Transport Stream

Related terms:

Synchronization, Multiplexing, Elementary Stream, Video Compression Standard

View all Topics

Video Communication Networks

Dan Schonfeld, in Handbook of Image and Video Processing (Second Edition), 2005

2.4.4 MPEG-2 Transport Stream

A transport stream (TS) permits multiplexing streams (PESs and PSs) that do not necessarily share a common time-base for transmission in noisy environments. The TS is designed for broadcasting over communication networks such as ATM networks. The TS uses small fixed-length packets (188 bytes) that make them more resilient to packet loss or damage during transmission. The TS provides the input to the transport layer in the OSI reference model.[9]

The TS packet is composed of a 4-byte header followed by 184 bytes shared between the variable-length adaptation field and the TS packet payload. An illustration of the TS header is depicted in Fig. 5. The corresponding glossary of the TS header is provided in Table 3. Note that the unshaded box appearing in Fig. 5 is used to represent the optional adaptation field (AF).

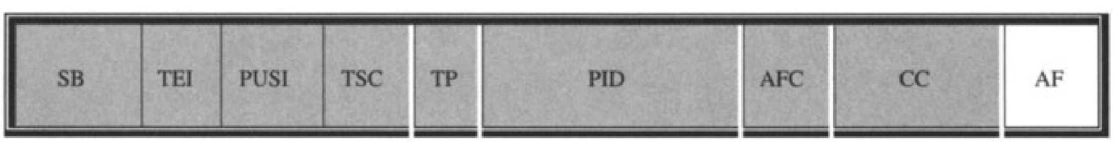

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FIGURE 5. Transport stream header.

TABLE 3. Transport stream header glossary

Acronym	Function
SB	Synchronization byte
TEI	Transport error indicator
PUSI	Payload unit start indicator
TSC	Transport scrambling control
TP	Transport priority
PID	Packet identifier
AFC	Adaptation field control
CC	Continuity counter
AF	Adaptation field (optional)

The TS header includes a synchronization byte (SB) designed for detection of the beginning of each TS packet. The transport error indicator (TEI) points to the detection of an uncorrectable bit error in this TS packet. The payload unit start indicator (PUSI) is used to ascertain if the TS payload contains PES packets or program-specific information (PSI). The packet ID (PID) identifies the type and source of payload in the TS packet. The presence or absence of the adaptation field (AF) and payload is indicated by the adaptation field control (AFC). The continuity counter (CC) provides the number of TS packets with the same PID, which is used to determine packet loss.

The optional adaptation field (AF) contains additional information that need not be included in every TS packet. One of the most important fields in the AF is the program clock reference (PCR). The PCR is a 42-bit field composed of a 9-bit segment incremented at 27 MHz as well as a 33-bit segment incremented at 90 KHz.[10] The PCR is used along with a voltage-controlled oscillator as a time reference for synchronization of the encoder and decoder clock.

A PES header must always follow the TS header and possible AF. The TS payload may consist of the PES packets or PSI. The PSI provides control and management information used to associate particular ESs with distinct programs. A program is once again defined as a collection of ESs that share a common time-base. This is accomplished by means of a program description provided by a set of PSI associated signaling tables (AST): program association tables (PAT), program map tables (PMT), network information tables (NIT), and conditional access tables (CAT). The PSI tables are sent periodically and carried in sections along with cyclic redundancy check (CRC) protection in the TS payload.

An example illustrating the formation of the TS packets is depicted in Fig. 6. The choice of the size of the fixed-length TS packets—188 bytes—is motivated by the fact that the payload of the ATM Adaptation Layer-1 (AAL-1) cell is 47 bytes. Therefore, four AAL-1 cells can accommodate a single TS packet. A detailed discussion of the mapping of the TS packets to ATM networks is presented in the next section.

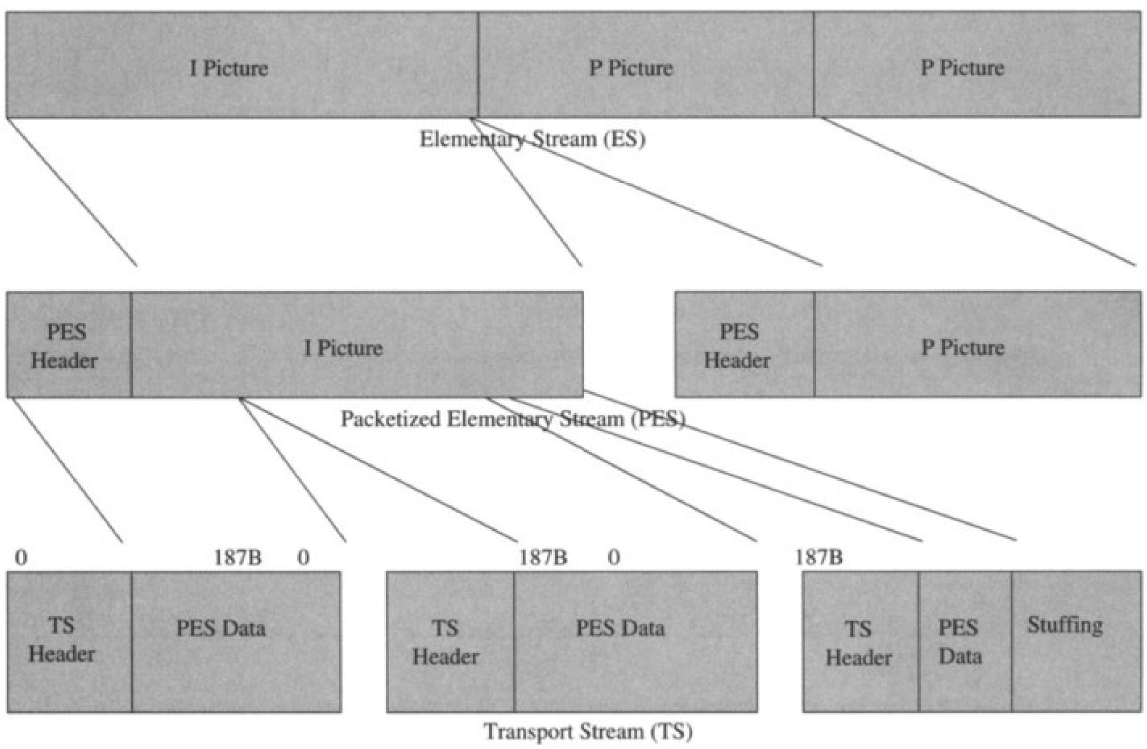

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FIGURE 6. Transport stream packets.

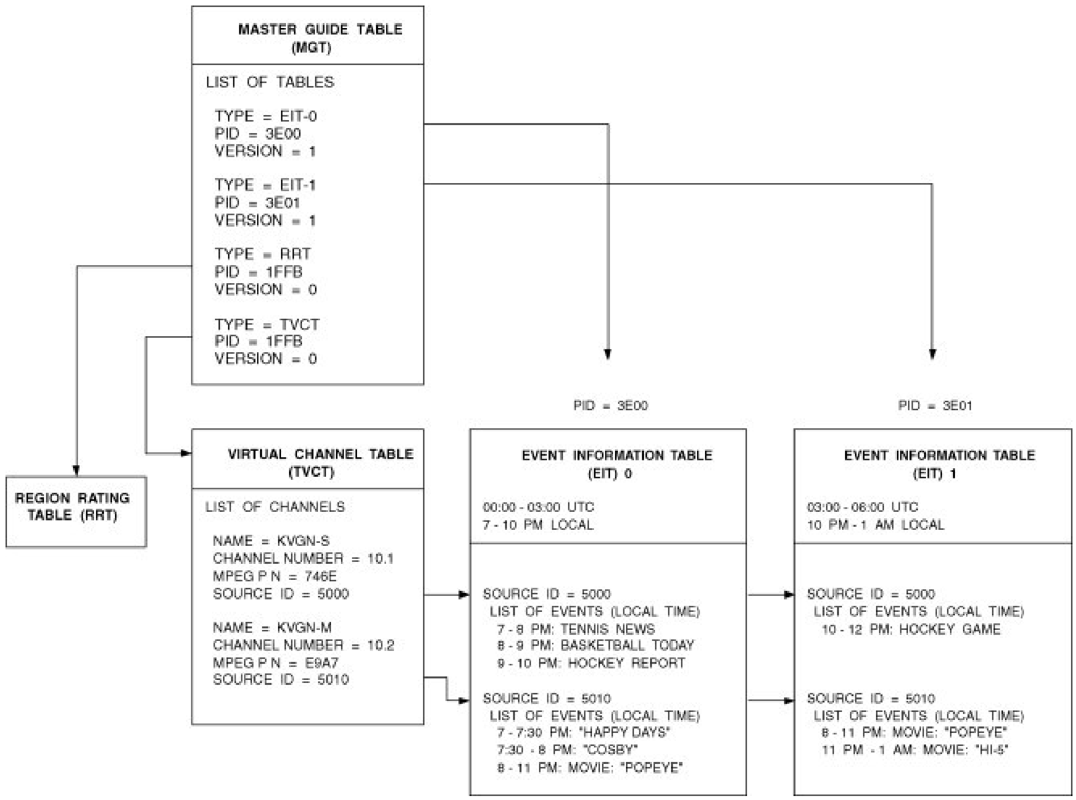
Digital Television (DTV)

Keith Jack, in Digital Video and DSP, 2008

Program and System Information Protocol (PSIP)

Enough bandwidth is available within the MPEG-2 transport stream to support several low-bandwidth non-television services such as program guide, closed captioning, weather reports, stock indices, headline news, software downloads, pay-per-view information, etc. The number of additional non-television services (virtual channels) may easily reach ten or more. In addition, the number and type of service will constantly be changing.

To support these non-television services in a flexible yet consistent manner, the Program and System Information Protocol (PSIP) was developed. PSIP is a small collection of hierarchically associated tables (see Figure 8.1 and Table 8.2) designed to extend the MPEG-2 PSI tables. It describes the information for all virtual channels carried in a particular MPEG-2 transport stream. Additionally, information for analog broadcast channels may be incorporated.



Download full-size image

Figure 8.1. ATSC PSIP Table Relationships.

Table 8.2. List of ATSC PSIP Tables, Descriptors, and Descriptor Locations

Descriptor	Descriptor Tag	Terrestrial Broadcast Tables									
		PMT	MGT	VCT	RRT	EIT	ETT	STT	DCCT	DCCSCT	CAT
PID		per PAT	0x1FFB	0x1FFB	0x1FFB	per MGT	per MGT	0x1FFB	0x1FFB	0x1FFB	0x0001
Table_ID		0x02	0xC7	0xC8	0xCA	0xCB	0xCC	0xCD	0xD3	0xD4	0x80, 0x81 (ECM) 0x82 – 0x8F (EMM)
Repetition rate		400 ms	150 ms	400 ms	1 min	0.5 sec	1 min	1 sec	400 ms	1 hour	
AC-3 audio stream	1000 0001	M				M					
ATSC CA	1000 1000			O		O					
ATSC private information*	1010 1101										
CA	0000 1001	M									M

Descriptor	Descriptor Tag	Terrestrial Broadcast Tables									
		PMT	MGT	VCT	RRT	EIT	ETT	STT	DCCT	DCCSCT	CAT
Caption service	1000 0110	M				M					
Component name	1010 0011	M									
Content advisory	1000 0111	M				M					
Content identifier	1011 0110	O				M					
DCC arriving request	1010 1001								M		
DCC departing request	1010 1000								M		
Enhanced signaling	1011 0010	M PMT-E									
Extended channel name	1010 0000			M							
Genre	1010 1011					M					
Redistribution control	1010 1010	M				M					
Service location	1010 0001			M							
SRM reference	0000 1001										M
Stuffing*	1000 0000										
Time-shifted service	1010 0010			M							

Note: 1. M=when present, required in this table. O=may be present in this table also.*=no restrictions.

Required Tables

Event Information Table (EIT)

There are up to 128 EITs, EIT-0 through EIT-127, each of which describes the events or TV programs associated with each virtual channel listed in the VCT. Each EIT is valid for three hours. Since there are up to 128 EITs, up to 16 days of programming may be advertised in advance. The first four EITs are required (the first 24 are recommended) to be present.

Information provided by the EIT includes start time, duration, title, pointer to optional descriptive text for the event, advisory data, caption service data, audio service descriptor, and so on.

Master Guide Table (MGT)

This table provides general information about the other tables. It defines table sizes, version numbers, and packet identifiers (PIDs).

Rating Region Table (RRT)

This table transmits the rating system, commonly referred to as the “V-chip.”

System Time Table (STT)

This table serves as a reference for the time of day. Receivers use it to maintain the correct local time.

Terrestrial Virtual Channel Table (TVCT)

This table, also referred to as the VCT although there is also a Cable VCT (CVCT) and Satellite VCT (SVCT), contains a list of all the channels in the transport stream that are or will be available, plus their attributes. It may also include the broadcaster's analog channel and digital channels in other transport streams.

Attributes for each channel include major/minor channel number, short name, Transport/Transmission System ID (TSID) that uniquely identifies each station, etc. The Service Location Descriptor is used to list the PIDs for the video, audio, data, and other related elementary streams.

Optional Tables

Extended Text Table (ETT)

For text messages, there can be several ETTs, each having its PID defined by the MGT. Messages can describe channel information, coming attractions, movie descriptions, and so on.

Directed Channel Change Table (DCCT)

The DCCT contains information needed for a channel change to be done at a broadcaster-specified time. The requested channel change may be unconditional or may be based upon criteria specified by the viewer.

Directed Channel Change Selection Code Table (DCCSCT)

The DCCSCT permits a broadcast program categorical classification table to be downloaded for use by some Directed Channel Change requests.

Descriptors

Much like MPEG-2, ATSC uses descriptors to add new functionality. In addition to various MPEG-2 descriptors, one or more of these ATSC-specific descriptors may be included within the PMT or one or more PSIP tables to extend data within the tables. A descriptor not recognized by a decoder must be ignored by that decoder. This enables new descriptors to be implemented without affecting receivers that cannot recognize and process the descriptors.

AC-3 Audio Stream Descriptor

This ATSC descriptor indicates Dolby® Digital or Dolby® Digital Plus audio is present.

ATSC CA Descriptor

This ATSC descriptor has a syntax almost the same as the MPEG-2 CA descriptor.

ATSC Private Information Descriptor

This ATSC descriptor provides a way to carry private information. More than one descriptor may appear within a single descriptor.

Component Name Descriptor

This ATSC descriptor defines a variable-length text-based name for any component of the service.

Content Advisory Descriptor

This ATSC descriptor defines the ratings for a given program.

Content Identifier Descriptor

This ATSC descriptor is used to uniquely identify content with the ATSC transport.

DCC Arriving Request Descriptor

This ATSC descriptor provides instructions for the actions to be performed by a receiver upon arrival to a newly changed channel:

Display text for at least 10 seconds, or for a less amount of time if the viewer issues a "continue," "OK," or equivalent command.

Display text indefinitely, or until the viewer issues a "continue," "OK," or equivalent command.

DCC Departing Request Descriptor

This ATSC descriptor provides instructions for the actions to be performed by a receiver prior to leaving a channel:

Cancel any outstanding things and immediately perform the channel change.

Display text for at least 10 seconds, or for a smaller amount of time if the viewer issues a "continue," "OK," or equivalent command.

Display text indefinitely, or until the viewer issues a "continue," "OK," or equivalent command.

Enhanced Signaling Descriptor

This ATSC descriptor identifies the terrestrial broadcast transmission method of a program element.

Extended Channel Name Descriptor

This ATSC descriptor provides a variable-length channel name for the virtual channel.

Genre Descriptor

This ATSC descriptor provides genre, program type, or category information for events., and may appear in the descriptor() loop for the given EIT event. It references entries in the Categorical Genre Code Assignments Table and may include references to expansions to that table provided by the DCC Selection Code.

Redistribution Control Descriptor

This ATSC descriptor conveys any redistribution control information held by the program rights holder for the content.

Service Location Descriptor

This ATSC descriptor specifies the stream type, PID, and language code for each elementary stream. It is present in the VCT for each active channel.

SRM Reference Descriptor

This ATSC descriptor is a specific implementation of the MPEG-2 CA Descriptor. It is used to signal that a System Renewability Message is present for the System Renewability Message Table (SRMT). It is present in the CAT.

Time-Shifted Service Descriptor

This ATSC descriptor links one virtual channel with up to 20 other virtual channels carrying the same programming, but time-shifted. A typical application is for Near Video On Demand (NVOD) services.

Multimedia Systems: Content-Based Indexing and Retrieval

Faisal Bashir, ... Dan Schonfeld, in The Electrical Engineering Handbook, 2005

MPEG-2: Coding of High-Quality Moving Pictures (MPEG-2)

The MPEG-1 standard was targeted for coding of audio and video for storage, in which the media error rate is negligible. Hence, MPEG-1 bitstream was not designed to be robust to bit errors. In addition, MPEG-1 was aimed at software-oriented image processing, where large and variable length packets could reduce the software overhead. The MPEG-2 standard, on the other hand, is more generic for a variety of audio–visual coding applications. It has to have the error resilience for broadcasting and ATM networks. The aim of MPEG-2 is to produce broadcast-quality video compression and support higher resolutions including high definition television (HDTV).[11] MPEG-2 supports four resolution levels: low (352 x 240), main (720 x 480), high-1440 (1440 x 1152), and high (1920 x 1080) (ISO/IEC, 1994). The MPEG-2 compressed video data rates are in the range of 3 to 100 Mbps.[12] Although the principles used to encode MPEG-2 are very similar to MPEG-1, they provide much greater flexibility by offering several profiles that differ in the presence or absence of B-pictures, chrominance resolution, and coded stream scalability[13] MPEG-2 supports both progressive and interlaced modes.[14] Significant improvements have also been introduced in the MPEG-2 system level. The MPEG-2 systems layer is responsible for the integration and synchronization of the elementary streams (ES): audio and video streams as well as an unlimited number of data and control streams that can be used for various applications such as subtitles in multiple languages. This is accomplished by first packetizing the ES, thus forming the packetized elementary streams (PES). These PES contain timestamps from a system clock for synchronization. The PES are subsequently multiplexed to form a single output stream for transmission in one of two modes: program stream (PS) and transport stream (TS). The PS is provided for error-free environments, such as storage in a CD-ROM. They are used for multiplexing PES that share a common time base, using long variable-length packets.[15] The TS is designed for noisy environments, such as communication over ATM networks. This mode permits multiplexing streams (PES and PS), which do not necessarily share a common time-base, using fixed-length (188 bytes) packets.

In the MPEG-2 standard, pictures can be interlaced, whereas in MPEG-1, the pictures are progressive only. The dimensions of the units of blocks used for motion estimation/compensation can change. Because the number of lines per field is half the number of lines per frame in the interlaced pictures, for motion estimation it might be appropriate to choose blocks of 16 × 8 (i.e., 16 pixels over 8 lines) with equal horizontal and vertical resolutions. The second major difference between the two is **scalability.** The scalable modes of MPEG-2 video encoders are intended to offer interoperability among different services or to accommodate the varying capabilities of different receivers and networks upon which a single service may operate. MPEG-2 also has a choice of a different DCT coefficient scanning mode **alternate scan** as well as a zigzag scan.

MPEG-1 and MPEG-2 Video Standards

Supavadee Aramvith, Ming-Ting Sun, in Handbook of Image and Video Processing (Second Edition), 2005

2.1 Introduction

2.1.1 Background and Structure of MPEG-2 Standards Activities

The MPEG-2 standard represents the continuing efforts of the MPEG committee to develop generic video and audio coding standards after their development of MPEG-1. The idea of this second phase of MPEG work came from the fact that MPEG-1 is optimized for applications at about 1.5 Mb/s with input source in SIF, which is a relatively low-resolution progressive format. Many higher quality higher bit rate applications require a higher resolution digital video source such as ITU-R BT 601, which is an interlaced format. New techniques can be developed to code the interlaced video better.

The MPEG-2 committee started working in late 1990 after the completion of the technical work of MPEG-1. The competitive tests of video algorithms were held in November 1991, followed by the collaborative phase. The Committee Draft (CD) for the video part was achieved in November 1993. The MPEG-2 standard (ISO/IEC 13818) [8] consists of nine parts. The first five parts are organized in the same fashion as MPEG-1: systems, video, audio, conformance testing, and simulation software technical report. The first three parts of MPEG-2 reached International Standard (IS) status in November 1994. Parts 4 and 5 were approved in March 1996. Part 6 of the MPEG-2 standard specifies a full set of Digital Storage Media Control Commands (DSM-CC). Part 7 is the specification of Advanced Audio Coding (AAC). Part 8 was originally planned to be the coding of 10-bit video but was discontinued. Part 9 is the specification of real-time interface (RTI) to transport stream decoders which may be utilized for adaptation to all appropriate networks carrying MPEG-2 transport streams. Part 10 is the specification of conformance testing part of DSM-CC. Part 6 and Part 9 have already been approved as International Standards in July 1996. Like the MPEG-1 video standard, MPEG-2 video coding standard specifies only the bit stream syntax and the semantics of the decoding process. Many encoding options were left unspecified to encourage continuing technology improvement and product differentiation.

MPEG-3, which was originally intended for HDTV (high definition digital television) at higher bit-rates, was merged with MPEG-2. Hence there is no MPEG-3. MPEG-2 video coding standard (ISO/IEC 13818-2) was also adopted by ITU-T as ITU-T Recommendation H.262 [9].

2.1.2 Target Applications and Requirements

MPEG-2 is primarily targeted at coding high-quality video at 4-15 Mb/s for video on demand (VOD), digital broadcast television, and Digital Storage

Media such as DVD (digital versatile disc). It is also used for coding HDTV, cable/satellite digital TV, video services over various networks, 2-way communications, and other high-quality digital video applications

The requirements from MPEG-2 applications mandate several important features of the compression algorithm. Regarding picture quality, MPEG-2 needs to be able to provide good NTSC quality video at a bit rate of about 4–6 Mbits/s and transparent NTSC quality video at a bit rate of about 8–10 Mbits/s. It also needs to provide the capability of random access and quick channel-switching by means of I-pictures in GOPs. Low-delay mode is specified for delay-sensitive visual communications applications. MPEG-2 has scalable coding modes in order to support multiple grades of video quality, spatial resolutions, and frame-rates for various applications. Error resilience options include intramotion vector, data partitioning, and scalable coding. Compatibility with the existing MPEG-1 video standard is another prominent feature provided by MPEG-2. For example, MPEG-2 decoders should be able to decode MPEG-1 bit streams. If scalable coding is used, the base layer of MPEG-2 signals can be decoded by a MPEG-1 decoder. Finally, it should allow reasonable complexity encoders and low-cost decoders be built with mature technology. Since MPEG-2 video is based heavily on MPEG-1, in the following sections, we will focus only on those features which are different from MPEG-1 video.

MPEG-2 storage and transport

In Digital Video and HD (Second Edition), 2012

MPEG-2 transport stream

An MPEG-2 *transport stream* (TS) is a part of the MPEG-2 suite of standards that specifies a relatively complex mechanism of multiplexing video and audio for one or more programs into a data stream, typically having short packets, suitable for transmission through error-prone media where relatively powerful forward error-correction (FEC) is required. A transport stream is suitable for applications where a player connects to atransmission in progress (like television), as opposed to reading a file from its beginning. For terrestrial over-the-air (OTA) or cable television, TS packets are expected to be suitably protected; however specification of the FEC and channel coding lies outside the MPEG standards and ordinarily lies within the realm of digital television standards (for example, ATSC standards in North America, and DVB standards in Europe).

ATM: Asynchronous transfer mode, a protocol for high performance networking.

A *transport stream packet* (TSP) comprises 188 bytes – a 4-byte header (whose first byte has the value 47_h), including a 13-bit *packet identifier* (PID), and 184 bytes of payload. Packet size was designed with ATM in mind: One TS packet fits into four ATM cells (48 bytes each). Owing to a lack of external interfaces for program streams, a *single program transport stream* (SPTS) may be used to carry one program. For some applications, a *multiple program transport stream* (MPTS) is used.

Transport stream packets with PID 0 contain the *program association table* (PAT), repeated a few times per second. The PAT lists one or more PIDs of subsequent packets containing *program map tables* (PMTs). APMT lists PIDs of video and audio elementary streams associated with a single program.

ATSC Standard A/65, *Program and System Information Protocol.*

An ATSC DTV transport stream contains a set of packets implementing the *program and system information protocol* (PSIP). PSIP identifies channels and programs, and conveys time-of-day and station callsign information. PSIP enables a receiver to provide an electronic program guide (EPG).

On a computer, 188-byte transport stream packets typically have a 4-byte timecode appended (resulting in 192-byte packets); a file comprising a sequence of such packets typically has the extension *m2t, m2ts,* or just *ts.*

TOD is reported to stand for *transport stream on disk.*

MPEG-2 transport streams are used in applications such as these:

- The TOD consumer video format (essentially an MPEG-2 MP@HL HD transport stream)
- The BDAV container of Blu-ray
- H.264 compressed video
- AVCHD compressed video (in computing, the file extension *mts* is usual)

SDI and HD-SDI interfaces

In Digital Video and HD (Second Edition), 2012

ASI

Within a broadcast facility, an MPEG-2 transport stream can be serialized onto a dedicated asynchronous *serial interface* (ASI). A serialized ASI stream for broadcast has a payload bit rate of around 20 Mb/s; however, the ASI interface bit rate is 270 Mb/s, chosen so that SDI distribution infrastructure can be used. The ASI interface uses BNC connectors and coaxial cable. ASI is polarity sensitive (unlike SDI), though modern ASI receivers typically detect and correct polarity inversion.

> ETSI EN 50083-9, *Cable networks for television signals, sound signals and interactive services – Part 9: Interfaces for CATV/SMATV headends and similar professional equipment for DVB/MPEG-2 transport streams.*Standards are not clear on whether transformer coupling is required or whether capacitive coupling suffices.

Although the SDI physical layer is used, the serialized ASI stream has no TRS codes and the interface does not use SDI scrambling. Instead, channel data is encoded according to the 8b/10b scheme borrowed from Fibre Channel standards. (ASI interface data rate is therefore at most 216 Mb/s.) An 8b/10b bitstream never has more than four consecutive 0s or 1s, so clock recovery is simple. An 8b/10b encoder minimizes low frequency (“DC”) content on the media.

> Some people write 8B/10B; however, the elements involved are bits, not bytes, so lowercase *b* is apt.

Since the ASI payload rate is typically far lower than the channel capacity, stuffing codes are inserted to occupy idle time. Stuffing codes – Fibre Channel *comma* codes, denoted *K28.5* – are inserted either at the earliest opportunity (bytewise, “spaced byte mode”), or at the completion of the current packet (packetwise, “burst mode”). MPEG packets are separated by at least two comma codes.

> The *synchronous serial interface* (SSI) was standardized by SMPTE for the purpose of conveying MPEG transport streams between equipment, but SSI has largely fallen into disuse.

It is increasingly common to convey transport streams using IP protocols across Ethernet.

Application

Vinod Joseph, Srinivas Mulugu, in Deploying Next Generation Multicast-enabled Applications, 2011

6.6.6.4.2 Media Loss Rate

Media loss rate is the count of the lost or out-of-order packets over a pre-specified time interval. It is common to measure this on a per second basis. It is important to note that the packets referred to here are not IP packets, but the MPEG Transport stream packets. So the loss of a single IP packet could result in the loss of seven MPEG transport stream packets.

As a practical example in a given service provider network when measured at different points, there may be different MDI values. Normally the lower the MDI values the better the QoE is for the end user. However, since MDI is a combination of two parameters, it gets difficult to compare. For instance, an MDI of 100:5 is better than an MDI of 150:20. However, due to transient conditions, if one has two MDI values of 50:20 and 100:5, then it would be a little more complex to analyze which would yield better QoE.

Image and Video Coding—Emerging Standards and Beyond

Barry G. Haskell, ... Patrick Haffner, in Readings in Multimedia Computing and Networking, 2002

C. MPEG-2 Coding

The MPEG-2 standard was designed to provide the capability for compressing, coding, and transmitting high-quality, multichannel, multimedia signals over terrestrial broadcast, satellite distribution, and broad-band networks, for example, using ATM (asynchronous transmission mode) protocols. The MPEG-2 standard specifies the requirements for video coding, audio coding, systems coding for combining coded audio and video with user-defined private data streams, conformance testing to verify that bit streams and decoders meet the requirements, and software simulation for encoding and decoding of both the program and the transport streams. Because MPEG-2 was designed as a transmission standard, it supports a for encoding and decoding of both the program and the transport streams. Because MPEG-2 was designed as a transmission standard, it supports a variety of packet formats (including long and variable-length packets of from 1 up to 64 kbits), and provides error correction capability that is suitable for transmission over cable TV and satellite links.

1) *MPEG-2 Systems:* The MPEG-2 systems level defines two types of streams: the program stream and the transport stream. The program stream is similar to that used in MPEG-1, but with a modified syntax and new functions to support advanced functionalities. Program stream decoders typically use long and variable-length packets, which are well suited for software-based processing and error-free environments. The transport streams offer the robustness necessary for noisy channels, and also provide the ability to include multiple programs in a single stream. The transport stream uses fixed-length packets of size 188 bytes, and is well suited for delivering compressed video and audio over error-prone channels such as CATV networks and satellite transponders.

   The basic data structure that is used for both the program stream and the transport stream data is called the packetized elementary stream (PES) packet. PES packets are generated by packetizing the compressed video and audio data, and a program stream is generated by interleaving PES packets from the various encoders with other data packets to generate a single bitstream. A transport stream consists of packets of fixed length, consisting of 4 bytes of header followed by 184 bytes of data obtained by chopping up the data in the PES packets. The key difference in the streams is that the program streams are intended for error-free environments, whereas the transport streams are intended for noisier environments where some type of error protection is required.

2) *MPEG-2 Video*: MPEG-2 video was originally designed for high-quality encoding of interlaced video from standard TV with bit rates on the order of 4–9 Mbits/s. As it evolved, however, MPEG-2 video was expanded to include high-resolution video, such as HDTV, as well as hierarchical or scalable video coding for a range of applications. Since MPEG-2 video does not standardize the encoding method, but only the video bit stream syntax and decoding semantics, there have evolved two generalized video codecs, one for nonscalable video coding and one for scalable video coding. Fig. 7 shows a block diagram of the MPEG-2 nonscalable video coding algorithm. The video encoder consists of an interframe/field DCT encoder, a frame/field motion estimator and compensator, and a variable-length encoder (VLE). The frame/field DCT encoder exploits spatial redundancies in the video, and the frame/field motion compensator exploits temporal redundancies in the video signal. The coded video bit stream is sent to a systems multiplexer, Sys Mux, which outputs either a transport or a program stream.

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Fig. 7. Generalized codec for MPEG-2 nonscalable video coding.

The MPEG-2 decoder of Fig. 7 consists of a variable-length decoder (VLD), interframe/field DCT decoder, and the frame/field motion compensator. Sys Demux performs the complementary function of Sys Mux and presents the video bit stream to VLD for decoding of motion vectors and DCT coefficients. The frame/field motion compensator uses a motion vector decoded by VLD to generate motion-compensated prediction that is added back to a decoded prediction error signal to generate decoded video out. This type of coding produces nonscalable video bit streams since, normally, the full spatial and temporal resolution coded is the one that is expected to be decoded.

A block diagram of a generalized codec for MPEG-2 scalable video coding is shown in Fig. 8. Scalability is the property that allows decoders of various complexities to be able to decode video of resolution/quality commensurate with their complexity from the same bit stream. The generalized structure of Fig. 8 provides capability for both spatial and temporal resolution scalability in the following manner. The input video goes through a preprocessor that produces two video signals, one of which (called the base layer) is input to a standard MPEG-1 or MPEG-2 nonscalable video encoder, and the other (called the enhancement layer) is input to an MPEG-2 enhancement video encoder. The two bit streams, one from each encoder, are multiplexed in Sys Mux (along with coded audio and user data). In this manner, it becomes possible for two types of decoders to be able to decode a video signal of quality commensurate with their complexity from the same encoded bit stream.

Download full-size image

Fig. 8. Generalized codec for MPEG-2 scalable video coding.

Video Interfaces

Keith Jack, in Digital Video and DSP, 2008

Pro-Video Transport Interfaces

Serial Data Transport Interface (SDTI)

SMPTE 305M and ITU-R BT.1381 define a Serial Data Transport Interface (SDTI) that enables transferring data between equipment. The physical layer uses the 270 or 360 Mbps BT.656, BT.1302, and SMPTE 259M digital component video serial interface. Figure 4.31 illustrates the signal format.

Download full-size image

Figure 4.31. SDTI Signal Format.

A 53-word header is inserted immediately after the EAV sequence, specifying the source, destination, and data format. Table 4.30 illustrates the header contents.

Table 4.30. DVI-D Connector Signal Assignments

Pin	Signal	Pin	Signal	Pin	Signal
1	D2–	**9**	D1–	**17**	D0–
2	D2	**10**	D1	**18**	D0
3	shield	**11**	shield	**19**	shield
4	D4–	**12**	D3–	**20**	D5–
5	D4	**13**	D3	**21**	D5
6	DDC SCL	**14**	+5V	**22**	shield
7	DDC SDA	**15**	ground	**23**	CLK
8	reserved	**16**	Hot Plug Detect	**24**	CLK–

The payload data is defined within BT.1381 and by other application-specific standards such as SMPTE 326M. It may consist of MPEG-2 program or transport streams, DV streams, etc., and uses either 8-bit words plus even parity and $\overline{D8}$, or 9-bit words plus $\overline{D8}$.

High Data-Rate Serial Data Transport Interface (HD-SDTI)

SMPTE 348M and ITU-R BT.1577 define a High Data-Rate Serial Data Transport Interface (HD-SDTI) that enables transferring data between equipment. The physical layer uses the 1.485 (or 1.485/1.001) Gbps SMPTE 292M digital component video serial interface.

Figure 4.32 illustrates the signal format. Two data channels are multiplexed onto the single HD-SDTI stream such that one 74.25 (or 74.25/1.001) MHz data stream occupies the Y data space and the other 74.25 (or 74.25/1.001) MHz data stream occupies the CbCr data space.

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Figure 4.32. HD-SDTI Signal Format. LN=line number (two 10-bit words), CRC=line number CRC (two 10-bit words).

A 49-word header is inserted immediately after the line number CRC data, specifying the source, destination, and data format.

The payload data is defined by other application-specific standards. It may consist of MPEG-2 program or transport streams, DV streams, etc., and uses either 8-bit words plus even parity and $\overline{\mathrm{D8}}$, or 9-bit words plus $\overline{\mathrm{D8}}$.

IPTV Architecture

James Farmer, ... Weyl Wang, in FTTx Networks, 2017

MPEG-4

MPEG-4 is the work-horse compression algorithm of Internet video transmission today, and finds some use in satellite and cable transmission. MPEG-4 is also known as advanced video coding (AVC), or by its ITU designation, H.264. As a rule of thumb, assume a bandwidth of 2 Mb/s for SD and 8 Mb/s for HD using MPEG-4. It is optional to put MPEG-4 packets in an MPEG-2 transport stream before putting them in TCP or UDP packets. MPEG-4 is specified as a decoding format for Blu-ray players. Oh, what happened to MPEG-3? Originally it was to be the high-definition companion to MPEG-2. But people found that MPEG-2 was a perfectly competent high-definition decoding system, so MPEG-3 was never developed.

You will see so-called HD streams transmitted over the Internet at speeds approximately one-quarter to one-half those cited above. True, you can do that, but the amount of compression required is such that a good video engineer would argue about the picture really being HD. We recently referred one such stream to an experienced video engineer with a major programmer, and he described it as being “a little better than SD, and wide screen.” This seems to be the story with much of what passes for “HD” on the Internet today, but many subscribers don’t know the difference.

Some OTT programmers detect the device being used to receive the stream, and send a stream with resolution (and speed) appropriate to the device in use. Thus, they will send a faster stream to an STB, and a much slower stream to a phone. The phone screen is so small that you would not see the higher resolution if you could get it into the phone. People will call them all “HD” streams, because that is what the public is expecting, and frankly many don’t know the difference.