# DIGITAL VIDEO BROADCASTING VIA SATELLITE (DVB-S)

**M V Raghavendra**
Associate Professor,
Dept of Electrical Engineering
Adama Science & Technology University, Ethiopia.

**K.A.Lathief**
Dept of Electrical & Computer Engineering
JIT, Jimma University, Ethiopia.

**Vuda Sreenivasarao**
Dept of Computer and I T
Defence University College, Ethiopia.

**ABSTRACT**
This paper provides a brief introduction to the DVB-S system based on [EN-300-421]. The DVB-S system provides direct-to-home (DTH) services for consumer integrated receiver decoders (IRD), as well as collective antenna systems (satellite master antenna television SMATV) and cable television head-end stations. The overview covers the physical layer that comprises adaptation, framing, coding, interleaving and modulation, and discusses error performance requirements to achieve quality of service (QoS) targets.

**Keywords**: system provides direct-to-homey, satellite master antenna television and achieves quality of service.

## 1. INTRODUCTION

The European Telecommunications Institute (ETSI) is a non-profit organization that creates standards for different areas of telecommunications. A standardized radio interface enables a mass market for consumer reception devices. Having in mind the many past issues resulting from the different analogue TV standards and its multiple variations, most of the actors (broadcasters, service providers, operators, equipment and chips manufacturers, etc.) worked together at the end of the 1980s to define a digital video broadcasting (DVB) standard. This standard has been broken down into different versions depending on the specific properties of the transmission channel which conditions the physical layer characteristics: DVB-T for terrestrial digital TV, DVB-C for cable, DVB-S for satellite. Later standards have been introduced: DVB-RCS for the return channel, DVB-S2 (the second generation of DVB-S), DVB-H for handheld terminals, DVB-SH for satellite handheld terminals, etc.

Although the DVB-S standard was designed initially for satellite digital television services, the physical layer of the DVB-S can carry streams of packetized data of any kind. Mass-market production, and the availability of different equipment and related building blocks, makes the standard appealing for a lot of applications other than the transmission of TV signals, such as Internet traffic.

### *A. Transmission System*

The transmission system consists of the functional block of equipment to transport baseband TV signals in the format of the MPEG-2 transport stream over the satellite channel. The transmission system carries out the following processes on the data stream:

- transport multiplex adaptation and randomization for energy dispersal;
- outer coding (i.e. Reed–Solomon);
- convolution interleaving;
- inner coding (i.e. punctured convolution code);
- baseband shaping for modulation;
- Modulation.

Digital satellite TV services have to be delivered to home terminals with rather small antennas (around 0.6 m) which translate typically into a power-limited downlink. To achieve a high power 148 Digital Communications Techniques efficiency without excessively penalizing the spectrum efficiency, the DVB-S uses QPSK modulation and the concatenation of convolutional and RS codes. The convolutional code can be configured flexibly, allowing the optimization of the system performance for a given satellite transponder bandwidth.

DVB-S is directly compatible with MPEG-2 coded TV signals (defined by ISO/IEC DIS 13818-1). The modem transmission frame is synchronous with the MPEG-2 multiplex transport packets. If the received signal is above the considered threshold for the carrier-to-noise power ratio, C/N, the FEC technique can provide a quasi-error-free (QEF) quality target. The QEF means BER less than 10 -10 to 10-11 at the input of the MPEG-2 demultiplexer.

### 1.1.1. Input stream scrambling:

The DVB-S input stream is the MPEG-2 transport stream (MPEG-TS) from the transport multiplexer.The packet length of the MPEG-TS is 188 bytes. This includes one sync-word byte (i.e.47HEX). The processing order at the transmitting side starts from the most significant bit (MSB). In order to comply with ITU-R Radio Regulations and to ensure adequate binary transitions, the data of the input MPEG-2 multiplex is randomized.

The randomization is in accordance with the configuration depicted in Figure 1. The polynomial for the pseudorandom binary sequence (PRBS) generator is defined as:

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

Loading the sequence 100101010000000 into the PRBS registers is initiated at the start of every eight transport packets. To provide an initialization signal for the descrambler, the MPEG-2 sync byte of the first transport packet in a group of eight packets is bit-wise inverted from 47HEX to B8HEX. This process is referred to as 'transport multiplex adaptation'.

The first bit at the output of the PRBS generator is applied to the first bit (i.e. MSB) of the first byte following the inverted MPEG-2 sync byte (i.e. B8HEX). To aid other synchronization functions, during the MPEG-2 sync bytes of the subsequent seven transport packets, the PRBS generation continues but its output is disabled, leaving these bytes un randomized. Thus, the period of the PRBS sequence is 1503 bytes. The randomization process is also active when the modulator input bit stream is non-existent, or when it is non-compliant with the MPEG-2 transport stream format (i.e. 1 sync byte +187 packet bytes). This is to avoid the emission of a un modulated carrier from the modulator, as energy concentrated on the carrier frequency can cause interference with neighbouring satellites.

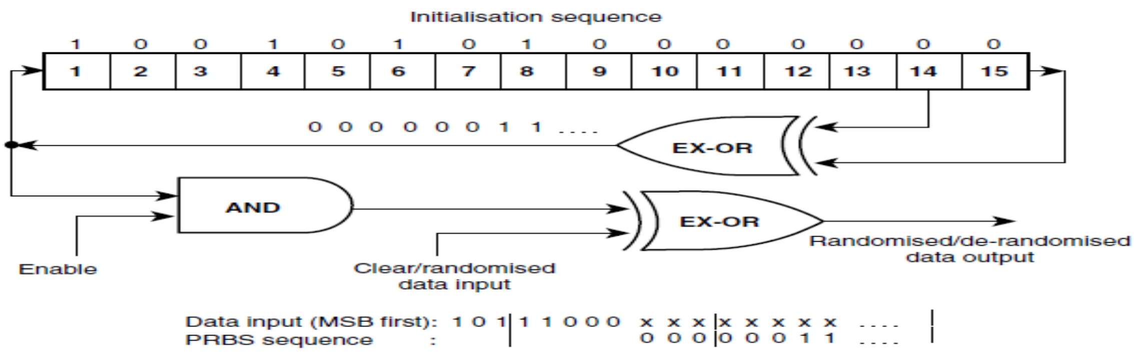


**Figure 1 Randomizer/de-randomizer schematic diagram.**

### 1.1.2 Reed–Solomon outer coding, interleaving and framing

The framing organization is based on the input packet structure shown in Figure 4.32a. The Reed–Solomon RS(204, 188, T¼8) shortened code, from the original RS(255, 239, T¼8) code, is applied to each randomized transport packet (188 bytes) of Figure 2b to generate an error protected packet (see Figure 2c). Reed–Solomon is applied to the packet sync byte, either non inverted (i.e. 47HEX) or inverted (i.e. B8HEX). The code generator polynomial is:

$$g(x) = (x+\lambda^0)\,(x+\lambda^{10})\,(x+\lambda^2)\,(x+\lambda^{15})\ldots\ldots\ldots\,(x+\lambda^0)$$

Where $\lambda=02_{Hex}$

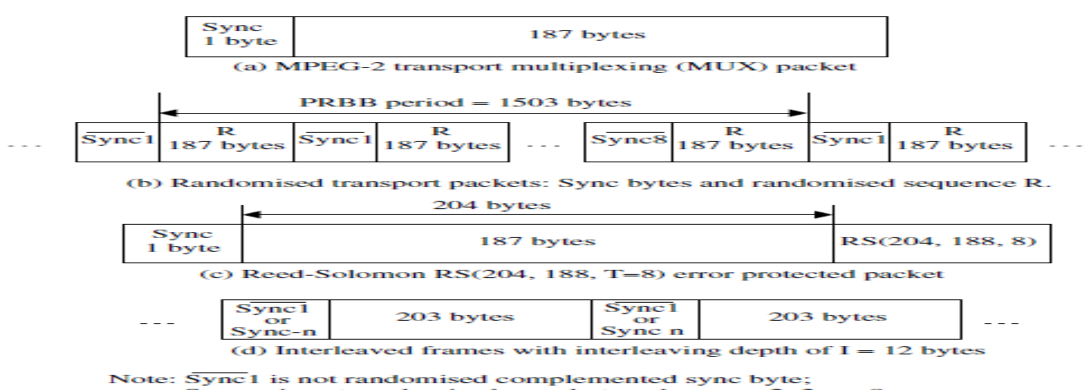


**Figure 2 Framing structure.**

**Table 1 Punctured code definition. Original code: K=7; G1(X) =171OCT; G2 (Y) =133OC**

| | P | | | | |
|---|---|---|---|---|---|
| Code rate | X | Y | I | Q | $d_{free}$ |
| 1/2 | 1 | 1 | $X_1$ | $Y_1$ | 10 |
| 2/3 | 1 0 | 1 1 | $X_1\ Y_2\ Y_3$ | $Y_1\ X_3\ Y_4$ | 6 |
| 3/4 | 1 0 1 | 1 1 0 | $X_1\ Y_2$ | $Y_1\ X_3$ | 5 |
| 5/6 | 1 0 1 0 1 | 1 1 0 1 0 | $X_1\ Y_2\ Y_4$ | $Y_1\ X_3\ X_5$ | 4 |
| 7/8 | 1 0 0 0 1 0 1 | 1 1 1 1 0 1 0 | $X_1\ Y_2\ Y_4\ Y_6$ | $Y_1\ Y_3\ X_5\ X_7$ | 3 |

1 transmitted bit
0 non-transmitted bit

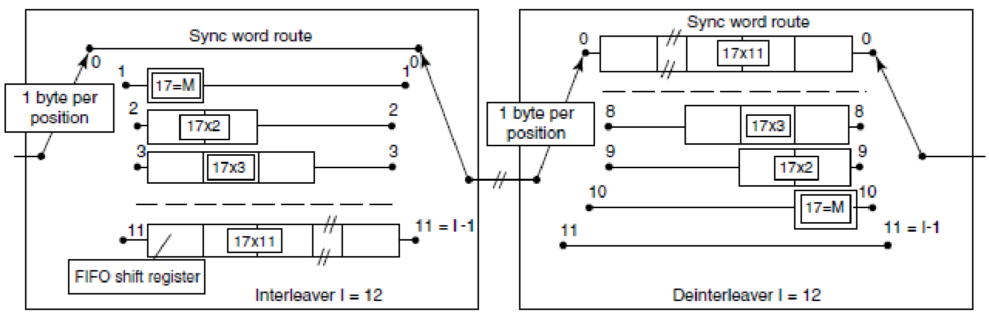


**Figure 3 Convolutional interleaver and deinterleaver.**

The field generator polynomial is:
$p(x) = x^8 + x^4 + x^3 + x^2 + 1$

The shortened Reed–Solomon code is implemented by adding 51 bytes, all set to zero, before the information bytes at the input of a (255,239) encoder. After the RS coding procedure these null bytes are discarded. Following the conceptual scheme of Figure 3, convolutional interleaving with depth I=12 is applied to the error protected packets (see Figure 2c). This produces an interleaved frame (see Figure 2d). The interleaved frame consists of overlapping error protected packets and is delimited by inverted or non-inverted MPEG-2 sync bytes (preserving the periodicity of 204 bytes).

The inter leaver consists of I=12 branches, cyclically connected to the input byte stream by the input switch. Each branch is a first-in–first-out (FIFO) shift register, with depth (Mj) cells (where M=17N/I, N=204 (error protected frame length), I=12 (interleaving depth) and j is the branch index). The cells of the FIFO contain 1 byte and the input and output switches are synchronized. For synchronization purposes, the sync bytes and the inverted sync bytes are always routed in branch 0 of the interleaver (corresponding to a null delay).

The deinterleaver is similar, in principle, to the interleaver, but the branch indexes are reversed (i.e. j=0 corresponds to the largest delay). The deinterleaver synchronization is carried out by routing the first recognized sync byte in the 0 branch.

### 1.1.3 Inner convolutional coding:

DVB-S allows for a range of punctured convolutional codes, based on a rate 1/2 convolutional code with constraint length K=7. This allows selection of the most appropriate level of error correction for a given service data rate. It allows convolutional coding with code rates of 1/2, 2/3, 3/4, 5/6 and 7/8. Table 1 gives the definition of punctured convolutional code.

### 1.1.4 Baseband shaping and modulation:

DVB-S employs conventional Gray-coded QPSK modulation with direct mapping (no differential coding). Prior to modulation, the I and Q signals (mathematically represented by a succession of Dirac delta functions spaced by the symbol duration Ts=1/Rs, with appropriate sign) is square root raised cosine filtered. The value of 0.35 is selected as the roll-off factor

**Table 2** BER versus $E_b/N_0$ performance requirements

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

### 1.2 Error performance requirements

Table 2 gives the modem BER versus Eb/N0 performance requirements. The figures of Eb/N0 refer to the useful bit rate before RS coding and include a modem implementation margin of 0.8 dB and the noise bandwidth increase due to the outer code (10 log 188/204¼0.36 dB). Quasi- error-free (QEF) means less than one uncorrected error event per hour, corresponding to
B ER=$10^{-10}$ to $10^{-11}$ at the input of the MPEG-2 demultiplexer.

## 2. SECOND GENERATION DVB-S

The DVB-S standard uses QPSK modulation and concatenated convolutional and Reed Solomon channel coding. It has been adopted by most satellite operators worldwide for television and data broadcasting services. Digital satellite transmission technology has evolved significantly in several areas since the first publication of the DVB-S standard in 1994. Without going into too many details of the standard, this section provides a brief summary of DVB-S2's new technology, transmission system architecture and performance.

### 2.1. New technology in DVB-S2

DVB-S2 makes use of the new developments in technology and future applications of broadband satellite applications. The main features can be summarized as the following:

- new channel coding schemes to achieve a capacity gain in the order of 30%;
- variable coding and modulation (VCM) to provide different levels of error protection to different service components (e.g. SDTV and HDTV, audio, multimedia);
- Extended flexibility to cope with other input data formats (such as multiple transport streams or generic data formats in addition to the single MPEG transport stream (MPEG-TS) in DVB-S) without significant complexity increase.

In the case of interactive and point-to-point applications, the VCM functionality is combined with the use of return channels to achieve adaptive coding and modulation (ACM). This technique provides dynamic link adaptation to propagation conditions, targeting each individual receiving terminal. ACM systems promise satellite capacity gains of more than 30%. Such gains are achieved by informing the satellite uplink station of the channel condition (e.g. the value of carrier power to-noise and interference power ratio, C/N þ I) of each receiving terminal via the satellite or terrestrial return channels.
DVB-S2 makes use of new technology in the following functions:

- stream adapter, suitable for operation with single and multiple input streams of various formats (packetized or continuous);
- forward error correction based on LDPC codes concatenated with BCH codes, allowing QEF operation at about 0.7 dB to 1 dB from the Shannon limit;
- a wide range of code rates (from 1/4 up to 9/10);
- Four constellations (QPSK, 8PSK, 16APSK, 32APSK), ranging in spectrum efficiency from 2 bits$^{-1}$ Hz $^{-1}$ to 5 bits$^{-1}$ Hz$^{-1}$, optimized for operation over non-linear transponders;
- three spectrum shapes with roll-off factors 0.35, 0.25 and 0.20;
- ACM functionality, optimizing channel coding and modulation on a frame-by-frame basis.

DVB-S2 has also been designed to support a wide range of broadband satellite applications including:

**Broadcast Services** (BS): Digital multi-program television (TV) and high definition television (HDTV) for primary and secondary distribution in the Fixed Satellite Service (FSS) and the Broadcast Satellite Service (BSS) bands. BS has two modes: non-backwards-compatible broadcast services (NBC-BS) allows exploitation of the full benefit of the DVB-S2 but is not compatible with DVB-S; backwards-compatible broadcast services (BC-BS) is backwards-compatible with DVB-S to give time for migration from DVB-S to DVB-S2.

**Interactive Services** (IS): Data services including Internet access for providing interactive services to consumer integrated receiver decoders (IRD) and to personal computers, where DVB-S2's forward path supersedes the current DVB-S for interactive systems. The return path can be implemented using various DVB interactive systems, such as DVB-RCS (EN-301-790), DVBRCP (ETS-300-801), DVB-RCG (EN-301-195), and DVB-RCC (ES-200-800).

**Digital TV Contribution and Satellite News Gathering** (DTVC/DSNG): Temporary and occasional transmission with short notice of television or sound for broadcasting purposes, using portable or transportable uplink earth stations. Digital television contribution applications by satellite consist of point-to-point or point-to-multipoint transmissions, connecting fixed or transportable uplink and receiving stations. They are not intended for reception by the general public.

**Professional Services** (PS): Data content distribution/trunking and other professional applications for point-to-point or point-to-multipoint, including interactive services to professional head-ends, which re-distribute services over other media. Services may be transported in (single or multiple) generic stream format.Digital transmissions via satellite are affected by power and bandwidth limitations. DVB-S2 helps to overcome these limits by making use of transmission modes (FEC coding and modulations), giving different trade-offs between power and spectrum efficiency. For some specific applications (e.g. broadcasting), modulation techniques, such as QPSK and 8PSK with their quasi-constant envelope, are appropriate for operation with saturated satellite power amplifiers (in a single carrier per transponder configuration). When higher power margins are available, spectrum efficiency can be further increased to reduce bit delivery cost. In these cases, 16APSK and 32APSK can also operate in single carrier mode close to satellite HPA saturation if linearization by pre-distortion techniques is implemented. DVB-S2 is compatible with MPEG-2 and MPEG-4 coded TV services (ISO/IEC 13818-1), with a transport stream packet multiplex. All service components are time division multiplexed (TDM) on a single digital carrier.

**2.2 Transmission system architecture:**

The DVB-S2 system consists of a number of functional blocks of equipment performing the adaptation of the baseband digital signals from the output of one or more MPEG transport stream multiplexers (ISO/IEC 13818-1) or one or more generic data sources to the satellite channel characteristics. Data services may be transported in transport stream format according to (EN-301- 192) (e.g. using multi-protocol encapsulation (MPE)) or generic stream (GS) format. DVB-S2 provides a QEF quality target of ‘less than one uncorrected error event per transmission hour at the level of a 5 Mbit/s single TV service decoder’, approximately corresponding to a transport stream packet error ratio (PER) of less than $10^{-7}$ before de-multiplexer.

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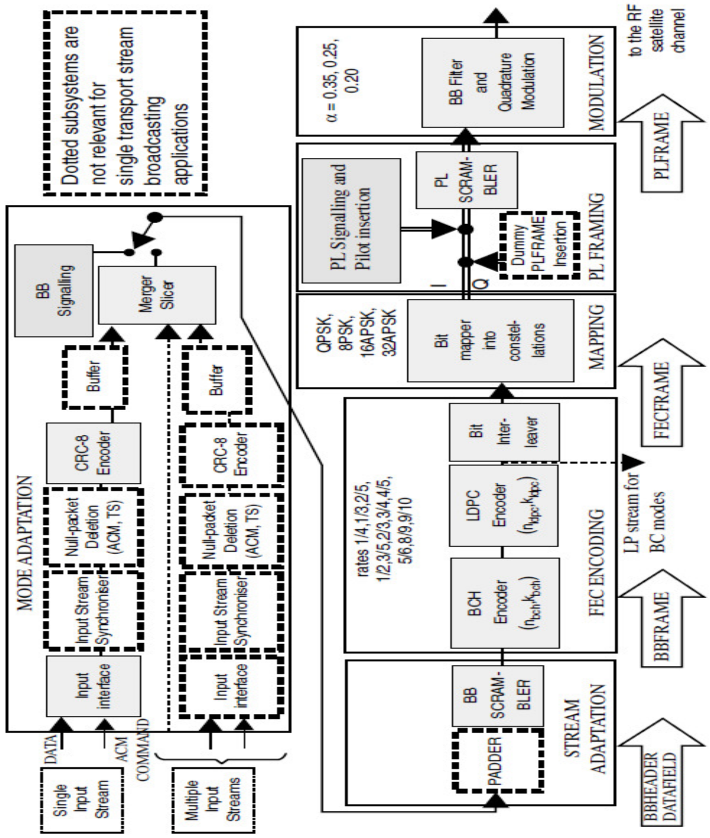


**Figure 4: Function Blocks in a DVB-S2 System:**

## 3. CONCLUSION

The standard discussed above also suggests that, for short FEC frames (FECFRAME), an additional degradation of 0.2 dB to 0.3 dB has to be taken into account; for calculating link budgets, specific satellite channel impairments should be taken into account. Spectral efficiencies (per unit symbol rate) are computed for normal FEC frame length and no pilots. This paper concludes by giving examples from digital transmission of telephony and broadcasting of television.

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**AUTHORS PROFILE:**

**Dr. M.V.Raghavendra** is currently working as Professor, Electrical Engineering Dept, Adama Science & Technology University, Ethiopia. He received his PhD from Singhania University, Rajasthan, India in the field of Optical Communication and he is a Research scholar, Department of Instrument Technology, College of Engineering, Andhra University, Andhra Pradesh, India. He has received his M.Tech Degree from ECE Dept, Vishakhapatnam College of Engineering, Andhra University. His main research includes signal estimation & evaluation of optical communication, Satellite communication & Microwaves. He has published papers in reputed national & international Journals. He has participated in different national & international conferences. He is a life member of ISTE & ISOI, MIAENG, MIACSIT, and MAIRCC.

**Dr. K.A.Lathief** received his M.Tech degree in Electronics& Communication Engineering from IASE University Rajasthan from 2005.He received PhD degree in Communication engineering from Magadh University from 2007. Currently working as Professor in the department of Electronics and Communication, Jimma University, jimma Ethiopia. His main research Interests are Communication areas of Satellite, Cellular Mobile, VSAT Networking and Wireless Environments.. He has got 12 years of teaching experience. He has published two research papers in various international journals. He has 4 memberships in
various international journals.

**Dr. Vuda Sreenivasarao** received his M.tech degree in computer science and engineering from Sathyabama University from 2007.He received PhD computer science and from Singhania University, Rajasthan, India from 2010. Currently working as Professor in the department of Computer and Information Technology, Defence University College, Deberezeit, Ethiopia. His main research interests are Data mining, Fuzzy logic, Mobile communication and Network Security. He has got 12 years of teaching experience. He has published 27 research papers in various international journals and one Springer international conference paper. He has Editorial memberships in various international journals. He is a life member of various professional societies like IEEE, ACM, MAIRCC, MCSI, SMIACSIT, MIAENG, MCSTA, MAPSMS, MSDIWC and MISTE