | One-bit error flag: If present the value 1, indicates that there is at least one incorrect bit which cannot be fixed in the TS packet. The price of the field may be changed by entities outside of it transport level. The reset from 1 to 0 is done only if the error has been corrected. If the error did not fix the decoder must ignore the whole package. |
| 3 | Payload_unit_start_ indicator | 1 | The meaning of a specific field varies based on the type of data being transported by the TS packet, particularly if it's of the PES or PSI variety. When dealing with a PES type, a value of 1 indicates the commencement of a PES packet. Similarly, in the context of the PSI type, a value of 1 in this field suggests that the TS packet is initiating the data of the PSI. |
| 4 | Transport_priority | 1 | Field which concerns the priority of the packet to relative to other packets that have the same PID, in case the specific field has the value 1. The value of this field can be changed by encoder or decoder (that is, at the ends of the chain and not in between). |

| | Name | Number of bits | Description |
|---|---|---|---|
| 5 | PID | 13 | The PID field shows the type of data contained within the package. There are reserved values of the PID but also available to are assigned to the elementary streams (ES) |
| 6 | Transport_ scrambling_control | 2 | Field which shows how much the content is scrambled. In case of value 00, the content is not encrypted. |
| 7 | Adaptation_field_ control | 2 | Field indicating whether the packet header of TS followed by adaptation field along with the data section. The adjustment field is header field extension and included only if required. |
| 8 | Continuity_counter | 4 | Counter whose value increases for each packet by same PID. The counter returns to 0 after the maximum value. The counter remains the same just in case identical packages. |

Notice that among other fields the header includes the Packet ID (PID), which consists of 13 bits. The PID field specifies, with the help of a table that provides information specifically for the respective program (Program Specific Information, PSI), the type of content of the TS packet. Program information is used to operate the multiplexing mechanism (at the transmitter) and demultiplexing (at the receiver). In addition to audio and video streams, there are also streams that contain metadata, that is, information about the remaining flows. There are two basic types of metadata, which compose relates tables compose related tables. Specifically:

- Program Specific Information (PSI) tables which are defined from MPEG.
- Program and System protocol information tables (Program and System Information Protocol, PSIP) which concern the DVB standard.

The two fundamental PSI tables are the Program Association Table (PAT) and the Program Map Table (PMT).

The primary role of PAT is to convey details about the programs present within the transport stream and to supply the PID for tables that further elucidate those programs.

The PMT tables provide descriptions of the programs. When decoding, the system searches for packets whose PID matches the value specified in the PAT. PMT tables outline the components of a program, including their associated PID. Modifications to the programs are depicted through revisions in the PMT tables. Additionally, there are private sections and tables employed by MPEG to address requirements that aren't catered to by the standard. Some tables, for instance, aren't standardized by MPEG but by DVB, and include the following:

- NIT: Network Information Table
- SDT: Service Descriptor Table • BAT Bouquet Association Table
- EIT: Event Information Table
- RST: Running Status Table
- TDT: Time & Date Table

Details about the MPEG protocol can be found on international bibliography as[32],[33]. Mpeg-2 is the signal feed for Digital Video Terrestrial Broadcasting.

# CHAPTER 7:
# DVB—The Family of International Standards for Digital Video Broadcasting

## *7.1 Fields of Application of DVB Technologies*

DVB, short for digital video broadcasting, initially focused on establishing technical specifications for conventional broadcasting of audio and video over various platforms like satellites, cable networks, and terrestrial transmitters. As it evolved, DVB expanded its scope beyond the realm of traditional broadcasting. Within DVB's terminology, broadcasting generically denotes the distribution of media content from a single source to multiple recipients, regardless of the transport medium. After the DVB Technical Module drafts a specification, and upon its endorsement by the Steering Board, it's publicly released as a "DVB Blue Book." For these specifications to achieve the status of international standards, they're handed to the Joint Technical Committee (JTC) Broadcast. This committee, a collaboration between the European Broadcasting Union, the European Telecommunications Standards Institute (ETSI), and the Comité Européen de Normalization Électrotechnique (CENELEC), then transforms these specifications into recognized standards. DVB often releases accompanying implementation guidelines alongside these specifications. In 2000, DVB re-evaluated its objectives, envisioning a future that harmoniously integrates the stability and interconnectivity of broadcast media with the dynamism, inventiveness, and service diversity of the internet world.

**Table 8:** DVB Deliverables by Field of Application.

| Field of application | Number of standards | Number of guidelines |
|---|---|---|
| Source coding | 0 | 3 |
| Service information | 1 | 2 |
| TV-Anytime, Teletext, subtitling, VBI information, software download | 5 | 0 |
| Conditional access | 1 | 2 |
| Data broadcasting | 1 | 1 |
| Transmission to the home | 12 | 2 |
| Interaction channels | 8 | 5 |
| Multimedia Home Platform (MHP), Globally Executable MHP (GEM) | 2 | 0 |
| In-home networks, professional network interfaces, home terminal interfaces | 10 | 3 |
| DVB content over broadband IP | 1 | 0 |
| Digital satellite news gathering | 2 | 1 |
| Measurement guidelines | 0 | 2 |
| Miscellaneous | 0 | 1 |
| Total | 43 | 22 |

From that period onwards, several innovative technical solutions emerged based on the demands of the member organizations. As a result, the roster of DVB outputs somewhat mirrors the prevailing spirit of the times. Table 8 categorizes these outputs based on their application area [35]. An analysis of Table 8 shows that there are 12 specifications dedicated to detailing technologies that facilitate content delivery to households. These specifications encompass a comprehensive range of transmission mediums.

## *7.2 DVB goals*

The initial goal set by the DVB team for the TV was to standardize a system that to ensure:

- Transmission of high-definition television (HDTV).
- Transmission of standard definition television (SDTV) on narrowband channels zone.
- Reception TV from portable, low-cost receivers.
- Receiving a television program from receivers installed in vehicles, even if these move at high speed.
- Stable quality of service even on channels with strong interruptions and under presence of interference.
- Stable quality of service in a well-defined coverage
- Capability of distribution of the content from existing telecommunications networks.

Along the way, DVB has evolved into a comprehensive standard for broadcasting digital content, the new goals that set DVB was:

- Multiplication of television programs that will be able to be transmitted, in the same frequency range available for transmitting one analog TV program.
- The radio program broadcast support.
- The support for the ability to transfer data (for information, entertainment etc.)
- The possibility of variable sound and image quality.
- Support for subscription services.
- Support through interactive services (requires the existence of a channel turn).
- Internet access through the television set.

Now days, all the goals are succeeded, and the digital television has all these capabilities.

## *7.3 Digital TV Systems*

Broadcasting of digital audio and video can be done using different standards depending of course on the medium used for transmission and the receiver that receives the transmitted data stream. The standards which have been established for these cases are the following:

- **DVB-S** (Digital Video Broadcasting - Satellite): This is the initial generation of the digital satellite system used for the transmission and reception of TV signals via satellites, predominantly using MPEG-2. The necessary components comprise an LNB - Low Noise Block Downconverter, a satellite dish, and a digital satellite receiver.
- **DVB-S2** (Digital Video Broadcasting - Satellite 2): This is the upgraded second version of the DVB satellite system.
- **DVB-C** (Digital Video Broadcasting - Cable): This pertains to the digital transmission of TV shows via terrestrial cables. Although the frequency bandwidth is capped at 8MHz, the data transfer speed sustains at 38Mbit/s.
- **DVB-T** (Digital Video Broadcasting - Terrestrial): Dedicated to the terrestrial broadcast of digital signals, it was established in 1997 and operates on both VHF and UHF bands. Each channel operates at an 8MHz frequency, maintaining a data bandwidth of 38Mbit/s.
- **DVB-T2** (Digital Video Broadcasting - Terrestrial 2): A more advanced version tailored for terrestrial digital broadcasting. Introduced in 2008, it's the successor to the DVB-T system.
- **DVB-H** (Digital Video Broadcasting – Handheld): This standard offers a robust and adaptable approach for digital broadcasting, specifically designed for handheld devices. The system is designed for compatibility with mobile devices. It builds upon the DVB-T terrestrial framework, leveraging the infrastructure of terrestrial digital broadcasting networks.

# CHAPTER 8:
# The DVB-T standard

## *8.1 DVB-T system - outline*

The DVB-T framework is tailored for the terrestrial transmission of TV signals encoded using MPEG-2. This demands a specialized adaptation of the digitally encoded transport stream to align with terrestrial channel properties. The solution emerges in the form of a versatile transmission system that employs a multi-carrier modulation, known as the Orthogonal Frequency Division Multiplex (OFDM) method, coupled with a robust concatenated error correction coding, termed as Coded Orthogonal Frequency Division Multiplex (COFDM)[34]. To optimize spectrum utilization in the UHF bands, OFDM provides options for varying numbers of carriers, modulation schemes, and guard intervals, facilitating both small and expansive Single Frequency Networks (SFN). In terms of bandwidth, an 8 MHz channel spacing is ideal, but adaptability to 7 MHz or 6 MHz is possible through proportionate adjustments to all system elements [35].

The concatenated error correction consists of two segments: the external coding and interleaving (shared with Satellite and Cable baseline specifications) and the internal coding (common to the Satellite baseline). The deployment of internal interleaving is exclusive to the DVB-T framework. To cater to diverse transmission speeds, besides the five code rates, users can choose from three non-differential modulation schemes: QPSK, 16-QAM, and 64-QAM. The latter two also support either uniform or varied mapping rules, enabling input data streams to be split into high and low priority streams, each receiving varied error protection for hierarchical broadcasting purposes. This capability permits the concurrent broadcasting of multiple programs with varying error protection and coverage scopes. However, while the DVB-T system facilitates hierarchical transmission, it doesn't support hierarchical coding. The attributes of this highly adaptable transmission system are further elaborated in the subsequent sections, as depicted in Figure 36.

## *8.2 DVB-T SYSTEM Block presentation - and choices*

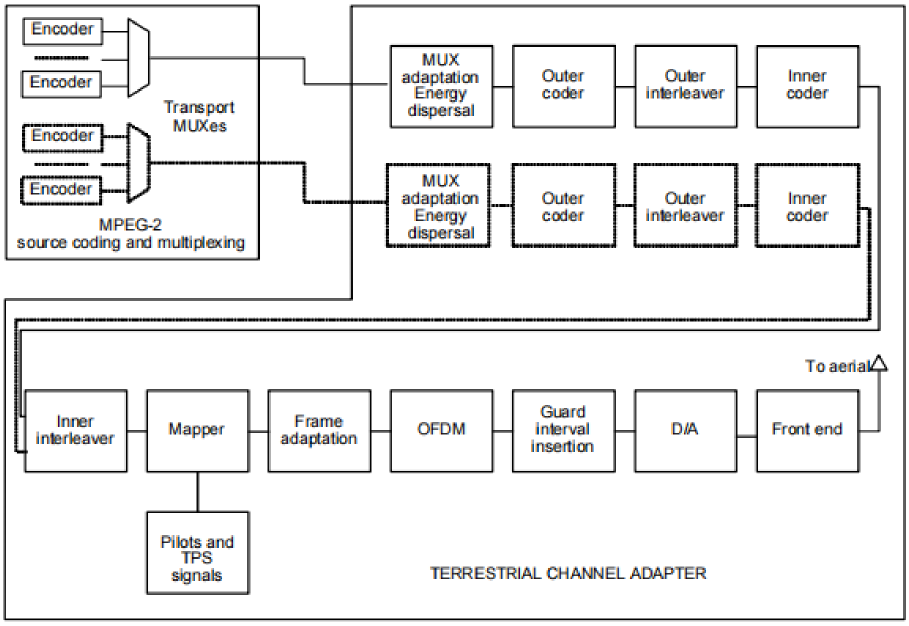



**Figure 36:** Functional block diagram of the DVB-T system.

### 8.2.1 Transmitter input signal

The DVB-T standard provides support for hierarchical configuration with two independent - different MPEG-2 TS data streams can be configured and transmitted together. A stream is called a high priority data stream (High Priority-HP) and is integrated into the other, called low-priority data stream (Low Priority - LP). When the reception conditions of the DVB-T signal are good, the receiver can demodulate and decode both streams. However, in unfavorable reception conditions, only HP is received successfully. The two data streams - HP and LP - can correspond to completely different services or in similar ones, for example a station may broadcast HDTV on the LP and SDTV, with the same content, on the HP, so when the receiving conditions allow it, the viewer watches the program in HDTV, while in unfavorable conditions download, watches the same program, on SDTV. LP usually has a higher tempo of transmission, butmore sensitive to errors and interference, while HP is transmitted with lower transmission rate, but more durable. As it presented to MPEG Chapter, initially, the encoding and multiplexing of content are performed using MPEG-2. The streams of video, audio, and image data are encoded using the MPEG-2 Program Stream (PS) format. Subsequently, one or more of these MPEG-2 PS streams are combined to produce the MPEG-2 Transport Stream (TS) format. This resulting stream is the fundamental digital data that is both broadcasted and received by DVB-T household devices.

8.2.2 MUX adaptation- Energy dispersal

The incoming system stream consists of an MPEG-2 transport multiplexer (MUX) packet, which uniformly maintains a length of 188 bytes. The packet's initial byte serves as a SYNC sequence, exhibiting a value that could either be 47HEX or B8HEX. To guarantee suitable binary transitions within the stream, the data is randomized in alignment with a specific system [37]. The configuration for this randomization process can be visualized as depicted in Figure 37.

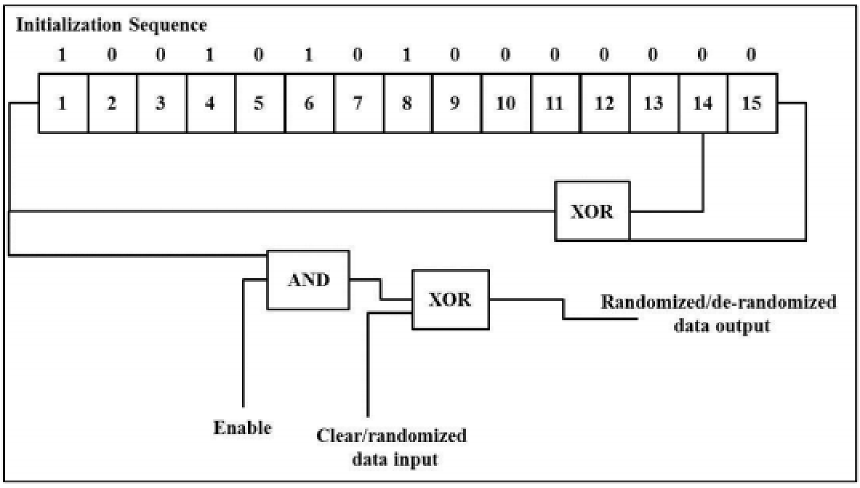



**Figure 37:** Scrambler / descrambler Schematic Diagram.

This system, known as the Pseudo Random Binary Sequence (PRBS) generator, is defined by the polynomial :

$$1+X^{14}+X^{15} \quad (22)$$

The operation is executed across eight transport packets, with the sequence "100101010000000" set in the PRBS registers, facilitating the descrambling initialization. For energy dispersion, a PRBS is generated. In DVB-T systems, this sequence is repeatedly reset based on specific conditions, particularly when a sync byte is flipped. The data flow undergoes an amalgamation with the PRBS using an Exclusive OR process, ensuring the disruption of extended strings of identical bits. When this dispersed data is combined with the identical PRBS at the receiver, the original data is restored.

### 8.2.3 Outer Encoder and Interleaver

Outer encoding and interleaving processes use the packet structure as seen in Figure 40a and it happens after energy dispersal. DVB-T system uses Reed-Solomon Encoding as its outer encoder (Figure 38). The main purpose is to generate an error corrected packet as seen in Figure 40c. The outer encoder has a Code Generator Polynomial, g*(x)* and a Field Generator Polynomial, *p(x)* which is

$$g(x) = (x+\lambda^0)(x+\lambda^1)(x+\lambda^2) \dots (\chi+\lambda^{15}) \quad (23)$$

$$p(x) = x^8 + x^4 + x^3 + x^2 + 1 \quad (24)$$

The Reed-Solomon encoding process consists of the following steps: :

1. For each randomized transport packet, which has a length of 188 bytes, the original RS(255,239,t=8) code is utilized. This is done by appending 51 bytes to the packet, all of which are set to zero, before processing the information bytes.

2. These 51 null bytes are then removed from the original RS code, resulting in a shortened RS(204,188,t=8) code with a length of N=204.

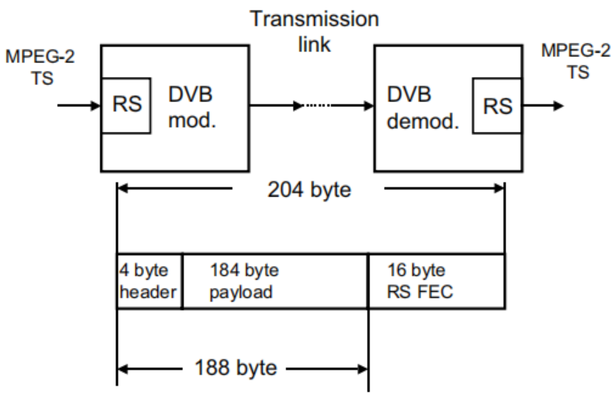



**Figure 38:** Reed-Solomon coding.

During transmissions, burst errors are a common occurrence. If these errors lead to more than 8 faults in a packet safeguarded by Reed-Solomon coding, the block error protection becomes ineffective. To address this, the data undergoes interleaving – a process where the data is dispersed over a specific time duration. This outer interleaving is achieved using a convolutional byte-wise method with an interleaving depth of I=12. The input to this process is the RS output packets, each being 204 bytes in length. The sequence that emerges from this interleaving can be observed in Figure 40d.

Interleaving follows the conceptual design shown in Figure 39. The interleaver is typically constructed of 12 branches, each of which is cyclically linked to the input stream via an input switch. Each of these branches operates as a First-In-First-Out (FIFO) shift register.[39].

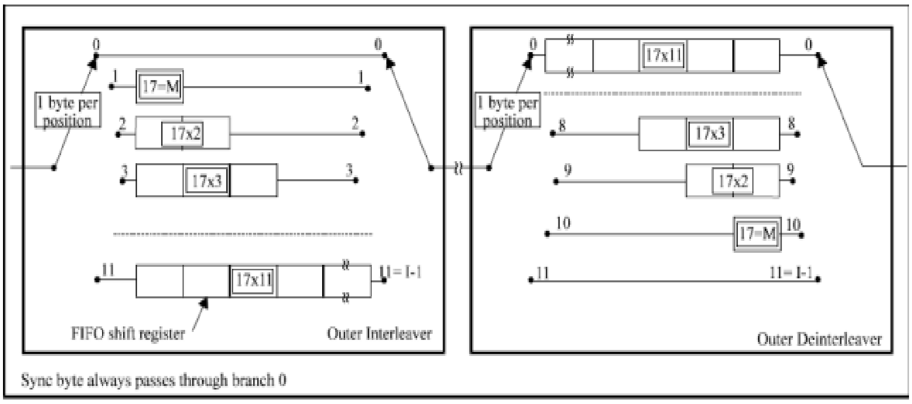



**Figure 39:** Representation of the outer interleaver and de interleaver processes.

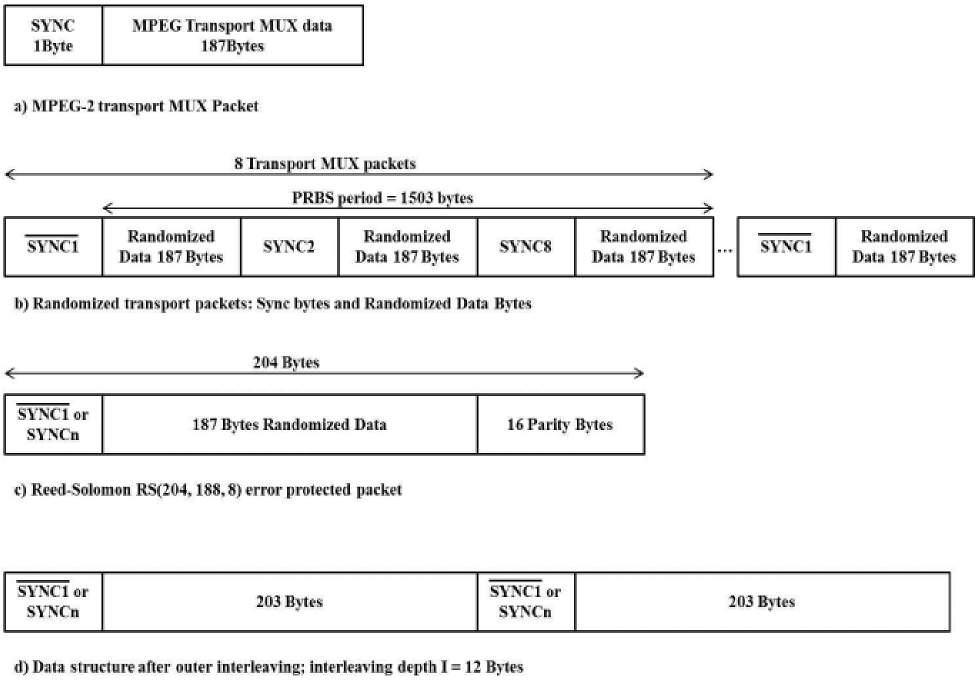



**Figure 40:** Process of adaptation, energy dispersal, outer coding and interleaving.

### 8.2.4 Inner Encoder

The subsequent component of the modulator is the sequence encoder, commonly referred to as the trellis coder. This component embodies the second layer of protection against errors, referred to as the inner defense. The sequence encoder is constructed in a straightforward manner. The DVB-T protocol utilizes a perforated sequence encoder for its primary encoding module. The inner encoding procedure involves two phases:

1. Primary encoding using a base sequence encoder possessing a 1/2 code rate

2. Puncturing according to system code rate.

The fundamental principle behind sequence coding can be observed in Figure 41. The defining polynomials of the base code are G1 = $171_{OCT}$ for output X, G2 = $133_{OCT}$ for output Y.

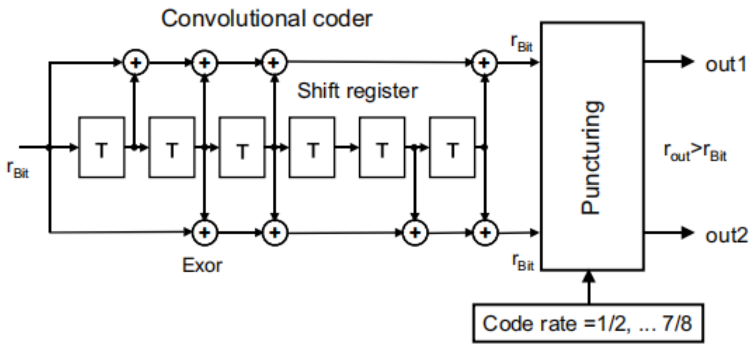



**Figure 41:** Convolutional coder in DVB-T.

The sequence encoder is made up of a 6-level shift register, complemented by two signaling routes where the incoming signal intertwines with specific tapped points from the shift register. The primary data flow bifurcates into three separate streams. Initially, the data enters the shift register, impacting the top and bottom streams of the sequence encoder via an Exclusive OR (EXOR) operation lasting six clock cycles. This procedure diffuses one bit's information across six bits. At designated junctions in both the upper and lower data pathways, EXOR gates are present, integrating the data streams with the contents stored within the shift register. Consequently, two intertwined data streams emerge from the sequence encoder, maintaining the same data rate as the original input. Moreover, this data stream is endowed with a six-clock cycle span of memory. Consequently, the cumulative output data rate becomes double the input, equating to a coding rate of 1/2. This process appends a 100% overhead to the original data stream. Beyond the foundational coding rate of 1/2, the system supports punctured rates such as 2/3, 3/4, 5/6, and 7/8. This punctured convolutional coding is detailed in Table 9.

**Table 9:** Puncturing pattern and transmitted sequence after parallel-to-serial conversion for possible code rates.

| Code Rate | Puncturing Pattern | Transmitted Sequence |
|---|---|---|
| 1/2 | X: 1<br>Y: 1 | $X_1$ $Y_1$ |
| 2/3 | X: 1 0<br>Y: 1 1 | $X_1$ $Y_1$ $Y_2$ |
| 3/4 | X: 1 0 1<br>Y: 1 1 0 | $X_1$ $Y_1$ $Y_2$ $X_3$ |
| 5/6 | X: 1 0 1 0 1<br>Y: 1 1 0 1 0 | $X_1$ $Y_1$ $Y_2$ $X_3$ $Y_4$ $X_5$ |
| 7/8 | X: 1 0 0 0 1 0 1<br>Y: 1 1 1 1 0 1 0 | $X_1$ $Y_2$ $Y_3$ $Y_4$ $X_5$ $Y_6$ $X_7$ |

In the mentioned table, X and Y represent the dual outputs emerging from the sequence encoder. Among the two, X1 always takes precedence in transmission, ensuring that the initial bit encoded symbolically aligns with X1.

### 8.2.5 Inner Interleaver

Within DVB-T systems, the inner interleaving comprises three primary components before progressing to the mapping and constellation stages as seen in Figure 42:

I. *Demultiplexer.*

II. *Bitwise Interleaver.*

III. *Symbol Interleaver*

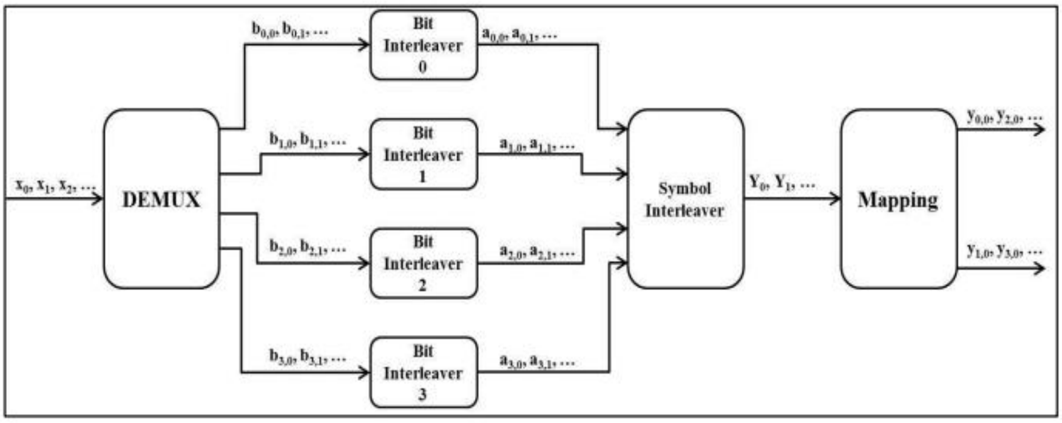



**Figure 42:** Mapping of input bits onto output 16QAM modulation symbols.

In simple terms, this means that the input bits are sorted and distributed across several sub-streams based on their modulation scheme. For example, with QPSK, there are two sub-streams, with 16QAM there are four, and with 64QAM, there are six. The exact way these bits are mapped or distributed is detailed in the given Table 10. The indices essentially help in determining the position or sequence of the bits as they are demultiplexed into their respective sub-streams [36] :

$d_i$ , Input bit number

e , The demultiplexed bit stream $(0 \leq e < v)$

$d_0$ , The bit number of a given stream at the output of the demultiplexer

**Table 10:** Demultiplexing bit mappings.

| | |
|---|---|
| **QPSK** | x0 maps to b0,0 |
| | x1 maps to b1,0 |
| **16QAM** | x0 maps to b0,0 |
| | x1 maps to b2,0 |
| | x2 maps to b1,0 |
| | x3 maps to b3,0 |
| **64QAM** | x0 maps to b0,0 |
| | x1 maps to b2,0 |
| | x2 maps to b4,0 |
| | x3 maps to b1,0 |
| | x4 maps to b3,0 |
| | x5 maps to b5,0 |

The individual sub-streams produced from the demultiplexer undergo further processing through distinct bit interleavers. Depending on the value of v, which is determined by the modulation scheme, you can have up to six separate bit interleavers operating concurrently. Each of these interleavers works on blocks that are 126 bits in size. Interleaving is an essential process as it spreads the data over the transmission frame, making the transmission more robust against burst errors or data loss. This operation is iterative and aligns with the size of the OFDM symbols, ensuring that the data is structured correctly for subsequent transmission steps.

The input bit vector $B_{(e)} = ( b_{e,0}, b_{e,1}, \ldots b_{e,125} )$.

The interleaved output vector $A_{(e)} = ( a_{e,0}, a_{e,1}, \ldots a_{e,125} )$ is :

$$a_{e,w} = b_{e,H_e(w)} \quad ,w=0,1,2,3,\ldots,125 \tag{25}$$

The permutation functions $H_e(w)$ are different for each interleaver and defined as:

$$H0(w) = w$$

$$H1(w) = (w + 63) \bmod 126$$

$$H2(w) = (w + 105) \bmod 126$$

$$H3(w) = (w + 42) \bmod 126$$

$$H4(w) = (w + 21) \bmod 126$$

$$H5(w) = (w + 84) \bmod 126$$

The primary objective of the symbol interleaver is to rearrange sets of v bbits into configurations suitable for the active carriers in an OFDM Symbol, specifically tailored for either 1512 carriers (in 2K Mode) or 6048 carriers (in 8K Mode). Initially, the data coming from the bit-level interleaver is transformed into groups, each constituting a bit word. This reorganization ensures that the modulation symbols are distributed across the OFDM carriers, enhancing the system's resilience against localized interference or noise in the frequency domain:

y'w = (a0,w, a1,w, ..., av-1,w)

The interleaved vector *Y = (y0, y1, y2,...yNmax-1)* is defined by:

*yH(q) = y'q* , for even symbols for q = 0,...,Nmax-1

*yq = y'H(q)* , for odd symbols for q = 0,...,Nmax-1

For a given system, Nmax represents the maximum number of carriers. Specifically, $N_{max}$ takes a value of 1512 in the 2K mode and 6048 in the 8K mode. The function H(q) serves as a permutation function. This function is crucial for ensuring that data is distributed across the carriers in a specific pattern, which can optimize reception under various conditions. The exact permutation, and the mathematical expression for H(q), is determined based on the system's requirements and the desired properties of the interleaving as shown in Figure 43.

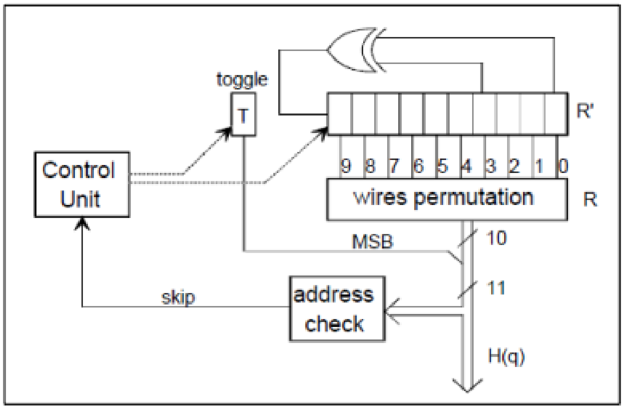



**Figure 43:** Symbol interleaver address generation scheme for the 2K mode.

## 8.2.6 Constellation and Mapping

The system employs OFDM (Orthogonal Frequency Division Multiplexing) for data transmission. In this technique, data is transmitted over multiple carriers, each modulated at a different frequency. The modulations used in the system can be QPSK (Quadrature Phase Shift Keying), 16QAM (16-Quadrature Amplitude Modulation), or 64QAM (64-Quadrature Amplitude Modulation). QPSK utilizes two bits per symbol, effectively conveying two bits of data with each transmitted symbol. In the QPSK constellation, each point represents a unique combination of two bits. 16QAM, as the name implies, uses a 16-

point constellation diagram. This means that it conveys 4 bits of data with each symbol, allowing for a higher data rate compared to QPSK but at the expense of greater susceptibility to noise. 64QAM further increases the data rate by using a 64-point constellation diagram, thus encoding 6 bits per symbol. However, the increased data rate comes at the cost of decreased noise tolerance. The Gray mapping ensures that successive symbols in the constellation differ by only one bit, minimizing the error probability. The specific details of the Gray mapping can be visualized in Figure 44, which provides a graphical representation of how the bits are mapped to the constellation points.

The exact values of the constellation points are $z \in \{n + jm\}$ with values of n, m given in the Table 11 for the various constellations.

The output data stream from the inner interleaver is organized into sequences of v bits, where v represents the number of bits in each word. This configuration is determined based on the modulation scheme employed. These are mapped onto complex number z, according to Figure 44.The constellation points, when represented in a complex plane, need to be normalized to ensure consistent energy levels across different modulation schemes. This ensures that the energy of the transmitted symbol, represented as E, remains consistent, irrespective of the modulation method. The normalization factors yield $E = [c \times c^*] = 1$. For the system to be effective, it's crucial that E equals 1. As such, specific normalization factors are provided for each modulation scheme (like QPSK, 16QAM, 64QAM) inTable 12 to ensure this consistent energy level across different modulation alphabets.

**Table 11:** Prices of the mapping points

| Modulation | n | m |
|---|---|---|
| QPSK | {-1, 1} | {-1, 1} |
| 16QAM | {-3, -1, 1, 3} | {-3, -1, 1, 3} |
| 64QAM | {-7, -5, -3, -1, 1, 3, 5, 7} | {-7, -5, -3, -1, 1, 3, 5, 7} |

**Table 12:** Scaling coefficients for data symbols.

| Modulation | Normalization factor |
|---|---|
| QPSK | $c = \frac{1}{\sqrt{2}}$ |
| 16QAM | $c = \frac{1}{\sqrt{10}}$ |
| 64QAM | $c = \frac{1}{\sqrt{42}}$ |

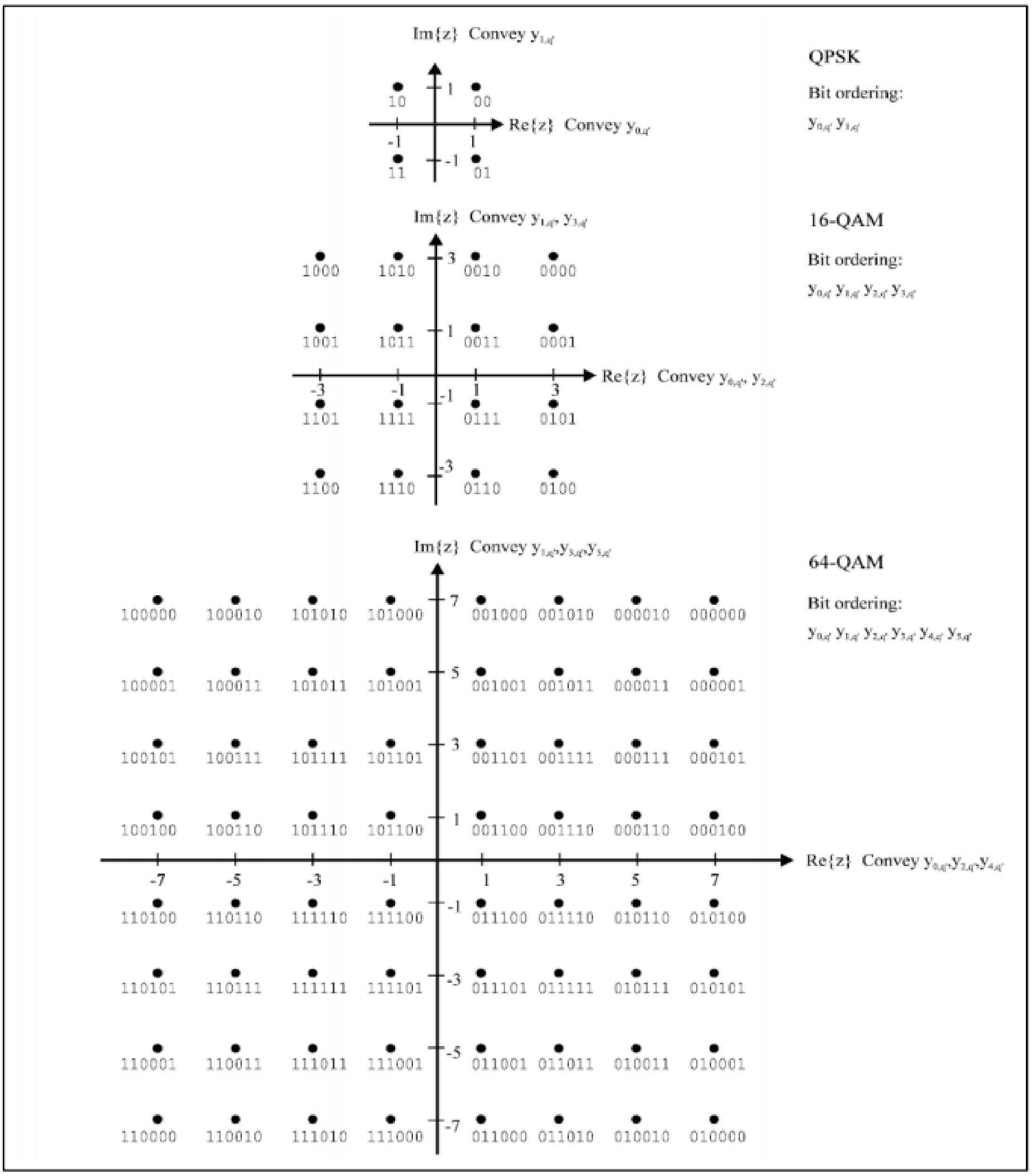



**Figure 44:** The bit patterns corresponding to QPSK, 16QAM, and 64QAM.

### 8.2.7 Pilots & TPS Signals

An OFDM Frame is made up of both data symbols and pilot symbols. Within DVB-T systems, there are three distinct types of pilot symbols, namely:

- ✓ Scattered pilots.
- ✓ Continual Pilots.
- ✓ Transmitter Parameter Signaling (TPS) pilots.

Pilots play a crucial role in frame alignment, frequency alignment, temporal synchronization, gauging the channel, identifying the transmission mode, and tracking phase noise. Both continual and scattered pilots are modulated based on a PRBS sequence, denoted as Wk , where k stands for the carrier index. The generation of this PRBS sequence is depicted in Figure 45. The generator for the PRBS is defined by a specific polynomial equation:

$$X^{11} + X^{2} + 1 \tag{26}$$

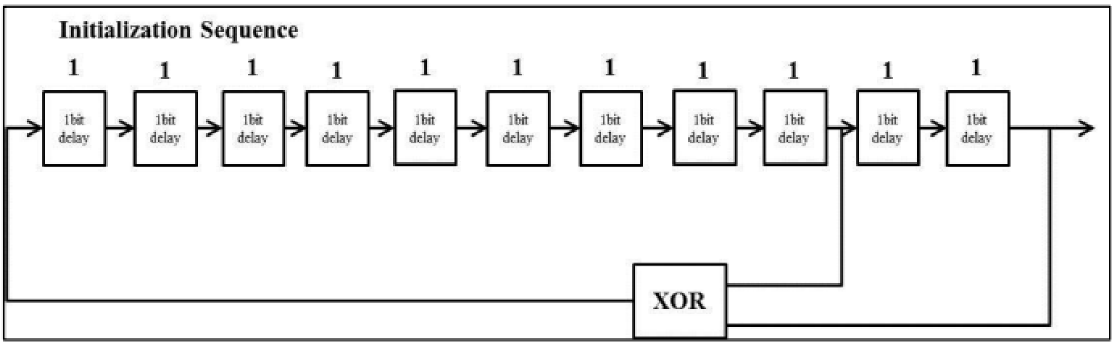



**Figure 45:** Generation of PRBS sequence.

Pilot signals within the OFDM frame are positioned based on the frame length and the chosen transmission mode. As highlighted earlier, every data cell is normalized such that its energy E equals the value derived from the equation E = [c x c* ] = 1. However, for cells designated as continual and scattered pilots, they're sent with enhanced power, setting their energy to E = [c x c*] = 16/9. The modulation specifics for these pilot cells can be found in Table 13.

**Table 13:** Modulation of Pilot Cells.

| | | |
|---|---|---|
| **Scattered Pilots** | $Re\{c_{m,l,k}\}$ | $\frac{4}{3} \times 2\left(\frac{1}{2} - w_k\right)$ |
| | $Im\{c_{m,l,k}\}$ | 0 |
| **Continual Pilots** | $Re\{c_{m,l,k}\}$ | $\frac{4}{3} \times 2\left(\frac{1}{2} - w_k\right)$ |
| | $Im\{c_{m,l,k}\}$ | 0 |
| **TPS Pilots** | $Re\{c_{m,l,k}\}$ | $2\left(\frac{1}{2} - w_k\right)$ |
| | $Im\{c_{m,l,k}\}$ | 0 |

The TPS carriers play a vital role in transmitting specific details regarding the transmission setup. In every symbol, each TPS carrier conveys a consistent differentially encoded data bit.

They relay crucial details such as:

- The type of modulation used.
- Information about the hierarchical structure.
- The designated guard interval.
- Inner coding rates.
- The chosen transmission mode.
- The count of the frame within a super frame.

In every OFDM frame, there's a transmission of 68 bits specifically allocated for the Transmission Parameter Signaling (TPS). These bits are broken down as:

- A single bit meant for initialization.
- 16 bits dedicated to synchronization.
- 37 bits that convey the actual information.
- An additional 14 bits which act as redundancy for the purpose of error correction.

To provide error correction, the 53 bits (consisting of both the synchronization and information bits) are supplemented with 14 additional parity bits, derived from the BCH (67, 53, t = 2) code. The polynomial responsible for generating this is denoted by h(x):

$$h(x) = x^{14} + x^{9} + x^{8} + x^{6} + x^{5} + x^{4} + x^{2} + 1 \qquad (26)$$

### 8.2.8 OFDM Process

The sent signal is structured into frames, with each frame lasting for $T_f$ and containing 68 OFDM symbols. A set of four frames forms a super frame. Depending on the mode, each symbol is made up of 6817 carriers in 8K mode or 1705 carriers in 2K mode, each being transmitted over a time span of Ts.

An OFDM symbol has two segments: a primary segment that lasts for Tu and a protective interval that lasts for Δ. This protective interval is essentially a repeated segment of the primary part Tu and is placed before it. Specific details about the guard intervals and their applications can be found in Table 14.

**Table 14:** Duration of symbol part for the allowed guard intervals.

| Mode | 8K Mode | | | | 2K Mode | | | |
|---|---|---|---|---|---|---|---|---|
| Guard Interval | 1/4 | 1/8 | 1/16 | 1/32 | 1/4 | 1/8 | 1/16 | 1/32 |
| $T_U$ | 896 μs | | | | 224μs | | | |
| Δ | 224 μs | 112 μs | 56 μs | 28 μs | 56 μs | 28 μs | 14 μs | 7 μs |
| $T_S=T_U+\Delta$ | 1120 μs | 1008 μs | 952 μs | 924 μs | 280 μs | 252 μs | 238 μs | 231 μs |

For the OFDM system, carriers have an assigned index denoted as K, which spans from Kmin to Kmax. Specifically, for the 2K mode, Kmin starts at 0 and extends to Kmax in 1704. For the 8K variant, Kmax is increased to 6816. A detailed breakdown of the numerical values associated with OFDM parameters for both these modes can be referenced in Table 15.

**Table 15:** Detailed parameters for the OFDM system in both the 8K and 2K modes.

| Parameter | 8K mode | 2K mode |
|---|---|---|
| Number of carriers, K | 6817 | 1705 |
| Value of carrier number, $K_{min}$ | 0 | 0 |
| Value of carrier number, $K_{max}$ | 6816 | 1704 |
| Duration $T_U$ | 896 μs | 224 μs |
| Carrier Spacing, $1/T_U$ | 1 116 Hz | 4464 Hz |
| Spacing between carriers $K_{min}$ and $K_{max}$ | 7,61 MHz | 7,61 MHz |

Mathematical representation of the OFDM signal in the time domain:

$$s(t)=\mathrm{Re}\left\{e^{j2\pi f_c t}\sum_{m=0}^{\infty}\sum_{l=0}^{67}\sum_{k=K_{min}}^{K_{max}} c_{m,l,k}\times\psi_{m,l,k}(t)\right\} \qquad (27)$$

Where,

$$\psi_{m,l,k}(t)=\begin{cases} e^{j2\pi\frac{k'}{T_U}(t-\Delta-l\times T_s-68\times m\times T_s)} & (l+68\times m)\times T_s\le t\le(l+68\times m+1)\times T_s \\ 0 & else \end{cases} \qquad (28)$$

where:

k : Carrier number, which ranges betweenKmin and Kmax

l : Represents the OFDM symbol number.

m : Denotes the transmission frame number.

K: Total number of transmitted carriers.

Ts: Duration of an OFDM symbol.

Tu: Inverse of the carrier spacing Tu=1/Δf, where Δf is the spacing between two consecutive carriers.

Δ : Duration of the guard interval, a portion of the symbol duration Ts which prevents interference between consecutive symbols.

fc : Central frequency of the RF signal

k': Carrier index relative to the centre frequency. It's defined as k' = k - (Kmax + Kmin) / 2

$c_{m,0,k}$ : complex symbol for carrier k of the Data symbol no. 1 in frame number m

$c_{m,1,k}$ : complex symbol for carrier k of the Data symbol no. 2 in frame number m

…

$c_{m,67,k}$ complex symbol for carrier k of the Data symbol no. 68 in frame number m

This equation provides a complete description of the OFDM signal in the time domain, showcasing the individual contributions of each carrier to the overall signal and accounting for the various parameters and time indices.

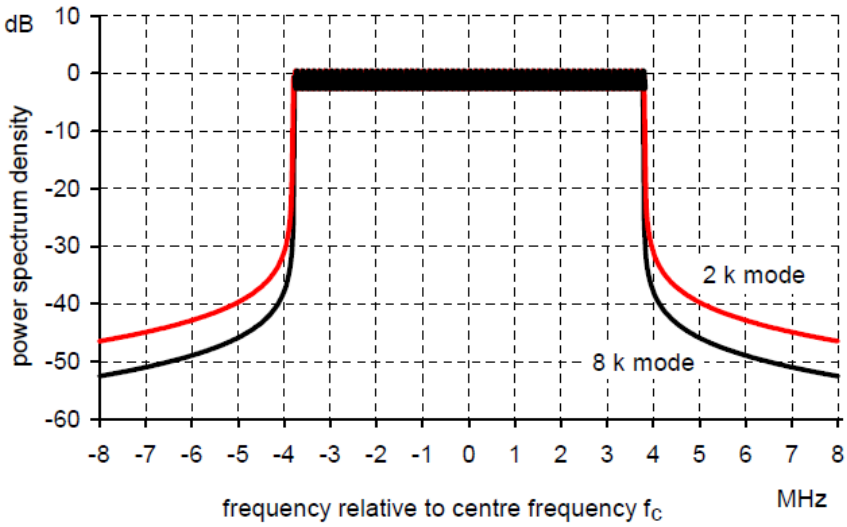



**Figure 46:** OFDM signal spectrum.

# CHAPTER 9:
# Analysis of DVB-T Protocol via MATLAB

## *9.1 Block Diagram of DVBT*

The DVB-T coder and modulator's block diagram is depicted in Figure 47. The coder comprises an outer Forward Error Correction (FEC) block, executed by a Reed-Solomon coder, outer interleaving at the byte level, and inner FEC employing a convolutional coder, with the option to use punctured code rates and an outer interleaver working at both bit and symbol levels. These four blocks function as FEC, offering protection against transmission errors. The Reed-Solomon and convolutional coders add redundancy to aid in error rectification at the receiving side, whereas interleaving is designed to counteract burst errors. The blueprint doesn't lay out the specifics of the demodulator and decoder, entrusting the implementation to the device manufacturer, such as those producing set-top boxes. Clearly, the receiver will have components executing operations opposite to the transmitter, along with components for aligning and rectifying the incoming signal.

Hierarchical modulation distinguishes itself from non-hierarchical modulation by splitting the input stream into two distinct streams - High Priority (HP), which has a lower data rate and can be received under subpar receiving conditions (low Carrier-to-Noise ratio), and Low Priority (LP) stream, which offers higher data rates but demands a higher Carrier-to-Noise ratio. Each stream is treated with a separate Forward Error Correction (FEC) and they are combined in an inner interleaver (as indicated by slashed blocks in Fig.47). The FEC difference lies in the code rate of the convolutional interleaver.

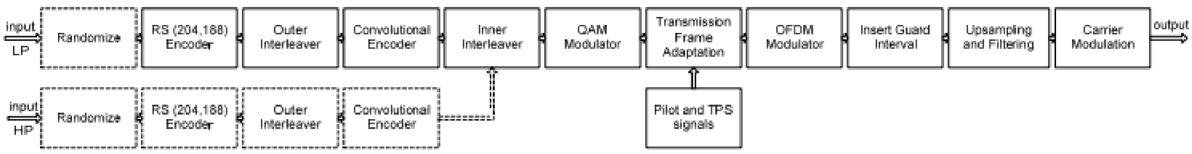



**Figure 47:** Block Diagram of DVB-T Coder/Modulator.

The Low Priority (LP) stream can have a code rate of 1/2, 2/3, 3/4, 4/5, 5/6, and 7/8, while the High Priority (HP) stream can have a rate of 1/2, 2/3, and 3/4. Both streams are transmitted simultaneously: the value of the QPSK modulated HP stream dictates the quadrant where the LP constellation points (16-QAM or 64-QAM) are located. Also defined is the α parameter (constellation ratio), which determines the distance between quadrants in the constellation diagram. The constellation rate can have α values of 1, 2, or 4. Constellation diagrams of a performed simulation of hierarchical transmission with α = 2 can be observed in Figure 48.

The main application window for simulations is shown in Figure 48.

The application incorporates blocks of coder and modulator as illustrated in Figure 47, functioning exactly in accordance with the DVB-T specification. It also contains a transmission channel simulator and demodulator and decoder blocks, which facilitate accurate decoding and demodulation of the received signal, as well as execution of error corrections and Bit Error Rate (BER) calculation. The final simulation's constellation plots can be seen at the window's lower section.

When simulating the transmission medium over which the signal travels, we have the option to choose from three primary channel types: the AWGN channel (a direct signal path between the sender and receiver influenced only by noise), the Rice channel (with both a direct and reflected signal paths), and the Rayleigh channel that involves signals solely from reflected paths.

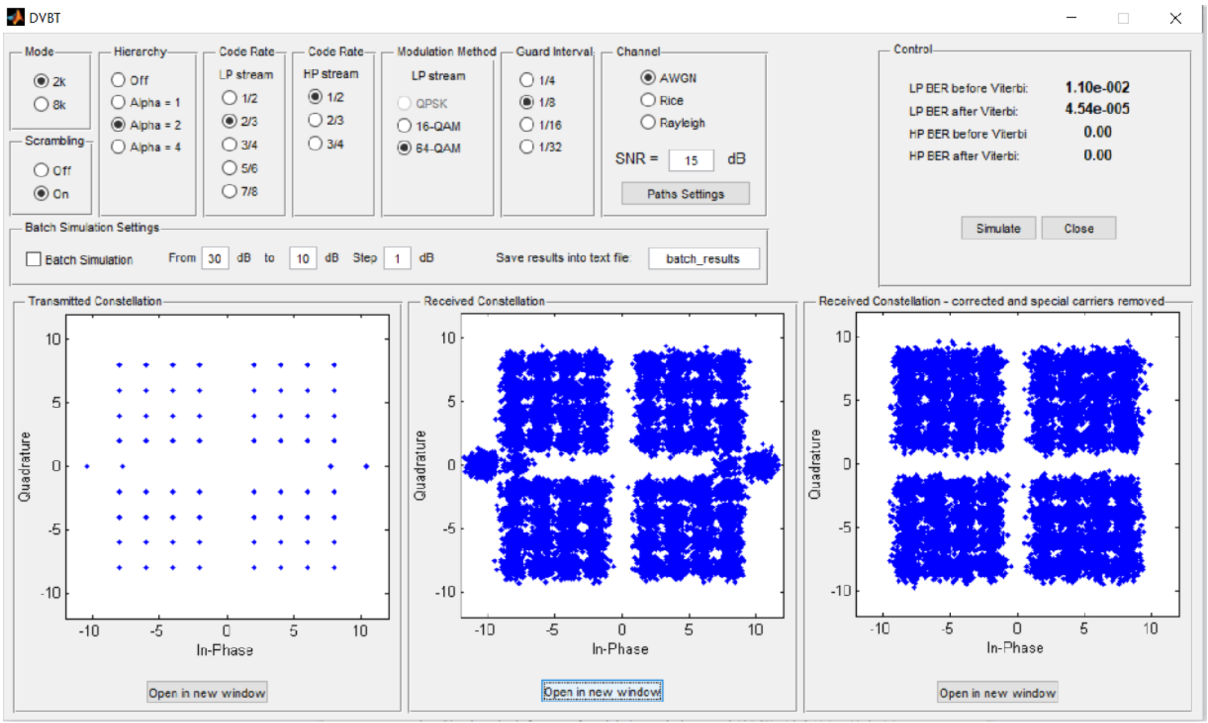



**Figure 48:** Main Window of DVB-T Simulator Application.

## *9.2 DVBT Blocks via MATLAB codes Highlights*

According to the descriptions of the DVB-T Blocks, some of the blocks via MATLAB are presented below:

- Scrambler is built with the initial sequence, the generation of PRBS and the randomization of the initial input shown in Figure 49. The process has been described in 8.2.2

```
%*************************************************************************
%   DVB-T scrambler using PRBS                                            *
%   Input:  row vector of numbers in range 0-255                          *
%   Output: row vector of numbers in range 0-255                          *
%*************************************************************************

function randomised = scrambler(input)

binary = zeros(1,length(input)*8);

% convert decimal input into serial binary
for n = 0:length(input)-1
    binary((n*8+1):(n*8)+8) = bitget(input(n+1), 1:8);
end;

% load intial sequence into registers
sequence = [1 0 0 1 0 1 0 1 0 0 0 0 0 0 0];
pseudorandom = zeros(size(binary));

% generate PRBS sequence
for n = 1:length(input)                                  % for each input bit
    pseudorandom(n) = xor(sequence(14), sequence(15)); % compute PRBS bit
    sequence = circshift (sequence,[0 1]);               % shift sequence to the right
    sequence(1)= pseudorandom(n);                        % place output bit to the first register
end

% randomise binary input using PRBS sequence
randomised = double(xor(binary, pseudorandom));

% reshape 8 each bits into one column (decimal numbers in rows)
reshaped = reshape(randomised', 8, length(randomised)/8);

% convert to decimal
decimal = zeros(1, length(reshaped));

for symbol = 1:length(decimal)                             % for each symbol (column)
    for bit = 1:8                                          % for each bit (row)
        C = reshaped(bit,symbol)*(2^(8-bit));              % count value of one bit
        decimal(1,symbol) = decimal(1,symbol)+C;           % add it to the total value of symbol
    end
end

% output of function
randomised = decimal;
```

**Figure 49:** Scrambler block via Matlab.

- Reed Solomon(204,188) encoder (outer encoder)and its Matlab code for LP stream (No Hierarchical mode) is highlighted in the Figure 50. The code word is 204 while the length of the message is 188. Thus, this Reed-Solomon encoder adds 16 bits (parity symbols). Therefore, the maximum number of errors that can be corrected is 16/2=8 erroneous bits. This is a Reed-Solomon that does not have a puncture code and is not shortened. Matlab code is performed and for HP stream (Hierarchical mode).

```
function [rs_encoded, rs_encoded_hp] = rs_encode(input, input_hp)
global values;
global settings
%*************************************************************************
% Performs RS(204,188)
% Input: column matrix
% Output: column matrix
%*************************************************************************
if settings.debug assignin('base', 'a_input_data', input); end;
if settings.debug && settings.hierarchy assignin('base', 'a_HP_input_data_hp', input_hp); en

k = 188;                                % input vector length
n = 204;                                % output vector length
m = 8;                                  % number of bits per symbol

% LP stream or no hierarchical mode
data = input;

if rem(length(data),k)                  % check dividing with 188
    x = 188*ceil(length(data)/k);
    values.zeros_added_RS = x - length(data); %amount of zeros added
    data = zeros(1,x);
    data(1:length(input)) = input;      % add zeros to be dividable
else values.zeros_added_RS = 0;         % no zeros added
end;

x = length(data)/k;                     % number of rows
data = reshape(data,x,k);               % make data into x by 188 matrix
msg = gf(data,m);                       % generate Galois Field
code = rsenc(msg,n,k);                  % perform RS(204,188)
code = double(code.x);                  % Galois Field into double

rs_encoded = reshape(code, x*n,1);      % form output into column matrix
```

**Figure 50:** Matlab Code for Reed Solomon encoding.

- Outer/convolution interleaving block is following. The highlight of its Matlab code in Figure 51 performs convolution interleaving with I=12 and slope 17. The I define how many rows of shift registers we will have and the slope determines the step with which the shift registers will increase for each row.

Both previous blocks are parts of the outer encoder/interleaver. The detailed characteristics and functions are described in the previous chapter 8.2.2

```
function [interleaved, interleaved_hp] = outer_interleaving(input, input_hp)
global values;
global settings;
%*************************************************************************
% Performs convolutional interleaving with I=12 slope=17
% Input: column or row matrix, length dividable with 204 (output from RS(204,188))
% Output: column matrix
%*************************************************************************

% LP stream of no hierarchical mode
l = length(input)/204;
reshaped = reshape(input,l,204);      % arrange data in groups of 204

interleave = zeros(1,204);      % create empty field for interleaved data

A = [1:204];                    % order generation
B = reshape(A,12,17);          % order before interleaving
B = B';
C = reshape(B,1,204);            % order after interleaving

for j = 1:l
    for i = 1:204
        interleave(j,C(1,i)) = reshaped(j,i);   % interleave
    end
end

l = size(interleave);
interleaved = reshape(interleave',1,l(1,1)*l(1,2))';  % rearrange into serial form
```

**Figure 51:** Outer Inter-leaver Matlab code with depth I=12.

Convolutional Encoder in Figure 52 is the block of the convolutional code. It is the block where the process of poly2trellis (polynomial to trellis) occurs. This is a few shift registers that are lined up in a row. That is, the output of the first shift register is the input of the second, the output of the second is the input of the third, and so on.

```
% Performs convolutional coding with G1=171oct G2=133oct and
% variable puncturing
% Input: column matrix (decimal)
% Output: column matrix (binary)

%**************************************************************************

global settings
global values

% LP stream or no hierarchical mode
binary = zeros(length(input)*8,1);   % create matrix for serial binary data

tic
for n = 0:length(input)-1           % convert decimal data to serial binary
    binary((n*8+1):(n*8)+8) = bitget(input(n+1), 1:8);
end;
toc

binary = flipdim(binary,2);   % MSB first instead of LSB returned by bitget

if settings.debug  assignin('base', 'e_converted_to_binary', binary); end;

switch settings.puncturing                          % prepare puncturing patterns
    case '1/2'
        puncpat = [1 1];
    case '2/3'
        puncpat = [1 1 0 1];
    case '3/4'
        puncpat = [1 1 0 1 1 0];
    case '5/6'
        puncpat = [1 1 0 1 1 0 0 1 1 0];
    case '7/8'
        puncpat = [1 1 0 1 0 1 0 1 1 0 0 1 1 0];
end;

switch settings.decision
    case 'soft'
        delay = 48;
    case 'hard'
        delay = 0;
end;

data = zeros(length(binary)+delay %TOO SOON!!!!! add zeros for soft decision delay
data(1:length(binary)) = binary;
binary = data;

if rem(length(binary), length(puncpat))  % input binary sequence has to be dividable by puncpat
    x = length(puncpat)*ceil(length(binary)/length(puncpat));
    bin = zeros(x,1);
    bin(1:length(binary)) = binary;               % add zeros to be dividable
    values.zeros_added_conv_encoding = length(bin)-length(binary);
    binary = bin;
else
    values.zeros_added_conv_encoding = 0;
end;

trel = poly2trellis(7,[171 133]);                          % define trellis
encoded = convenc(binary,trel,puncpat);  % encode the data with puncturing
```

**Figure 52:** Convolutional coder via Matlab code.

In the Matlab code is defined the Trellis structure, the poly2trellis (7, (171,133) is used. It has constraint length 7 (6 shift registers + 1input) and the 171 and 133 are the generator polynomials and are in octal.

The convolutional code is punctured so the puncturing patterns are defined to be able for DVBT.m to be selected different code rates 1/2, 2/3, 3/4, 5/6, 7/8. An additional Matlab code is created for HP stream (Hierarchical mode). The convolutional coder is also called inner coder and its function, and its characteristics are shown in previous chapter 8.2.4.

- Inner interleaver block is highlighted in 3 parts of the code. According to its parts and functions that have been presented in chapter 8.2.5, the Matlab code is structured to apply demultiplexing of the LP and HP streams as shown in Figure 53, bitwise interleaving (Figure 54) and symbol interleaving(figure 55).

```
demultiplexed=zeros(rows,symbols);    % prepare matrix for demultiplexing

%demultiplexing of HP stream
demultiplexed(1,:) = input_hp(1:2:end); %1st bit to 1st row
demultiplexed(2,:) = input_hp(2:2:end); %2nd bit to 2nd row

%demultiplexing of LP stream
switch settings.modulation
    case '16-QAM'
        demultiplexed(3,:) = input(1:2:end,1); %1st bit to 3rd row
        demultiplexed(4,:) = input(2:2:end,1); %2st bit to 4th row
    case '64-QAM'
        demultiplexed(3,:) = input(1:4:end,1); %1st bit to 3rd row
        demultiplexed(5,:) = input(2:4:end,1); %2nd bit to 5th row
        demultiplexed(4,:) = input(3:4:end,1); %3rd bit to 4th row
        demultiplexed(6,:) = input(4:4:end,1); %4th bit to 6th row
end
```

**Figure 53:** Demultiplexing of HP and LP Matlab code.

```
% HP and LP streams are joined together, hereinafter the processing is the
% same for both hierarchical and non-hierarchical modulation
bit_interleaved = zeros(rows,length(demultiplexed));

for w=1:126                                   % prepare permutations
    H(1,w) = w;
    H(2,w) = mod(w+63,126)+1;
    H(3,w) = mod(w+105,126)+1;
    H(4,w) = mod(w+42,126)+1;
    H(5,w) = mod(w+21,126)+1;
    H(6,w) = mod(w+84,126)+1;
end;

for e=1:rows;                                 % bit interleaver
    for w=1:126
        for n=0:length(bit_interleaved)/126-1;
            bit_interleaved(e,n*126+w)=demultiplexed(e,n*126+H(e,w));
        end;
    end;
end;

if settings.debug assignin('base', 'h_bit_interleaved', bit_interleaved'); end;

switch settings.mode                          % settings according to OFDM mode
    case '2k'
        perm_table=[4 3 9 6 2 8 1 5 7 0];
        Nmax = 1512;
        Mmax = 2048;
    case '8k'
        perm_table=[7 1 4 2 9 6 8 10 0 3 11 5];
        Nmax = 6048;
        Mmax = 8192;
end
```

**Figure 54:** Bit Interleaving Matlab code highlight.

```matlab
% add zeros to fill all OFDM symbols needed
values.OFDM_symbols_count = ceil(length(bit_interleaved)/Nmax);      % compute number of OFDM symbols needed
values.zeros_added_inner_interleaving = values.OFDM_symbols_count * Nmax - length(bit_interleaved); % compute number of zeros to add
temp = zeros(Nmax*values.OFDM_symbols_count, rows);                  % empty field with size of bits for integer number of OFDM symbols
[l1, l2] = size(bit_interleaved);
temp(1:l1,:) = bit_interleaved;                                      % fill the prepaired field with data
bit_interleaved = temp;                                              % replace original data

interleaved = zeros(Nmax*values.OFDM_symbols_count, rows);           % prepare field for interleaved data

% perform symbol interleaving
for s = 0:values.OFDM_symbols_count - 1
    for i = 1:Nmax
        if rem(s, 2) == 0
            interleaved(s*Nmax+1+Hq(i),:) = round(bit_interleaved(s*Nmax+i,:));       % even OFDM symbol
        else
            interleaved(s*Nmax+i,:) = round(bit_interleaved(s*Nmax+1+Hq(i),:));       % odd OFDM symbol
        end
    end
end

if settings.debug assignin('base', 'i_symbol_interleaved', interleaved); end;
```

**Figure 55:** Symbol Interleaving Matlab code.

- QAM modulator is the next block in DVBT implementation. It is a critical part of preparing signals for OFDM transmission and is used to modulate each of the individual carriers in a OFDM system. The code implements the filter shift which is part of the equalization process used to mitigate channel effects such as inter-symbol interference. Filter shift compensation involves estimating and correcting the phase and amplitude of the received symbols to account for the changes imposed by the channel. Then, the Root Raise Cosine (RRS) is because the RRS primary function is to shape the signal before the transmission. It is designed to minimize Inter Symbol Interference (ISI). ISI happens when pulses spread out and overlap, causing errors in detecting the transmitted symbols. Carriers' generation and the modulation of I and Q components codes are taken place to produce the final QAM signal (Figure 56).
- OFDM modulator block and its parts are built with code that create TPS sequence, apply PRBS for each carrier, place test sequence into scattered carriers, define the position of the continuous and TPS carriers as defined at ETSI EN 300 744 specification for DVBT standard. At the end, applied IFFT and generate the OFDM signal as it described in 8.2.7 and 8.2.8 chapter. Highlights of the Matlab code are shown in Figure 57,Figure 58, Figure 59 and is created both for 2K and 8K mode.

```
function transmit=quadrature_modulate(input)
global values
global settings

fs = 9.143e6;          %%symbol frequency
fvs = 82.287e6;        %sampling frequency
Tvs = 1/fvs;           %sampling period
fn = 30e6;             %carrier wave frequency in MHz
rolloff=0.35;          %roll-off factor of the transmitter and receiver filter
delay = 10;

input_real = real(input);
input_imag = imag(input);

%compensation for filter shift
input_real_compensated = [input_real, zeros(1,2*delay)];
input_imag_compensated = [input_imag, zeros(1,2*delay)];

% upsample
n = fvs/fs;

upsampled_real = upsample(input_real_compensated, n);
upsampled_imaginary = upsample(input_imag_compensated, n);

% filtering with RRC filter
RRC_filter = rcosine(fs, fvs, 'sqrt', rolloff, delay);
filtered_real = filter(RRC_filter,1,upsampled_real);
filtered_imaginary = filter(RRC_filter,1,upsampled_imaginary);

% prepare time vector
t = 0:1/fvs:(length(filtered_real))/fvs;
t = t(1:end-1);

% generate carrier
carrier_real = cos(2*pi*fn*t);
carrier_imag = sin(2*pi*fn*t);

% modulate to the carrier
signal_real = filtered_real.*carrier_real;
clear carrier_real;
signal_imag = filtered_imaginary.*carrier_imag;

% create resulting IQ modulated signal
transmit = signal_real+signal_imag;
if settings.debug assignin('base', 'n_IQ_modulated', transmit); end;
```

**Figure 56:** Quadrature Modulation Matlab Code.

```
% define positions of continual carriers (ETSI EN 300 744 page 28)
continual_positions = [0 48 54 87 141 156 192 201 255 279 282 333 432 450 ...
                       483 525 531 618 636 714 759 765 780 804 873 888 918 ...
                       939 942 969 984 1050 1101 1107 1110 1137 1140 1146 ...
                       1206 1269 1323 1377 1491 1683 1704];
```

**Figure 57:** Define positions of continual carriers (2K) as ETSI EN 300 744 defines.

```
% define positons of TPS carriers (ETSI EN 300 744 page 29)
tps_positions = [34 50 209 346 413 569 595 688 790 901 ...
                 1073 1219 1262 1286 1469 1594 1687];
```

**Figure 58:** Define position of TPS carriers (2K) as ETSI EN 300 744.

```
% transfer to the time domain and demultiplexing
OFDM_time=ifft(OFDM_frequency_domain);              % IFFT - to the time domain
[l,m]=size(OFDM_time);
OFDM_time = reshape(OFDM_time,1 ,l*m);             % create resulting OFDM signal

% store variables into workspace in debug mode
if settings.debug assignin('base', 'k_OFDM_frequency_domain', OFDM_frequency_domain); end;
if settings.debug assignin('base', 'l_OFDM_time_domain', OFDM_time); end;
```

**Figure 59:** IFFT and OFDM signal code.

- Guard Interval block is used to prevent interference or overlap between separate transmissions, and it is part of OFDM symbol. Its characteristics and duration are shown in chapter 8.2.8. The code is highlighted below in Figure 60.

```
function guarded = guard(input)
global values
global settings

switch settings.guard_interval
    case '1/4'
        guard = 1/4;
    case '1/8'
        guard = 1/8;
    case '1/16'
        guard = 1/16;
    case '1/32'
        guard = 1/32;
end

guarded = zeros(1,length(input)*(guard+1));

switch settings.mode
    case '2k'
        symbol_length = 2048;
        guard_length = 2048 * guard;
    case '8k'
        symbol_length = 8192;
        guard_length = 8192 * guard;
end
```

**Figure 60:** Guard Intervals code.

- Carriers' modulation block is defined with a Matlab code (Figure 61) to prepare modulator objects according to modulation and to achieve the scatter plot on simulation.

```
%prepare modulator objects according to modulation
switch settings.modulation
    case 'QPSK'
        modulator = modem.qammod(4);
        modulator.SymbolOrder =  ('user-defined');
        modulator.symbolmapping = [2 3 0 1];
    case '16-QAM'
        modulator = modem.qammod(16);
        modulator.SymbolOrder =  ('user-defined');
        modulator.symbolmapping = [8 9 13 12 10 11 15 14 2 3 7 6 0 1 5 4];
    case '64-QAM'
        modulator = modem.qammod(64);
        modulator.SymbolOrder =  ('user-defined');
        modulator.symbolmapping = [32,33,37,36,52,53,49,48,34,35,39,38,54,55,51,50,42,4
end;

% perform modulation
modulated = modulate(modulator, decimal);

if settings.hierarchy
    alpha = settings.hierarchy-1;
    % hierarchical scaling - real part of signal
    modulated(real(modulated)>0)=modulated(real(modulated)>0) + alpha;
    modulated(real(modulated)<0)=modulated(real(modulated)<0) - alpha;

    % hierarchical scaling - imaginary part of signal
    modulated(imag(modulated)>0)=modulated(imag(modulated)>0) + j*alpha;
    modulated(imag(modulated)<0)=modulated(imag(modulated)<0) - j*alpha;
end

digitally_modulated=modulated;
%scatterplot(digitally_modulated);
if settings.debug assignin('base', 'j_digitally_modulated', digitally_modulated); end;
```

**Figure 61:** Digital modulation code Highlight

- Channel simulation code in Figure 62 is the code that determines the type of noise that transmission could have. This code gives as the opportunity to simulate and to examine the transmission of the signal, the interference that could exist from the transmitter to the receiver.

The AWGN (Additive White Gaussian Noise) channel is a model that posits the sole distortion comes from the linear introduction of white noise with a consistent spectral density throughout its frequency range. This noise is termed 'additive' since it combines with any pre-existing noise and signal, and it's labeled 'white' due to its uniform spectral distribution.

The Rician channel is a model designed to represent scenarios where a signal reaches the receiver via a combination of paths: a direct Line of Sight (LOS) route and multiple indirect non-LOS routes. In this model, the direct path signal is conceptualized as a stable component, while the indirect scattered routes are depicted as fluctuating components.

The Rayleigh fading channel is a model that represents variations in a signal's amplitude resulting from multipath propagation. Specifically, this model is relevant in scenarios where there is no direct Line of Sight (LOS) connection between the transmitter and receiver. In such conditions, the signal often encounters significant fluctuations in strength, occasionally leading to severe signal dropouts.

```
function output = channel_simulation(input, tau, pdb, snr)
global settings

        fs = 9.143E6;                    % Symbol frequency FFT: 9.143 MHz
        fn = 30e6;                       % Carrier frequency: 626MHz (3*)
        fvz = 9*fs;                      % Sampling frequency of OFDM signal: 6 (at least 3*fn)
        v = 0;                           % Receiver movement [km/h]
        fd = (v/3.6)*(fn/3E8);            % Maximum Doppler shift
        tvz = (1/fvz);                     % Sampling period of input signal (duration of one sample)
        ps = 10.^(pdb./10);


switch settings.channel
        case 'AWGN'
             output = awgn(input, snr, 'measured');   % just add noise
        case {'Ricean', 'Rayleigh'}
            pocCest = length(tau);
            signal_length = length(input);
            shift = fix(tau./tvz);
            maxshift = max(shift);
            input = input';
            output = zeros(1,signal_length+maxshift);

            for path = 1:pocCest
                onepath = zeros(1,signal_length+maxshift);

                onepath(shift(path)+1:shift(path)+signal_length) = input;
                gain = ps(path);                     %*exp(-j*theta(path));
                onepath = onepath*gain;

                %Adding the signal to the total
                output = output + onepath;

            end
            %Loss of the shifted signal at the end: return to the original signal length and normalisation:
                output = output(1:signal_length);
                norm = sqrt(sum(ps.^2));
                output = output/norm;
                output = awgn(output, snr, 'measured');   % just add noise
end
```

**Figure 62:** AWGN, Ricean and Reyleigh model channels

In this chapter has been covered, by highlights, the major parts of Matlab code for each block of the DVBT Simulation.

# CHAPTER 9: Simulated Transmission of Digital Television According to DVB-T Standard

## *9.1 Simulation of the influence of modulation technique used (QPSK, 16-QAM a 64-QAM – all non-hierarchical).*

Next, it simulated the influence of modulation technique used (QPSK, 16-QAM a 64-QAM –all non-hierarchical). It is used these settings: mode 8k, guard interval 1/4, channel type AWGN, code rate 1/2. It switched the batch simulation on, the noise sweep is set from 25 to 5 dB with step 0.5 dB. The simulations are performed for QPSK, 16-QAM a 64-QAM modulation. The graph in Figure 63 containing all three dependencies of BER on the C/N ratio before Viterbi decoding. The Quadrate Consellation for QPSK is shown in **Error! Reference source not found.** and .

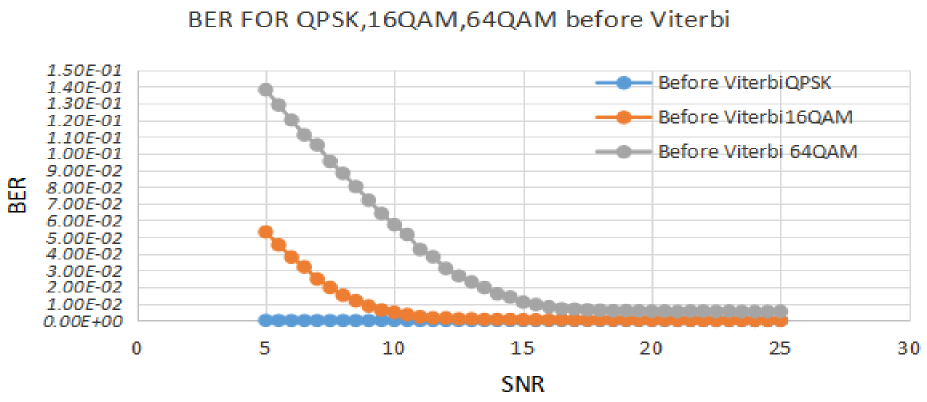



**Figure 63:** BER graph for QPSK,16QAM,64QAM before Viterbi

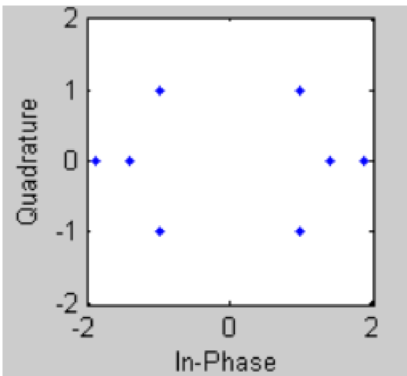



**Figure 64a:** Transmitted Constellation

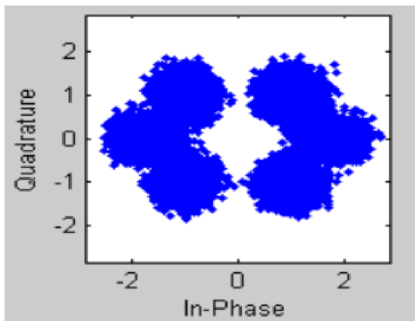



**Figure 64a:** Transmitted Constellation: Received Constellation before Viterbi

As it noticed from the graph in figure 63, a QPSK modulation, can tolerate higher noise levels. Therefore, for a given SNR, the BER for QPSK is lower than for the other two modulation schemes. 16QAM have a higher BER than QPSK but lower than 64QAM. 64QAM carries even more bits per symbol but as it noticed is also the most susceptible to noise. 64QAM has the highest BER.

## *9.2 Simulation of the influence of convolutional encoder code rate.*

The settings for this case of simulation are mode 8k, guard interval 1/4, channel type AWGN, and 64-QAM modulation. In simulation the code rate is changed for each code rate: 1/2, 2/3 a 3/4. The noise sweep is set from 25 to 5 dB with step 0.5 dB. The graph in figure containing all three dependences of BER on the C/N ratio after Viterbi decoding. In Figure 65 the graph shows how code rate values influences the BER for 64QAM modulation.

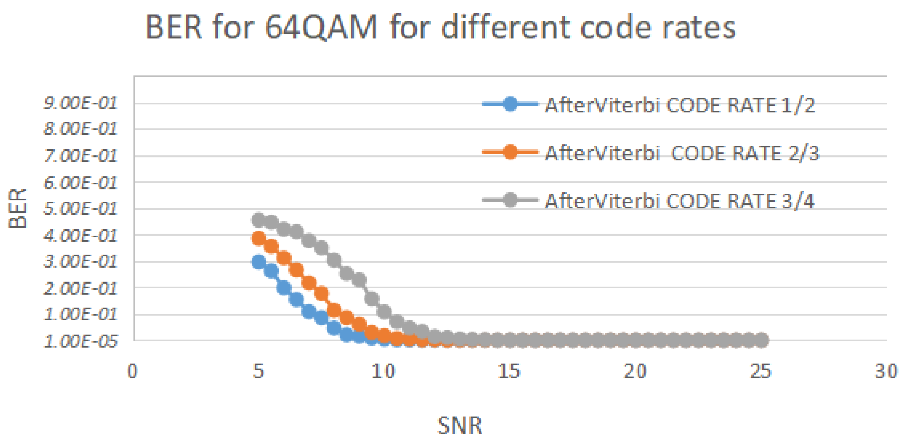



**Figure 65:** BER graph for different code rates

As it noticed, the choice of convolutional code rate significantly affects the system's bit error rate (BER), especially when the modulation scheme is 64QAM and the FFT size is 8K. Code Rate 1/2 means that for every 1 bit of data, there are 2 bits of encoded data. It provides the highest redundancy, and consequently, the highest level of error correction. It can withstand a higher degree of noise and interference. However, this comes at the cost of a reduced data rate since the number of bits transmitted is doubled. Code Rate 2/3 for every 2 bits of data, there are 3 bits of encoded data. This represents a balance between error protection and data rate. It provides less error correction capability than a 1/2 code rate, but it allows a higher data rate. Code Rate 3/4 means that for every 3 bits of data, there are 4 bits of encoded data. This has the least redundancy and therefore provides the least error protection, but it allows the highest data rate.

In terms of BER, as the code rate increases (from 1/2 to 2/3 to 3/4), the BER is noticed that is increased due to the lower redundancy, hence less error correction capability. However, it must be noted that the actual BER will also heavily depend on the signal-to-noise ratio (SNR) of the system. In a 64-QAM modulation scheme, the choice of a lower code rate (like 1/2) can be beneficial as it can help to correct more errors in the presence of noise, especially considering that 64-QAM is more sensitive to noise than simpler modulation schemes like QPSK or 16-QAM. But it comes with the trade-off of a lower data rate. Is also noticed that for high SNR, that means low noise, is achieved high data rate and low BER and this is beneficial case if the transmission happens in low noise environment.

## *9.3 Simulation of hierarchical modulation*

In this simulation the setting that are used are: mode 8k, hierarchical modulation QPSK in 64-QAM, HP code rate 1/2, LP code rate 2/3. The simulation has the parameter α (Alpha) = 2. The graph in Figure 66 contains dependence of BER in HP and LP stream on the C/N ratio before Viterbi decoding and the graph in Figure 67 containing dependence of BER in HP and LP stream on the C/N ratio after Viterbi decoding.

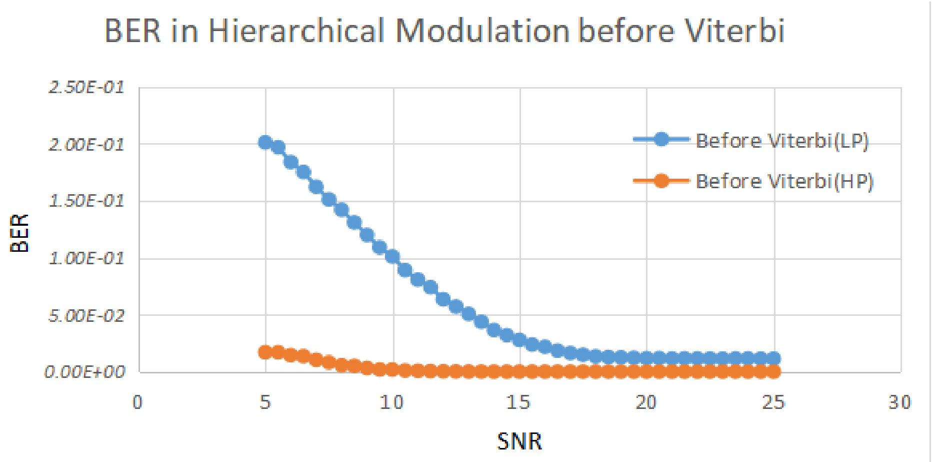



**Figure 66:** BER in HP and LP before Viterbi decoding

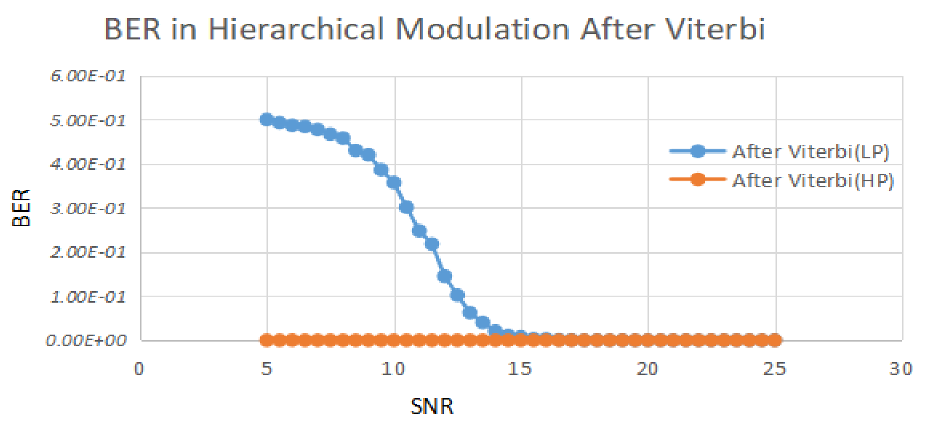



**Figure 67:** BER in HP and LP after Viterbi decoding

As it mentioned in previous chapter Hierarchical modulation, allows the transmission of two (or more) data streams within one signal, with each stream given a different level of robustness against noise and interference. This technique is particularly useful for broadcasting scenarios where you want some content to be received reliably even under poor conditions, while other content requires a higher-quality signal to be received correctly. In such a scenario, the "high priority" stream is more robust, and can be received even when the signal quality is poor. The "low priority" stream carries additional data but requires a better signal to be received without errors.

The parameter "alpha" determines the power ratio between these two layers. A higher alpha means more power (and thus more robustness) is given to the high priority stream, and less to the low priority stream. In the DVB-T system, the alpha values that can be chosen are 1 (meaning the power is divided equally), 2, 4, or unequal error protection (UPE) without any specific ratio. In our simulation the value of a is 2. In both figures is noticed that BER for LP stream is much higher as it is expected.

In the DVB-T standard, the Viterbi algorithm is used as the decoder for the convolutional encoder at the transmitting side. The Viterbi algorithm is widely used in DVB-T because of its ability to correct errors that might have been introduced during the transmission of digital signals. It is used to ensure that the digital information (e.g., video, audio) is received with as few errors as possible, which results in higher quality video and audio for the end user.

From the graphs is noticed that Viterbi decoder corrects errors and reduce the BER on SNR ratio especially on the LP stream and gives an acceptable BER about 7 dB lower of SNR.

## *9.4 Simulation of the influence of the transmission channel*

The settings for this simulation are mode 8k, non-hierarchical 64-QAM, code rate 2/3, guard interval 1/4). First, the simulation is performed in the AWGN channel. Then we switch to the Rice channel, open signal propagation paths settings and use “ETSI EN 300 744” button to perform automatic setting of testing paths according to the DVB-T specification. Next, we choose the Rice channel and perform the simulation. The characteristic of each channel is mentioned in page 81. The graph in Figure 68 contains all three dependences of BER on the C/N ratio after Viterbi decoding.

From the graph AWGN channel emerges that has the lower BER due to the lower noise in it. The BER on Rician and Rayleigh channel are roughly the same but the drops on Rayleigh channel because of the fading is also noticed.

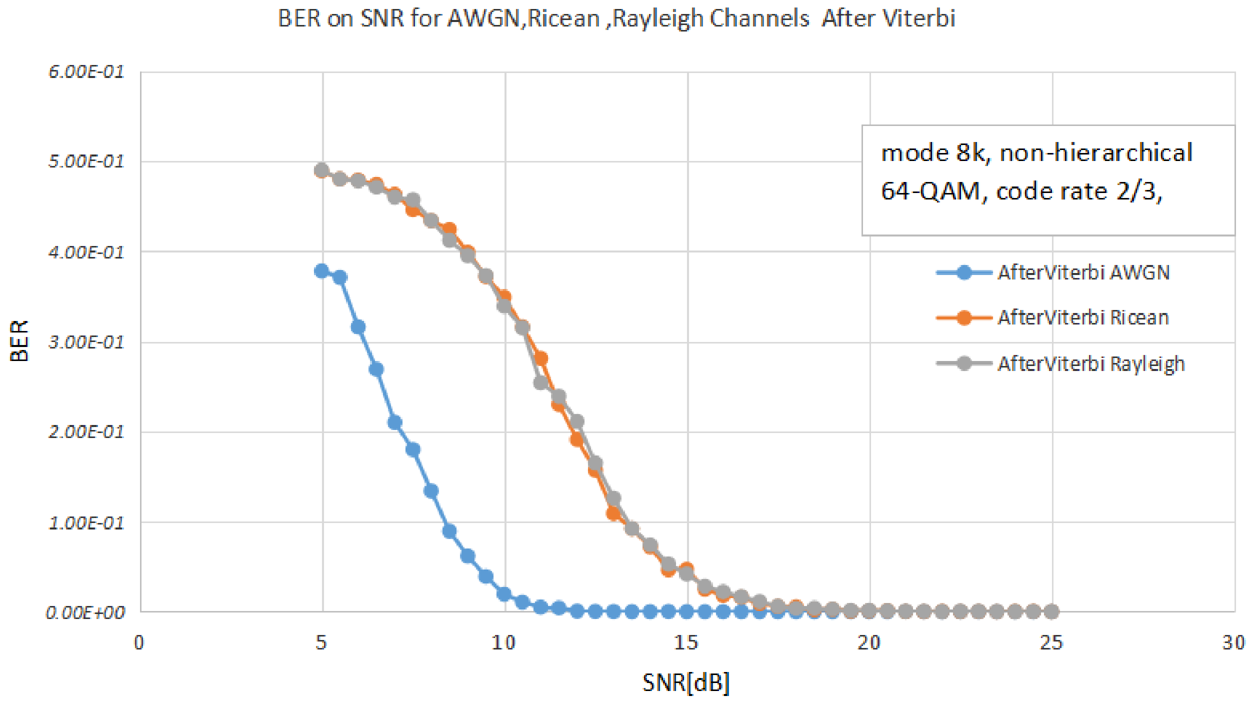



**Figure 68:** Simulation of BER for AWGN, RICIAN and Rayleigh Channels

### *9.5 General Conclusion s in Simulation of the DVBT System*

- Reed-Solomon coding and Convolutional coding are the two error correction methods. From the results of the simulation is noticed that among the two, Reed-Solomon decoding is generally more computationally intensive and can consume more processing time, especially in cases where the error rates are high, and many corrections need to be made. The RS coding consuming the most computing time. However, it's worth noting that both methods are essential to maintain the quality of transmission and the computational cost is a trade-off for the resilience they provide against data errors. Also, the exact computational cost can vary depending on the specific implementation and system parameters.
- There are disadvantages of robust error correction and modulation such as: effective data reduction, more power is needed for the additional processing required for error correction and modulation, In COFDM, a guard interval is introduced between symbols to prevent inter-symbol interference. This slightly reduces the useful data rate, also is noticed that trade-off in DVB-T between robustness to multipath effects (by using longer guard intervals and more redundancy in FEC - Forward Error Correction) and the capacity of the channel. More robust settings lead to a lower bit rate, and less robust settings lead to a higher bit rate but might suffer more from errors. In general, it's crucial to strike a balance between robustness and efficiency when designing and implementing digital communication systems like DVB-T. These disadvantages should

be weighed against the benefits of more reliable signal reception and the potential for improved coverage and reception quality.

- As it mentioned, QPSK, as a phase modulation scheme, encodes two bits per symbol, which means it has four possible states. On the other hand, 16-QAM and 64-QAM encode 4 and 6 bits per symbol, respectively, and have 16 and 64 states, respectively. Each state corresponds to a specific phase and amplitude of the signal. The more states there are (i.e., the more bits per symbol), the closer these states are to each other in the signal space. This makes the signal more susceptible to noise and interference because it's more difficult to distinguish between the different states. Thus, QPSK, with only four states, is more robust against noise and interference than 16-QAM or 64-QAM because its states are further apart and easier to distinguish. However, this robustness comes at the cost of a lower data rate since fewer bits are transmitted per symbol.
- Examine the Hierarchical vs Non-Hierarchical modulation in the simulation, it is made it cleared, that are some advantages and disadvantages on Hierarchical modulation.

✓ Positive, is that the high-priority stream is more protected and can be received even in poor signal conditions. The low-priority stream can provide extra data or higher quality when the signal conditions are good. Also offers Service Flexibility as it allows for a variety of services to be offered, from mobile TV (which would only need the high-priority stream) to HDTV (which would take advantage of both streams). By layering data streams, the same channel can be used to deliver different services to different users, leading to more efficient use of the spectrum.

✓ As disadvantages are considered the Reduced Capacity: The process of protecting the high-priority stream reduces the overall capacity of the system. The total data rate of the two streams in hierarchical modulation is less than it would be in a system using only non-hierarchical modulation. Additionally, the performance of the low-priority stream is closely tied to the state of the signal. Under unfavorable signal situations, the low-priority stream might become completely unreceivable.

- Beyond the observations related to transmission channels, it's evident that the real error rate is influenced by elements like the chosen modulation method, the coding strategy, the average SNR, and the predominant strength of the LOS component in a Rician channel.

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