Primary distribution of TV signals using MPEG-2 technologies

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Summary

This document summarizes the requirements for the primary distribution of MPEG-2 television signals. Nowadays, this often implies the use of various types of standardized digital telecom links, rather than dedicated broadcast networks.

In view of the ongoing transition from analogue to digital delivery, two basic scenarios are considered, namely using analogue signals at the source and/or destination, plus the fully-digital scenario.

The advantages and constraints of the various transmission systems are analyzed. The avoidance of degradation due to cascaded encoding/decoding is also given due consideration.

The document was prepared by a working group of EBU Project Group N/MT.

Introduction

The increasing use of techniques such as server-based editing and storage in the broadcast environment has led to an integration of several services in order to ensure fast and random access to all the programme elements (picture, sound, data, etc.) needed to produce a programme.

As far as networks are concerned, these changes mean that the integrated transport of all relevant data – together with additional information needed for production, storage and editing of programme material – becomes necessary. Furthermore, the introduction of different compression formats, together with different and possibly changing bit-rates for the transport of programme material, demands flexibility in networks – both in their transport capacities for contribution and distribution, and in their ease of topological adaptation.

A very important issue is the need for a seamless combination of internal networks (campus nets or intranets) with wide-area networks (WANs). This is to avoid the unnecessary cascading of compression processes just in order to adapt data rates to network access rates.

A network should be able to cope with both fast file transfers and real-time transmission of programme material. In the case of contribution material, the signal quality must be maintained at a level which is in line with the studio requirements for post-processing.

Looked at from a purely economic perspective, networks that have been specifically designed for broadcast applications, and adapted for emerging broadcast technologies, are no longer affordable. This makes it increasingly attractive for broadcasters to make use of ordinary, commonly-available, telecommunications equipment.

Using standard telecommunications networks means having to accept the behaviour of these networks, which does have certain immediate disadvantages for broadcasters. These networks are optimized for telephony and data transport only, and measures need to be taken in the network adapters or video/audio terminals to fulfil the requirements for the real-time transmission of video/audio signals.

Frame-store synchronizers for video signals, or sample-rate converters for audio signals, are currently used to enable the safe operation of these telecom networks for broadcast purposes. Nevertheless, in complex scenarios which involve the cascading of network connections, their use may introduce sig-nificant delays to the signals (i.e. latency) which makes interactive programmes difficult.

New concepts – such as the re-synchronization of clocks at all studio sites, using the Global Positioning System (GPS) – should be considered in order to minimize the delays, even within synchronizers. It should be emphasized that these new synchronization concepts offer potential improvements when compared with the existing practice of using free-running studio clocks throughout the network.

The adoption by broadcasters of new network technologies may well result in improvements in functionality and costs. However, broadcasters should also be made aware of the risks that these new network technologies may present to current broadcast applications.

With regard to the new digital networks, this document offers EBU member organizations:

- a comprehensive description of the possible scenarios for the primary distribution of MPEG-2 signals;
- the requirements and parameters for safe operation;
- guidelines for the transport of MPEG-2-based signals.

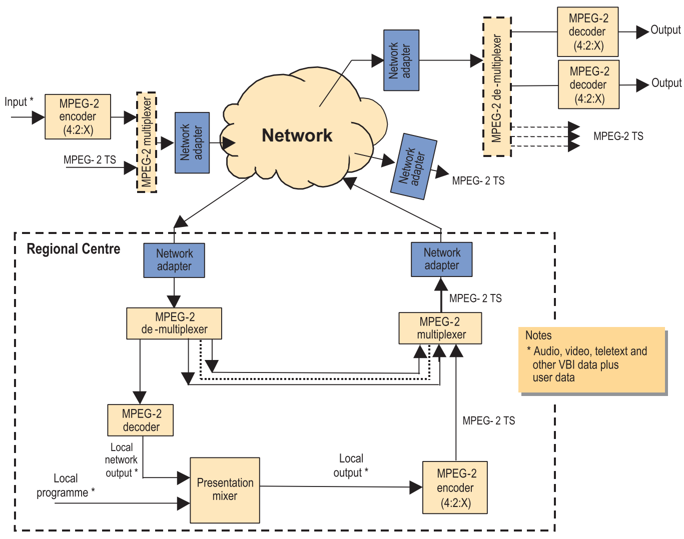
Chapter 1
Analogue television signals

1.1. Terrestrial distribution networks for the analogue environment

1.1.1. Reference model

The reference model for terrestrial distribution within the analogue environment is summarized in *Fig. 1.1*. Television broadcast signals are conveyed between two ends of a link, via a telecommunications network. The television broadcast signal may contain a number of components including:

- video;
- audio;
- teletext;



This figure takes account of the programme(s) routing. In addition, communication facilities for co-ordination, monitoring and control are necessary. These facilities could be organized using the same network.

Figure 1.1
Reference model for the analogue environment.

- non-teletext signals which would normally be carried in the vertical blanking interval (VBI) of the video component;
- user data streams.

Prior to entry to the network, the television broadcast signal is compressed using an MPEG-2 encoder. The single-programme MPEG-2 transport stream generated by the encoder may be combined with other MPEG-2 transport streams in a multiplexer, and then a network adapter maps the resulting multiplexed MPEG-2 transport stream to a form that is suitable for the network.

At the receiving ends of the link, a network adapter extracts the required data stream from the network. If a multiplex of programmes was created at the input to the network, the adapted data stream is presented to a de-multiplexer to extract the individual MPEG-2 transport streams. These are then each passed to an MPEG-2 decoder for conversion into television broadcast signals.

Within a regional centre, there may be a need to switch between national and regional versions of a particular programme, i.e. *regional insertion*. In this case, the national variant of the programme is decoded at the regional centre and a mixer or switch is used to select between this and the regional version of the programme. The selected programme is then encoded and passed back onto the network in a similar manner to the original network variant of the programme.

1.1.2. *Operational requirements*

1.1.2.1. Interfaces

Video	**Signal format options:** *General case:* - PAL; - SECAM. *For specific cases:* - PALplus; - Analogue components (YUV / RGB); - Enhanced SECAM. **Lines in the vertical blanking interval:** - A facility should be provided to convey teletext, VITS and other user-specified VBI lines between the two ends of the link **Synchronization:** - For some specific applications, provision should be made for an analogue black-and-burst reference which is linked to the source (e.g. using GPS) at the receive end of the link.
Audio	**Signal format options:** - Analogue stereo: 0.5 dB bandwidth (40 Hz to 15 kHz); - Digital stereo: AES / EBU @ 32, 44.1 or 48 kHz sampling rate, with 16, 20 or 24 bits-per-sample. **Number of stereo channels:** - Minimum of 2. **Synchronization:** - Differential video-to-audio delay: +40ms / –60ms (a "+" sign means the audio is in advance of the video). See [1].

Optional user data channels	**Options:** ❋ Unidirectional, asynchronous data channel with RS232- or RS422-compatible interfacing. Data rate: up to 38.4 kbit/s. ❋ Unidirectional, synchronous data channel with RS232- or RS422-compatible interfacing. Data rate: up to 38.4 kbit/s.
MPEG-2 Transport Stream input	**Interface formats:** ❋ DVB-ASI: asynchronous serial MPEG-2 transport stream; ❋ DVB-SPI (LVDS) : synchronous parallel MPEG-2 transport stream [2]. **Packet length:** ❋ 188 bytes without forward error correction; ❋ 204 bytes with forward error correction. **Transport Stream data rate:** ❋ up to 10 Mbit/s per programme.
Optional facilities	❋ Bi-directional, asynchronous data channel, with RS232- or RS422-compatible interfacing. Data rate: up to 38.4 kbit/s. ❋ When the distribution network is used for radio in parallel with television, additional audio channels are required with an interface as defined above.
Network adapter	**Interface options (mapping):** ❋ ATM [3]; ❋ PDH [4]; ❋ SDH [5].

1.1.3. Operational performance

End-to-end (video input to video output) delay	**Required characteristics:** ❋ Constant, fixed and deterministic (specified during installation); ❋ Maximum values for different coding parameters. (It should be noted that if standard terrestrial telecommunication links and techniques (e.g. SDH, ATM) are used for the transport of the MPEG-coded signals, the main contribution to the end-to-end delay is the coding / decoding delay of the MPEG codecs. If satellite links are used, the additional delay due to the propagation delay on the satellite link has to be taken into account.) **Calculated values for the codec delay, assuming the use of the maximum buffer size specified in MPEG:** ❋ 4 seconds for 4:2:2P@ML and 3 Mbit/s coded video bit-rate; ❋ 2 seconds for 4:2:2P@ML and 8 Mbit/s coded video bit-rate; ❋ 1 second for MP@ML and 3 Mbit/s coded video bit-rate. Where the main delays of a MPEG-2 codec are: Buffer: MP@ML 1.835 Mbit ÷ coded video bit-rate 4:2:2P@ML 9.4 Mbit ÷ coded video bit-rate GOP: e.g. IBBP 160 ms Frame DCT usage: 40 ms Field comb for PAL decoding: 20 ms Processing delays: 20 ms (approximately) ATM adaptation (buffer, interleaver): 0.2 Mbit ÷ TS bit-rate

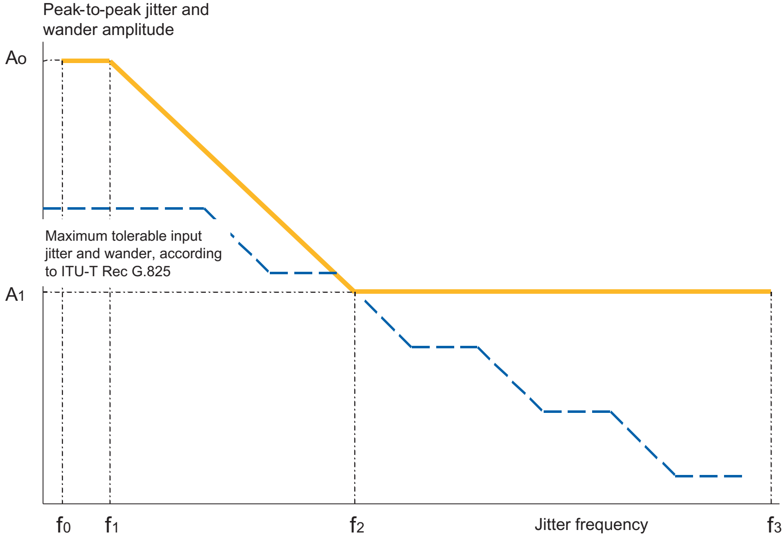
Recovery following interruption	Following an interruption on the link, a stable signal with no visible effects should be obtained at the remote-end video output, within 5 seconds of the link being re-established.
Reliability	It is expected that reliability will be at least as good as that achieved using comparable analogue contribution and distribution.

1.1.4. Signal quality parameters

The analogue signals delivered at the end of the transmission chain are subject to impacts from the transport mechanisms used in the network (e.g. jitter and wander). As these signals may be used for processing in a local studio (insertion of local programmes), the signal parameters of the video and audio signals should conform to the following requirements:

1.1.4.1. MPEG-TS

The MPEG-2 transport streams delivered by the network adapter at the receiving end should conform to the requirements set out in [6] and, in particular, the tolerances must lie within the limits defined in the template given in *Fig. 1.2*.



Parameter values for jitter and wander tolerance:

A_0 : 3 ms (maximum cell delay variation in ATM networks according to [7]

A_1 : 1 µs (maximum peak-to-peak PCR jitter due to multiplexing/remultipexing according to [6]

f_0 : 2 µHz

f_1 : 10 µHz

f_2 : 31 mHz

f_3 : 100 kHz

Figure 1.2
Maximum tolerable PCR jitter and wander for asynchronous serial interface.

1.1.4.2. Video signal parameters

Line jitter	❋ ± 10ns [a]
Wander	Under study by ETSI and ITU-T.
Subcarrier phase drift	❋ ≤ ± 20 ppb or 0.1 Hz/sec (referenced to the subcarrier frequency). See [8]

a. This value is defined in an ARD / ZDF specification for the transport of PAL signals.

1.1.4.3. Audio signal parameters

Jitter	No standard available
Wander	No standard available

1.1.5. Audio compression

Compression type	❋ MPEG-1 layer-II; ❋ MPEG-1 layer-III. For radio applications which demand low delay compression, other compression techniques may be used.
Compressed signal bit-rate	**The following range of data rates should be supported:** ❋ 32 – 384 kbit/s.

1.1.6. Video compression

Compression type	**The following options should be supported:** ❋ MPEG-2 Main Profile @ Main Level; ❋ MPEG-2 4:2:2 Profile @ Main Level.
Luminance resolution	**The following options should be supported:** ❋ Up to 720 x 576.
Phasing of luminance samples	The spatial phasing of the digital samples to be a configurable parameter where appropriate, and consistent between input and output (should be covered by target background grid and video window descriptors within the PES descriptors contained in the PMT of the generated stream)
Chrominance sub-sampling	**Depending upon the compression type, the following options should be supported:** ❋ 4:2:0 (using e.g. MP@ML which is defined up to 15 Mbit/s) ❋ 4:2:2 (using e.g. 4:2:2P@ML which is defined up to 50 Mbit/s)
Compressed signal bit-rate	A user-specified value in the range 4 – 8 Mbit/s.

1.1.7. *Guidelines for technical implementation*

1.1.7.1. Coding and decoding of video and audio

Nominal compressed SDTV video bit-rates for distribution.

Signals not subject to further cascaded MPEG-2 coding:	❋ 4 – 8 Mbit/s
Signals which will be subject to further cascaded MPEG-2 coding (this could be considered as a contribution application):	❋ 8 – 24 Mbit/s [a]

a. This bit-rate may be reduced if it is possible to make use of a "helper" signal (which minimizes the undesirable effects of cascaded coding). See [9] and [10].

1.1.7.2. Processing of VBI signals (teletext and non-teletext lines)

It is recommended that teletext, VITS, ancillary lines and lines containing timecode are conveyed in the manner defined in [11]. Note that ETS 300 472 [12] is relevant for system B teletext only,

1.1.7.3. Synchronization of video output

In respect of the synchronization of the video output at the remote end, three operational scenarios are possible.

1) The video output is synchronized to a black-and-burst reference supplied at the remote end, which is not necessarily synchronized to the video at the sending end. In this case, the decoder at the remote end may preserve the integrity of its output signal by skipping input frames and / or by displaying frames for multiple frame periods such that synchronization with the local black-and-burst reference is maintained.
2) The video output is synchronized to a black-and-burst reference supplied at the remote end, which is synchronized to the video at the sending end (by, for example, employing a GPS-based reference signal). In this case, the decoder at the remote end should not skip input frames and / or display frames for multiple frame periods in order to maintain synchronization with the local black-and-burst reference. Instead, it may apply an appropriate amount of buffering.
3) The video output at the remote end is free-running. In this case, the decoder should display all frames in the correct order.

1.1.7.4. Processing of optional user data

In the case of uni-directional data links, it is recommended that the user data be conveyed in MPEG-TS packets using ***transport_private_data*** (see Section 2.4.3.4, "Adaptation Field", in [6]). Note, however, it is thought that the transport of user data by this means is appropriate to the analogue scenario but not necessarily to the digital scenario where the information would probably be committed to a separate network channel.

1.1.7.5. Optional facilities

For the carrying of the additional audio channels, a processing based on the DVB architecture is possible. One audio stereo channel is conveyed in a MPEG-2 TS; this bitstream is multiplexed in an MPEG-2 multiplexer with the television programme(s) following the same path.

Another way to carry the additional audio channels following the same path is based on a specific multiplexing technique (e.g. DAB structure). It is recommended that access to the aggregated bitstream is made according to a standardized interface such as that defined in ITU-T Rec. X.21 [13] and ITU-T Rec. G.703 [14] with a bit-rate up to 2 Mbit/s. This bitstream is considered as a user data channel and is to be conveyed transparently.

In addition to the processing of user data within the TS as defined in *Section 1.1.7.4.*, a routing via a direct access to the network adapters is possible. Depending on the configuration, this access may allow bi-directional communication.

1.1.7.6. Processing of the input MPEG-2 transport stream

The configuration shown in *Fig. 1.3* is recommended for situations where it is required to carry the signal along with one or more programmes which have already been MPEG-2 encoded, and which are available in a transport-stream format. Here, the components of the input programme are first encoded in order to produce a single-programme transport stream (SPTS). An MPEG-2 multiplexer is then used to combine this with the additional transport stream inputs. The resulting MPTS, containing the input programme, is then fed to a network adapter.

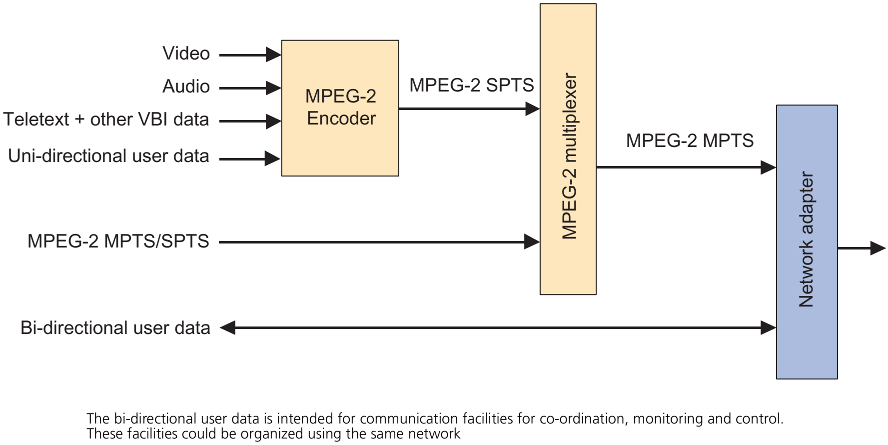


Figure 1.3
Processing of MPEG-2 transport stream input.

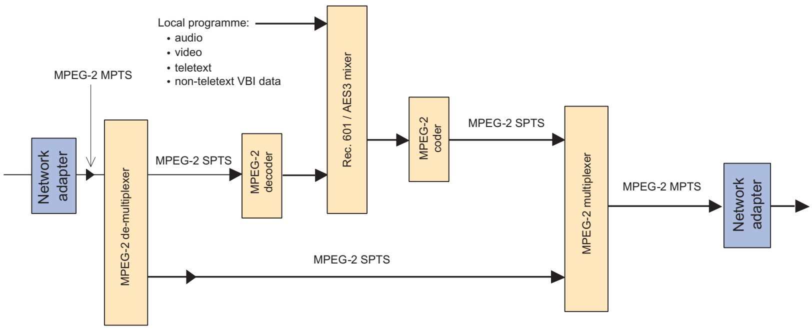
The re-multiplexer should be configured to re-author the Programme Association Table [6] from its input transport streams, and insert details of the new programme.

Depending upon the format of the transport streams, it may also be necessary for the re-multiplexer to modify the transport-stream packet identifiers (PIDs) and the programme number assigned by the MPEG-2 encoder, in order to avoid clashes with corresponding PIDs and programme numbers in the input transport streams. Where this may occur, there should be sufficient information for the remote end to uniquely locate the input programme in the MPTS, which is passed into the network. This could be achieved by having the re-multiplexer modify the Service Description Table (SDT) in its input transport streams. This will involve inserting details of the new programme and supplying a unique Service Name and Service Provider Name [15].

1.1.7.7. Regional insertion

The configuration shown in *Fig. 1.4* is recommended for situations where it is required to insert local variants of a programme into an MPTS at the remote end of a link, . Here, an MPEG-2 MPTS is delivered to the remote centre along with the programme material for which a revised regional version is required. Both of these are extracted from the network. The programme is decoded and the decoder's output is mixed with a locally-generated version of the programme. The resulting programme is encoded in order to generate an SPTS. An MPEG-2 multiplexer is then used to combine this with the MPTS extracted from the network. The resulting transport stream is then passed back onto the network.

In order to maintain the quality of the encoded video which results from this cascading process, one of the following approaches should be adopted:



This figure takes account of the programme(s) routing. In addition, communication facilities for co-ordination, monitoring and control are necessary, and these facilities could be organized using the same network.

Figure 1.4
Regional insertion.

- a sufficiently high bit-rate (at least 8 Mbit/s) should be employed to convey the encoded video to the Regional Centre;
- a “helper” signal (which minimizes the undesirable effects of cascaded coding) should be employed [9][10].

The re-multiplexer should be configured to re-author the Programme Association Table (PAT) [6] from its input MPTS, and insert details of the new programme.

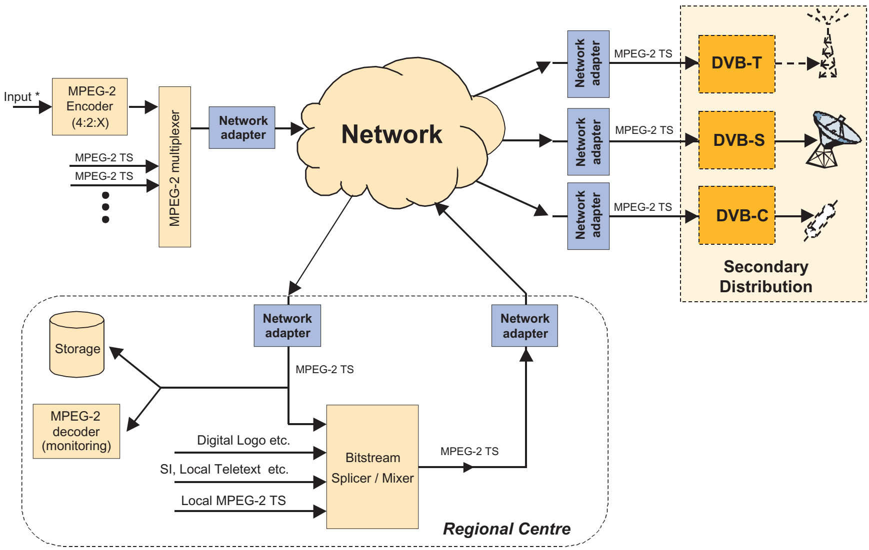
Depending upon the format of the input MPTS, it may also be necessary for the re-multiplexer to modify the transport-stream packet identifiers (PIDs) and programme number assigned by the MPEG-2 encoder, in order to avoid clashes with corresponding PIDs and programme numbers in the input MPTS. Where this may occur, there should be sufficient information for the remote end to uniquely locate the input programme in the MPTS, which is passed into the network. This could be achieved by having the re-multiplexer modify the Service Description Table (SDT) in its input transport stream. This will involve inserting details of the new programme and supplying a unique Service Name and Service Provider Name [15].

Chapter 2
Digital television signals

2.1. Terrestrial distribution networks for the digital environment

2.1.1. Reference model

The reference model for terrestrial distribution within the digital environment is summarized in *Fig. 2.1*.



This figure takes account of the programme(s) routing. In addition, communication facilities for co-ordination, monitoring and control are necessary. These facilities could be organized using the same network.

Figure 2.1
Reference model for the digital environment.

2.1.2. Operational requirements

2.1.2.1. Interfaces

Video	**Signal format options:** ❋ SDI [16]; ❋ Any additional programme-related data (e.g. teletext), if inserted into the SDI signal, should be in ancillary data format [17].
Audio	**Signal format options:** ❋ Digital stereo: AES/EBU @ 32, 44.1 or 48 kHz sampling rate, with 16, 20 or 24 bits per sample; ❋ Embedded audio according to [18]. **Number of stereo channels:** ❋ Minimum of 2 **Synchronization:** ❋ Differential video-to-audio delay: +40ms / –60ms (a "+" sign means the audio is in advance of the video). See [1].
Optional facilities	❋ Bi-directional, asynchronous data channel, with RS232- or RS422-compatible interfacing. Data rate: up to 38.4 kbit/s.
MPEG-2 Transport Stream input	**Interface formats:** ❋ DVB-ASI: asynchronous serial MPEG-2 transport stream; ❋ DVB-SPI: synchronous parallel MPEG-2 transport stream. **Packet length:** ❋ 188 bytes without forward error correction; ❋ 204 bytes with forward error correction. **Transport Stream data rate:** ❋ Up to 68 Mbit/s (the maximum bit-rate for a transponder, given in EN 300 421 [19]).
MPEG-2 Transport Stream composition	❋ Video :- MPEG-2 compressed; ❋ Audio :- MPEG-1 / MPEG-2; ❋ Teletext :- Ref. [12]; ❋ Subtitles / closed-captioning :- Ref. [20]; ❋ Service information (SI) :- Ref. [15]; ❋ User data :- Ref. [21].
Network adapter	**Interface options (mapping):** ❋ ATM [3]; ❋ PDH [4]; ❋ SDH (such as ITU-T Rec. J.132 [5])

2.1.3. Operational performance

Delay	**Required characteristics:** ❋ Constant, fixed and deterministic (specified during installation); ❋ Maximum values for different coding parameters. (It should be noted that if standard terrestrial telecommunication links and techniques (e.g. SDH, ATM) are used for the transport of the MPEG-coded signals, the main contribution to the end-to-end delay is the coding / decoding delay of the MPEG codecs. If satellite links are used, the additional delay due to the propagation delay on the satellite link has to be taken into account.) **Calculated values for the codec delay, assuming the use of the maximum buffer size specified in MPEG:** ❋ 4 seconds for 4:2:2P@ML and 3 Mbit/s coded video bit-rate; ❋ 2 seconds for 4:2:2P@ML and 8 Mbit/s coded video bit-rate; ❋ 1 second for MP@ML and 3 Mbit/s coded video bit-rate. Where the main delays of a MPEG-2 codec are: Buffer: MP@ML — 1.835 Mbit ÷ coded video bit-rate 4:2:2P@ML — 9.4 Mbit ÷ coded video bit-rate GOP: e.g. IBBP — 160 ms Frame DCT usage: 40 ms Processing delays: 20 ms (approximately) ATM adaptation (buffer, interleaver): 0.2 Mbit ÷ TS bit-rate
Recovery following interruption	Following an interruption to the link, a valid [a] MPEG-2 TS should be available at the output of the (remote) Network Adapter, within 5s of the link being re-established.
Reliability	It is expected that reliability will be at least as good as that achieved using comparable analogue contribution and distribution.

a. Valid = fulfilling the signal quality parameters given in *Section 2.1.4.*

2.1.4. Signal quality parameters

The MPEG-transport streams delivered by the network adapter at the receiving end should conform to the requirements set out in [6], and in particular, the tolerances must lie within the limits defined in the template given in *Fig. 1.2*.

2.1.5. Audio compression

Compression type	❋ MPEG-1 layer-II; ❋ MPEG-1 layer-III.
Compressed signal bit-rate	**The following range of data rates should be supported:** ❋ 32 – 384 kbit/s.

2.1.6. Video compression

Compression type	**The following options should be supported:** ❋ MPEG-2 Main Profile @ Main Level; ❋ MPEG-2 4:2:2 Profile @ High Level.

Luminance resolution	**The following options should be supported:** ❋ Up to 720 x 576; ❋ others to be defined.
Phasing of luminance samples	The spatial phasing of the digital samples to be a configurable parameter where appropriate, and consistent between input and output (should be covered by target background grid and video window descriptors within the PES descriptors contained in the PMT of the generated stream).
Compressed signal bit-rate	A user-specified value in the range 2 - 30 Mbit/s, as follows: ❋ SDTV distribution 2 - 8 Mbit/s; ❋ HDTV distribution 15 - 30 Mbit/s.

2.1.7. Guidelines for technical implementation

2.1.7.1. Benefits of using a digital environment

The widespread adoption of the DVB system makes the digital distribution of broadcast signals to the consumer's home a more and more common practice. Digital television and radio production facilities are also becoming much more commonly used. The use, therefore, of a digital environment for the contribution and primary distribution of broadcast signals is also becoming a real requirement – both to maintain the quality of services across the complete chain from the broadcaster to the consumer, and also to maximize the efficient use of transport capacity. A major advantage of going to an **all-digital** environment is the avoidance of cascaded coding / decoding processes.

Other advantages are:

- handling of programme bouquets;
- direct feeding of the different DVB secondary distribution chains;
- general infrastructure to feed all kinds of DVB secondary distribution media;
- telecommunications networks can be used for the transport, based on different physical and transport layers.

2.1.7.2. DVB references

a) The DVB "cookbook"

DVB environments – including the transport of DVB signals (i.e. primary and secondary distribution) – are considerably complex. In order to cope with all the system aspects, many different documents have to be taken into consideration when planning the services or designing the equipment. As a starting point for a more detailed study of the system parameters, a DVB document *"A Guideline for the Use of DVB Specifications and Standards"* is available [22]. It presents an overview of existing DVB documents and is regarded as a "cookbook", in that it lists all the ingredients needed for setting up a DVB system.

The DVB "cookbook" provides appropriate references to relevant ETSI documents as well as to basic ISO / IEC documents on the generic coding of moving pictures and associated audio information. The "cookbook" also includes short chapters on baseband processing, transmission, conditional access, interactive services and other relevant issues.

b) DVB and telecommunications networks

Telecommunications networks will play an important role in connecting, for example, the playout centre of a broadcaster to the satellite uplink station, perhaps in another city. Different types of networks will be used for this purpose (e.g. PDH, SDH, ATM etc.). The DVB Project has designed an interface which will be used for connecting the world of DVB signals to PDH networks (described in [4]). A comparable interface for connecting to SDH networks is described in [5].

c) *Testing and evaluation of DVB systems*

ETSI document ETR 290 [23] offers useful guidelines for the testing and evaluation of DVB systems. It has been designed to enable us to distinguish meaningful measurements from useless ones, and to help us to understand how the measuring should be carried out.

2.1.7.3. Processing of the input MPEG-2 transport stream

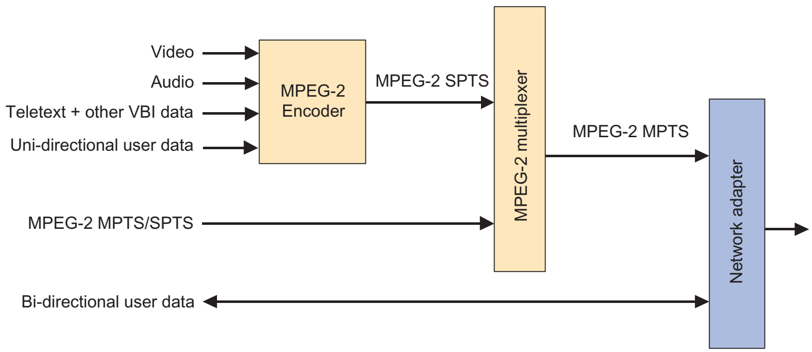
The configuration shown in *Fig. 2.2* is recommended for situations where we require a signal to be carried along with one or more other programmes which have already been MPEG-2 encoded, and which are available in a transport-stream format. Here the components of the input programme are firstly encoded in order to produce a single-programme transport stream (SPTS). An MPEG-2 multiplexer is then used to combine this with the additional transport stream inputs. The resulting multiple-programme transport stream (MPTS) – containing our input programme – is then fed to a network adapter.

2.1.7.4. Transport of an MPEG-2 TS in distribution networks

The transport of DVB signals – for example, (i) from a broadcaster's playout centre to the satellite uplink station for a DVB-S service, or (ii) from a central programme provider to local stations where a local programme may periodically be inserted into the DVB signal stream – will in many cases include the use of telecommunications networks. Different types of transport networks (e.g. PDH, SDH, ATM etc.) may be used for this purpose. In order to map / demap our DVB signals onto / from these telecommunication networks, appropriate network adaptation techniques have been defined and standardized.

By way of example, a network adapter has been defined by DVB (described in [4]) for the transmission of MPEG transport streams between two DVB interfaces within PDH networks working at the ITU-T Rec. G.702 [24] hierarchical bit-rates of 1 544, 2 048, 6 312, 8 448, 34 368, 44 736 and 139 264 kbit/s.

A similar network adapter has been defined for the transmission of MPEG-2 transport streams between two DVB interfaces within SDH networks working at the ITU-T Rec. G.707 [25] hierarchical bit-rate of 155 520 kbit/s, or at a bit-rate of 51 840 kbit/s as given in [5].



The bi-directional user data is intended for communication facilities for co-ordination, monitoring and control. These facilities could be organized using the same network

Figure 2.2
Processing of MPEG-2 transport stream input.

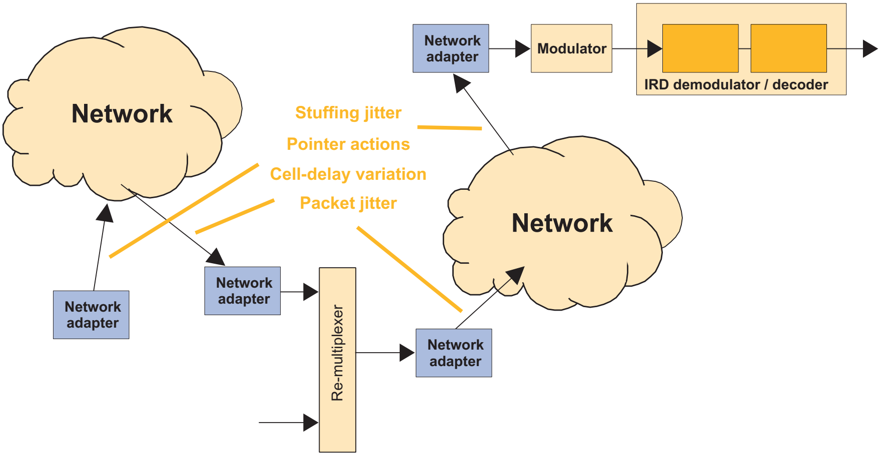


Figure 2.3
Potential timing impairments due to the distribution on telecom networks.

For the transmission of DVB signals over ATM-based B-ISDN networks, another adaptation mechanism is defined in ITU-T Rec. J.82 [3].

These adapters accept DVB signals via DVB-ASI interfaces and map one or more MPEG transport streams onto the standard telecommunication interface formats. At the receiving end, the MPEG transport streams are de-mapped from the network and provided again via a DVB-ASI interface.

a) Timing impairments from the network and their effect on DVB streams

Telecommunication networks which use PDH-, SDH-, ATM- or IP-based transmission techniques inevitably add timing impairments, i.e. jitter and wander, to signals carried on them. The reasons are, for example, stuffing jitter and pointer actions in PDH- and SDH-based networks, respectively, as well as cell delay variation in ATM-based networks, and packet jitter in IP-based networks. Similar effects to varying packet delays have to be taken into account if using IP-based networks.

Potential timing impairments – due to the primary distribution of DVB signals using telecommunication networks – are indicated in *Fig. 2.3*.

The potential effects may include increased TS-PCR jitter and wander, increased phase modulation (COFDM modulator) as well as video jitter and wander at the output of an IRD. It should be mentioned in particular that the feeding of Single Frequency Networks (SFNs) may be affected by network impairments.

The requirements for jitter / wander values accepted by DVB equipment were not available at the time this report was prepared. These issues are still under study within DVB.

2.1.7.5. Regional insertion

a) Use of a bitstream splicer / mixer

The configuration shown in *Fig. 2.4* is recommended for situations where local variants of a programme have to be inserted into an MPTS at the remote end of a link. An MPEG-2 MPTS is delivered to the remote centre, along with the programme material for which a revised regional version is required. Both of these are extracted from the network. The transport stream which will have the local programme variant inserted into it is mixed with a

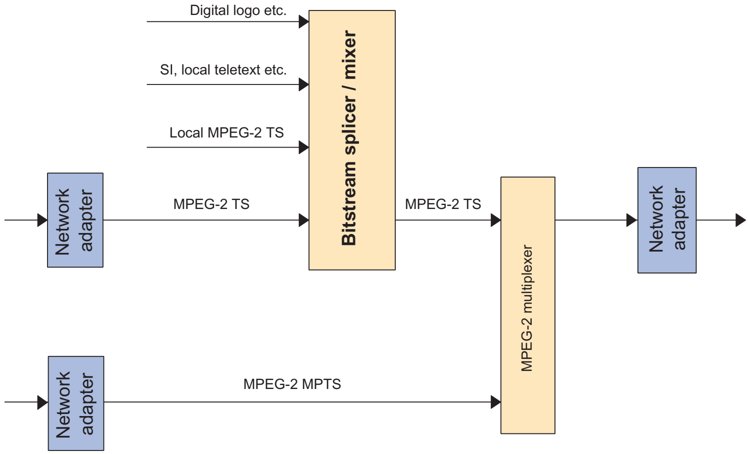


Figure 2.4
Regional insertion using a bitstream splicer / mixer.

locally-generated version of the programme in a bitstream splicer / mixer device. The resulting programme (SPTS) is then multiplexed in an MPEG-2 multiplexer in order to combine it with the MPTS extracted from the network. The resulting transport stream is passed back onto the network.

It should be mentioned that the use of a bitstream splicer / mixer avoids the cascading of decoding /recoding processes, and preserves the picture quality as delivered by the incoming MPEG-2 TS. In this case, any bit-rate overhead for cascading may not be necessary in the feeding programme signal.

b) Fallback solution

If no bitstream splicer / mixer is available, the configuration shown in *Fig. 2.5* could be used for regional insertion of a local programme.

Here, an MPEG-2 MPTS is delivered to the remote centre along with the programme material for which a revised regional version is required. Both of these are extracted from the network. The programme is decoded and the decoder's output is mixed with a locally-generated version of the programme. The resulting programme is encoded in order to generate an SPTS. An MPEG-2 multiplexer is then used for combining this stream with the MPTS extracted from the network. The resulting transport stream is passed back onto the network.

In this case the consequences of cascading the coding schemes should be considered.

Where a signal may be subject to multiple MPEG-2 coding operations in cascade, there is a risk of deterioration of the quality of the decoded signal. In order to overcome this, the decoder at the remote end may optionally output "helper" signals which contain information on how the signal at its input had been compressed. These "helper" signals can then be employed in subsequent MPEG-2 coding operations in order to minimize the undesirable effects of unassisted cascade coding. The main benefit of such "helper" signals is that, for a final target signal quality, they allow signals which will be subject to subsequent coding to be transferred at lower bit-rates than would be required if "helper" signals were not employed.

In the case of video, standards for this "helper" information, and how it can be conveyed through 4:2:2 component digital interfaces, are being defined by the SMPTE [9][10]. It is recommended that these standards be adopted for "helper" signals.

In order to maintain the quality of the encoded video which results from this cascading process, one of the following approaches should be adopted:

- a sufficiently high bit-rate (at least 8 Mbit/s) should be employed to convey the encoded video to the Regional Centre;
- a "helper" signal (which minimizes the undesirable effects of cascaded coding) should be employed [9][10].

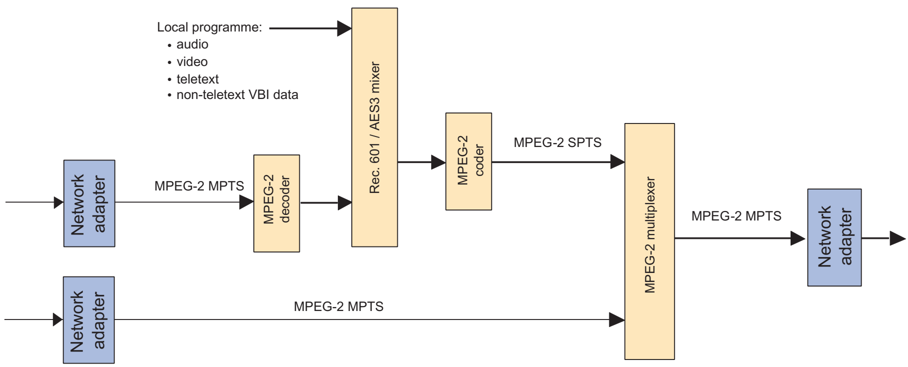
The re-multiplexer should be configured to re-author the Programme Association Table [6] from its input transport streams, and insert details of the new programme.

Depending upon the format of the transport streams, it may also be necessary for the re-multiplexer to modify the transport-stream packet identifiers (PIDs) and programme number assigned by the MPEG-2 encoder, in order to avoid clashes with corresponding PIDs and programme numbers in the input transport streams Where this may occur, there should be sufficient information for the remote end to uniquely locate the input programme in the MPTS, which is passed onto the network. This could be achieved by having the re-multiplexer modify the Service Description Table (SDT) in its input transport streams. This would involve inserting details of the new programme and supplying a unique Service Name and Service Provider Name [15].

2.1.7.6. Handling of MPEG-2 Tables and Service Information

A specification for the handling of MPEG-2 Tables and Service information is given in [15]. That document specifies the Service Information data which forms a part of DVB bitstreams for the assistance of users in selection of services and/or events within the bitstream, and for the automatic configuration of the Integrated Receiver Decoder (IRD) for the selected service. SI data for automatic configuration is mostly specified within ISO/IEC 13818-1 [6] as Programme Specific Information (PSI).

It should be mentioned here that it is important to comply with the rules and information given in these documents, particularly when regional programme content is inserted in programme streams.



This figure takes account of the programme(s) routing. In addition, communication facilities for co-ordination, monitoring and control are necessary. These facilities could be organized using the same network.

Figure 2.5
Regional insertion using decoding / recoding.

Chapter 3
Network requirements

3.1. Physical layer

Transport media available today for the distribution of MPEG-2 transport streams within local-area networks (LANs), metropolitan-area networks (MANs) and wide-area networks (WANs) include:

	LAN (e.g. Intra-studio)	MAN (e.g. Campus)	WAN
Twisted pair	X		
Coaxial cable	X	X	
Fibre optics	X	X	X
Satellite			X
Microwave		X	X

3.2. Transport / network layers

For the different physical transport media, the network technologies given below can be used:

	PDH	SDH	ATM [a]	DVB [b]
Twisted pair			X	
Coaxial cable	X	X	X	X
Fibre optics [c]	X [d]	X	X	X
Satellite	X	X	X	X
Microwave	X	X	X	X

a. For the transport of ATM cells at the present time, usually SDH or PDH is used as the transport/network layer.
b. See Ref. [26].
c. Including WDM technology.
d. For PDH, fibre optic interfaces are not standardized.

Further transport / network layers – such as IP, SDTI and Fibre Channel – are currently under study.

Considering the existing standards for the transport of MPEG-2 TSs on different networks, the user bit-rates are given in the following table:

	PDH [a] [kbit/s]	SDH [kbit/s]	ATM [kbit/s]	DVB [Mbit/s]
User data rates for the MPEG-2 TS	E4: 139 264 DS3: 44 736 E3: 34 368 E1: 2 048	STM-1(C4): 149 760 STM-1(C3): 48 384 STM-1(C12): 2 176	Up to 128 655 (STM-1)	e.g. 18 to 68 [b] depending on the channel adaptation and transponder bandwidth
Standards	ITU-T G.Series, ETSI	ITU-T G.Series, ETSI	ITU-T I.Series, ITU-T J.Series (ATM-Forum)	ITU-T J.Series, ETSI

a. The user data rates may be lower due to framing overhead.

b. According to EN 300 421 [19].

3.3. Mapping of MPEG-2 transport streams into PDH, SDH and ATM

Recommendation ITU-T J.131 [4] provides the requirements for a piece of equipment called a “PDH network adapter” for the transport of MPEG-2 signals over PDH networks. It describes the necessary operations to adapt the MPEG-2 transport stream into the interfaces of PDH networks, and the functional characteristics associated with this equipment.

PDH hierarchy	PDH link transmission capacity	Transmission capacity for MPEG-2 TS / RS-coded MPEG-2 TS [a]
E1	2 048 kbit/s	1 649 kbit/s
E3	34 368 kbit/s	29 140 kbit/s
DS3	44 736 kbit/s	37 980 kbit/s
E4	139 264 kbit/s	118 759 kbit/s

a. Depending on the application, these figures may be slightly reduced.

Recommendation ITU-T J.132 [5] provides the requirements for a piece of equipment called an “SDH network adapter” for the transport of MPEG-2 signals over SDH networks. It describes the necessary operations to adapt the MPEG-2 transport streams into an STM-1 or sub-STM-1 frame, and the functional characteristics associated with this equipment.

Type of container	Container capacity	Example of transmission capacity for MPEG-2 TS / RS-coded MPEG-2 TS [a]
C-4	149 760 kbit/s	128 655 kbit/s
C-3	48 384 kbit/s	41 565 kbit/s
C-2	6 784 kbit/s	5 828 kbit/s
C-12	2 176 kbit/s	1 869 kbit/s
C-11	1 600 kbit/s	1 374 kbit/s

a. Depending on the application, these figures may be slightly reduced.

3.4. Network performance

	PDH	SDH	ATM (Ref. [7])	DVB serial data interface
Jitter	See Ref. [27]	See Ref. [28]		
Wander	See Ref. [27]	See Ref. [28]		
Cell delay variation	N/A	N/A	1.5 ms	N/A
Quality of service	YES	YES	YES	YES
Availability	See Ref. [29]	See Ref. [29]	See Ref. [28]	?
Error performance	See Ref. [30]	See Ref. [30]	See Ref. [29]	?
Cell loss ratio	N/A	N/A	4.8×10^{-8}	N/A
Cell error ratio	N/A	N/A	5.2×10^{-7}	N/A
End-to-end delay	Network dependent	Network dependent	Network dependent + up to 10 ms for CTD	Network dependent

Chapter 4
Bibliography

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[4] ITU-T Rec. J.131: **Transport of MPEG-2 signals in PDH networks**.

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[10] SMPTE standard 319M-2000: **Transporting MPEG-2 Recoding Information through 4:2:2 Component Digital Interfaces**.

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[19] EN 300 421: **Digital Video Broadcasting (DVB); Framing structure, channel coding and modulation for 11/12 GHz satellite services**.

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[29] ITU-T G.827: **Availability parameters and objectives for path elements of international constant bit-rate digital paths at or above the primary rate**.

[30] ITU-T G.826: **Error performance parameters and objectives for international, constant bit-rate digital paths at or above the primary rate**.

2. ITU-R BT.799: **Interfaces for digital component video signals in 525-line and 625-line television systems operating at the 4:4:4 level of Recommendation ITU-R BT.601 (Part A).**

Chapter 5
Glossary of terms

4:2:0	Luminance and chrominance sampling used in D2-MAC, SECAM and MPEG-2 MP@ML (DVB)
4:2:2	Luminance and chrominance sampling used in MPEG-2 422P@ML (e.g. Eurovision Network)
ADM	Add Drop Multiplexer (used in synchronous networks)
ASI	Asynchronous Serial Interface
ATM	Asynchronous Transfer Mode
BER	Bit Error Ratio
BW	Bandwidth
CCR	Carrier and clock recovery sequence
CDV	Cell Delay Variation (ATM specific)
CER	Cell Error Ratio (ATM specific)
CLR	Cell Loss Ratio (ATM specific)
CMF	Co-ordination and Monitoring Functions
CTD	Cell Transfer Delay (ATM specific)
Digital-S	Trade name of JVC
DTH	Direct To Home
DV	Digital Video
DVB	Digital Video Broadcasting
DVB-C	Digital Video Broadcasting via Cable
DVB-S	Digital Video Broadcasting via Satellite
DVB-T	Digital Video Broadcasting via Terrestrial Transmitters
DVC PRO	Trade name of Panasonic – a digital tape format
DVCAM	Trade name of Sony Corporation a digital tape format
EUTELSAT	European Telecommunications Satellite Organization
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
GOP	Group of Pictures
GPS	Global Positioning System
HDTV	High Definition Television (something approximating four times the spatial resolution of SDTV)
IDS	Insertion Data Signal
Interleaver	A device for interleaving, or shuffling samples such that adjacent samples are separated during transmission, and are therefore less likely to be corrupted by bursts of errors
ITS	Insertion Test Signal

Jitter	The short-term variations of the significant instants of a digital signal from their ideal positions in time
MCPC	Multiple Carrier Per Channel (this can describe a multiplex)
MPEG	Moving Picture Expert Group
MPEG-2	A family of (lossy) compression systems developed by MPEG for coding video at bit-rates between 1.5 Mbit/s - 80 Mbit/s, encompassing SDTV to HDTV applications
MPTS	Multiple Programme Transport Stream.
MUX	Multiplexer
N/A	Not Applicable
NA	Network Adapter
PAL	Phase Alternating Line – a composite analogue television format
PALplus	Development of PAL with improved colour and vertical resolution and (often) 16:9 aspect ratio
PCR	Programme Clock Reference
PDC	Programme Delivery Control
PDH	Plesiochronous Digital Hierarchy
PID	Programme identifier
ppb	Parts per billion (10^9)
PSI	Programme Specific Information
QPSK	Quaternary Phase Shift Keying
SCPC	Single carrier per channel
SDH	Synchronous Digital Hierarchy
SDI	Serial Digital Interface (Rec. ITU-R BT.656-4)
SDTI	Serial Data Transport Interface
SDTV	Standard Definition Television (something approximating to the quality of 525 / 625-line television)
SI	Service Information
SPI	Synchronous Parallel Interface
SPTS	Single Programme Transport Stream
SX	Trade name of Sony Corporation – a digital tape format
TDMA	Time Division Multiple Access
TS	Transport Stream
UW	Unique Word
VPS	Video Programme System
Wander	The long-term variations of the significant instants of a digital signal from their ideal positions in time