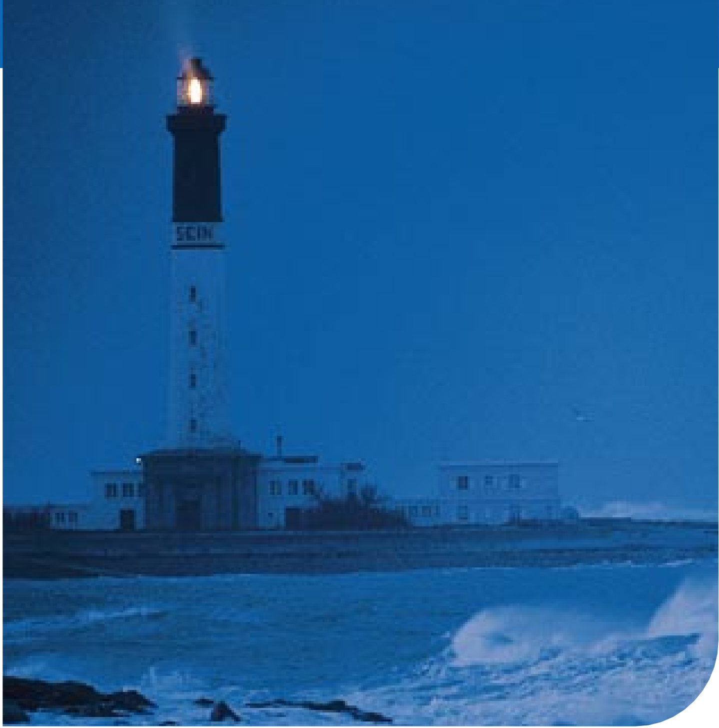
MPEG-2 Pocket Guide Digital Broadcast Systems



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MPEG-2
Pocket Guide

Digital Broadcast Systems

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1 Introduction

In 1936, the year of the Berlin Olympic Games, spectators crowded into specially built viewing rooms called Fernsehstuben (literally, television rooms) to catch a glimpse of one of the first-ever television broadcasts. In black and white, 180 lines/frame, and 25 frames/second, it would hardly compare to television by today's standards; however, it became the progenitor of modern-day broadcasting, one of the most powerful tools of the Information Age.

Now, nearly seven decades later, the international community stands at the dawn of a new millennium. The great technological advances of the past century propel us toward the next great advancement or invention at a frantic pace. More than ever before, international organizations of experts are pooling their resources—working together for the greater good of the entire world. Digital broadcasting is one of the newest technological advances to rest on the shoulders of international collaboration. In 1990, the Moving Pictures Experts Group, commissioned by the ISO/IEC international standards organizations, organized a multinational group of experts to begin work on the MPEG-2 specification. This standard has become the backbone for digital television as we know it.

This pocket guide will help you to become familiar with the basics of digital broadcast transmission. It discusses, in some detail, MPEG-2 audio and video compression and the MPEG-2 system layer. It also provides an overview of the DVB and ATSC standards, which are extensions of MPEG-2. In addition to outlining the standards, it discusses the need for test and verification in the digital broadcast environment, provides several test scenarios, and discusses why and how to test MPEG-2 transport streams in order to ensure the highest-quality transmission now and in the future.

The pocket guide ends with a Glossary of Terms, a Reference Material List and an Index. Because this guide is not exhaustive, we encourage you to consult the Reference Material list for continued study.

2 MPEG History

In 1987 the International Electrotechnical Commission (IEC) and the International Organisation for Standardisation (ISO) joined forces to create JTC 1 (Joint Technical Committee 1). The JTC's mission was to coordinate international standardization for Information Technology (IT). To handle this task, the Committee split its resources into various subcommittees, one of which was Subcommittee 29 (SC 29), formed to investigate the standardization of audiovisual coding. SC 29, entitled "Coding of audio, picture, multimedia, and hypermedia information", was further divided into several working groups. Working Group 11 (WG 11), "Coding of Moving Pictures and Audio," later became known as MPEG, or the Moving Pictures Expert Group.

When the first official MPEG meeting was held on 10 May 1988, digital television broadcasting was no more than a vision. The development of audio CDs had already proven that analog signals could be digitized to produce better quality sound that required less bandwidth than traditional analog storage methods. The implications of digitization stretched as far as television, where the lower bandwidth would make room for more programs, internet services and interactive applications. Still, the compression and transmission of digital sound and video together would require extensive research. For broadcasters to make the transition from analog to digital, they would need entirely new methods of broadcasting, including new technology, new equipment and new international standards.

MPEG consists of a family of standards that specify the coding of video, associated audio and hypermedia. These standards include MPEG-1, MPEG-2 and MPEG-4. Though this guide deals mainly with MPEG-2, the digital broadcasting standard, we will discuss MPEG-1 and MPEG-4 briefly.

For more information about the Moving Pictures Experts Group or the MPEG standards, see http://www.mpeg.org.

2.1 MPEG-1

MPEG-1 is the original MPEG standard for audio and video coding. First published in 1993, this standard defines digital audio and video compression for storage at up to approximately 1.5 Mbps. Like all MPEG standards, MPEG-1 is flexible in that it defines only the syntax and semantics of the encoded bit stream, and not the encoding process itself. Common applications for MPEG-1 include the storage of sound and pictures for interactive CDs such as video games and movie CDs. MPEG-1 has also been used for digital radio broadcasts.

Soon after work on MPEG-1 began, champions of the "digital television" concept realized that MPEG-1's syntax and structure would not support the complexity and versatility of digital TV. Hence, in 1990, work began on MPEG-2, the standard that would make digital television broadcasting a reality. We will discuss MPEG-2 in greater detail in the next chapter.

2.2 MPEG-4

MPEG-4 represents the next breakthrough in audiovisual coding. It allows for simultaneous coding of synthetic and natural objects and sound, giving service providers more options for creating games and other multimedia applications. It extends interactive possibilities by allowing the user to manipulate, among other things, views and the viewing perspective.

MPEG-4 also allows for the coding of arbitrarily shaped objects, instead of the standard rectangle-shaped video frame. Because of this, the standard allows even greater compression than MPEG-1 or MPEG-2 and will be used for applications with especially limited data capacity. Though digital broadcast will continue to use the MPEG-2 standard, MPEG-4 will serve a variety of applications including networked video applications, computer games and wireless services.

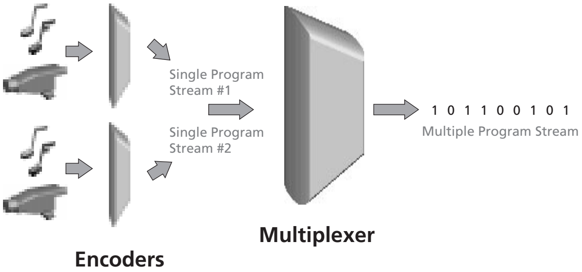
3 MPEG-2 Compression and Transport

MPEG-2 refers to the set of standards developed by the Moving Pictures Experts Group for audio and video compression, transportation of multiple compressed programs in a single multiplex, and encapsulation of data into the multiplex. It defines a mechanism to compress a video signal to as much as 30 times less than its original size.

Because of MPEG compression, broadcasters no longer need as much bandwidth to send a single TV program. This gives them the option to include other programs and services in their commercial offering. These programs and services might include high definition television (HDTV), more channels to choose from, programming in different languages, internet services and interactive TV.

The MPEG-2 standard defines the structure and syntax of the encoded bit stream but does not specify the method for encoding the bit stream. This flexibility gives manufacturers and content providers room to develop improved solutions for specific applications with the knowledge that their equipment or content is based on an international standard that is unlikely to change. The goal of MPEG-2 video and audio compression is to reduce redundancy without affecting quality. However, compression involves compromise; therefore, it does not necessarily produce an original-quality sound or image. The more a signal is compressed, the lower the quality of the resulting signal. To offer broadcasters and consumers flexibility to choose between high compression (more available services—lower quality signal) and high quality, MPEG specifies different profiles and levels, each offering a different degree of compression vs. quality.

The following figure shows a simplified example of how MPEG-2 transport streams are created from audio and video signals.



3.1 Video Compression

Video compression depends on (1) redundancy within and between pictures or frames and (2) the characteristics of the human visual system. Two types of compression, spatial encoding and temporal encoding, allow encoders to reduce redundant data significantly, thereby greatly decreasing the bandwidth required to transmit a video stream. Video compression also makes use of the eye's inability to detect certain visual degradations including noise in a "busy" picture area and reduced color spatial frequency in a picture.

3.1.1 Spatial Encoding

Spatial redundancy deals with similarities between adjacent pixels in plain areas of a picture. For instance, a picture that contains a blue-sky background will likely contain several rows of identical blue pixels. Spatial encoding codes only one of these pixels, significantly reducing redundancy in the bit stream. This type of encoding involves a series of steps including Discrete Cosine Transform (DCT), weighting, scanning and entropy coding.

The first step in spatial encoding, Discrete Cosine Transform (DCT), requires that the video waveform be transformed into a matrix of spatial frequencies. This matrix includes blocks of 8x8 pixels which, when transformed, provide a series of coefficients that indicate the prevalence of a given frequency in each pixel. Since the transform requires multiplication by fractions, the resulting coefficients often have a longer word length (or number of bits needed to express a value) than the pixel values themselves. Hence, DCT itself does not compress the picture block; rather, it actually expands it. However, because of spatial redundancy, many coefficients end up with zero or near-zero values, which means that the corresponding pixels can be dropped from transmission as the differences are not visible to the viewer. This allows for considerable compression with minimal degradation in quality.

If even greater compression is needed, then the word length of the remaining coefficients must be expressed in fewer bits. Reducing the word length reduces the accuracy of these coefficients and introduces degradation into the picture. With the process of weighting, degradation in a picture can be strategically placed where the viewer is least likely to notice it.

Weighting takes advantage of the fact that more noise can be tolerated at higher spatial frequencies. In other words, where the images on the screen are more detailed or complex, imperfections in the video transmission are less noticeable to the viewer. Weighting re-quantizes DCT coefficients based on their perceptual importance to the viewer. This process ensures that coefficients with the highest frequency are accompanied by the greatest amount of noise. In this way, noise is most likely to occur where it is least likely to be perceived by the viewer.

After quantization and weighting, the matrix is reorganized with the most significant DCT coefficients placed in the top-left corner of the matrix; coefficients with insignificant values are changed to zero. Scanning makes use of this new organization to reorder coefficients so they will be sent in order of importance. In this way, the most significant coefficients are sent first, followed by an indication within the code that the remaining coefficients are all zero.

The final step in spatial coding deals with entropy coding, which resizes coefficients based on the number of times they occur. Frequently repeated coefficients are expressed in the smallest word length, thus greatly decreasing the total amount of bandwidth used to transmit a single frame.

3.1.2 Temporal Encoding

Temporal encoding eliminates redundancy between individual frames in the video stream. This can be accomplished through motion estimation and inter-frame prediction. Imagine, for instance, that you are encoding video that shows the bird's-eye view of a soccer game. Though the players move from frame to frame, the background scenery - the field itself - doesn't change. Temporal coding takes advantage of the similarities between sequential frames and encodes only the differences from one frame to the next. Two types of temporal coding are Inter-frame Prediction and Motion Prediction, which are discussed below.

❑ **Inter-Frame Prediction**

Inter-frame prediction uses one complete frame, called an Intra-coded frame (I-frame), as a basis from which to reproduce other frames in the video stream. Redundant information is transmitted only once, allowing for major bit rate reduction.

I-frames represent only one of three types of frames used in Interframe prediction. Predicted frames, or P-frames, are predicted from either a previous I-frame or a previous P-frame. Instead of transmitting all the transform coefficients for a P-frame, the encoder transmits only those coefficients that differ from the preceding I- or P-frame. At the decoder, P-frames are re-created using the preceding I- or P-frame as a reference, and applying the differentials. B-frames are bi-directionally predicted in the same fashion from either preceding or subsequent I- or P-frames. Where a P-frame generally requires 1/2 of the data of an I-frame, a B-frame requires only 1/4.

Of course, using only one I-frame as a basis for creating all other frames in a video stream would leave the stream extremely vulnerable to error, since an error in the I-frame would propagate indefinitely. For this reason, frames are divided into Groups of Pictures (GOPs), usually 12-15 pictures long. GOPs always begin and end with a complete I-frame, providing for rapid error correction when the preceding I-frame has become corrupted. GOPs also contain P-frames and B-frames. Below is one example of a Group of Pictures.

Encoding/ Presentation sequence $I_1 \rightarrow B_1 \rightarrow B_2 \rightarrow P_1 \rightarrow B_3 \rightarrow B_4 \rightarrow P_2 \rightarrow B_5 \rightarrow B_6 \rightarrow I_2$

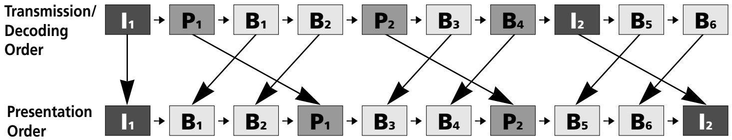
❑ **Motion prediction**

Though objects may change location on the screen, their appearance oftentimes remains the same. Motion Prediction takes advantage of this similarity by measuring an object's motion at the encoder. This measurement is sent as a motion vector to the decoder. The decoder then uses this vector to shift the specified image from its location in the previous frame to a new location in the next frame. Typically, motion continues across several frames, so even greater compression can be attained when vectors are transmitted differentially. For instance, if an object's speed is constant, the motion vectors do not change; only a vector differential of zero is transmitted.

3.1.3 Decoding

The decoding of MPEG-2 audio and video streams reverses the encoding process one for one. An inverse DCT process restores frequency coefficients according to the accuracy of the encoder. The decoder then uses transmitted macroblocks from I- and P-frames to replace redundant macroblocks discarded from P- and B-frames during encoding. Motion vectors specify the location of these macroblocks within the predicted frames.

As explained above, in order to decode a B-frame, data from both the previous and next pictures must be available in the decoder. Therefore, inter-frame prediction requires that frames be sent out of sequence and stored temporarily in the decoder. Consider the order in which the frames in the above sequence must be decoded before they can be presented to the viewer:



Decoding Time Stamps (DTS) and Presentation Time Stamps (PTS) within the header of each frame ensure that the frames are decoded on time and presented in the proper order. For more information about time stamps, see section 3.4 Timing: PCR, PTS and DTS.

3.2 Audio Compression

Similar to the compression of video, MPEG audio compression also capitalizes on the characteristics of a human sensatory organ-this time, it's the human ear. The compression takes into account both auditory masking, where a louder sound will hide a softer sound, and time uncertainty, where a sound in the past or future will interfere with the ear's ability to hear a current sound. One example of auditory masking occurs when you try to carry on a quiet conversation in a train station. Passing trains drown out your conversation each time they speed by. In the presence of the sound generated by the train, the quiet voices in the conversation become imperceptible.

Auditory masking is most prevalent among sounds with similar frequencies. Its effect depends on the frequency separation between a loud sound and the quieter sound being masked. The closer two signals are in frequency, the more likely it is that one sound will drown out the other, though it may be only slightly louder. For example, if two horns are playing at two similar high frequencies, the quieter horn cannot be heard, but a bass drum playing at the same sound level as the quieter horn will likely be heard, since its frequency is significantly different from that of the louder horn. Since the sensitivity of the ear is frequency dependent, the effect of masking is also frequency dependent. Sounds at lower frequencies must be even closer together in order to be masked than sounds at higher frequencies. The dynamic range is also frequency dependent, and the area of largest dynamic range is the range of normal speech.

In order for sounds to mask each other, they need to occur at the same time or nearly the same time. In fact, there is a range of time several milliseconds long before and after a loud masking sound during which its masking effects will still be present.

In order to capitalize on these auditory characteristics, the audio compression algorithms break the audio spectrum into many subbands. The dynamic range in each subband is reduced separately such that the effects of a dynamic range's compression are not noticeable. This means that instead of 16 bits per audio sample in each subband, there might only be 2-4 bits per sample. A scaling constant for each band is also used. The allocation of bits per subband is divided such that the important frequency ranges receive more weighting. The size of a subband also varies by frequency in order to match the masking by frequency in the human ear.

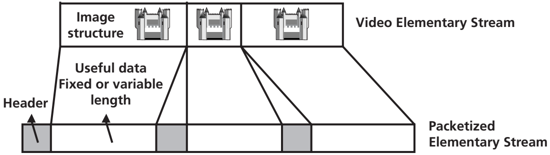
An audio signal is compressed in blocks such that the allocation of frequency information can be changed over time, and time masking can be used effectively. The typical size of an audio block is 24 milliseconds.

3.2.1 AC-3

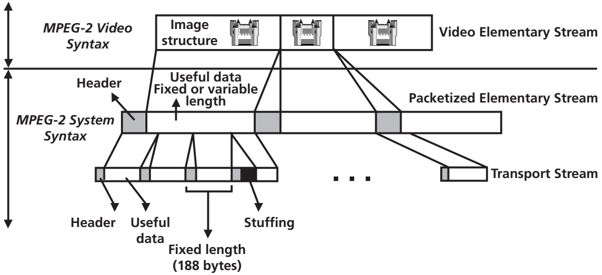
The Dolby AC-3 audio compression algorithms use the same humanear characteristics described above, but the methods they use to divide the frequencies and process the data are different than the ones used by MPEG. The AC-3 algorithm also uses audio blocks of data.

3.3 From ES to PES and from PES to Transport Stream

Once an audio or video elementary stream has been compressed, it is divided into variable-length packets and converted into a Packetized Elementary Stream (PES). Video stream packets include one frame each, while audio stream packets usually include approximately 24 milliseconds of sound each. Each packet contains a header and a payload. The header gives timing information so the decoder will know at what time to decode and present the specified frame.



From here, PESs are further divided into transport packets of 188 bytes each and multiplexed with other elementary streams and data to form a transport stream containing audio and video for multiple programs. The transport stream also contains system information in the form of PSI/SI/PSIP tables, which we will discuss later. It may also contain data for interactive applications. The use of fixed-size transport packets simplifies the multiplexing and transmission processes along the broadcast chain.



Each packet in the transport stream, whether it contains audio, video, system information, or data, is identified by a number called a PID, or Packet Identifier. PIDs help the decoder find and sort information in the transport stream. This will become more apparent when we discuss MPEG-2's system layer and the use of tables. These tables, called PSI, SI, or PSIP tables, enable the decoder to locate the various programs and their components in a multiple-program stream. They also provide Electronic Program Guide (EPG) information. The tables are organized into a hierarchical tree structure.

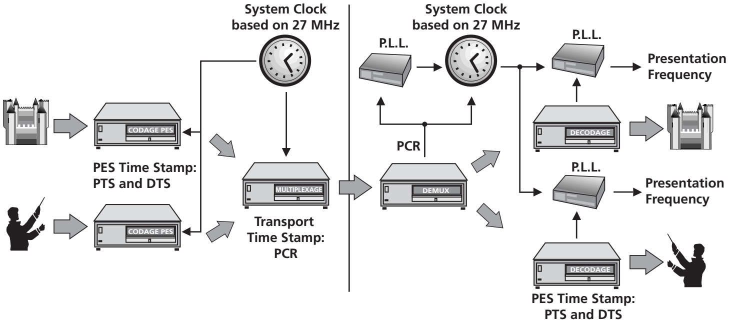
3.4 Timing: PCR, PTS and DTS

End-to-end synchronization, as well as synchronization between components in a transport stream, is based on the system clock of the encoder. The encoder uses this clock to periodically send timing information in the transport stream with a Program Clock Reference (PCR). Elementary stream packets contain time stamps called Presentation Time Stamps (PTSs) and Decoding Time Stamps (DTSs).

Since frames are not always transmitted in the order in which they appear on the screen, the PTS allows the decoder to determine, based on the PCR, the time at which a frame must be presented to the viewer. As we mentioned earlier, MPEG-2 compression creates dependencies between frames.

For instance, a B-frame cannot be decoded without referencing the previous and next I or P frames. Because of this dependency, some frames may need to be decoded well before their presentation time. To manage this condition, a DTS in the header of each PES packet notifies the decoder of the time at which a frame must be decoded. If the DTS precedes the PTS for a specific frame, the frame is decoded and held in a buffer until its presentation time arrives.

The following figure shows the timing sequence in the transport stream. At the encoder, a stamp of the system clock is put into the PES with a PTS and/or DTS. Samples of the system clock are added to the transport stream via PCRs. On the decoder side, the PCRs are put through a Phase Lock Loop (PLL) to recover the original system clock. This ensures that the decoder is synchronized to the encoder so that data buffers in the decoder do not overflow or underflow. Once the original system clock is recovered, programs are then decoded and presented as specified by the PTS and DTS time stamps for each individual frame.



3.5 PSI Tables

MPEG-2 Program Specific Information (PSI) tables provide the data required for a decoder to demultiplex a program from a transport stream for presentation to a user. This information can include location of audio and video for a certain program, access rights, and information regarding the compression and characteristics of audio and video signals. The tables are repeated periodically (for example, 10 times/second) in the transport stream to support random access required by a decoder turning on or switching channels.

3.5.1 PAT

Think of the Program Association Table (PAT) as a root directory for the transport stream. This table lists all the programs in the stream and provides the PID (Packet Identifier) value for the Program Map Table (PMT) associated with each program. The PAT can also contain the PID number for the Network Information Table (NIT), which provides access to other transport streams in the network. The decoder is always able to find the PAT (if present in the stream) since it is always in packets with a PID value of 00.

PAT (PID 00)

Prog. 1	PMT PID 65
Prog. 2	PMT PID 32
Prog. 3	PMT PID 56
Prog. 4	PMT PID 120
NIT	PID 16

3.5.2 CAT

The Conditional Access Table (CAT) provides a mechanism for decoders to find Entitlement Management Messages (EMMs) in the transport stream. EMMs update the subscription options or pay-per-view rights for each subscriber or for groups of subscribers. The CAT lists the EMMs in the transport stream and gives the associated PID values.

CAT (PID 01)

EMM A	PID 61
EMM B	PID 76
EMM C	PID 38
EMM D	PID 109

3.5.3 PMT

The Program Map Table (PMT) provides a description for each program in a transport stream. This table literally acts as the map for a program, listing the PID values for its video, audio, clock reference and data components. With this information, the decoder can locate the audio and video for the program and display them synchronously.

PMT Program 1 (PID 65)

Components of Program 1	
Video	PID 131
Audio English	PID 132
Audio German	PID 133

3.5.4 NIT

The Network Information Table (NIT) provides information regarding other transport streams in the network. This table is not defined in MPEG-2, but DVB and ATSC have either defined it or identified another method for providing information similar to it.

3.6 Summary

PAT (PID 00)

PMT Program #1 (PID 65)

Video Program #1 (PID 131)

Audio German Program #1 (PID 132)

Audio English Program #1 (PID 133)

...

PMT Program #N (PID XY)

Video Program #N (PID YZ)

Audio German Program #N (PID ZZ)

Audio French Program #N (PID XY)

Audio Italian Program #N (PID XZ)

The following steps outline the process followed by a decoder to display a certain program, in this case, Program 1:

1. Locate the PAT (always found on PID 00) and use it to find the PMT information for Program 1.

2. Use the PMT for Program 1 (defined as PID 65 in the PAT) to locate the PIDs for the audio, video and clock reference in Program1. The PID for the video stream often carries the PCR.

3. Find the packets that contain the video (PID 131) for Program 1.

4. Find the packets that contain the audio for Program 1. If the user has selected the sound track in German, locate the audio track on PID 132. If the user has requested the English sound track, locate the audio on PID 133.

5. Use the PTS and DTS in the header of audio and video packets to decode and present them at the proper time.

4 Digital Video Broadcasting (DVB)

4.1 DVB History

The DVB Project began in September 1993 when public and private television organizations from across Europe signed an agreement to work together for the creation of a digital broadcasting standard that would bring digital television to the home. Because the DVB Project united major players in the European broadcast market, it provided a forum through which a truly unified digital television system could be created. In time, the organization developed international standards for satellite, cable and terrestrial transport.

The Project now includes over 220 participants in more than 30 nations worldwide. Because the DVB standard applies to all types of transmission links - satellite, cable and terrestrial - it eliminates redundancy in research and design, reducing costs to manufacturers. Because the standard is employed throughout Europe and in many countries across the world, digital transmission to all countries under the DVB umbrella need only be tested to fit DVB parameters. Even for terrestrial broadcast, where information can be transmitted in multiple ways by different service providers, the DVB standard limits the variations in transmission from one provider to the next. This minimizes testing costs for network operators, system integrators, service providers and broadcasters.

Though this pocket guide focuses mainly on the System Information specified by DVB, the standard also addresses other parts of digital transmission, such as transmission mechanisms and data services. For more information on these aspects of the DVB standard, see http://www.dvb.org.

4.1.1 DVB System Information (SI) Tables

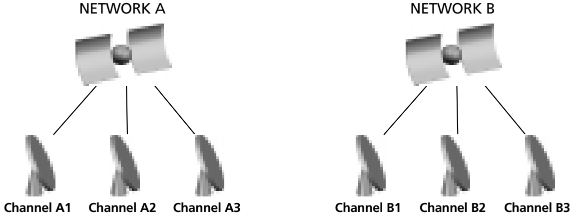
While MPEG-2 PSI tables organize the transmission of compressed audio and video in a single transport stream, they are not designed to provide information for the large number of programs and services available on a network of multiple transport streams. DVB Service Information (SI) tables give service providers the tools they need to offer programs and services across a large network that may include many transport streams. SI tables function much like the Table of Contents in a multi-chapter book. They enable receivers and set-top boxes to access information anywhere in the MPEG/DVB network, or bouquet of multiplexes, by providing information for events and services available on all transport streams in the system. SI tables also provide information for the Electronic Program Guide (EPG), which supplies viewers with a list of all the programs and services available, along with their duration and a description of their content.

SI tables have a specific structure similar to a file-management tree structure on a PC with a root directory and subdirectories. They use specific PID values for transmission and identification by any receiver. There are four mandatory DVB SI tables: the Time and Date Table (TDT), the Network Information Table (NIT), the Service Description Table (SDT), and the Event Information Table (EIT).

- ❑ **Time and Date Table (TDT)** — This table provides the present UTC date and time, which can be adjusted according to time zone and presented on screen for the viewer.

- ❑ **Network Information Table (NIT)** — A Network is a system of one or more transport streams controlled by a single content provider. The Network Information Table (NIT) contains information regarding the physical organization of the transport streams carried on a single network, including the tuning parameters that enable the receiver to change channels at the viewer's command with little or no delay. Tuning parameters are specific to the type of transmission assigned to the network - terrestrial, cable or satellite. This table also includes characteristics about the network itself.

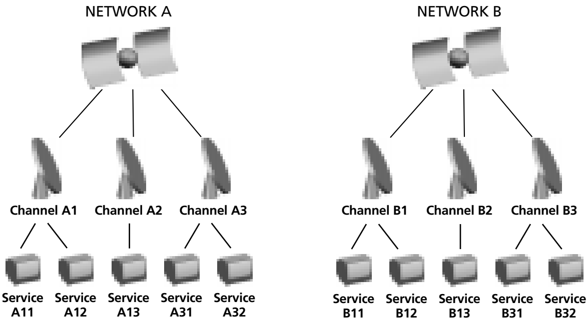
The NIT defines tuning parameters for the channels in the network.



- **Service Description Table (SDT)**

A Service is a sequence of programs under the control of a broadcaster, which can be broadcast as part of a schedule. The Service Description Table (SDT) provides information about the services available in the network, including the name of the service provider and the textual description of the channel. Two types of SDTs, "Actual" and "Other", are required by DVB. The SDT Actual describes the services available on the transport stream currently being accessed by the viewer, while the SDT Other describes services available on other transport streams in the network. This information is used in the Electronic Program Guide (EPG) to provide the viewer access to all the services available on the network.

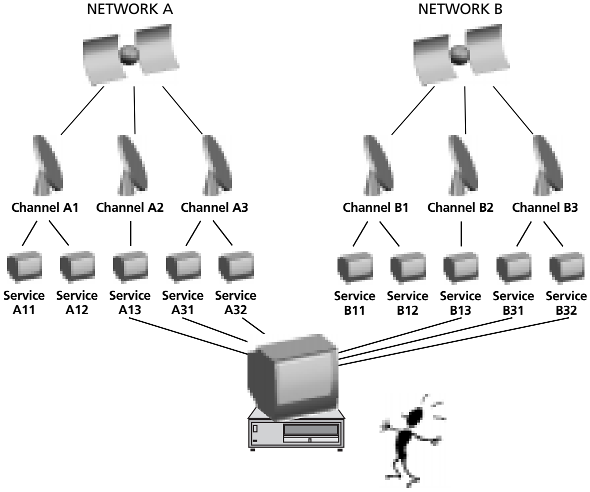
The SDT provides information about the services in the network.



- ❑ **Event Information Table (EIT)**

An event is a collection of elementary streams with a common time base set to start and end at the same time. (We often refer to events as “TV programs”.) The Event Information Table (EIT) provides information about events in the network, such as their start time and duration. Three different types of EITs can be transmitted at the same time: EIT Present and Following and the EIT Schedule. The EIT Present describes the events currently being broadcast on the transport stream accessed by the viewer, while the EIT Following provides information about the next events to be broadcast on that transport stream. The EIT Schedule lists events available on the network for anywhere from the next few hours to the next few days, depending on the service provider's implementation.

The EIT is used to create the EPG, which gives the viewer access to all the events available on the network.



- ❑ **Optional DVB Tables**

Optional DVB tables include:

Bouquet Association Table (BAT)—A Bouquet is a commercial offering, or a group of services that can be purchased as a single product. The Bouquet Association Table (BAT) describes the services available in a given bouquet.

Running Status Table (RST)—This table carries information used to update the timing status of events in the system when scheduling changes occur. This saves broadcasters from having to retransmit an entire table when only a portion of the content changes.

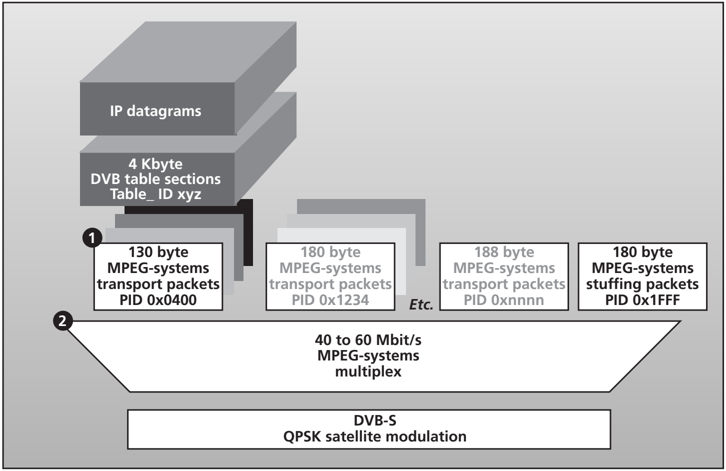
Timing Offset Table (TOT)—This table contains the UTC time and date, along with the local time offset.

Stuffing Table (ST)—The Stuffing Table invalidates the remaining sections in a table when one section has been overwritten. This maintains the integrity of the section_number field.

4.2 DVB and Internet Protocol (IP)

DVB's recently developed Data Broadcasting specification allows for high-speed data transfer via satellite, cable and terrestrial television channels. Some potential applications for data broadcasting include data-casting, downloading software, providing Internet services over broadcast channels, and interactive TV. Data casting and surfing the web via satellite offer consumers internet services at much greater speeds than the typical telephone modem can offer. Where the transmission of a 10MB file can take 30–45 minutes via telephone modem, the same file can be downloaded via satellite to a high-performance system in less than 20 seconds.

There are different implementations for data-broadcast over DVB. The following figure summarizes the operation for multi-protocol encapsulation (MPE) of IP as defined in the DVB standard, which uses table section encapsulation into transport packets from MPEG-2.


IP datagrams
4 Kbyte
DVB table sections
Table_ ID xyz
1
130 byte
MPEG-systems
transport packets
PID 0x0400
180 byte
MPEG-systems
transport packets
PID 0x1234
Etc.
188 byte
MPEG-systems
transport packets
PID 0xnnnn
180 byte
MPEG-systems
stuffing packets
PID 0x1FFF
2
40 to 60 Mbit/s
MPEG-systems
multiplex
DVB-S
QPSK satellite modulation

- **Fragmentation into MPEG Transport Packets**

  Step ❶ benefits from a predefined data-fragmentation mechanism. Multi-kilobyte pieces of data, such as IP datagrams in DVB tables, are fragmented in fixed-size (188-byte) MPEG transport packets with minimal overhead. A series of transport packets for a given data stream is identified by a user-defined PID (Packet ID) value, such as PID 0x0400, as seen in the example. Notice that each packet will automatically benefit from a standard Reed Solomon forward-error correction (FEC) mechanism later during the digital modulation.

- **Multiplexing the various Data-Streams**

  Step ❷ in the process is the multiplexing of the transport packets from multiple data streams—possibly along with video and audio streams for digital TV—in a single MPEG transport stream. A transport stream is a flagged bit stream with a fixed bit-rate corresponding to the performance of the digital modulation through the satellite transponder. Null packets are used for stuffing as the total bandwidth is rarely used for Quality of Service (QoS) reasons.

5 Advanced Television Systems Committee (ATSC)

5.1 ATSC History

In 1987, the U.S. Federal Communications Commission (FCC) appointed a special Advisory Committee as counsel regarding the technical and political issues of Advanced Television (ATV). The formation of this committee sparked a fierce competition among broadcast industry leaders who hoped to propose to the Advisory Committee a system that would be accepted as the nation's standard. Twenty-three proposals were originally brought before the Advisory Committee, but by 1993, only four remained. The Committee tested these remaining systems extensively and found deficiencies in each. In order to eliminate these deficiencies and take advantage of the strengths found in each system, the Advisory Committee encouraged the remaining competitors to form a consortium by which they could work together to create a U.S. standard for ATV broadcasting.

In response to this request, the remaining companies formed the HDTV Grand Alliance in May of 1993. Within the Alliance, they worked together to build a final prototype system based on specifications approved by the Committee. The Committee called upon the Advanced Television Systems Committee (ATSC) to develop and document the detailed specifications of the new ATV standard based on the Grand Alliance system. ATSC Digital Television Standard (A/53), the document produced by the ATSC membership, was accepted in 1996 by the FCC for digital terrestrial television broadcast in the U.S. Though ATSC was initially a North American organization, its charter has been modified to include members from other countries. It now serves more than 200 members in several nations worldwide.

5.1.1 ATSC Program and System Information Protocol (PSIP) Tables

Like DVB SI tables, ATSC PSIP tables act as extensions to the MPEG-2 system layer, allowing broadcasters to make a larger number of products and services available to the viewer. Unlike DVB, however, ATSC was created mainly for terrestrial broadcast where services are made available through local TV stations that offer a limited number of programs. Up to this point in time, the ATSC standard has been used mainly for terrestrial applications, though it also provides parameters for Cable TV (CATV) transmission.

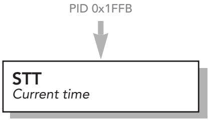
PSIP tables are organized in a hierarchy similar to their DVB counterparts. They are identified by specific PID values and can thus be identified by any ATSC receiver. PSIP tables include: the Master Guide Table (MGT), the Virtual Channel Table (CVCT for cable, TVCT for terrestrial), the System Time Table (STT), the Rating Region Table (RRT), the Event Information Table (EIT), and the Extended Text Table (ETT).

These tables are organized into three types:

- Base tables, which are always found on PID 0x1FFB. These include the MGT, the VCT, the STT and the RRT;
- EITs whose PID values are defined in the MGT;
- ETTs whose PID values are also defined in the MGT.

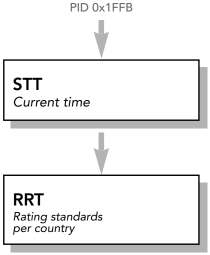
❑ System Time Table (STT)

The System Time Table consists of only one packet that serves as a reference for the current time of day. This information enables the receiver to start advertised events on schedule.



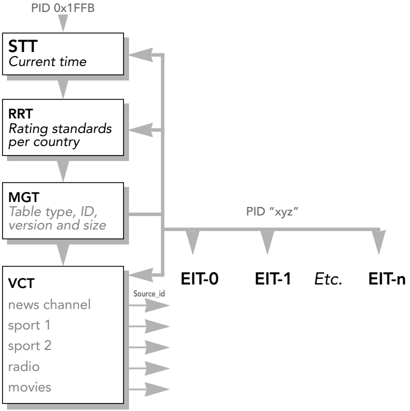
- ❑ **Rating Region Table (RRT)**

This table transmits program-rating systems for each country that uses a rating standard. The information in this table allows viewers to filter certain programs based on their content.

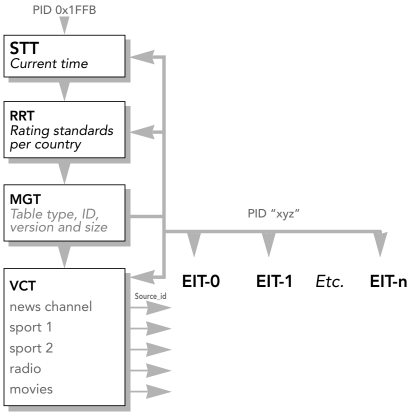


❑ **Master Guide Table (MGT)**

The Master Guide Table acts as an index for all other tables in the PSIP standard. It defines table sizes (necessary for proper decoding), version numbers (which help to identify the tables that need to be updated), and PID values (which enable the decoder to locate the EITs and ETTs in the system).

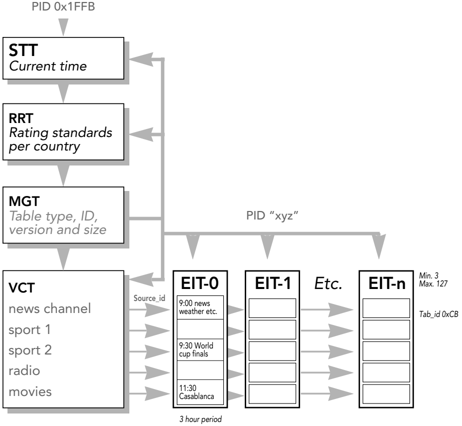


- **Virtual Channel Table (TVCT - terrestrial, CVCT - cable)** This table lists all the channels in the transport stream and defines their characteristics. It includes information such as channel name, stream components and types, and navigation identifiers. It identifies a source_id for each program, which the EIT uses to locate and display information for the Electronic Program Guide.



❑ **Event Information Table (EIT)**

The Event Information Table defines the events (TV programs) associated with each of the virtual channels listed in the VCT. Information contained in this table might include an event's start time and duration. According to the PSIP specification, between 4 and 128 EITs must be in the transport stream at any given time. Each EIT provides event information for a three-hour time period, so up to 16 days of programming can be advertised in advance. EIT-0 always contains information for the current 3-hour time block, while EIT-1 defines programming for the next 3 hours.



- **Extended Text Table (ETT)** Extended Text Tables carry text messages describing both channels and events; hence, there are two types of ETTs: Channel ETTs and Event ETTs. ETTs give the viewer more detailed information than is available in the EIT. For example, Channel ETTs may contain information about the price of a channel or its coming attractions. Event ETTs might include a short paragraph describing a specific event, such as a movie. As with EITs, the PID number for ETTs is also identified in the MGT. ETTs are optional.

6 Transport Stream Analysis/Simulation: Why, How and What to Test?

6.1 Why Should I Test?

So, why should you test? The answer should be simple enough, right? You test to make sure that the quality of your products will withstand the rigors of the real-world environment. It seems simple enough, but because testing can be costly and time consuming, validation teams often limit the scope of their testing to include only those parameters and events that are most likely to occur with their systems. Naturally, these test scenarios are the easiest to create. But the constantly changing digital broadcast environment forces all players to be much more broad-minded in their testing procedures. They must test the limits of their equipment based on all possible permutations of the MPEG-2, DVB or ATSC standards. They must also ensure their systems' interoperability with other manufacturers' equipment and have confidence that their products will resolve error conditions successfully. Rigorous testing under both standard and extreme conditions allows companies to keep pace with the market and consistently outperform the competition.

Testing to this degree allows companies to:

- Ascertain that their equipment is robust enough to handle broader use of the specified standards as the industry evolves.
- Avoid expensive recall and customer service issues when equipment fails to meet protocol standards.
- Ensure that their equipment can be trusted to work effectively in integrated systems with equipment from several different manufacturers.
- Earn and keep their customers' respect by supplying equipment that continues to perform in spite of error conditions.
- Maintain their reputation for quality in the industry.

- Improve their processes to ensure that their equipment outperforms the competition.

The following sections use the creation and implementation of the Electronic Program Guide as one example of why testing under extreme conditions is absolutely necessary in the digital broadcast environment. Though the examples deal mainly with the EPG, they can be applied to many test scenarios.

6.1.1 Testing the Limits of the Protocol

Because digital-broadcasting technology is relatively new, current transmissions don't even approach many of the limits specified by the protocol—but this may soon change as the industry evolves rapidly to support a growing number of available services and potential customers. In order to develop equipment that will be compliant in a year or more, manufacturers must test their products according to all the boundaries and limits of the specification. Take for example the creation and implementation of software for the Electronic Program Guide (EPG).

Since it is not easy to upgrade the software in digital televisions within the home, developers need to be confident that their EPG software will provide top-quality service both now and several years from now, as more advanced features of SI/PSIP are implemented in broadcast

SI/PSIP generators. In order to have high confidence in their EPG software, developers must test the software with transport streams that include all permutations of the different DVB and ATSC tables that make up the EPG. This also extends to the many descriptors defined by the standards. These may require support for decompression of text strings, handling of text from many different character formats, and reading the ratings values for each program. Because the tables are linked, the permutations between tables must also be tested. Current real-world streams are not complex enough to offer sufficiently extreme test inputs, so many developers use emulation devices that allow them to simulate transport streams with a maximum number of permutations. For more information on test emulators, see section 6.2.1 Emulation or Simulation.

6.1.2 Interoperability

Because MPEG-2, DVB and ATSC are open systems, they specify only the syntax of the output transport stream, not the manner in which the stream is generated. This means that different manufacturers often have different methods of encoding, modulating, or multiplexing the input, while still remaining within the specification. Because complex broadcasting systems often use equipment provided by different manufacturers, these manufacturers must be sure their systems can properly transmit, receive and interpret data from many different brands of equipment.

Testing for interoperability becomes especially critical when it comes to the Electronic Program Guide (EPG). In the ATSC world, information for the PSIP data is usually created by each local television station from information it receives from the national network feed or from local programming. A given station might use any brand of PSIP generator, and every home viewer may select from a number of brand names of ATSC-compliant digital televisions. Because of this, digital television manufacturers must be sure their systems can properly receive and interpret PSIP data from many different manufacturers' PSIP generators. This makes the task of developing the EPG more difficult for an open system since the digital television must handle all possibilities within the A/65 protocol. In order to test for interoperability, the developer of the EPG for a digital television must either have all PSIP generators available for testing or have a piece of test equipment that can emulate all the possible scenarios within the A/65 specification.

6.1.3 Handling Errors

Because digital transmission is especially vulnerable to errors, developers and manufacturers must produce equipment that effectively handles dropped packets or erroneous bits without crashing. In the case of the EPG, developers need to make sure that their software can not only parse all possible variations to the specification, but also resolve any error conditions that may be present in a stream. This means making sure that the software does not crash when an error occurs, thereby causing interruption in the video or audio or forcing the user to turn the power off and on to recycle the system.

One common example of such an issue arises for ATSC systems when an error occurs in the MGT. The MGT includes the length of the PSIP tables being transmitted. However, if the value in the MGT does not match the true length of a table, and the EPG software uses the length value from the MGT, the table referred to in the MGT may not fit in the memory allocated. If this occurs, the software might either discard it or overwrite other critical memory. Either of these results could cause the software to crash or provide erroneous information to the user. A reasonable EPG software package would detect this problem and work around it, causing minimal impact to the EPG display. In order to avoid this and other similar problems, the software developer must test the digital television with as many erroneous transport streams as possible. Again, a test PSIP simulator would allow you to easily simulate this and many other error conditions.

6.2 How Do I Test? What Equipment Do I Need?

Testing relies on two complementary functions: emulation or simulation and analysis. Emulation of a transport stream refers to embedding the audio/video components and data. Analysis of a transport stream involves checking the output transport stream for proper timing, PSI/SI/PSIP, and elementary stream information.

6.2.1 Emulation or Simulation

The test and verification of a digital television or set-top box includes the feeding of many different variations of transport streams into the equipment over time. This is a time-consuming process, and if any changes are made to the hardware or software in the receiver during the process, the testing must be repeated to test the areas affected by the changes. Because the MPEG-2/DVB/ATSC specifications are so complex, the number of different test scenarios required to test a digital television or receiver to a sufficient degree of confidence is quite large. In order to obtain a collection of transport streams for testing, manufacturers generally use one of three methods: live transmissions, captured transport streams stored on disk, or simulated test streams.

Over the air (live) transmissions - Continuous feed is the main advantage to using live transport streams for testing purposes. You can connect a TV or IRD to an input source and let it run for days or weeks without the cost of storing huge amounts of data on a hard drive. The problem with this method, however, is that you cannot determine the conditions of the test stream. And with live feed, these conditions probably do not test the limits of what the equipment will be required to do either now or in the future, according to the specification.

Captured transport streams - In the past, testing has generally been accomplished with transport streams stored on a disk and played out, repeating once they ended in order to mimic continuous feed. Unlike over-the-air transmissions, broadcasters using captured transport streams can select and store any number of test streams, each representing a different test scenario.

These captured streams can then be run through a transport stream generator or other emulation device that sends the stream as input to the unit under test. In spite of the possible variety available with this method, however, the storage space required to keep these streams severely limits the number of test scenarios that most manufacturers store and use. For example, reasonable test coverage would require 10s to 100s of transport streams, each one occupying 100s of megabytes or more of disk space. This database of streams is not only difficult to obtain, but also time-consuming to manage.

Further, most captured streams do not test the limits of the specified standards for items such as output rate, table size and repetition rate. For example, U.S. terrestrial markets only transmit at 19.3 Mbps, so most ATSC streams will be transmitted at this rate. But the future may quite possibly see ATSC streams broadcast at 50 or 100 Mbps, as specified by the standard. Manufacturers must know how their equipment will react to these output rates. Table section lengths provide another example. These days, most table sections are only 100 to 200 bytes long, but according to the ATSC specification, they can be up to 1024 bytes long. Actual transport streams with table sections of this size are difficult to obtain, but manufacturers must test this condition for the future.

Valid test streams should also present the unit under test with changing conditions, such as changes in PID or PCR values. In order to test these scenarios with real-world streams, manufacturers must encode and multiplex a multitude of test streams that represent extreme conditions. This requires expensive video compression equipment and multiplexers. Once the stream has been created and sent as input to the unit under test, a transport stream generator may be used to record erroneous portions of the stream for analysis or testing.

Simulated test streams - The emergence of test multiplexers has largely changed the process for gathering test streams, since these machines can create multiplexes that cover the complete range of parameters specified by the standards. With test multiplexers, the possibility for creating different test parameters is virtually limitless. Manufacturers can easily create test streams with output rates of 100 Mbps and/or table section lengths of 1024 bytes. Simulated test streams can also include error conditions and on-the-fly changes to System Information so they represent near-to-real-life scenarios that test the equipment's reaction to extreme situations. Some test multiplexers even simulate days or weeks of continuous feed by replaying a relatively small piece of audio and video and updating timing and table information continuously. There can be 100s or 1000s of such test streams in the disk space required by just one captured stream. This allows manufacturers to test the robustness and longevity of their systems without requiring the large amounts of disk space necessary for captured streams.

This real-time creation of streams also permits time discontinuities, allowing the emulation of a day passing in just a few minutes. For example, in ATSC, the STT time can increase for 3 minutes and then jump to just under 3 hours later to test the changing of the EIT/ETT information every three hours. These tests can also be run one after another, in an automated fashion, to reduce the time and effort needed to perform the tests. A test multiplexer allows you to create a limited number of test streams that cover the range of conditions for which you want to test.

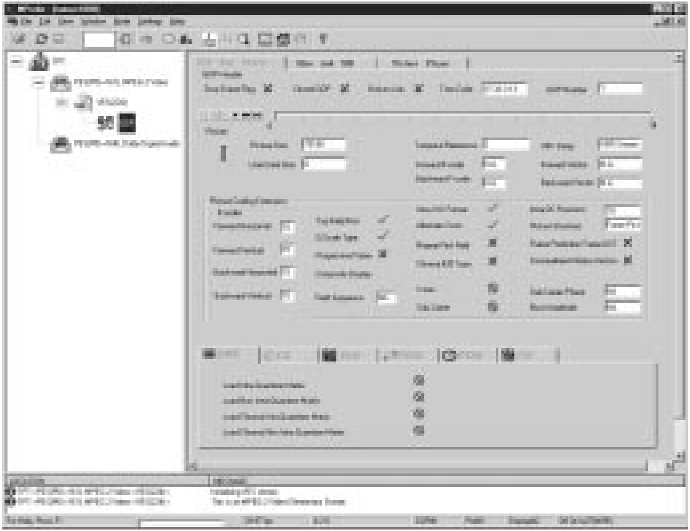
6.2.2 Elementary Stream Analysis

For broadcasters, developers and manufacturers, it often becomes necessary not only to analyze the multiplexed transport stream, but also the various components that make up the stream.

These may include compressed audio and video, data and interactive applications.

Specific test tools have been developed for the validation of audio and video elementary streams. These tools enable verification teams to isolate and troubleshoot errors efficiently by fully validating video and audio data before transmission or distribution.

Elementary stream analyzers can help validate GOP structures, slices and macroblocks.



They can also be used to check the size, quantizer scale, and motion vectors for any macroblock in the video stream.

Elementary stream analyzers also provide buffer models used to validate encoders. The validation against a model ensures that the buffers in decoders and set-top boxes will not overflow or underflow due to encoding or multiplexing issues when receiving the transport stream.

On the other hand, if a stream is not compliant to the buffer model, a decoder is not required to display the video or play the audio to match the original encoded program material, and the decoder could show discontinuous jumping or blocking video or have discontinuities in the audio.

6.2.3 Analysis of a Transmission System

Validating any one of the different parts of the digital transport chain requires not only the rigorous equipment testing discussed above, but also high-to-low-level transport stream analysis and validation. For system integrators, this includes checking and rechecking the output of the various pieces of equipment that make up a system. Transport stream analysis helps system integrators verify that a particular combination of encoders, modulators, multiplexers, and decoders can effectively reproduce the input signal at the required level of quality.

Broadcasters who control large networks of programs and services must be able to constantly monitor the output of their systems for Quality of Service. Essentially all players in the digital transmission chain must verify the output of their systems or services in order to ensure quality performance for their customers. This kind of verification requires some type of transport stream analyzer.

Transport stream analyzers allow testers to view and verify everything from transport stream syntax to system-level information, timing related issues, and audio and video information. Test analyzers can also identify errors in the transport stream and capture parts of the stream for later analysis off line. Manufacturers and system integrators use transport stream analyzers to print reports regarding PIDs, Tables, Timing, Programs and Errors detected on the stream.

Transport stream generators are also used during validation for recording erroneous segments of a transport stream for off-line analysis. This allows users to isolate problems in a system and resolve them without delay. Transport stream generators can also be used in the absence of a test multiplexer to play captured transport streams as input to the unit under test.

6.3 What do I Look For?

Many equipment manufacturers recognize the need to test their equipment and even have the test tools necessary to do so, but are unsure exactly what to look for during testing and validation. Because testing requirements are often application specific, we've divided our answer to this question into a few different test scenarios. Of course, the information provided here is not all inclusive, but we hope this section will give you a better idea of some of the most important issues involved in testing the different elements of the digital broadcast system.

❑ Television or Set-top-box Manufacturer

Manufacturers of receivers, whether they be television sets or set-top boxes, are primarily concerned with two issues: (1) Can my equipment receive and decode MPEG-2/DVB/ATSC transport streams? and (2) Is the output free of errors? Does it match expected video, audio and EPG results?

Compliance issues

Compliance with the standards is essential when building digital broadcast equipment of any kind. Because a single system often uses equipment from different manufacturers, each piece of equipment must be governed by the same protocol—and each must be able to perform up to the limits specified by that protocol. The following questions will help you create a more complete set of testing requirements.

Can the equipment handle the specified boundaries for items such as output rate, table section size and PID number? If a high-definition stream arrives at 45 Mbps, will the equipment process it properly? What about table section sizes of 1024 bytes or PID values of 8190? These and other protocol limits may become commonplace in the future, so you will want to know how your equipment reacts to these conditions.

After you have tested all boundary limits, consider possible changes to the transport stream, both expected and unexpected. For example, what if an audio or video stream changes PIDs? How will that affect your system? What will happen if the PCR value changes unexpectedly? What if extra bytes appear in the stream—will synchronization be lost? For DVB systems, what happens when a new transport stream is added to the network (or to the NIT)? How will that affect your system? This event might be rare, but you want to know what effect it will have on your system when and if it does occur.

The EPG represents the intermediary between service providers and their customers, so its quality is critical. To verify EPG software, you need to make sure the information you input is displayed correctly on the screen (channel name, event description, start time). Can the viewer use the EPG to navigate through the entire network? Does the receiver have all the tuning information it needs to change channels without delay? Can event information be displayed in different languages?

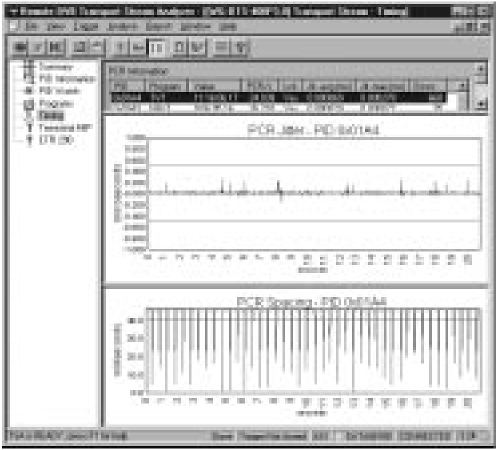
For ATSC systems, you need to ensure that rating filters function properly. Does the receiver handle information about ratings in the EPG? Does it allow the viewer to customize the rating filters?

Error conditions

Of course, when you develop broadcast equipment your main concern is that your equipment produces an error-free transmission stream. Knowing your system's limits can help you avoid unnecessary errors. The main question here is, "What will it take to make the system fail?"

Some questions that might help you understand your systems' limitations include: Bit errors - How does my receiver handle bit errors? What if bits become corrupted? What if they are dropped? What if the bit errors occur in tables? What if they occur in audio or video? How does my receiver handle Forward Error Correction (FEC)?

Timing - How does the unit handle bit errors that affect timing? How does it handle incorrect timing values in general? For instance, what happens when the transport stream indicates discontinuity in the PCR and the time base changes? What happens when the application discontinuity bit (which notifies the system of a PCR change) is lost or corrupted and the time changes unexpectedly? How does my system react to this condition? Does it reset the clock? Does it ignore the discontinuity? How does your system handle PCR jitter? How much can it handle before failure occurs? What are its limits? In these situations, a test analyzer can be especially helpful in determining time discontinuities and measuring jitter.



Tables - What happens when a CRC error occurs? For instance, what if the PMT says the video PID is 50, but there is a bit error in the table? Does the system process the table or discard it?

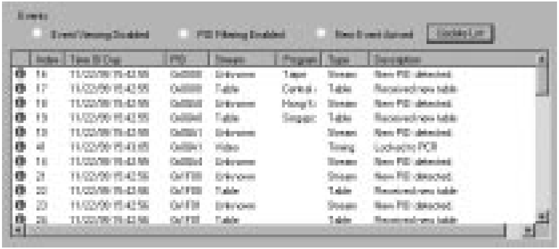
What happens when extra bytes are inserted every now and then? Does the system maintain synchronization?

❑ **Encoder or Multiplexer Joint System Testing**

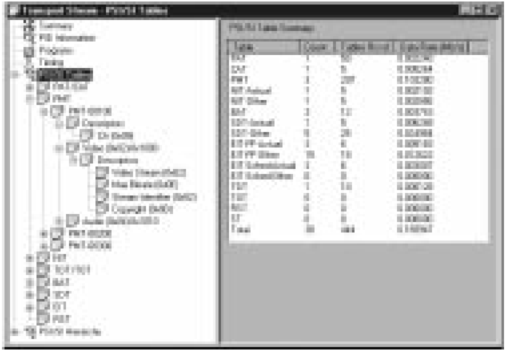
When verifying a joint encoder/multiplexer system, you want to confirm that the system performs according to your expectations. Is it doing what you designed it to do? Other helpful questions include:

Are all the pieces in the transport stream as they should be? For instance, are all the tables present in the stream? Are they on the right PIDs? Do they reference the right PIDs? Are they being sent fast enough? Are the data rates correct? Are the rates for video and audio correct? Are PCR values correct?

Does the transport stream syntax conform to the MPEG-2/DVB/ATSC specifications? For instance, does a sync byte come every 188 bytes? Are the bits ordered properly for table decode? Does the table start with a unit start indicator and include a table offset value and table length field? This is one area where a transport stream analyzer will perform the test for you. For instance, it will check for synchronization and verify that table bits and sequences match the specification. It will also check table structure and descriptor size and look for adaptation field errors and PCR bit errors. Depending on your test analyzer, it may also generate error warnings if these conditions are not met.



You will want to verify that system-level information is consistent. For instance, are all the PIDs referenced in the PMT present in the stream? Does the information in the SI/PSIP tables match the information in the PMT? For DVB systems, are the NIT and SDT Actual in the stream? Is the TSID in the SDT Actual the same as the one referenced by the PAT? Again, a test analyzer can verify this information for you.



You will also want to verify rate- and timing-related issues that measure the statistical parameters of the transport stream. For instance, do tables occur as often as expected? Does video arrive at the expected speed? How many transport errors does the stream contain per second? Some of these parameters are not exactly specified, so they must be monitored and controlled by the broadcast equipment to ensure proper decode and presentation. You can customize some transport stream analyzers to identify unspecified errors that may affect your system.

Check for basic structural consistency within the tables. If a descriptor_length indicates that the descriptor is 10 bytes long, is that value correct? This verifies consistency in the Cyclic Redundancy Check (CRC). Are the raw bits in the tables transformed into a message the user can identify? For instance, is the value in the time field decoded and displayed in a format the user will understand?

If you are writing software that creates tables, look at the tables and descriptors to confirm that the data was produced correctly from the software. For instance, do they produce the correct channel name and event description in the EPG? Is the tuning information correct?

You may also want to test the transport stream to see how data flows on the decoder. Do the PCRs occur with acceptable accuracy? Is the buffer model accurate? For DVB systems, do MIP packets contain the correct information?

In reality, there are a myriad of parameters that must be tested to ensure a system's reliability and longevity in the digital broadcast market. Test equipment can help you by checking rates and consistency to the specification. It can also help you check for consistency within your system's setup.

❑ **System Integration Verification**

In order to integrate complex systems, such as MPEG-2 digital television delivery systems, you need the proper tools to isolate and solve problems quickly and easily in order to reduce integration time. These same tools can be applied to smaller subsets of integration, such as verification of the output of an MPEG-2 compression system. In testing integrated systems, you should consider the following:

First you want to verify that your system is programmed properly. To do this, confirm that the output from the modulator to the multiplexer is generally correct. Look at each stream to verify the Transport Stream IDentifier (TSID). Are the services correct in each TSID? Are the ratings there? Is the configuration set up properly? Does it have the correct settings so that the table and descriptor information will emerge from the multiplexer correctly? From a customer standpoint, these questions are vital, since the decoder uses this information to create the EPG. Does the EPG contain the correct information (was the information entered correctly)? Are the tuning parameters right? Does the system channel hop without delay?

You will also want to check your configuration over time. Are changes to the system implemented correctly?

As an integrator, you can't assume that the equipment in your configuration will function as expected. Since you are often testing equipment produced by different manufacturers, you must verify interoperability between different pieces in the system. Does the equipment in your system work together? If not, which piece of equipment is causing the problem? Is it a stream problem? An interface problem? As you test an integrated system, different individual pieces of the system may also malfunction or fail. When this occurs, you must effectively isolate and resolve the problem as quickly as possible. If a piece of equipment has failed, it must be repaired or replaced immediately.

Is the video correct? To verify this, you will need to run the transport stream through a decoder. Are the rates right? Does the audio/video quality match your expectations? Are there lip sync issues? If so, you can use specialized software to help you determine which piece of equipment is causing the error.

Video/audio, IP data, Java data, OpenTV and other components of a multiplex have their own protocol and syntax issues, and each has its own buffer model. System integrators must be sure that each piece of the multiplex meets the expectations outlined in its individual specification. For instance, with video, you may want to measure the quality of the video or look for statistics about motion vectors. With IP you may want to know what IP traffic is occurring in the transport stream, or what data addresses are being transmitted.

Test equipment can help you quickly isolate hardware and software failures in integrated systems. Transport stream analysis helps you identify errors at the output. Emulation devices allow you to feed known good streams to the different pieces of your system in order to test their functionality and isolate errors. You may also need to capture streams and send them back to a manufacturer for analysis and repair of faulty equipment.

Test equipment can also be useful to system integrators when it comes to acceptance testing. During acceptance testing, you must prove to the customer that the system is set up properly and performs as expected. The customer will want to know that output transport streams meet the required specifications, that the program guide is set up properly, and that the streams are transmitted at the correct rate. Test analyzers allow integrators to print reports based on the content of the output transport stream. These reports verify the PIDs and tables in the stream at any given time and display their individual rates. They can be run periodically over time to prove that a system can run without error for days or weeks at a time.

7 Glossary

AC-3—Audio compression standard adopted by ATSC and owned by Dolby.

ADC—Analog to Digital Converter.

ASCII—American Standard Code for Information Interchange.

ASI—Asynchronus Serial Interface. A standard DVB interface for a transport stream.

ATM—Asynchronous Transfer Mode.

ATSC—Advanced Television Systems Committee. Digital boadcasting standard developed in North America.

ATV—Advanced Television. North American standard for Digital Broadcasting.

BAT—Bouquet Association Table. This DVB table describes a set of services grouped together by a broadcaster and sold as a single entity. It is always found on PID 0x0011.

BER—Bit Error Rate.

B-frames—Bidirectionally predicted pictures, or pictures created from references to past and future pictures.

Bitrate—The rate at which a bit stream arrives at the input of a decoder.

Block—A set of 8x8 pixels used during Discrete Cosine Transform (DCT).

Bouquet—A set of services sold as a single entity.

Broadcaster—Someone who provides a sequence of scheduled events or programs to the viewer.

CA—Conditional Access. This system allows service providers to control subscriber access to programs and services via encryption.

CAT—Conditional Access Table. This table identifies EMM streams with a unique PID value. The CAT is always found on PID 0x0001.

CATV—Community Access Television, otherwise known as Cable TV.

Channel—A digital medium that stores or transports an MPEG-2 transport stream.

COFDM—Coded Orthogonal Frequency-Division Modulation.

Compression—Reduction of the number of bits needed to represent an item of data.

Conditional Access—A system used to control viewer access to programming based on subscription.

CRC—Cyclic Redundancy Check. This 32-bit field is used to verify the correctness of table data before decoding.

CVCT—Cable Virtual Channel Table. This ATSC table describes a set of one or more channels using a number or name within a cable network. Information in the table includes major and minor numbers, carrier frequency, short channel name, and information for navigation and tuning. This table is located on PID=0x1FFB.

D/A—Digital to Analog Converter.

DAVIC—Digital Audio Visual Council.

DBS—Direct Broadcasting Satellite or System.

DCT—Discrete Cosine Transform. Temporal-to-frequency transform used during spatial encoding of MPEG video.

Decoding Time Stamp—Decoding Time Stamp. This stamp is found in the PES packet header. It indicates the time at which a piece of audio or video will be decoded.

DigiTAG—Digital Television Action Group.

Downlink—Communication link from a satellite to earth.

DTV—Digital Television. A general term used to describe television that has been digitized. It can refer to Standard-definition TV or High-definition TV.

DTS—See Decoding Time Stamp.

DVB—Digital Video Broadcasting. The DVB Project is a European consortium that has standardized digital TV broadcasting in Europe and in other countries.

DVB ASI—Asynchronous Serial Interface. This is a standard DVB interface for a transport stream.

DVB-C—Digital Video Broadcasting-Cable. The DVB standard for broadcasting digital TV signals by cable. The RF spectrum in digital cable TV networks has a frequency range of (approx) 46MHz to 850MHz.

DVB-S—Digital Video Broadcasting-Satellite. The DVB standard for broadcasting digital TV signals via satellite.

DVB SPI—Synchronous Parallel Interface. This is a standard DVB interface for a transport stream.

DVB-T—Digital Video Broadcasting-Terrestrial. The DVB standard for broadcasting digital terrestrial TV signals.

ECM—Entitlement Control Message. ECMs carry private conditional access information that allows receivers to decode encrypted information.

EIT (ATSC)—Event Information Table. This table is part of the ATSC PSIP. It carries the TV guide information including titles and start times for events on all the virtual channels within the transport stream. ATSC requires that each system contain at least 4 EIT tables, each representing a different 3-hour time block. The PIDs for these tables are identified in the MGT.

EIT Actual (DVB)—Event Information Table. This table is part of the DVB SI. It supplies the list of events corresponding to each service and identifies the characteristics of each of these events. Four types of EITs are defined by DVB : 1) The EIT Actual Present/Following supplies information for the present event and the next or following event of the transport stream currently being accessed. This table is mandatory and can be found on PID=0x0012. 2)The EIT Other Present/Following defines the present event and the next or following events of other transport streams in the system that are not currently being accessed by the viewer. This table is optional. 3) The EIT Actual Event Schedule supplies the detailed list of events in the form of a schedule that goes beyond what is currently or next available. This table supplies a schedule of events for the transport stream currently being accessed by the viewer. 4) The EIT Other Event Schedule supplies the detailed schedule of events that goes beyond what is currently or next available. This table supplies a schedule of events for other transport streams in the system that are not currently being accessed by the viewer. The EIT Schedule tables are optional.

EMM—Entitlement Management Message. EMMs specify authorization levels or services of specific decoders. They are used to update the subscription options or pay-per-view rights for an individual subscriber or for a group of subscribers.

EPG—Electronic Program Guide. This guide represents a broadcasting data structure that describes all programs and events available to the viewer. It functions like an interactive TV guide that allows users to view a schedule of available programming and select what they want to watch.

ES—Elementary Stream. A bit stream that includes video, audio or data. It represents the preliminary stage to the Packetized Elementary Stream (PES).

ETR—ETSI Technical Report.

ETR 290—ETSI recommendation regarding measurement of MPEG-2/DVB transport streams.

ETSI—European Telecommunication Standard Institute.

ETT—Extended Text Table. This table is part of the ATSC PSIP. It carries relatively long text messages for additional descriptions of events and channels. There are two types of ETTs, the Channel ETT, which describes a channel, and the Event ETT, which describes individual events in a channel. The PID for this table is identified in the MGT.

Event—A collection of elementary streams with a common time base and an associated start time and end time. An event is equivalent to the common industry usage of "television program."

FEC—Forward Error Correction. This method adds error control bits before RF modulation. With these bits, errors in the transport stream may be detected and corrected prior to decoding.

Frame—Lines of spatial information for a video signal.

GOP—See Group Of Pictures

Group of Pictures—a set of pictures usually 12-15 frames long used for temporal encoding of MPEG-2 video.

HDTV—High Definition Television. HDTV's resolution is approximately twice as high as that of Standard Definition Television (SDTV) for both horizontal and vertical dimensions. HDTV has an aspect ratio of 16x9 as compared to the 4x3 aspect ratio of SDTV.

IEC—International Electrotechnical Commission.

IEEE—Institute of Electrical and Electronics Engineers.

I/F—Interface.

I-frame—Intra-coded frame, or a picture encoded without reference to any other picture. I-frames provide a reference for Predicted and Bidirectionally predicted frames in a compressed video stream.

IRD—Integrated Receiver Decoder. This is a receiver with an MPEG-2 decoder, also known as a set-top box.

ISO—International Standardization Organization.

ITU—International Telecommunications Union (UIT).

LVDS—Low Voltage Differential Signal. An electrical specification used by some manufacturers, usually on a parallel interface. It is a balanced interface with a low signal voltage swing (about 300mV).

Macroblock—A group of 16x16 pixels used for motion estimation in temporal encoding of MPEG-2 video.

MFN—Multiple Frequency Network (DVB-T).

MGT—Master Guide Table. This table is part of the ATSC PSIP. It defines sizes, types, PIDs, and version numbers for all of the relevant tables within the transport stream. The PID value for this table is 0x1FFB.

MHEG—Multimedia & Hypermedia Expert Group.

MIP—Megaframe Initialization Packet. This packet is used by DVB-T to synchronize the transmitters in a multi-frequency network.

MP@HL—Main Profile at High Level. MPEG-2 specifies different degrees of compression vs. quality. Of these, Main Profile at High Level is the most commonly used for HDTV.

MP@ML—Main Profile at Main Level. MPEG-2 specifies different degrees of compression vs. quality. Of these, Main Profile at Main Level is the most commonly used.

MPEG—Moving Picture Experts Group, also called Motion Picture Experts Group.

MPEG-2—ISO/IEC 13818 standard defining motion video and audio compression. It applies to all layers of transmission (video, audio and system).

MPTS—Multiple Program Transport Stream. An MPEG-2 transport stream containing several programs that have been multiplexed.

Multiplex (n)—A digital stream including one or more services in a single physical channel. (v)—To sequentially incorporate several data streams into a single data stream in such a manner that each may later be recovered intact.

Network—The set of MPEG-2 transport streams transmitted via the same delivery system.

NIT—Network Information Table. This DVB table contains information about a network's orbit, transponder etc. It is always located on PID 0x0010. DVB specifies two types of NITs, the NIT Actual and the NIT Other. The NIT Actual is a mandatory table containing information about the physical parameters of the network actually being accessed. The NIT Other contains information about the physical parameters of other networks. The NIT Other is optional.

NTSC—National TV Standard Committee Colour TV System (USA and 60 Hz countries).

NVoD—Near Video on Demand. This service allows for a single TV program to be broadcast simultaneously with a few minutes of difference in starting time. For example, a movie could be transmitted at 9:00, 9:15 and 9:30.

Packet—See Transport Packet.

PAL—Phase Alternating Line.

PAT—Program Association Table. This MPEG-2 table lists all the programs contained in the transport stream and shows the PID value for the PMT associated with each program. The PAT is always found on PID 0x0000.

Payload—All the bytes in a packet that follow the packet header.

PCR—Program Clock Reference. A time stamp in the transport stream that sets the timing in the decoder. The PCR is transmitted at least every 0.1 seconds.

PES—Packetized Elementary Stream. This type of stream contains packets of undefined length. These packets may be comprised of video or audio data packets and ancillary data.

PES Packet—The structure used to carry elementary stream data (audio and video). It consists of a header and a payload.

PES Packet Header—The leading bytes of a PES packet, which contain ancillary data for the elementary stream.

PID—Packet Identifier. This unique integer value identifies elements in the transport stream such as tables, data, or the audio for a specific program.

PLL—Phase Lock Loop. This locks the decoder clock to the original system clock through the PCR.

PMT—Program Map Table. This MPEG-2 table specifies PID values for components of programs. It also references the packets that contain the PCR.

P-frame—Predicted frame, or a picture coded using references to the nearest previous I- or P- picture.

Program—See Service.

PSI—Program Specific Information. PSI refers to MPEG-2 table data necessary for the demultiplexing of a transport stream and the regeneration of programs within the stream. PSI tables include PAT, CAT, PMT and NIT.

PSIP—Program and System Information Protocol. The ATSC protocol for transmission of data tables in the transport stream. Mandatory PSIP tables include MGT, STT, RRT, VCT and EIT.

PTS—Presentation Time Stamp. This stamp indicates the time at which an element in the transport stream must be presented to the viewer. PTSs for audio and video are transmitted at least every 0.7 seconds. The PTS is found in the PES header.

QAM—Quadrature Amplitude Modulation. This is a type of modulation for digital signals used in CATV transmission (DVB-C). Amplitude and phase of a carrier are modulated in order to carry information.

QPSK—Quadrature Phase Shift Keying. A type of modulation for digital signals used in satellite transmission (DVB-S).

RRT—Rating Region Table. An ATSC PSIP table that defines ratings systems for different regions or countries. The table includes parental guidelines based on Content Advisory descriptors within the transport stream.

RS—Reed-Solomon Protection Code. This refers to the 16 bytes of error control code that can be added to every transport packet during modulation.

RST—Running Status Table. A DVB-SI table that indicates a change of scheduling information for one or more events. It saves broadcasters from having to retransmit the corresponding EIT. This table is particularly useful if events are running late. It is located on PID 0x0013.

SDT—Service Description Table. This DVB SI table describes the characteristics of available services. It is located on PID 0x0011. Two types of SDTs are specified by DVB, the SDT Actual and the SDT Other. The SDT Actual is a mandatory table that describes the services within the transport stream currently being accessed. The SDT Other describes the services contained in other transport streams in the system.

SDTV—Standard Definition Television. SDTV refers to television that has a quality equivalent to NTSC or PAL.

Section—A syntactic structure used for mapping PSI/SI/PSIP tables into transport packets of 188 bytes.

Service—A collection of one or more events under the control of a single broadcaster. Also known as a Program.

SFN—Single Frequency Network (DVB-T).

SI—Service Information. This DVB protocol specifies all the data required by the receiver to demultiplex and decode the programs and services in the transport stream. Mandatory DVB SI tables include TDT, NIT, SDT and EIT.

SMPTE—Society of Motion Picture and Television Engineers.

SNG—Satellite News Gathering. This refers to the retransmission of events using mobile equipment and satellite transmission.

SNMP—Simple Network Management Protocol. This is the standard protocol for system and network administration.

SPI—Synchronous Parallel Interface. This is a standard DVB interface for a transport stream.

SPTS—Single Program Transport Stream. An MPEG-2 transport stream that contains one unique program.

ST—Stuffing Table. An optional DVB-SI table that authorizes the replacement of complete tables due to invalidation at a delivery system boundary such as a cable headend. This table is located on PID 0x0014.

STB—Set-top box. A digital TV receiver (IRD).

STD—See System Target Decoder.

STT—System Time Table. An ATSC PSIP table that carries time information needed for any application requiring schedule synchronization. It provides the current date and time of day and is located on PID 0x1FFB.

System Target Decoder (STD)—A hypothetical reference model of the decoding process defined by MPEG-2.

Table—Service Information is transmitted in the form of tables, which are further divided into subtables, then into sections, before being transmitted. Several types of tables are specified by MPEG, DVB and ATSC. Refer to the Pocket Guide for more information on the different types of Service Information tables and their functions.

TDT—Time and Date Table. This mandatory DVB SI table supplies the time reference expressed in terms of UTC time/date. This enables joint management of the events corresponding to the services accessible from a single reception point. The PID for this table is 0x0014.

Time-stamp—An indication of the time at which a specific action must occur in order to ensure proper decoding and presentation.

TOT—Time Offset Table. This optional DVB SI table supplies the UTC time and date and shows the difference between UTC time and the local time for various geographical regions. The PID for this table is 0x0014.

Transponder—Trans(mitter) and (res)ponder. This refers to the equipment inside a satellite that receives and re-sends information.

Transport Packet—188-byte packet of information in a transport stream. Each packet contains a header and a payload.

Transport Stream—A stream of 188-byte transport packets that contains audio, video, and data belonging to one or several programs.

T-STD—See System Target Decoder.

TV—Television.

TVCT—Terrestrial Virtual Channel Table. This ATSC table describes a set of one or more channels or services using a number or name within a terrestrial broadcast. Information in the table includes major and minor numbers, short channel name, and information for navigation and tuning. This table is located on PID=0x1FFB.

Uplink—Communication link from earth to a satellite.

UTC—Universal Time, Co-ordinated.

VCT—Virtual Channel Table. This ATSC table describes a set of one or more channels or services. Information in the table includes major and minor numbers, short channel name, and information for navigation and tuning. There are two types of VCTs, the TVCT for terrestrial systems and the CVCT for cable systems.

VLC—Variable Length Coding. This refers to a data compression method (Huffmann).

VoD—Video on Demand.

VSB—Vestigial Sideband Modulation. This is the terrestrial modulation method used in the ATSC. It can have either 8 (8 VSB) or 16 (16 VSB) discrete amplitude levels.

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