EBU – TECH 3319

# The WAN Roadmap:

# The use of WANs to carry audiovisual content

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# Foreword

This document contains:

- Detailed descriptions of currently in broadcaster's contribution and distribution networks deployed wide area network technologies. They can be found in chapter 2 and should mainly be relevant to technical readers with some background in networking. Non-technical readers will find a short pro/con summary of most of the technologies at the end of each sub-section in chapter 2.
- Trends and new wide area network technologies that appear in the near future and may be of interest for future broadcast networks. They can be found in chapter 3 and are relevant for technical readers, especially when they are involved in the planning of future networks.
- Use cases, examples of real deployed wide area broadcast networks. These example networks can be found in chapter 4 and are relevant for both technical and non-technical readers and show the experiences and structures of networks deployed by/for broadcasters in different European countries.

This document can be used as a basis for the evaluation of WAN technologies for actual and planned contribution and distribution networks.

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# The use of WANs to carry audiovisual content

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## 1. Introduction

With the growing deployment of large fibre-based networks and the rising number of carriers offering wide area connectivity, Wide Area Networks (WANs) are already an important broadcasting tool for contribution and primary distribution of audio and video content. There are a number of different technologies that may be used in constructing a WAN, such as:

- ATM (Asynchronous Transfer Mode),
- SDH/SONET (Synchronous Digital Hierarchy/Synchronous Optical NETwork),
- DTM (Dynamic Synchronous Transfer Mode),
- WDM (Wave Division Multiplex),
- IP (Internet Protocol),
- MPLS (Multi-Protocol Label Switching), and
- GMPLS (Generalised Multi-Protocol Label Switching).

All vendors and carriers naturally say that their solutions are the best in all circumstances. These claims are often made in ignorance of broadcasters' special requirements of real-time transmissions (especially bi-directional transmissions). Such an application, requiring a high Quality of Service (QoS) and a known constant (low) latency, belongs to the most critical applications a network has ever seen. Testing a network infrastructure often shows disappointing results when judged against "our" very special requirements.
Mature technologies based on ATM and SDH still currently dominate broadcasters' networks for real-time applications, but IP-based networks (well suited for file transfer) and new technologies based on DTM, next generation SONET/SDH and several types of MPLS are becoming increasingly important.

Some EBU members have already gained experience with different WAN solutions; some are carrying out intensive tests on new technologies. This document describes the different technologies, the experiences that broadcasters have had with their current WANs together with the new emerging WAN technologies and their possible impact on broadcast networking.

Readers will learn which network structure and technology really fulfils broadcasters' requirements and moreover, what are the exact requirements that are so special for broadcasting applications.

# 2. Current WAN technologies used in broadcasters' networks

## 2.1 SDH/SONET

### 2.1.1 SDH basics

SDH was defined at the end of the 1980s by the ITU-T (Recommendations G.707, G.708, G.709 and several others) as an extension of the American digital hierarchy SONET, thus allowing a world harmonisation of the standards.

SDH allows the transport of a certain number of tributary streams having different bitrates. Multiplexing is achieved by inserting the tributary streams in pre-defined positions (Containers) to form a single stream. Add and drop functions allow the extraction of the desired tributary without demultiplexing the whole stream.

The first level of the SDH hierarchy, defined as STM-1 (Synchronous Transport Module 1), is a frame with a bitrate of 155.52 Mbit/s. The format of the frame is given in Figure 1. The frame repetition rate is 125 µs; each byte in the payload represents a 64 kbit/s channel (i.e. a phone call, at ITU G.711 PCM coding rates).

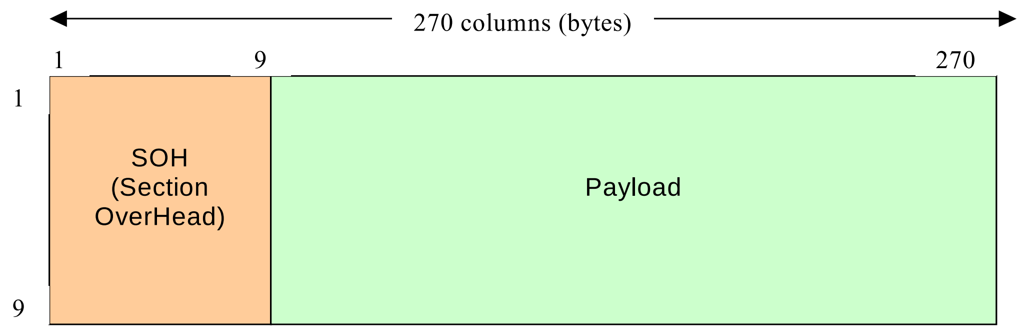


**Figure 1:- Schematic diagram of the STM-1 frame**

Higher levels of the hierarchy are also available, as reported in Table 1.

**Table 1: SDH hierarchy**

| SDH level | Bitrate (Mbit/s) |
|---|---|
| STM-0 (STS-1)[1] | 51.84 |
| STM-1 | 155.52 |
| STM-4 | 622.08 |
| STM-16 | 2488.32 |
| STM-64 | 9953.28 |

Various types of digital signal can be transported over SDH networks by mapping them into pre-defined tributary signals. For instance, PDH (Plesiochronous Digital Hierarchy) signals are mapped using a special Container (C-n). These containers are slightly larger than the payload; the remaining capacity is partly used for justification (stuffing) in order to equalise timing inaccuracies in the PDH signals. A Virtual Container (VC-n) is made up from the Container together with the POH (Path OverHead). This is transmitted unchanged over a path through the network.

[1] STS-1 (Synchronous Transport Signal) is the first level of the American SONET hierarchy, also called OC-1 (Optical Carrier) if transmitted over optical fibre.

Figure 2 shows a summary of possible mappings (yellow signifies those bitrates usually adopted for transport of audio/video signals).

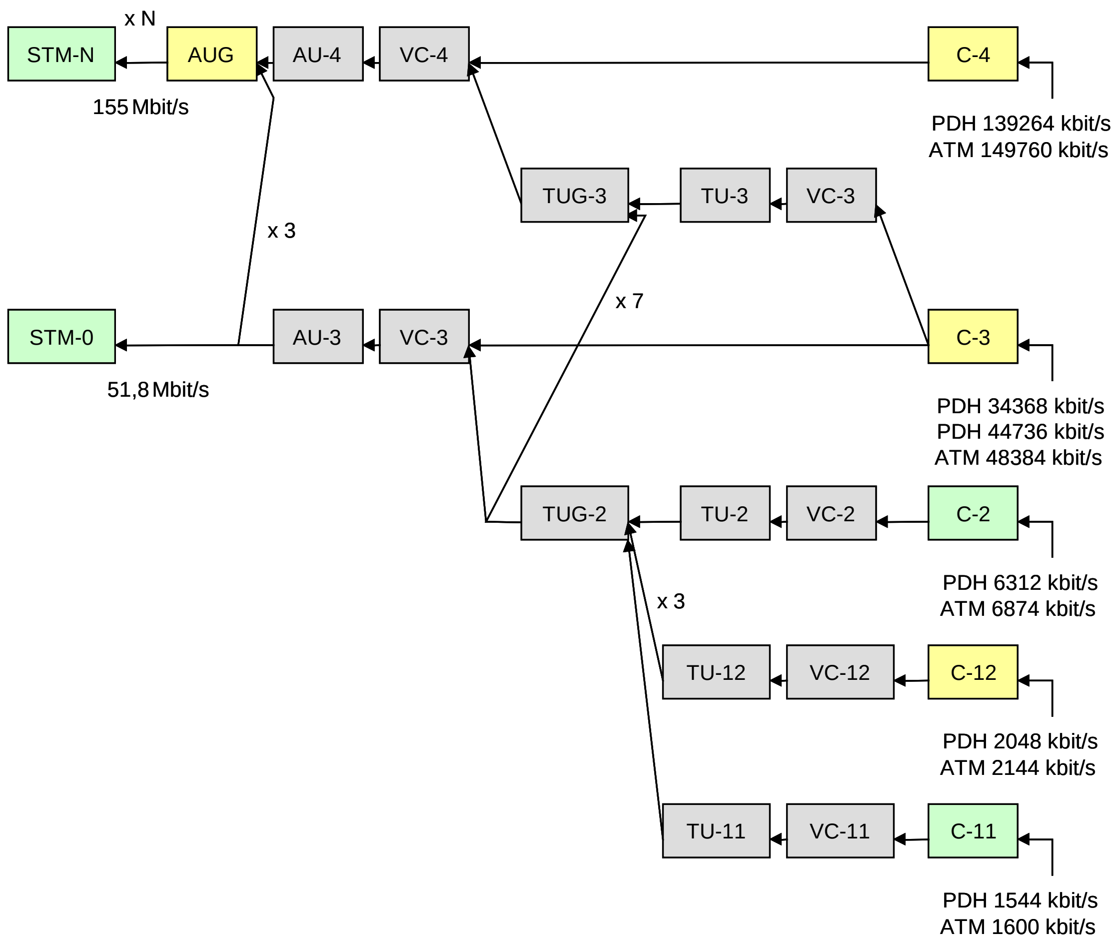


**Figure 2: Mapping in SDH**

Current SDH networks include the following elements:

- Regenerators, which regenerate the clock and amplitude relationships of the incoming signals, deriving the clock from the incoming data streams;
- Multiplexers (MUX), terminal equipments used to combine plesiochronous and synchronous input signals into higher bitrate STM-N signals;
- Add-Drop Multiplexers (ADM), used to switch lower level signals, extracting from or inserting into high speed SDH streams
- Digital Cross-Connect (DXC), with the widest range of functions, i.e. mapping of PDH tributaries into VCs as well as switching of various VCs.

The PDH hierarchy is shown below.

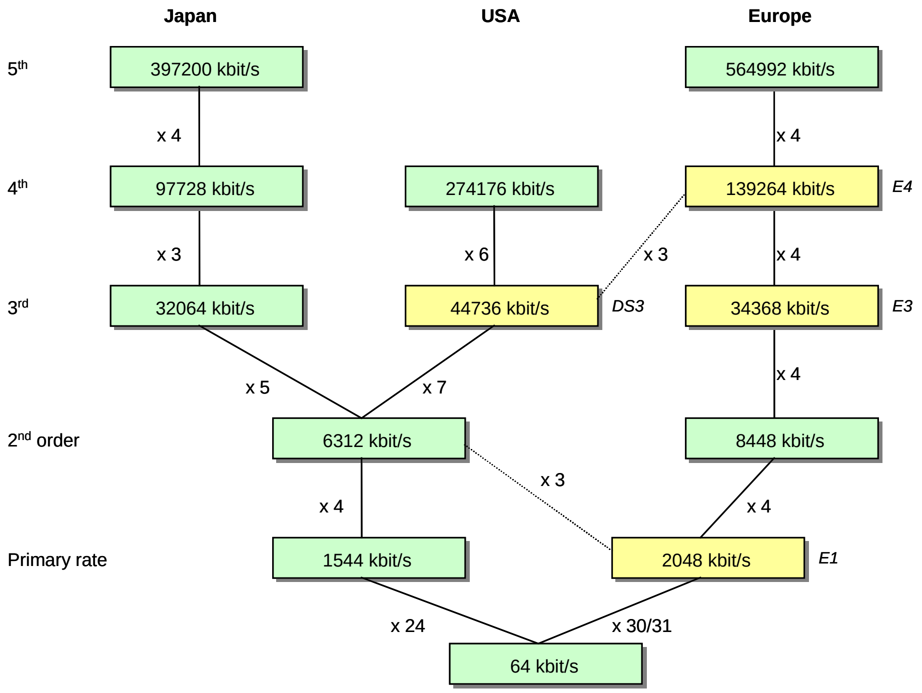


**Figure 3: The PDH hierarchy**[1]

### 2.1.2 Transport of video signals over SDH networks

Transmission of video streams over a SDH network requires a format adaptation.

This section deals only with direct mapping of video streams over SDH. Mapping using an intermediate switching layer (ATM, IP, etc.) is considered in other sections of this document.

Both uncompressed (SDI) or compressed (MPEG-2 TS) video streams can be transported over SDH networks:

- SDI signals (270 Mbit/s): a capacity of 2 x STM-1 per SDI signal are needed.
- MPEG-2 TS signals (i.e. ASI): the number of transported streams (i.e. in 1 x STM-1 capacity) depends on their bitrate, as shown in the following Table.

**Table 2: Possible allocation of MPEG-2 TS streams**

| SDH tributary | Max. gross MPEG-2 TS bitrate | No. of streams in 1 x STM-1 |
|---|---|---|
| VC3 (DS3) | 44.2 Mbit/s | 3 |
| VC4 (E4) | 137.6 Mbit/s | 1 |

An error protection scheme based on Reed-Solomon code RS(188,204) is generally applied, thus reducing the net TS bitrate in this case by about 8.5%.

The following block diagram illustrates this.

[1] Note that, even if the European hierarchy includes a 3rd order rate of 34 Mbit/s, (E3), the American 45 Mbit/s rate (DS3) is also often used in Europe for TV contribution applications, as its insertion in the SDH hierarchy implies a lower overhead.

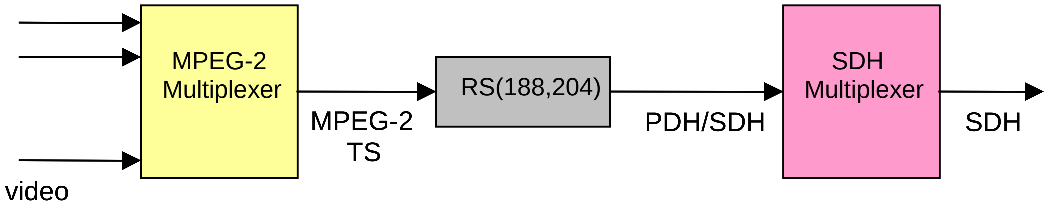


**Figure 4: Direct mapping of MPEG-2 TS over SDH networks**

This solution also allows:

- Transmission of more than one video stream in one SDH container (e.g. 2 video signals at 19 Mbit/s in the same VC3 stream); however these video streams can only be switched together in the SDH network (i.e. from the same source to the same destination), thus limiting the network flexibility.
- Transmission of streams at lower bitrates (e.g. a DVB-T bouquet, composed of 4-6 programmes, at 24 Mbit/s in a VC3 stream), but with a certain waste of capacity.

Also data streams can be transported over SDH networks, i.e. allocating fixed circuits over PDH tributaries, at bitrates according to the available hierarchy levels (i.e. 2 Mbit/s, 45 Mbit/s, etc.).

### 2.1.3 Pros and cons

| Pros | Cons |
|---|---|
| • SDH is a consolidated standard technology, and SDH networks are widely available all around the world | • Originally designed for two-way communications: some old equipment must be adapted in order to deal with uni-directional links |
| • Cheaper solution (if compared to solutions using intermediate switching layers) | • No automatic re-routing in case of link failure (N+1 protection or dual links in the network must be established) |
| • Transport of uncompressed or compressed video streams is possible | |
| • Point-to-point and point-to multipoint circuits | • Low bitrate flexibility, i.e. high overhead in case of bitrate not close to the VC bitrate |
| • Low end-to-end delay and jitter | • Low switching flexibility, i.e. only at the bitrates defined by the SDH/PDH hierarchies |
| • The network can be configured by a centralised Network Management System, allowing for a strict control of the circuits and the occupied bandwidth | • Automatic bandwidth allocation to data traffic, in case of available capacity, not possible |
| • Can be easily upgraded to NG-SDH, changing only the terminal nodes and not the backbone | • High circuit establishment times (i.e. tens of seconds in case of many ADMs involved along the path) |
| • In case of introduction of an intermediate switching layer (such as ATM, IP, DTM), the SDH physical layer can be kept | |

## 2.2 ATM

### 2.2.1 ATM basics

In 1986, ATM was chosen as the transmission technology for B-ISDN (see Figure 5) and its development was started in 1988. Since ATM is already well established and well known, only a short overview of its functionality is given here. A use case (the ARD HYBNET) is given in section 5.1.

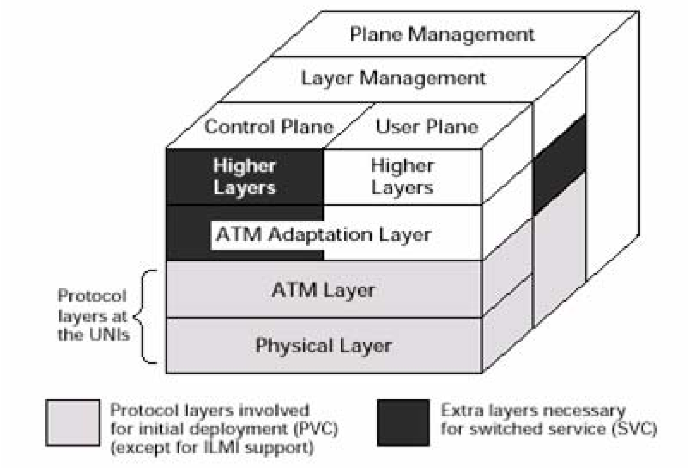


**Figure 5: B-ISDN Protocol Reference Layer model, (source: ITU-T)**

ATM is a fast packet switching with fixed length packets (cells of 53 byte length – see Figure 6). It is, in principle, a combination of elements of STM (Synchronous Transfer Mode) and PTM (Packet Transfer Mode).

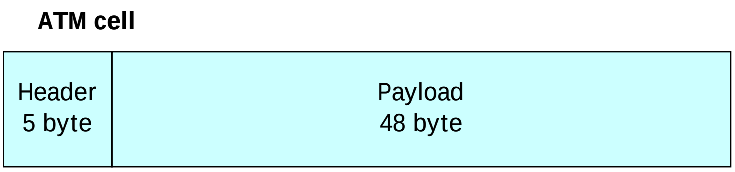


**Figure 6: ATM Cell**

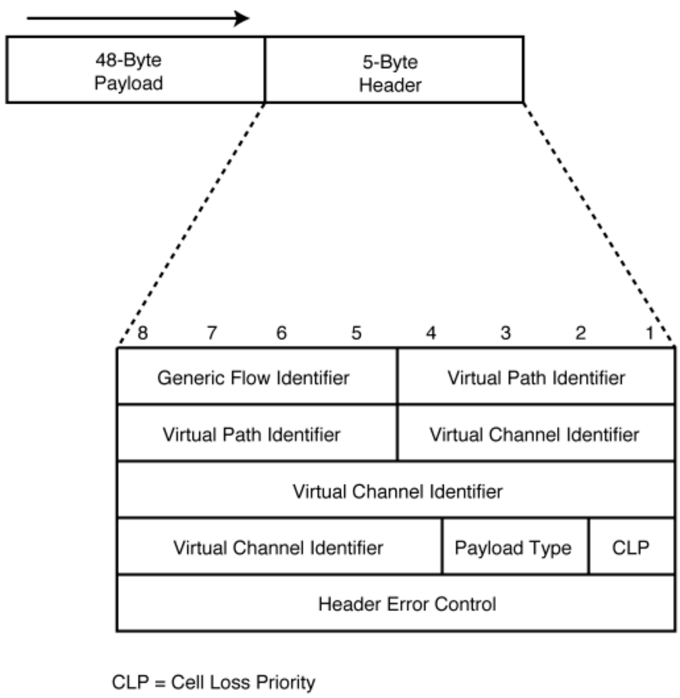


**Figure 7: ATM Cell Header**

The 5-byte header contains all necessary information to identify the ATM connection. The switching of cells in an ATM switch is based on the VPI/VCI, which identify a virtual connection. There are two types of connections:

- VCC: Virtual channel connection: Connection between two ATM end devices (with determined QoS), characterised by the VCI
- VPC: Virtual path connection: Connection between one or more end devices, characterised by the VPI, the QoS of the most demanding connection is valid

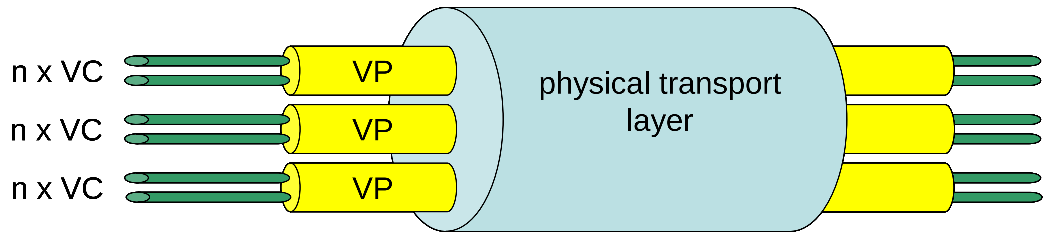


**Figure 8: ATM connections**

### 2.2.2 ATM Quality of Service

Packet networks can introduce certain kinds of signal impairment such as jitter, cell loss and bit errors. Since ATM uses quasi-statistical multiplexing in order to efficiently use network capacities, connections using the same physical path must be prioritised against each other in order to guarantee the QoS that each connection needs. In ATM, QoS parameters are established during connection set-up and are guaranteed during the whole duration of the connection.

In order to use ATM for the transport of application data, so-called ATM Adaptation Layers (AAL) are employed. These adapt the structure of the user data to ATM, cope with network impairments and are the link between the application and the ATM layer.

For real-time traffic (such as audio and video) needing a constant bitrate and timing compensation, ATM Adaptation Layer 1 (AAL 1) is used to cope with transmission errors and delay variations.

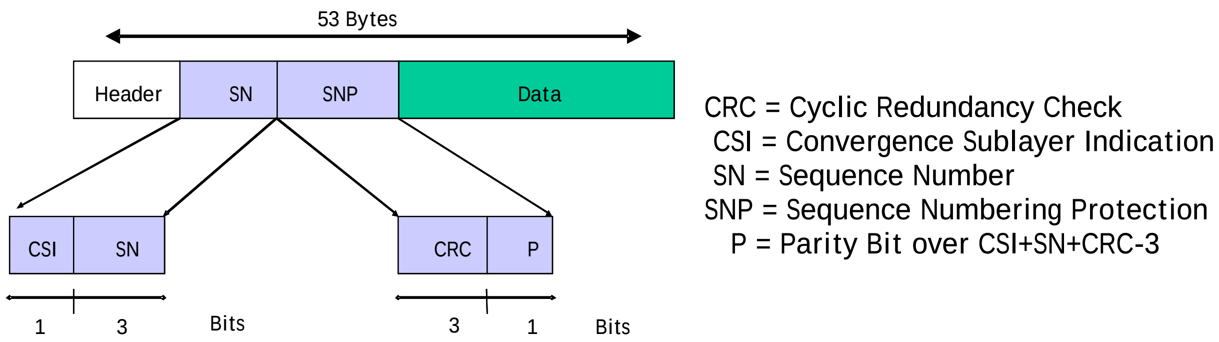


**Figure 9: AAL 1 Cell**

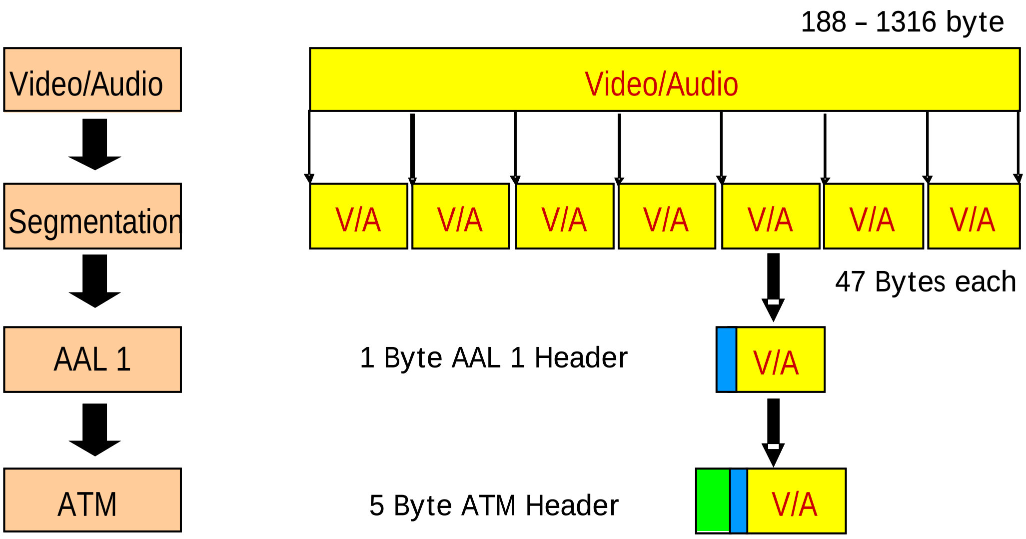


**Figure 10: Adaptation of MPEG-2 on ATM**

As an example, MPEG-2 adaptation is done by segmenting the MPEG-2 data into 47-byte parts, adding a sequence number field and the ATM header.

For non-real-time traffic such as file transfer and LAN traffic, AAL 5 is used.

The deployment of ATM is currently decreasing and some vendors have announced that they will no longer support their ATM products. Nevertheless, a huge amount of ATM is still used in backbones and a number of vendors still continue to produce ATM products.

### 2.2.3 Pros and cons

| Pros | Cons |
|---|---|
| • Provides guaranteed Quality of Service<br>• Low jitter and wander<br>• Provides almost every kind of broadcast audio and video interface<br>• Mature technology | • Some manufacturers will no longer support ATM products<br>• Complexity of transporting IP on ATM |

## 2.3 IP

### 2.3.1 IP Basics

IP stands for Internet Protocol; it is a packet delivery protocol in the family of network protocols. As it runs on Layer 3 (Network Layer) it is independent from the data transmission technology. Many IP adaptations exist for Layer 2 (Link Layer) networks such as Ethernet, ATM, SDH and DVB.

### 2.3.2 Delivery

Two main schemes of delivery exist in IP

- **Unicast:** for one-to-one delivery between hosts. Unicast is the simplest and most used delivery mode. Unicast is not appropriate for delivering the same content at the same time to a large number of hosts as the number of streams is equal to the number of hosts (one-to-one delivery).

  - For Unicast delivery, routing can be fixed or dynamic (choosing the best route or changing the route in case of link disruption). The main dynamic routing protocols are OSPF (Open Shortest Path First) for internal networks and BGP (Border Gateway Protocol) for larger networks.

- **Multicast:** for one-to-many or many-to-many delivery between hosts. Multicast is used in dedicated networks; it requires special signalling mechanisms and routing algorithms in order to control the delivery. Additionally, it requires special support and configuration of network routers. Currently only a very limited number of rather small isolated multicast-enabled networks exist.

  For multicast delivery the following protocols exist:

  - Group Membership Protocol (IGMP) - enables hosts to dynamically join/leave multicast groups; membership information is communicated to the nearest router (which must support multicast).
  - Multicast Routing Protocol (DVMRP, MOSPF, PIM) - enables routers to build a delivery tree between senders and receivers of a multicast group

### 2.3.3 IP packet

Figure 12 depicts the header of an IP packet (version 4, IPv4):

<table>
<tr><th>4-bit</th><th>8-bit</th><th colspan="2">16-bit</th><th colspan="2">32-bit</th></tr>
<tr><td>Ver.</td><td>Header Length</td><td colspan="2">Type of Service</td><td colspan="2">Total Length</td></tr>
<tr><td colspan="4">Identification</td><td>Flags</td><td>Offset</td></tr>
<tr><td colspan="2">Time to Live</td><td colspan="2">Protocol</td><td colspan="2">Checksum</td></tr>
<tr><td colspan="6">Source Address</td></tr>
<tr><td colspan="6">Destination Address</td></tr>
<tr><td colspan="6">Options and Padding</td></tr>
</table>

**Figure 12: IP packet header**

The maximum size of an IP packet is 64 kbyte. However it is important to consider the maximum frame size of the adaptation to the underlying data link layer (Layer 2). This is called Maximum Transmission Unit (MTU). If an IP packet is larger than the MTU it will be fragmented into multiple smaller IP packets. This is not optimal and should be avoided. In general on Ethernet networks the MTU is 1500 byte.

Most IP networks now use IP version 4 (IPv4). The next generation is IP version 6 (IPv6). It is designed for the Internet; in particular, the length of the addresses increase from 32 bits to 128 bits which means a huge number of available IP addresses. In some countries, IPv6 is already partly in use in mobile networks.

IPv6 adds many other enhancements, such as:

- New mechanisms for multicast delivery in the case of any source multicast (many-to-many)
- New mechanisms for IP mobility.

### 2.3.4 IP Network impairments

Among the main IP network impairments may be noted the following:

- **Packet losses:** These can be due to network congestion (e.g. buffer overruns in routers) but also transmission line errors. UDP and TCP transport protocols use a checksum to verify the integrity of the packet. In case of a single bit error the entire packet is discarded.
- **Out-of-sequence Packets:** This happens in networks with dynamic routing. In case of a route change some packets may arrive out of sequence at the receiving end because of different transmission path lengths.
- **Delay:** This is the latency due to the propagation time and buffering. Delay can change suddenly if dynamic routing is used (longer or shorter route).
- **Jitter:** This is the delay variation due to switching and routing time and also to sharing.

The magnitude of these impairments depends on how a network is managed regarding routing, transmission lines and sharing.

Impairments are negligible or absent when unshared, point-to-point IP links are used.

The Internet is an unmanaged (public) network, it uses different transmission lines and it is an interconnection of different networks with dynamic routing. In consequence, its IP impairments are substantial.

As noted below, IP network impairments can be partially controlled using Quality of Service mechanisms (QoS).

The ITU has issued a recommendation on network profiles defining classes with bounds on packet loss probability, delay and jitter. This recommendation (Y.1541) should help operators agree on classes of service and manufacturers to offer equipment scaled to these defined classes. An annex to Y.1541 has been proposed by the Video Service Forum to set out the network requirements to accommodate video over IP.

### 2.3.5 Quality of Service (QoS)

QoS mechanisms can be used to control IP network impairments. Two main approaches exist:

- “IntServ" (Integrated Services), whose principle is to reserve network resources by external signalling. This is the approach of the Resource ReSerVation Protocol (RSVP) that is defined in RFC2205.
- "DiffServ" (Differentiated Services), which consists of tagging IP packets and then using different priority policies in routers depending on the tag. The tag is put in the 8 bits TOS (Type of Service) field in the IP packet header.

  Definition of the Differentiated Services Field (DS Field) can be found in RFC2474 (DSCP).

### 2.3.6 Transport protocols

Two main transport protocols exist on top of IP, **UDP** (User Datagram Protocol) and **TCP (**Transmission Control Protocol).

UDP is very simple and consists of sending information "as is", without congestion control, session establishment or acknowledgements. The user must implement all these functions in the upper Layers.

TCP is a transport mechanism that requires bi-directional transmission.

- It is statefull (session establishment),

- It is reliable, using an acknowledgement and retransmission mechanism
- It uses congestion control to adapt the sending rate according to the congestion of the network.

It is the protocol used for HTTP, FTP and many other applications.

Because of congestion control and increased delivery delay it is not preferred for audio/video streaming. Congestion control might decrease the sending rate under that of the media codec rate, which will lead to send-buffer overrun and receive-buffer under run. A detailed description can be found in EBU document Tech 3318.

### 2.3.7 Problems of audio/video transport over IP networks.

IP networks are asynchronous whilst audio and video is synchronous.

There is no clock signal transported through an IP network, so it must be recovered at the receiver side or obtained externally. Jitter in the network means that the delivery time is not a constant, leading to delay variations. Whilst many clock recovery algorithms exist, the difficulty is to estimate the clock drift correctly and to separate it from the network jitter.

Audio and video require a guaranteed network bitrate.

The available bitrate varies on shared IP networks and congestion may occur, leading to packet loss and bandwidth reduction. Recovery mechanisms and, ideally, congestion control mechanisms should be used.

### 2.3.8 RTP for multimedia transport over IP

RTP (Real-time Transfer Protocol) has been designed for the transport of multimedia streams over IP networks. It works on top of UDP and has been standardized by the IETF[1] in RFC3550.

RTP defines fields to address the problems associated with the main IP network impairments.

Figure 13 displays the header of an RTP packet.

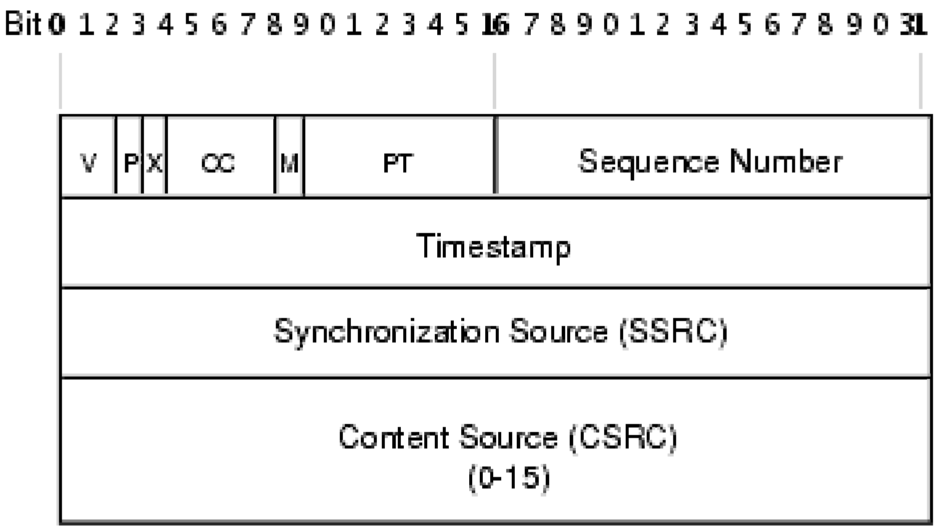


**Figure 13: RTP packet header**

[1] The Internet Engineering Task Force (IETF) is a large open international community of network designers, operators, vendors and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet.

Where:

| | |
|---|---|
| **V (Version)** | Identifies the RTP version. |
| **P (Padding)** | When set, the packet contains one or more additional padding octets at the end, which are not part of the payload. |
| **X (Extension bit)** | When set, the fixed header is followed by exactly one header extension, with a defined format. |
| **CC (CSRC Count)** | Contains the number of CSRC identifiers that follow the fixed header. |
| **M (Marker)** | The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream. |
| **PT (Payload Type)** | Identifies the format of the RTP payload and determines its interpretation by the application. Some standardized values exist for videoconference codecs but for other codecs the type may be described in a protocol such as RTSP or SDP (Session Description Protocol) |
| **Sequence Number** | Increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence |
| **Timestamp** | Is the sampling instant of the first media data byte in the packet. It is used to help to recover clock at the receiving side if it is not given externally. |
| **SSRC (Synchronization Source)** | Is a source identifier that is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier. It helps the receiver to differentiate packets arriving from different sources. |
| **CSRC (Contributing Source identifiers list)** | Identifies the contributing sources for the payload contained in this packet. |

Optional associated RTP protocols are:

| | |
|---|---|
| **RTCP (Real Time Control Protocol)** | Is used to send receiver reports to the sender who can then resend packets or adapt the transmission. |
| **RTSP (Real Time Session Protocol)** | Defines play, pause, rewind functionality and describes the session. |
| **SRTP (Secured RTP)** | Is a security profile for RTP that adds confidentiality, message authentication, and replay protection to that protocol. |

For more information see [Perkins].

### 2.3.9 RTP media frame encapsulation

This defines how media frames are encapsulated into RTP packets. Encapsulation must take into account possible variable frame length and possible lost packets.

The IETF has defined many types of media encapsulation into RTP, notably:

**RFC 2250:** A RTP Payload Format for MPEG-1 and MPEG-2 Video

**RFC 3016:** A RTP payload format for MPEG-4 Audiovisual streams.

Because of several layers of encapsulation this leads to overhead, as illustrated below.

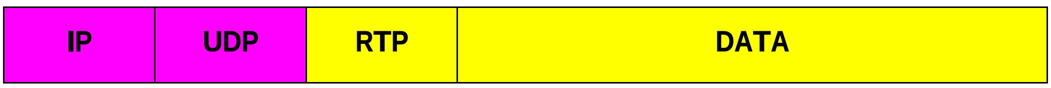


**Figure 14: Headers and data encapsulation**

The sum of the IP+UDP+RTP headers is 40 bytes. So it is preferable not to use too small a data payload in order to avoid too much overhead and also too high a network load.

On the other hand the total packet size (headers+ data) should not exceed the Link Layer (Layer 2) maximum transmission unit (MTU) in order to avoid fragmentation.

### 2.3.10 Packet loss recovery

Two approaches exist to packet loss recovery. These are active recovery (or retransmission) and passive recovery (or forward error correction). In the former, the receiver notifies lost packets to the sender, which resends them. In the latter there is no communication with the sender and lost packets are recovered using error correction techniques.

In general Forward Error Correction (FEC) is preferred as there is no need for a return channel and also because retransmission of packets leads to longer end-to-end delays. The Forward Error Correction must, however, be appropriately scaled to the occurrence of lost packets. A too-robust FEC scheme will lead to increased bandwidth overhead and delay and a too weak FEC scheme will lead to unrecovered packets.

The simple way to effect FEC is to make parity packets out of a certain number of packets. RFC2733 defines such an FEC scheme for RTP but it has its limitations, so the Pro-MPEG Forum proposed an improved standard (ProMPEG-COP3) that has been accepted as an SMPTE standard.

Active recovery can be performed on RTP using RTCP receiver reports. The receiver can indicate lost packets to the sender, which can then resend them. However this leads to bandwidth surges and it may not be adequate in cases of network congestion (in such a case the sending rate should temporarily be decreased).

In summary, audio and video transport over IP is a vast topic that includes many issues:

- Transport protocol: RTP.
- RTP Audio/Video Frame encapsulation.
- Network impairments that depend on network topology and on how the network is managed, on resource sharing but also on the underlying transmission technologies.
- Recovery mechanisms according to network characteristics.

The IETF, the Pro-MPEG Forum and other organizations have issued apposite standards. The DVB Project has also issued a recommendation for DVB transport over IP [ETSI06].

## 2.4 MPLS

MPLS (Multi-Protocol Label Switching) was introduced in order to improve traffic engineering and to support QoS in large IP networks. It eases the management of different protocols for the network operators.

MPLS combines the best aspects of high-speed switching with the intelligence of routing, uses labels to simplify packet forwarding, and adds a level of control without impacting performance.

### 2.4.1 Introduction to MPLS

MPLS label forwarding occurs when a Label Switch Router (LSR) performs a label lookup on an incoming packet, swaps the incoming label for an outgoing label, and forwards the packet to the next LSR along the Label Switch Path (LSP). The Label Edge Router (LER) sits on the entrance and exit of the LSP and respectively adds and removes the MPLS label to and from the packet. Label assignment occurs based on a common grouping or *Forwarding Equivalence Class (FEC)*; packets classified together based on common attributes, such as source addresses (policy based routing), source and destination address pairs, destination address, and even Type of Service (ToS) or Differentiated Service Code Point (DSCP) bits – all packets grouped into the same *FEC* receive similar treatment along the LSP. The LER inserts the MPLS label in between the MAC (Layer 2) and

Network (Layer 3) headers. The core LSRs simply receive packets, read the MPLS labels, swap labels, switch and forward the packets while simultaneously applying the appropriate service.

The MPLS lookup and forwarding scheme gives the service provider the ability to fine-tune and explicitly control packet routing based on source and destination addresses (or other *FEC* classification criteria), allowing for the seamless introduction of new IP services. MPLS provides the following beneficial applications: Virtual Private Networking (VPN), Traffic Engineering (TE), Quality of Service (QoS) and ATM over MPLS (AToM). Additionally, it decreases the forwarding overhead on the core routers. MPLS technologies are applicable to any network layer protocol. Each router of a connectionless network makes an independent forwarding decision for the packets travelling from one router to the next. The forwarding decision can be thought of as analyzing the packet's header and running a network layer routing algorithm. Based on the analysis and the results of running the algorithm, each router chooses the next hop for the packet. In conventional IP forwarding, a particular router will typically consider two packets to be in the same *FEC* if there is some address prefix, X, in that router's routing tables such that X is the longest prefix match for each packet's destination address.

In contrast, in MPLS, the assignment of a particular *FEC* is done just once, as the packet enters the network. The *FEC* to which the packet is assigned is encoded as a short fixed length value known as a label. When a packet is forwarded to its next hop, the label is sent along with it. At subsequent hops there is no further analysis of the packet's network layer header, but the label is used as an index into a table, which specifies the next hop and a new label. The old labels are replaced with the new one and the packet is forwarded to its next hop.

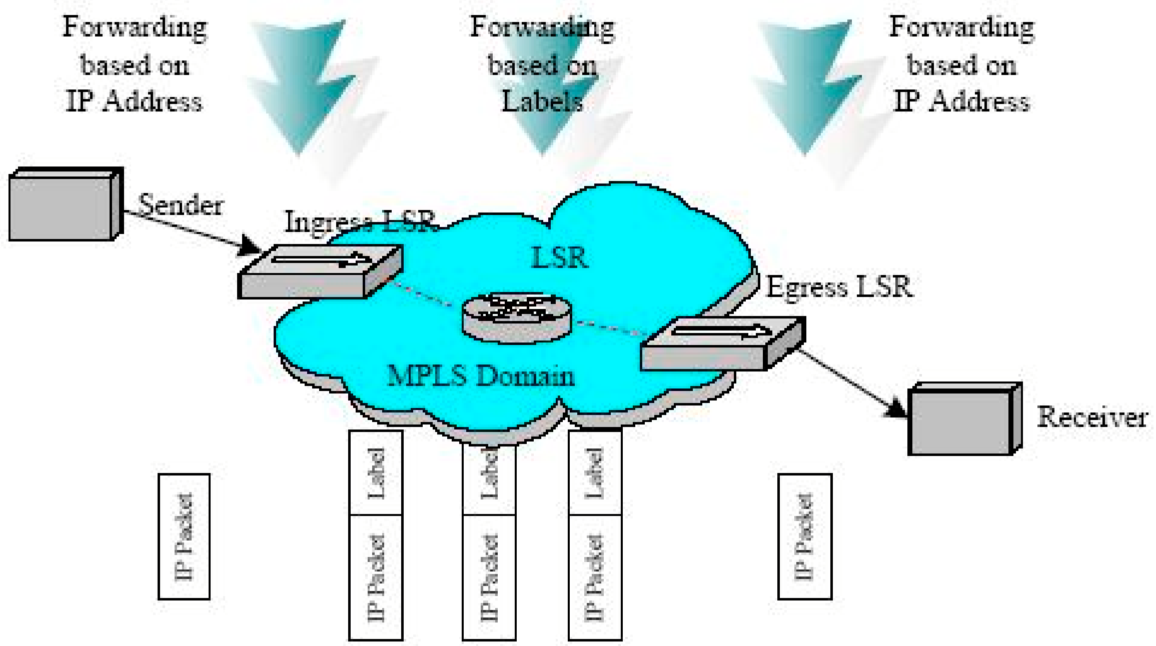


**Figure 15: MPLS structure**

## 2.4.2 Labels

### Label Value (Unstructured), 20 bits

Because MPLS is designed to use different data link layers, the label format will reflect the characteristics of the link layer used. An MPLS shim header can be seen in Figure 16.

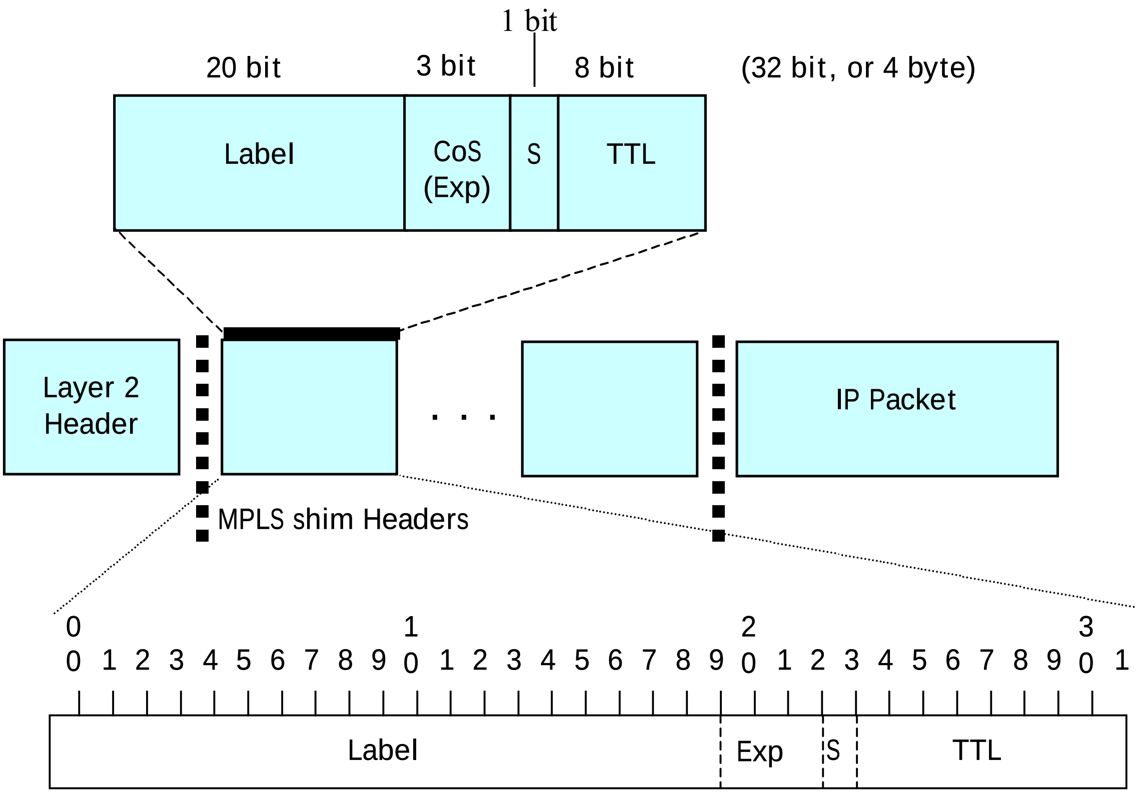


**Figure 16: MPLS label**

The Label field (20-bit) carries the actual value of the MPLS label. A label is a four-byte, fixed-length, locally significant identifier that is used to identify a Forwarding Equivalence Class (FEC). The label that is put on a particular packet represents the FEC to which that packet is assigned.

The label is imposed between the data link layer (Layer 2) header and network layer (Layer 3) header. The top of the label stack appears first in the packet, and the bottom appears last. The network layer packet immediately follows the last label in the label stack.

| | | | | |
|---|---|---|---|---|
| **Layer 2 Header** | **Top Label** | **- - - -** | **Bottom Label** | **Layer 3 Header** |

**Figure 17: Label field**

- Exp: Experimental Use, 3 bits; currently used as a Class of Service (CoS) field. The CoS field (3-bits) can affect the queuing and discard algorithms applied to the packet as it is transmitted through the network.
- S: Bottom of Stack, the Stack (S) field (1-bit) supports a hierarchical label stack.
- The TTL (time-to-live) field (8-bits) provides conventional IP TTL functionality

## Label Details

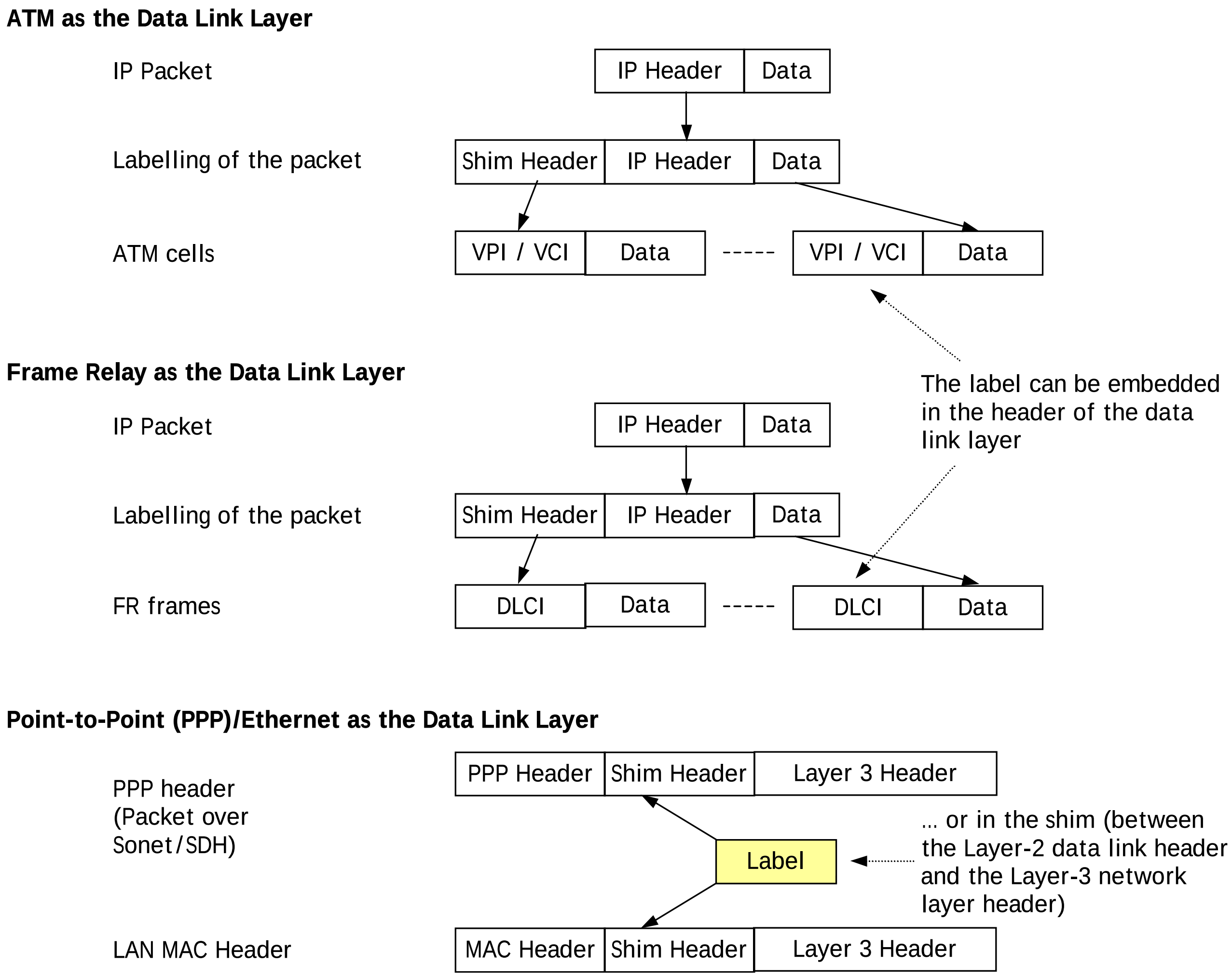


**Figure 18: Different data link layers**

## Label distribution

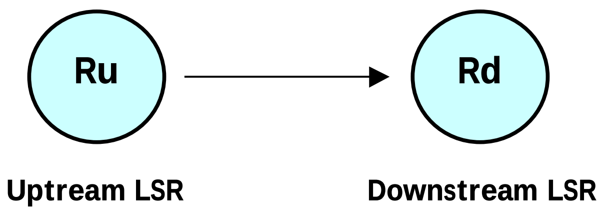


**Figure 19: Label distribution**

Agreement: "binding" label L to *FEC* F for packets moving from Ru to Rd. So, L becomes Ru's "outgoing label" representing *FEC* F, and L becomes Rd's "incoming label" representing *FEC* F. Note that L is an arbitrary value whose binding to F is local to Ru and Rd.

A Label Distribution Protocol (LDP) is used between the nodes in an MPLS network to establish and maintain the label bindings. In order for MPLS to operate properly, label distribution information need to be transmitted reliably, and the LDP messages pertaining to a particular *FEC* need to be transmitted in sequence. The label distribution protocol also encompasses negotiations in which two label-distribution peers need to engage in order to learn each other's MPLS capabilities.

### LDP Procedures

1. *Downstream LSR: Distribution Procedure*
   The Distribution Procedure is used by a downstream LSR to determine when it should distribute a label binding for a particular address prefix to its label distribution peers.
2. *Upstream LSR: Request Procedure*
   The Request Procedure is used by the upstream LSR for an address prefix to determine when to explicitly request that the downstream LSR bind a label to that prefix and distribute the binding.
3. *Upstream LSR: Release Procedure*
   Suppose that Rd is an LSR that has bound a label to address prefix X, and has distributed that binding to LSR Ru. If Rd does not happen to be Ru's L3 next hop for address prefix X, or has ceased to be Ru's L3 next hop for address prefix X, then Ru will not be using the label. The Release Procedure determines how Ru acts in this case.
4. *Upstream LSR: LabelUse Procedure*
   Suppose Ru is an LSR which has received label binding L for address prefix X from LSR Rd, and Ru is upstream of Rd with respect to X. Ru will make use of the binding if Rd is Ru's L3 next hop for X. The labelUse Procedure determines just how Ru makes use of Rd's binding.

### Label handling

It is useful if a labelled packet carries a number of labels, organized as a last-in, first out stack, referred to as a label stack. (An unlabeled packet can be thought of as a packet whose label stack is empty.) If a labelled packet is forwarded, the so-called Next Hop Label Forwarding Entity (NHLFE) is used. It contains the following information:

- The packet’s next hop;
- The operation to perform on the packet’s label stack. These operations can be one of the following:
  - Replace the label at the top of the label stack with a specified new label;
  - Or pop the label stack;
  - Or replace the label at the top of the stack with a specified new label and then push one or more given new labels onto the label stack.

<u>What to do with the NHLFE?</u>

- The Incoming Label Map (ILM) maps each incoming label to a set of NHLFEs. It is used when forwarding packets that arrived as labelled packets.
- The *FEC*-to-NHLFE (FTN) maps each *FEC* to a set of NHLFEs. It is used when forwarding packets that arrived unlabelled, but which are to be labelled before being forwarded. To forward a labelled packet, the procedures below should be followed:

1. The LSR examines the label at the top of the stack. Using the information in the NHLFE, it determines where to forward the packet.
2. Performs an operation on the packet’s label stack.
3. Encodes the new label stack onto the packet and forwards the result.

To forward an unlabelled packet, the procedures below should be followed:

1. The LSR analyses the network layer header to determine the packet’s *FEC*.
2. Using the information in the NHLFE, it determines where to forward the packet.
3. Performs an operation on the packet’s label stack.

4. Encodes the new label stack onto the packet and forwards the result.

### 2.4.3 Route selection

Route selection refers to the method used for selecting the LSP for a particular *FEC*. The architecture supports two options for route selection:

- Hop-by-hop routing allows each node to independently choose the next hop for each *FEC*.
- In an explicitly routed LSP the LSP ingress or the egress specifies several (or maybe all) of the LSRs in the LSP. If a single LSR specifies the entire LSP, the LSP is strictly explicitly routed. If a single LSP specifies only some of the LSRs, the LSP is loosely explicitly routed. The explicit route needs to be specified at the time that labels are assigned, but the explicit route does not have to be specified with each IP packet.

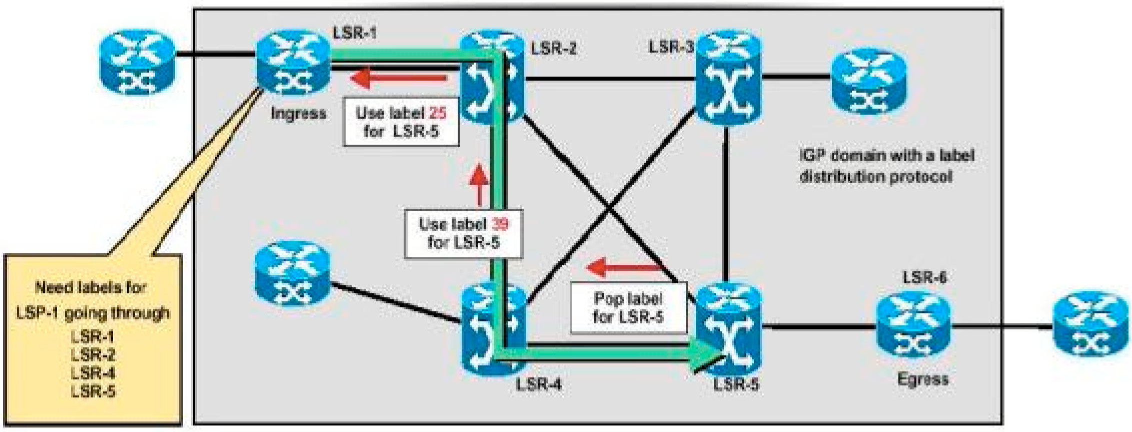


**Figure 20: Route selection**

The request travels hop-by-hop and when it reaches the egress point labels are advertised back to the ingress LSR.

#### Forwarding Equivalence Class (*FEC*)

*The FEC* is a group of IP packets that are forwarded in the same manner, over the same path, and with the same forwarding treatment. A *FEC* might correspond to a destination IP subnet, but it also might correspond to any traffic class that the Edge-LSR considers significant. For example, all traffic with a certain value of IP precedence might constitute a *FEC*.

#### Conventional IP Routers and Label-Switching Routers

The internal functionality of an IP router and an MPLS LSR can be split into two distinct parts: a management engine and a packet-forwarding engine. Signalling and topology/path discovery protocols run on the management engine. The forwarding (switching) engine executes specific packet forwarding rules (governing next-hop and CQS treatment).

#### Native IP Forwarding

The term IP routing is often applied to both the packet forwarding and route determination processes in an IP network. To avoid confusion, this article will use the term native IP forwarding (NIF) to specifically refer to hop-by-hop, destination-based packet forwarding. IP routing will be reserved for references to the topology and path discovery processes.

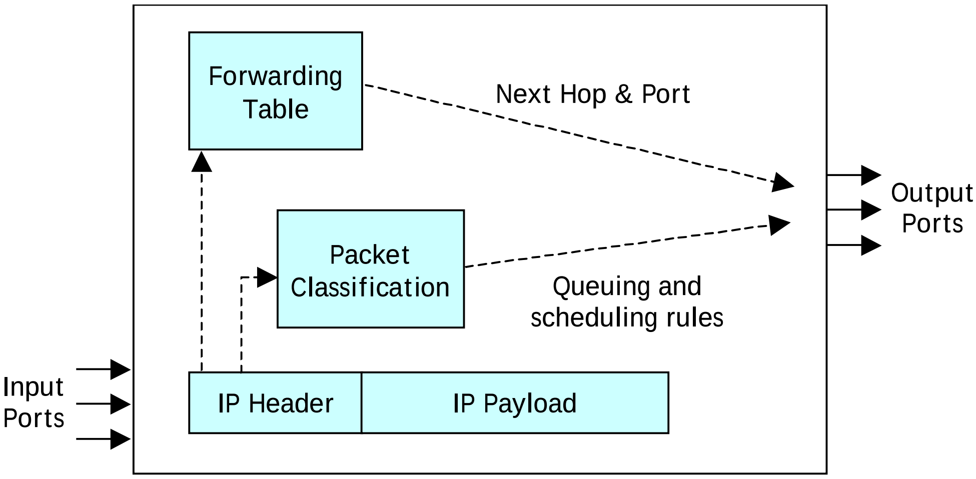


**Figure 21: NIF forwarding engine**

Figure 21 simplifies a NIF node-forwarding engine. Each packet's next hop and output port are determined by a longest-prefix-match forwarding table lookup with the packet's IP destination address as the key. Additional packet classification may also be performed in order to derive output port queuing and scheduling rules (if no such rules are derived, single-queue FIFO is assumed) or permute the forwarding table lookup (e.g. select one of multiple forwarding tables). Armed with this information, the packet is queued at the appropriate output port for transmission. The forwarding table is established and updated by the management engine based on the decisions of the active IP routing protocol(s). Rules for packet classification are installed in response to IP-level signalling protocols (e.g. RSVP) or administrative provisioning.

### Label-Based Forwarding

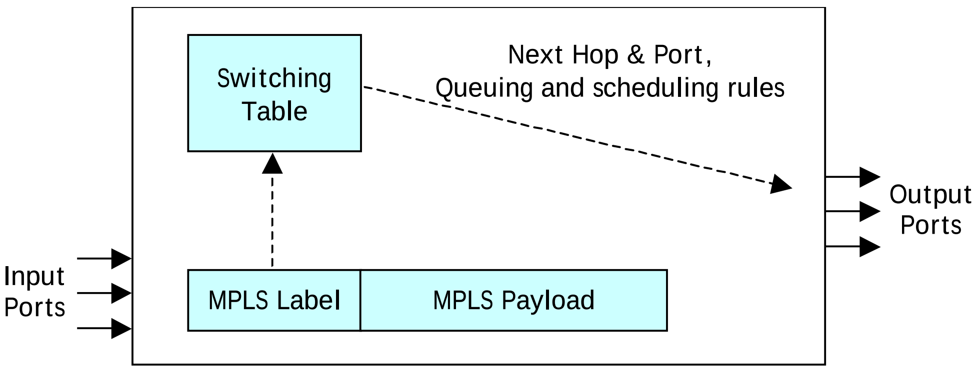


**Figure 22: LSR forwarding**

Figure 22 simplifies LSR forwarding. Each packet's forwarding treatment is entirely determined by a single index lookup into a switching table, using the packet's MPLS label (and possibly the arrival port ID) as the key. The packet's label is replaced with a new next-hop label retrieved from the switching table, and the packet is enqueued at the appropriate output port for transmission. The switching table is loaded a priori with unique next-hop label, output port, queuing, and scheduling rules for all current MPLS label values. This mapping information is established and managed by the management engine in response to external requests for a labelled path through the LSR, and is only modified when a new label needs to be activated or an old label removed.

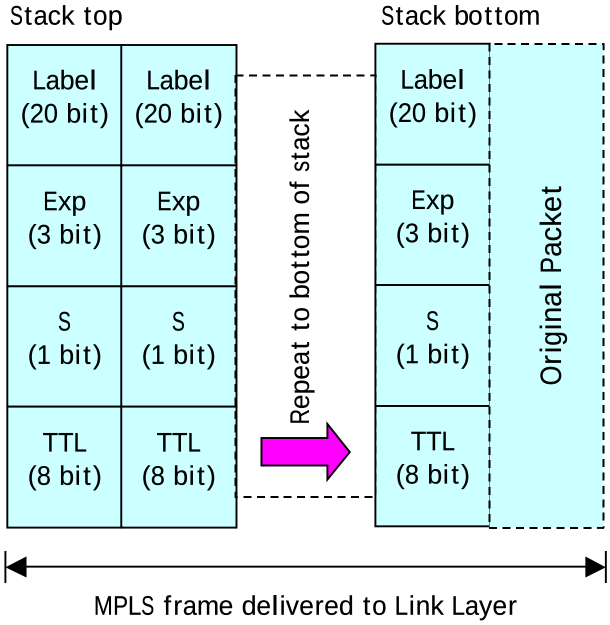


**Figure 23: MPLS frame**

Figure 23 shows the structure of the generic MPLS frame. An MPLS label stack of one or more 32-bit entries precedes the payload (e.g. an IP packet). The label is 20-bit wide, with 3 additional bits for experimentation (e.g., to indicate queuing and scheduling disciplines). An 8-bit time to live (TTL) field is defined to assist in the detection and discard of looping MPLS packets: the TTL is set to a finite value at the beginning of the LSP, decremented by one at every label switch, and discarded if the TTL reaches zero. The S bit is set to 1 to indicate the final (and possibly only) stack entry before the original packet; an LSR that pops a stack entry with S set to 1 must be prepared to deal with the original packet in its native format.

MPLS forwarding is defined for a range of link layer technologies, some of which are inherently label-switching (e.g., ATM and frame relay, FR) and others that are not, such as packet over SONET/SDH (POS) and Ethernet. Although switching logically occurs on the label in the top (and possibly only) stack entry, ATM and FR switch based on a link-layer copy of the top stack entry.

**Packet-Based MPLS:** For packet-based link layers the MPLS frame is simply placed within the link's native frame format; Figure 24 shows the example when running over Point-to-Point Protocol (PPP) links. Unique PPP code points identify the PPP frame's contents as an MPLS frame. A similar encapsulation scheme is used when transmitting over Ethernet, with unique Ether-Types identifying the payload as an MPLS frame.

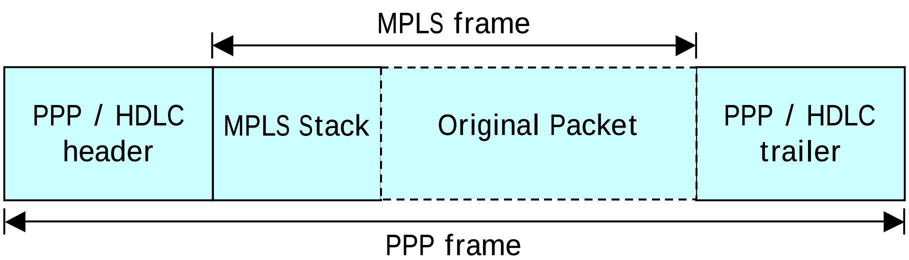


**Figure 24: PPP link**

**Cell and Frame-Relay-Based MPLS:** A core LSR's forwarding engine can be an ATM switch fabric operating purely at the cell level. At the edges of an ATM-based LSP you have hybrid packet/cell LSRs, segmenting ATM adaptation layer 5 (AAL 5)-encapsulated MPLS frames at the ingress to an ATM LSP segment, and reassembling them at the egress (Figure 25).

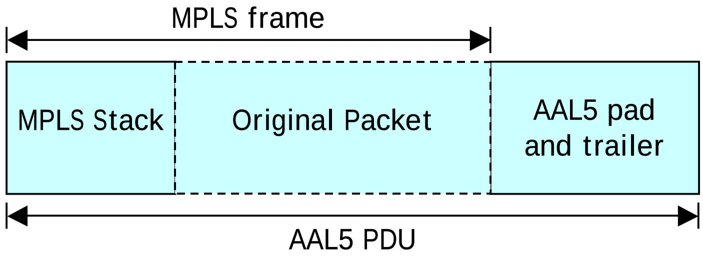


**Figure 25: ATM link**

Packet-to-cell conversions (and vice versa) may occur in edge or core LSRs where an LSP passes from an ATM based link to a POS-based or frame-based link. The top-level label may be carried in three ways across MPLS-ATM links in the VPI/VCI, the VCI alone, or indirectly associated with a switched virtual channel (SVC) or permanent virtual channel (PVC) crossing some non-MPLS ATM network elements. In all cases the MPLS frame still carries a placeholder label stack entry representing the top label, simplifying the design of packet-based LSRs that terminate ATM-based LSP segments. When FR switches are utilized as LSRs, the MPLS frame is mapped directly into the FR frame and the value of the top MPLS label is copied into the data link connection identifier (DLCI), which may support 10, 17, or 23 bits of label space depending on the specifics of each FR switch.

### 2.4.4 The Attraction of MPLS

MPLS solves the IP/ATM scaling problem by making every interior ATM switch an IGP peer with its neighbours (other directly attached ATM switches or the directly attached IP Routers originally "surrounding" the single LIS). ATM switches become IGP peers by having their ATM control plane replaced with an IP control plane running an instance of the network's IGP. With the addition of the Label Distribution Protocol (LDP), each ATM switch becomes a core (or interior) LSR, while each participating IP router becomes an edge LSR (or label edge router, LER). Core LSRs provide transit service in the middle of the network, and edge LSRs provide the interface between external networks and the internal ATM switched paths. The demands on the IGP drop dramatically, since each node now has only as many peers as directly ATM-attached neighbours.

Many packets follow to a great extend the same shortest paths across any given IP backbone regardless of their final destination(s). The MPLS Working Group gives the name forwarding equivalence class (FEC) to each set of packet flows with common cross-core forwarding path requirements. LDP dynamically establishes a shortest path VC (now known as a label-switched path, or LSP) tree between all the edge LSRs for each identifiable FEC. The label —virtual path/channel identifier (VPI/VCI) — at each hop is a local key representing the next-hop and QoS requirements for packets belonging to each FEC. VC utilization is no worse than the single LIS case, and with the introduction of VC-merge-capable ATM-based LSRs it can be much more efficient (only a single VPI/VCI is required downstream of the merge point, regardless of the number of VCs coming in from upstream).

Pure packet-based MPLS networks are a trivial generalization of the ATM model; simply replace the ATM-based core LSRs with IP router-based core LSRs, and use suitable packet based transport technologies to link the LSRs. Today there is significant interest in packet-only MPLS networks because packet LSRs have been demonstrated with OC-48c packet interfaces (with reasonable promise of OC-192c rates and beyond). By contrast, ATM-based MPLS solutions are limited to edge LSRs with OC-12c links, since OC-48c ATM segmentation and reassembly capabilities are proving troublesome for vendors to implement.

### 2.4.5 Centralised Network Management

In a "classical" IP network, packet switching is managed in each router in a decentralised way, according to the adopted routing protocol (e.g. OSPF, IS-IS). As a consequence, some links might saturate, while capacity is still available on alternative paths.

On MPLS networks, a centralised Network Management System can be adopted by means of RSVP-TE (Resource ReSerVation Protocol – Traffic Engineering), managing resource allocation, traffic prioritisation and optimal routing [RFC 3209].

At the moment, this centralised management based on RSVP-TE can be applied only to Unicast streams. However, RSVP TE is likely to deal also with multicast LSPs in the near future (an Internet Draft has already been proposed, and requirements are already available in RFC 4461).

### 2.4.6 Conclusions (MPLS)

Multiprotocol label switching is the convergence of connection-oriented forwarding techniques and the Internet's routing protocols. It can leverage ATM's existing cell switching capabilities, and new high-speed packet forwarding techniques. In its pure packet form it simplifies the mechanics of packet processing within core routers, substituting header classification and longest-prefix-match lookups with simple index label lookups.

However, MPLS is not alone. Advances in the design of conventional gigabit routers allow very similar performance and traffic differentiation goals to be attained. A number of architectures exist that support conventional packet forwarding with differentiated queuing and scheduling at rates sufficient for OC-12, OC-48, and faster pipes. Improved queuing and scheduling technologies may be equally applied to gigabit routers as to MPLS label-switching routers.

Topology driven MPLS builds label-switched paths that map out the same shortest-path trees the packets would have travelled had the network been built with conventional routers. Given the speed gains of conventional IP switch-routers, there is little to be gained by moving to topology-driven MPLS unless you desire to optimize a legacy ATM network currently carrying IP traffic. Interestingly, ATM solutions are not available at OC-48c rates and above because commercially viable OC-48c (and higher) ATM segmentation and reassembly is proving troublesome to implement.

The real selling point for MPLS is its ability to support constraint-routed LSPs from edge to edge using either CR-LDP or M-RSVP. This enables sophisticated load balancing, QoS, and MPLS-based VPNs to be deployed by service providers and large enterprise sites. However, such LSPs enable careful engineering of critical cross-core traffic patterns, and significant work needs to be done before complete solutions exist.

## 2.5 WDM technologies

### 2.5.1 Introduction to WDM

Wavelength Division Multiplexing (WDM) is a technology that combines different wavelengths on an optical fibre. Each wavelength can transport a separate service (SDH/SONET, SDI, Gigabit Ethernet, etc.). WDM is protocol and bitrate independent; hence a single fibre is transformed into multiple virtual fibres, which vastly enhances the payload capacity of the fibre.

Figure 26 gives an example of the different services that can be transported on a WDM system.

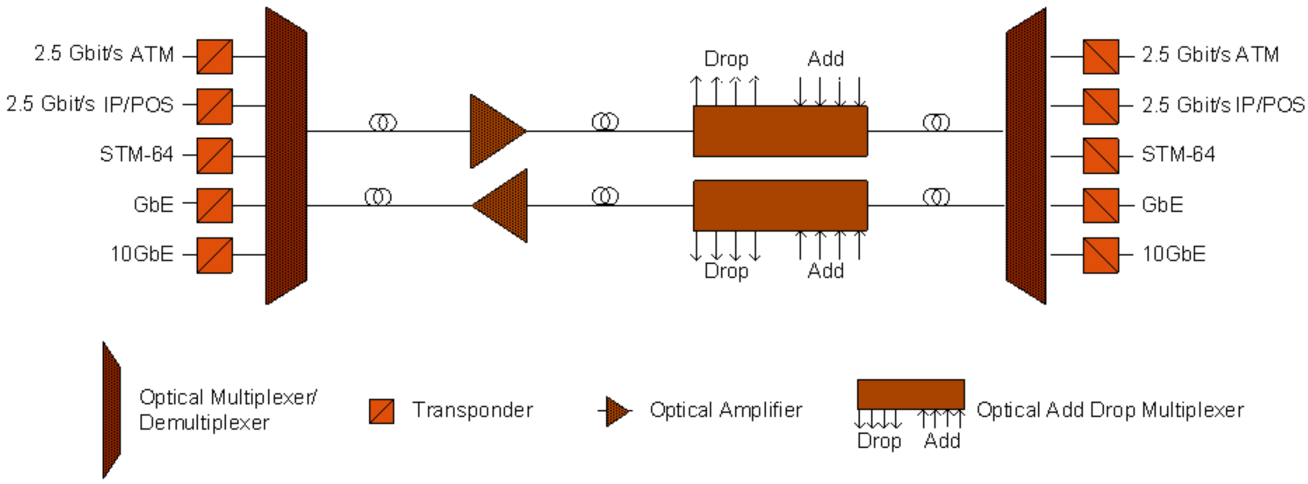


**Figure 26: WDM**

A generic WDM system consists of the following sub-systems:

- Transmit Transponder: Changes the incoming optical or electrical signal into a WDM optical signal / wavelength, normally according to one of the defined ITU-T grids.
- Receive Transponder: Changes the incoming WDM optical signal into an optical or electrical signal.
- Transceiver: Combination of a transmit and receive transponder in one sub-system.
- Optical Multiplexer: a.k.a. optical combiner. Optically combines the different WDM signals / wavelengths coming from the transmit transponders onto a fibre.
- Optical Demultiplexer: a.k.a. optical splitter. Optically splits the different WDM signals / wavelengths coming from the fibre.
- Optical Amplifier: Amplifies an optical signal / wavelength / wavelength band without a conversion to the electrical domain.
- Optical Add Drop Multiplexer (OADM): Allows at a node to drop and add a (limited) number of wavelengths. The wavelengths that are not added or dropped (pass-through) suffer a smaller optical loss then with an Optical Demultiplexer / Optical Multiplexer put back-to-back.

### 2.5.2 WDM

To some, the term WDM refers specifically to a system that combines two wavelengths on a fibre. The two wavelengths normally used are 1310 nm and 1550 nm.

### 2.5.3 DWDM

By definition, the wavelengths of a Dense Wavelength Division Multiplexing (DWDM) system have a small channel spacing. ITU-T Recommendation G.694.1 defines wavelength grids of 1.6 nm (200 GHz), 0.8 nm (100 GHz), 0.4 nm (50 GHz), and 0.2 nm (25 GHz). All grids are rooted around a base wavelength at 1552.52 nm (or 193.1 THz).

Most commercial DWDM systems work in the C-band (Conventional band), covering a wavelength range from 1530 to 1565 nm. This is also the range of wavelengths where optical fibres have the lowest attenuation.

Typically, commercial systems using the 100 GHz grid in the C-band support 40 wavelengths. With a 50 GHz grid the number of supported wavelengths is 80. High-end systems can transport STM-64 (10 Gbit/s) data rates on each wavelength. Most vendors of DWDM equipment are also developing STM-256 (40 Gbit/s) equipment.

To increase the total fibre capacity beyond 800 Gbit/s (80 x 10 Gbit/s) the following two approaches are mostly used:

- The channel spacing is decreased to 25 GHz
- In addition to using the C-band, the L-band (Long band, ranges from 1565 to 1620 nm) is used. Most systems only use a part of the L-band (up to 1610 nm) because the fibre attenuation beyond 1610 nm tends to be too high.

DWDM systems can be designed for Metropolitan, Long-Haul and Ultra Long-Haul applications. Optical amplifiers (normally Erbium Doped Fibre Amplifiers) are inserted at approximately every 60 to 80 km to transport the wavelengths typically 500 km (long haul) or even 2000 km and more (Ultra Long Haul) without optical/electrical/optical conversion.

One EDFA (Erbium Doped Fibre Amplifier) can amplify the whole C-band. For systems that utilize both the C- and the L-bands, two EDFAs are used.

### 2.5.4 CWDM

Coarse Wavelength Division Multiplexing is a cheaper WDM technology that has larger spacing between the different wavelengths used, allowing the use of cheaper, less stable (uncooled) lasers and wider filters. ITU-T recommendation G694.2 defines a 20 nm grid from 1271 to 1611 nm.

It should be noted that prior to October 2003, the ITU-T grid was defined from 1270 to 1610 nm, but it was decided to offset this by 1 nm to accommodate the many CDWM vendors who were using the 1271 to 1611 nm grid. In practice, equipment conforming to both grids is encountered.

CDWM mostly works without optical amplifiers and subsequently they are used in metropolitan and on-premises applications rather than for long-haul. Optically amplifying the whole range from 1271 to 1611 nm is currently prohibitively expensive.

### 2.5.5 Combined C- and DWDM approaches

Some vendors combine both CWDM and DWDM in their equipment. To minimize initial costs, CWDM can be employed, but when traffic requirements demand more capacity the equipment can be migrated either to a hybrid CWDM/DWDM or a full DWDM system.

### 2.5.6 Video

Most long-haul DWDM systems do not have video interfaces (PAL, SDI, …) and therefore a form of multiplexing and translation (to SDH, IP, ATM, …) is required.

Metropolitan systems based on CWDM and DWDM do exist with video interfaces. Some of these allow one video signal per wavelength, whilst others multiplex multiple video signals onto a wavelength.

### 2.5.7 Pros and cons

| Pros | Cons |
|---|---|
| • High capacities can be transported (tens or even hundreds of Gbit/s). Transport of uncompressed or compressed video streams is possible | • Dark fibre is required |
| • Many different client services can be transported (SDH, Ethernet, SDI, ASI, …) | • Static (can be made dynamic with optical switching technologies) |
| • Simple to install and operate (compared to technologies that provide switching) | • No re-routing, restoration or protection (this require optical switching and potentially GMPLS) |
| • Point-to-point and point-to-multipoint (with optical splitters) | • Limited multiplexing capabilities, hence most of the time needs to be used together with technologies that provide multiplexing (SDH, Ethernet, IP, …) |
| • Very low end-to-end delay and jitter | • Automatic bandwidth allocation to data traffic, in case of available capacity, not possible (needs GMPLS) |
| • Can be upgraded to an optical switched and GMPLS enabled network) | • High circuit establishment times |

## 2.6 DTM

DTM (Dynamic Synchronous Transfer Mode) is now standardised in ETSI. A Swedish based company originally developed the technology.

The system allows the construction of circuits for data (IP/Ethernet), voice (E1/T1), video (SDI, HDSDI or DVB-ASI), audio (AES/EBU) or even SDH/SONET. It runs directly on fibre using physical Gigabit Ethernet optics or run on existing standard SONET rings (OC-3/STM1, OC-12/STM4 and OC-48/STM16). With DTM it is possible to establish virtual circuits that provide permanent links between ports.

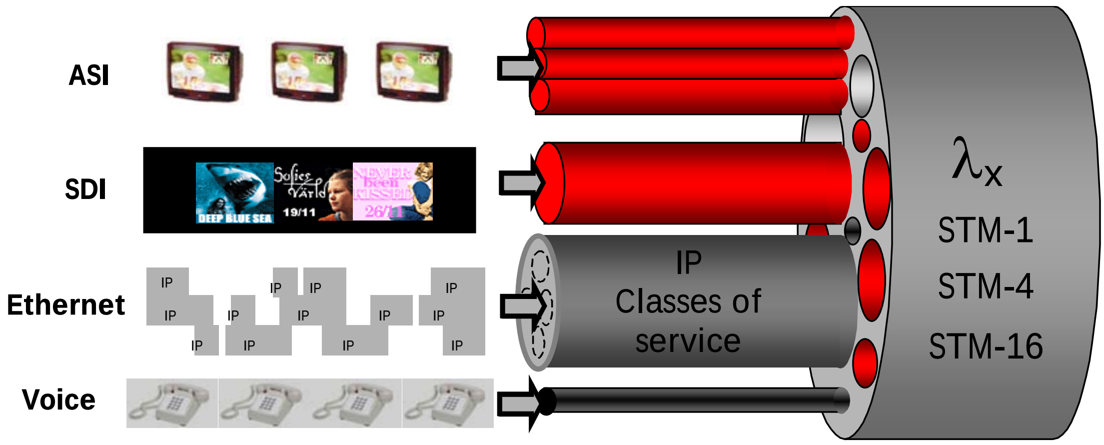


**Figure 27: DTM structure**

DTM is based on TDM, whereby the transmission capacity of a fibre link is divided into small time units. The total link capacity is divided into frames of 125 µs. These frames are further divided into a number of 64-bit time slots. The number of time slots per frame is dependent on the bit rate desired. For example, one slot per frame provides a capacity of 512 kbit/s. The selected use of a frame length of 125 µs and 64-bits per time slot enables simple adaptation to digital voice and Plesiochronous Digital Hierarchy (PDH) transport.

The time slots within each frame are separated into data slots and control slots. At any point in time, a slot is either a data slot or a control slot. However, if needed, data slots may be converted into control slots and vice versa.

# 3. Trends

## 3.1 Gigabit Ethernet in the WAN

Gigabit Ethernet (GbE) and 10-Gigibit Ethernet (10-GbE), despite being originally LAN (Local Area Network) technology, are starting to migrate to WANs. Therefore, a short description of Ethernet and its variants is provided in this section.

### 3.1.1 Ethernet

Ethernet (10 Mbit/s) is a LAN technology that implements the bottom two Layers (Physical and Data Link) of the OSI (Open System Interconnection) 7-layer model. Today the Ethernet family is the most used LAN protocol (more than 85% of LANs use Ethernet), because it is a quite cost efficient technology. Fast Ethernet (100 Mbit/s) is widely used for desktop applications. GbE (1 Gbit/s) and 10-GbE (10 Gbit/s) is the natural upgrade path for Ethernet services to higher data-rates.

All upgrades from Ethernet to 10-GbE preserve backward compatibility since all versions use the same frame; only the data-rate increases successively. The first IEEE Ethernet standard was published in 1985.

Please note that the words “frame” and “packet” are used interchangeably in this text.

### 3.1.2 Gigabit Ethernet

The IEEE ratified the 802.3z 1000Base-X (GbE) standard in July 1998. The standard allows for half-duplex operation in the LAN and full-duplex operation in the LAN and WAN. Both types of communication will be explained.

The Medium Access Control (MAC) layer is part of the data link layer (OSI-model) and it is responsible for initializing, controlling and managing the connection with peer stations. In other words it is responsible for transferring data to and from the Physical layer.

#### GbE in the LAN

In common with Ethernet and Fast Ethernet, GbE uses CSMA-CD (Carrier Sense Multiple Access-Collision Detection) in the LAN. This is an inherently half-duplex protocol.

With CSMA-CD, any device can try to send a frame to line at any time (‘bus’ model). Each device monitors whether the line is idle and therefore available for use. If the line is found to be idle, the device begins to transmit its first frame, if not, it again waits for the line to become idle.

If another device has tried to send at the same time, a collision occurs and the frames are discarded. Even after detection of a collision the sending devices must continue to send for a predefined period of time to ensure the propagation of the collision through the system. Afterwards each device waits a random amount of time (different for both) and retries until it is successful in getting its transmission sent. For each consecutive collision the waiting time gets longer, because successive collisions indicate a very busy medium.

Ethernet and Fast Ethernet have a minimum packet size of 64 bytes. The minimum packet size imposes a limitation on the physical size of the LAN, because of the necessity for collision detection. Any station must be able to detect a collision between the data it is sending and data coming from a station at the remote end of the cable before it stops transmitting. To be able to detect this collision the data from the remote end of the cable must have reached the sending station before it is finished with transmitting.

At 1 Gbit/s a minimum frame length of 64 bytes would limit the extent of the LAN to around 10 m. The minimum frame length for GbE has therefore been extended to 512 bytes, allowing the maximum size of the LAN to be roughly 100 m. To remain compatible with Ethernet the minimal packet size of 64 bytes can still be supported by padding smaller packets to 512 bytes, a technique known as carrier extension.

### GbE in the WAN

GbE in the WAN uses full-duplex operation; WAN GbE becomes a framing and encapsulation method for IP packets. CSMA-CD is not used, its imposed distance limitation of the LAN does not exist and the normal Ethernet 64 byte length frame is used. The payload length is limited to 1500 bytes.

Full duplex operation can only be deployed in a point-to-point configuration where separate send and receive circuits need to be implemented between the two nodes.

### Cabling

There are four different physical layer standards for gigabit Ethernet using optical fibre, twisted pair cable, or balanced copper cable.

IEEE 802.3z provides 3 cabling specifications:

1000Base-SX (Short Wavelengths): This specification targets Multi-Mode (MM) fibre only. The wavelengths of SX lasers are specified between 770 and 860 nm (referred to as 850 nm). The achievable working distance is 500 m.

1000Base-LX (Long Wavelengths): The target fibre types are both MM and Single Mode (SM) and the lasers should be between 1270 and 1355 nm (referred to as 1300 nm). The achievable working distance is 2 km.

1000Base-CX (Copper): defines transceivers or other physical layer devices for shielded balanced copper cables. The achievable working distance is 25 m or less.

IEEE 802.3ab, ratified in 1999, defines GbE transmission over unshielded twisted pair (UTP) category 5, 5e, or 6 cabling and became known as 1000Base-T (twisted pair). The achievable working distance is 100 m.

The 1000Base-X standards use an 8B/10B (8 bit to 10 bit) coding, taken from the ANSI X3.230-1994 (FC-PH) specification for Fibre Channel. Each 8 bits is encoded using 10 bits (consequently the line data-rate is inflated by 25%, from 1000 Mbit/s to 1250 Mbit/s) to ensure a sufficient density of transitions for clock recovery and a DC balance (to avoid heating of the uncooled lasers). The encoding also makes sure that idle periods (no packets being transmitted) are filled with symbols having sufficient transitions.

The coding requires either an equal number of zeros and ones or six zeros (or ones) and four ones (or zeros).

Ethernet did not provide ways of offering QoS (Quality of Service) or CoS (Classes of Service) in the past. To facilitate this QoS or CoS will be provided through IEEE 802.1Q, IEEE 802.1p or higher layers in the OSI stack. 802.1p and 802.1Q tag packets with an indicator of priority or class of service.

## 3.2 Next Generation SDH/SONET

### 3.2.1 New features

The Next-Generation SDH (NG-SDH) adds some new functionality to traditional SDH technology.

In particular, Virtual Concatenation (VCAT), defined in ITU-T G.707, allows the creation of groups of arbitrary VCs, enabling the transmission of streams with various bitrates whilst minimising the overhead (the minimum granularity is 2 Mbit/s).

For example, to transport a 10 Mbit/s stream (e.g. Ethernet), a group of 5 VC12 (11 Mbit/s) can be created, whereas in traditional SDH networks a whole VC3 (45 Mbit/s) would need to be used.

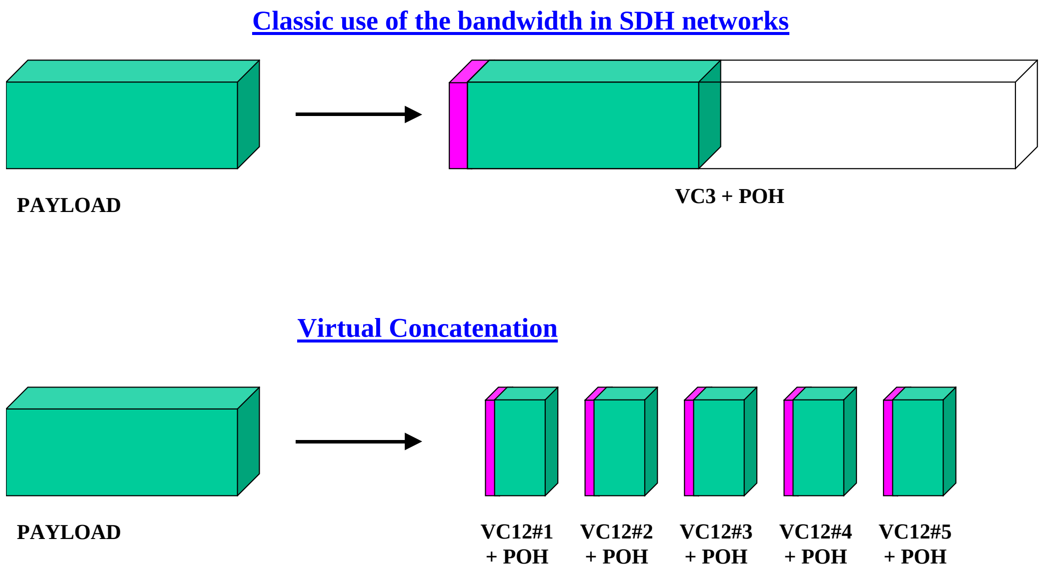


**Figure 28: Virtual Concatenation (VCAT)**

VCAT is transparent for “classic” SDH equipment; only terminal equipment must implement the grouping of VCs.

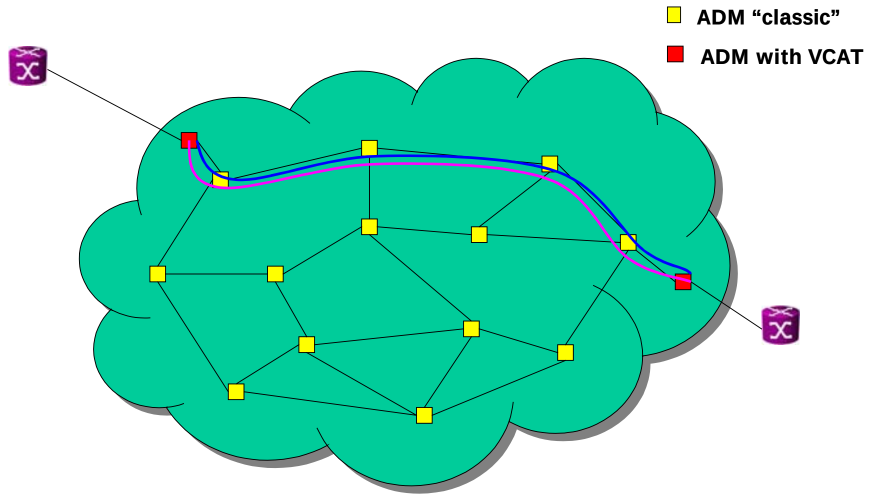


**Figure 29: VCAT in an existing SDH network**

Moreover, with VCAT the various VCs can have different routes in the network.

Generic Frame Procedure (GFP) is a new advanced encapsulation mechanism, specified in ITU-T G.7041, allowing the adaptation to various payload types on traditional VCs (with or without VCAT).

Mapping of the most common payload types have already been standardized, such as Ethernet, Fast Ethernet, GbE, Fibre Channel, RPR, DVB-ASI.

This procedure allows interoperability among equipment from different manufacturers.

Two types of GFP are defined:

- GFP-T (Transparent): is intended to facilitate the transport of 8B/10B block coded client signals for scenarios that require very low transmission latency.
- GFP-F (Frame mapped): adaptation of client signals using a frame-by-frame mapping of the client payload. In this approach only the useful packets are selected. The bandwidth efficiency is better but a certain degree of latency is introduced.

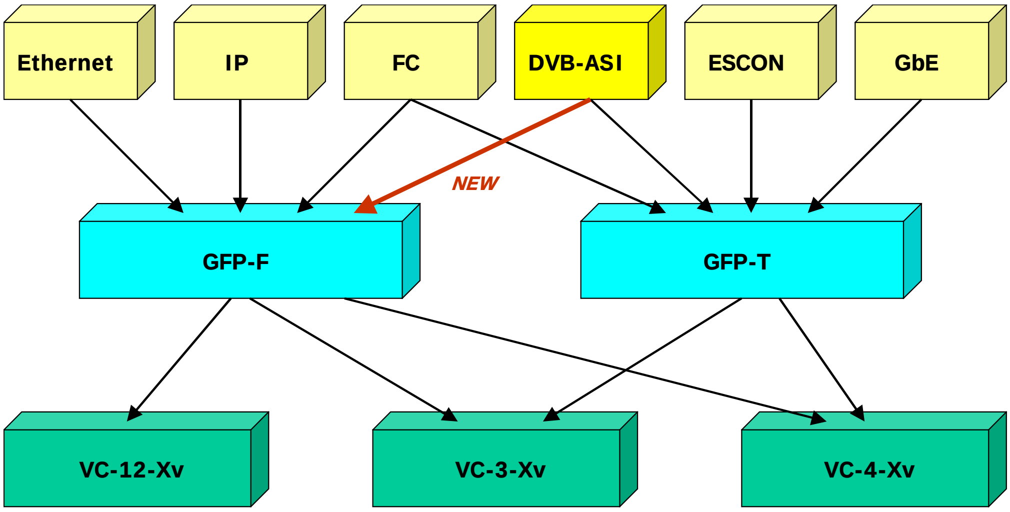


**Figure 30: Generic Framing Procedure**

### 3.2.2 Transport of video signals over NG-SDH networks

The transport of DVB-ASI signals over NG-SDH networks was already defined in ITU-T G.7041 using GFP T. More recently (May 2005) the Recommendation has been integrated with the inclusion of DVB-ASI mapping over GFP-F, following a proposal presented by Italy. Mapping of both 188 and 204 byte TS packets is now allowed.

Using full-rate GFP-T mapping, the DVB-ASI Transport Stream is mapped in 2 VC4 (about 310 Mbit/s), independently of the effective useful bitrate (Figure 31), with a consequent waste of bandwidth.

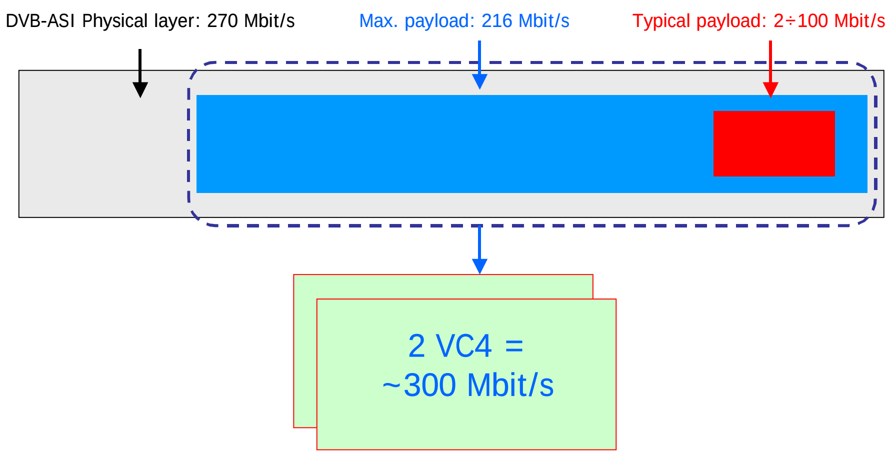


**Figure 31: Transport of DVB-ASI signals using full-rate GFP-T mapping**

An asynchronous sub-rate mapping of 8B/10B clients into GFP is also defined, allowing the use of only the effective bitrate. In the case of video applications, this transparent sub-rate mapping is less efficient in its use of bandwidth with respect to the GFP-F approach and could introduce some issues in terms of packet delineation and consequent jitter.

The transport of DVB-ASI signals using GFP-F, i.e. mapping the ASI payload into NxVC-12-Xv streams virtually concatenated, allows to transmit only the useful part of the DVB-ASI signal, thus increasing the network efficiency (Figure 32).

For example, with this method, a 20 Mbit/s ASI Transport Stream can be mapped in 10 VC12, with a minimum overhead.

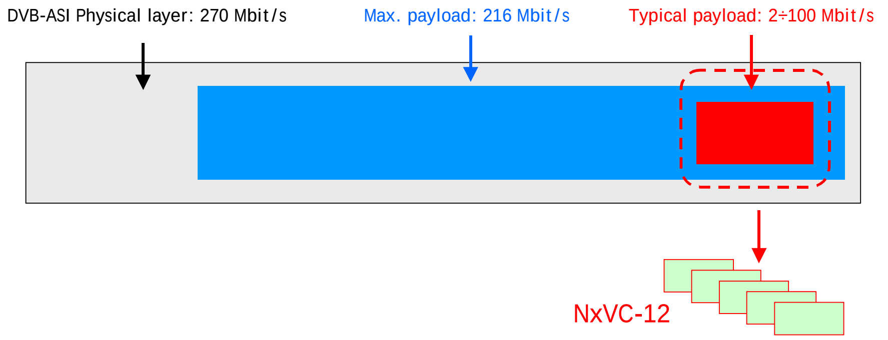


**Figure 32: Transport of DVB-ASI signals using GFP-F**

Point-to-point and point-to-multipoint video circuits can be established on NG-SDH networks, as well as in SDH networks.

### 3.2.3 Evolution of a SDH network towards NG-SDH

Current TV contribution and distribution networks based on SDH can easily and gradually evolve towards NG-SDH:

- Video streams are set-up as VCAT circuits;
- Only (some of) the terminal nodes (video insertion and extraction sites) have to be replaced with new NG-SDH ADMs;

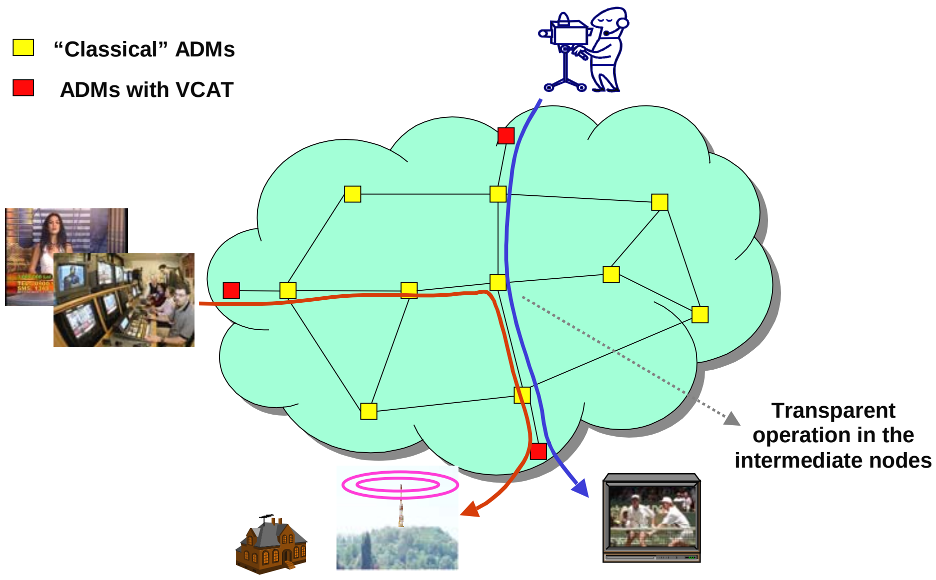


**Figure 33: Evolution of the TV contribution and distribution network towards NG-SDH**

- The existing trunk SDH ADMs are transparent with respect to virtually concatenated streams, which are switched as standard VC12 streams;
- The Network Management System has to include the VCAT functionality.

Moreover, new services can be efficiently transported by the same network using the same approach and with the same bandwidth efficiency, i.e. video-file transfer over IP connections (Ethernet).

### 3.2.4 Layer structure with respect to other technologies

NG-SDH is a very interesting solution for an operator already using a classic SDH network, but it could be an interesting solution for new networks also.

In fact, many networks use SDH as the lower layer, because SDH provides reliability and quality of service. Then, on top of SDH, many video networks currently use ATM to allow more flexibility: the layer structure in this case is represented in Figure 34.

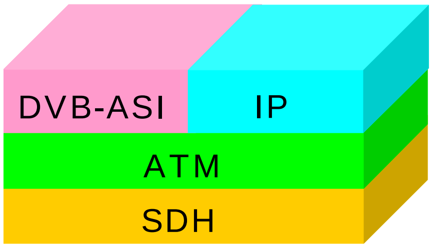


**Figure 34: Typical layer structure for networks based on ATM technology carrying video and IP services**

In this case, two different layers have to be managed, the SDH and ATM layers, with higher complexity and costs.

Some new video networks use the IP layer also for video applications. The layer structure in this case is represented in Figure 35, assuming that PoS (Packet over SDH/SONET) routers are used.

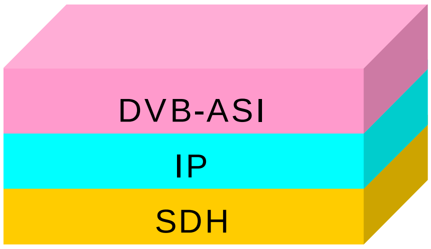


**Figure 35 - Layer structure of a network for video over IP applications**

This is the best solution for video file transfer, but, at the moment, there might be some issues with real-time applications.

Instead, using NG-SDH, both IP services and real-time video services can be transported guaranteeing satisfactory QoS, with very limited jitter and latency, and with sufficient bandwidth granularity. Moreover, as only one layer is involved, management of this network is easier and cheaper.

The layer structure in this case is represented in Figure 36.

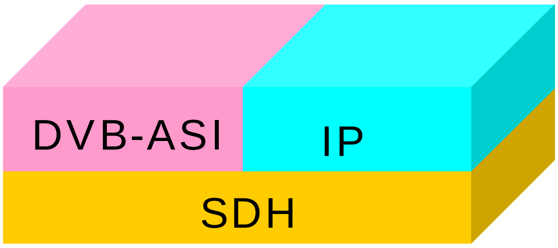


**Figure 36 - Layer structure for networks based on NG-SDH technology carrying video and IP services**

### 3.2.5 Pros and cons

| Pros | Cons |
|---|---|
| • NG-SDH is the natural evolution of SDH, widely available all around the world | • No automatic rerouting in case of link failure (N+1 protection or dual links in the network must be established) |
| • Easy and gradual upgrade from SDH networks, changing only the terminal nodes and not the backbone | • Automatic bandwidth allocation to data traffic, in case of available capacity, not possible |
| • Video streams can be routed in a flexible way, minimizing the network capacity allocation (thanks to the new GFP-F mapping and VCAT) | • High circuit establishment times (i.e. tens of seconds in case of many ADMs involved along the path) |
| • Cheaper solution (if compared to solutions using intermediate switching layers) | • First interface cards for DVB-ASI signals and GFP-F mapping expected on the market in 2006 |
| • Point-to-point and point-to multipoint circuits | • Transport of uncompressed video streams not yet standardised (proprietary solutions are available) |
| • Low end-to-end delay and jitter | |
| • The network can be configured by a centralised Network Management System, allowing for a strict control of the circuits and the occupied bandwidth | |
| • Other services (e.g. Ethernet) can be transported in the same network and with the same bandwidth efficiency | |
| • All SDH mechanisms still available | |

## 3.3 Optical routing and OTN (Optical Transport Network)

### 3.3.1 Introduction

CWDM/DWDM, creates a new layer called the optical layer (between the fibre and the SDH / Ethernet / etc. layers). The first basic CWDM/DWDM systems where developed for fibre relief and they were static.

Static systems do not:

- Provide protection or restoration in case of network failure
- Dynamically reconfigure bandwidth
- Groom capacity to make optimal use of the available bandwidth

To provide these functions the optical layer had to rely on higher layers such as IP, Ethernet, and SDH.

### 3.3.2 Optical routing/switching

#### ROADM

A first step towards truly optical switching is the Reconfigurable Optical Add Drop Multiplexer (ROADM). A ROADM is an OADM where the add, drop and pass-through wavelengths can be changed remotely.

#### Optical Switches

Optical switches can be divided in two categories depending on their switch fabric.

**Optical switches with an electrical switch fabric**

SDH switches and DXC (Digital Cross Connects) switch signals at the VC-12, VC-3 or VC-4 level. When these devices switch at, for instance, STM-16, STM-64, GbE or 10-GbE levels, they are called optical switches or OXC (Optical Cross Connects). In these kinds of switches light is converted from optical-to-electrical-to-optical, but normally they only tend to have optical interfaces.

**Photonic Switches**

When optical switches have an optical switch fabric they are called photonic or all optical switches. Light is not converted to the electrical domain.

Many different technologies exist to build photonic matrixes, but MEMS (tiny movable mirrors) attract the most interest.

Photonic switches have the benefit that they are truly data rate and protocol transparent. On the other hand they require a very careful design to minimise the optical impairments such as attenuation, dispersion and fibre non-linearity.

### 3.3.3 OTN

The ITU-T defined a set of standards to provide networking capability at the optical layer. One of the goals of the OTN is to provide an SDH type of management to optical networks.

The most important ITU-T OTN (Optical Transport Network) standards are:

- G.709/Y.1331: Interfaces for the Optical Transport Networks
- G.872: Architecture of Optical Transport Networks
- G.959.1, Optical Transport Network physical layer interfaces
- G.798: Characteristics of Optical Transport Network Hierarchy Equipment Functional Blocks.

OTN defines different layers: Optical Channel (OCh), Optical Multiplex Section (OMS) and Optical Transmission Section (OTS) layer as shown in Figure 37.

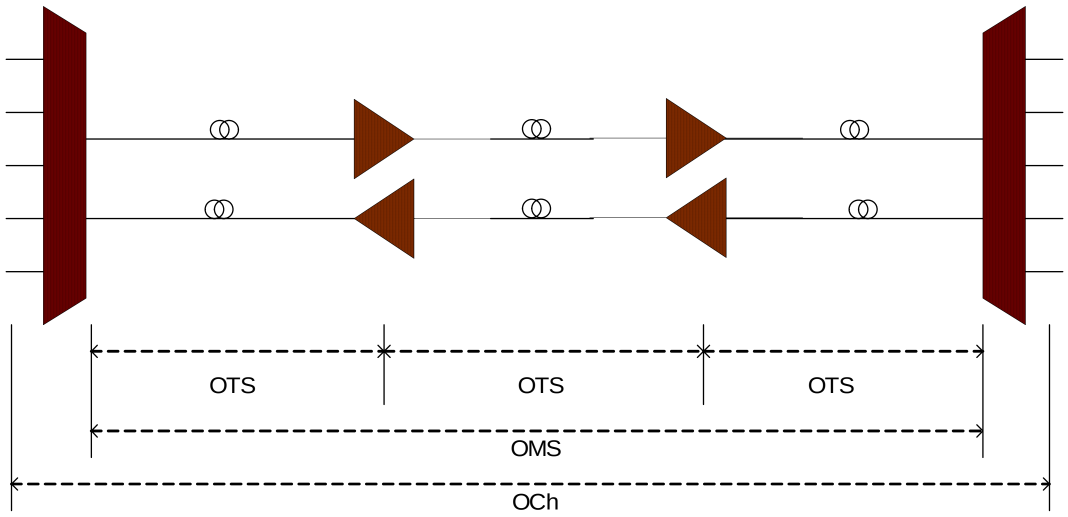


**Figure 37: OTN layer structure**

### 3.3.4 G.709

Recommendation G.709 provides management capabilities and error correction at the "wavelength layer". It is also referred to as digital wrapper.

Today it defines three Optical channel Payload Units (OPU):

- OPU1: 2.5 Gbit/s (STM-16)
- OPU2: 10 Gbit/s (STM-64)
- OPU3: 40 Gbit/s (STM-256)

The OPU can be almost any type of client signal for instance SDH, IP, GbE, ATM, Frame Relay.

The digital wrapper consists of an Optical Channel Overhead (OCh-OH) and Forward Error Correction (FEC). The OCh-OH allows for management at the "optical layer" similar to the SDH overheads. FEC provide for recovery of bit errors that occurred during transport. The data rates of the OTN signals are approximately 7 % higher than the corresponding SDH data rates.

An overhead (OH) is added to the client signal to form the Optical channel Payload Unit (OPU). This OH is needed to regulate the mapping of the client signals and it also gives information about the type of client signal.

The Optical channel Data Unit (ODU) OH provides additional management capabilities such as path monitoring, tandem connection monitoring (technique to monitor networks or sections of networks even if they are run by different operators) and Protection Switching.

The Optical Transport Unit (OTU) OH provides frame alignment, section monitoring and communication channels.

The OTU is carried by a CWDM/DWDM wavelength as an OCh.

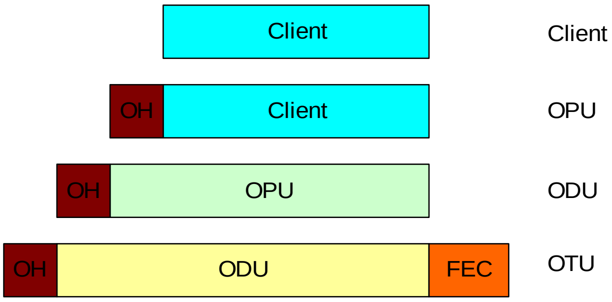


**Figure 38: G.709 Transport structure**

G.709 is very useful for carriers' carrier because it enables transparent transport of managed SDH signals.

## 3.4 MPλS and GMPLS

In optical networks optical switches or OXC provides switching and management is provided by Network Management Systems. Intelligence is needed in a network to provide, among others things, scalable and dynamic bandwidth provisioning and distributed restoration. For intelligence a control plane is needed.

### 3.4.1 Network Planes

Telecommunication equipment can generally be decomposed into different planes:

- The management plane provides the management of the equipment from a central place (NOC): SNMP, CMIP, etc.
- The forwarding or data plane is responsible for transporting data from one point in the network to another: SDH/SONET, G.709, etc.
- The control plane implements the intelligence of the intelligent transmission network: GMPLS

The goal of the control plane is to open, maintain, change and close circuits dynamically, in some cases even without human intervention.

### 3.4.2 MPλS

The first idea was to implement intelligence based on the concepts of MPLS (Multi-Protocol Label Switching) for wavelength (lambda) switched networks. Soon after the industry realised that the concept of wavelength switching could be expanded to SDH/SONET, fibre, Ethernet and other switching. Hence the IETF decided to call the control plane for transmission (and other) networks GMPLS - Generalized Multi-Protocol Label Switching.

### 3.4.3 GMPLS

Generalized MPLS (GMPLS) provides the control plane for packet (e.g. IP), cell (e.g. ATM) and frame (e.g. Ethernet), TDM (e.g. SDH), wavelength, and fibre switched networks. Five types of network (interfaces) are defined:

- PSC: Packet Switch Capable
- L2SC: Layer 2 Switch Capable
- TDM: Time Division Multiplexing capable
- LSC: Lambda Switch Capable
- FSC: Fibre Switch Capable

The GMPLS control plane reuses protocols from the IP/MPLS (based on MPLS- Traffic Engineering Extensions) world with adaptations for transmission networks. The control plane consists of two sub-planes a routing and a signalling sub-plane.

### Routing sub-plane

The routing sub-plane has as a main function to calculate the path when a new circuit is requested and it also maintains the routing information, more or less dynamically. Constraints can be added to the required path. The source node of the GMPLS enabled network computes the path (distributed intelligence). To facilitate finding a constraint-based path the routing plane also provides distributing topology and resource information, the Traffic Engineering (TE) extensions.

Two GMPLS routing protocols are specified by the IETF standards:

- OSPF (Open Shortest Path First) with TE
- IS-IS (Intermediate System to Intermediate System) with TE

It is left up to the equipment manufacturer to choose which of the two routing protocols to implement.

To be able to provide protection (fast recovery, typically sub-50 ms) or restoration (slower recovery) route diversity most be understood. To allow network operators to specify which circuits share the same physical resources (ducts, fibres, etc.) and risks, Shared Risk Link Groups (SRLGs) are defined. Circuits with the same SRLG share a same resource and are consequently not totally diverse.

### Signalling sub-plane

The signalling plane signals to the network devices a route set-up, reroute or teardown of a path (see SS7 in PSTN networks). The GMPLS signalling protocols defined are based on RSVP-TE (Resource Reservation Protocol with Traffic Engineering extensions) and CR-LDP (Constraint-based Routed Label Distribution Protocol). Similar to the routing protocols the choice between the two signalling protocols is not specified by GMPLS.

### Link Management Protocol

Link Management Protocol (LMP) is an important addition to GMPLS. It is used to manage the physical link layer. LMP provides the following capabilities:

- Control channel providing control plane information exchange
- Link property correlation to be able to combine data links in an aggregate TE link
- Physical connectivity verification of links in the data plane
- Manage link failures in the data plane.

### GMPLS Recommendations of the IETF

The most relevant GMPLS recommendations are:

- RFC3945: Generalized Multi-Protocol Label Switching (GMPLS) Architecture
- RFC3209: RSVP-TE: Extensions to RSVP for LSP Tunnels
- RFC3212: Constraint-Based LSP Setup using LDP
- RFC3471: Generalized Multi-Protocol Label Switching (GMPLS) Signalling Functional Description

- RFC3472: Generalized Multi-Protocol Label Switching (GMPLS) Signalling Constraint-based Routed Label Distribution Protocol (CR-LDP) Extensions
- RFC3473: Generalized Multi-Protocol Label Switching (GMPLS) Signalling Resource ReserVation Protocol-Traffic Engineering (RSVP-TE) Extensions
- RFC3946: Generalized Multi-Protocol Label Switching (GMPLS) Extensions for Synchronous Optical Network (SONET) and Synchronous Digital Hierarchy (SDH) Control
- RFC2702: Requirements for Traffic Engineering Over MPLS

# 4. Use cases and examples

## 4.1 ARD HYBNET

The HYBNET (HYBrides diensteintegrierendes BreitbandübertragungsNETz der ARD) is the production network of ARD used to transport TV, Radio, voice and data services. It is based on standard SDH, ATM and more recently DTM network technologies. It has been in full operation since the middle of 2002 in the Northern Ring and since January 2003 in the Southern Ring.

It connects the twelve main ARD production centres with SDH on fibre (DWDM) with a capacity of STM-16 (2.5 Gbit/s). Of this capacity, STM-4 (622 Mbit/s) is used for ATM. The regional networks are under construction (already partly in operation).

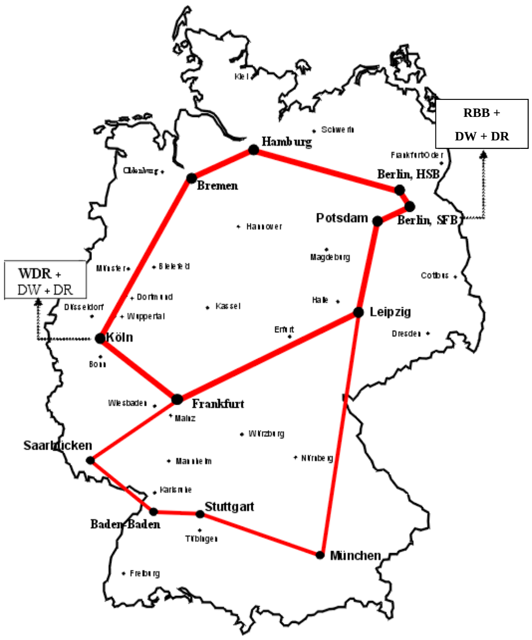


**Figure 39: ARD HYBNET.**

The applications and technologies are displayed in figure 40.

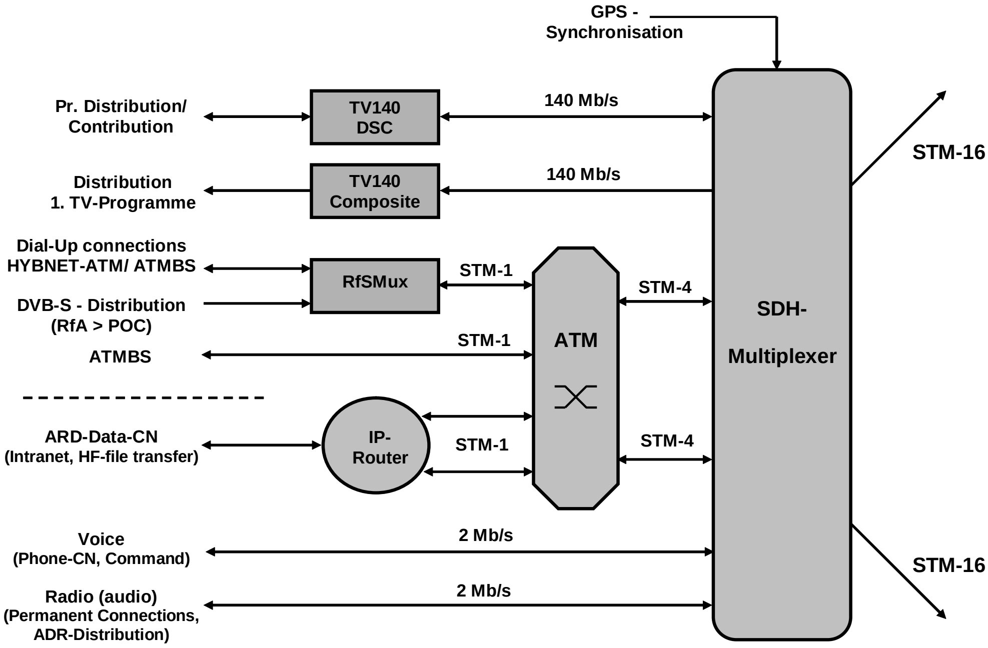


**Figure 40: Hybnet technologies and services**

## Experiences:

- SDH-, ATM-, and Codec-technology has proven to be highly stable
- Errors in the WAN are extremely rare and caused by:
    - planned switching in STM-16-Segment (agreed by the broadcaster)
    - fibre break (construction work…)
    - failure of equipment
- GPS-Synchronisation of SDH-, ATM-, and terminals
    - almost no pointer actions
    - very low Cell Delay Variation (ATM)
    - almost no timing problems in the broadcast applications
- signal quality for contribution and primary distribution improved
- low delay due to elimination of MPEG-Codecs

## Ongoing changes

A secondary DVB distribution via DTM (STM-1) was recently added.

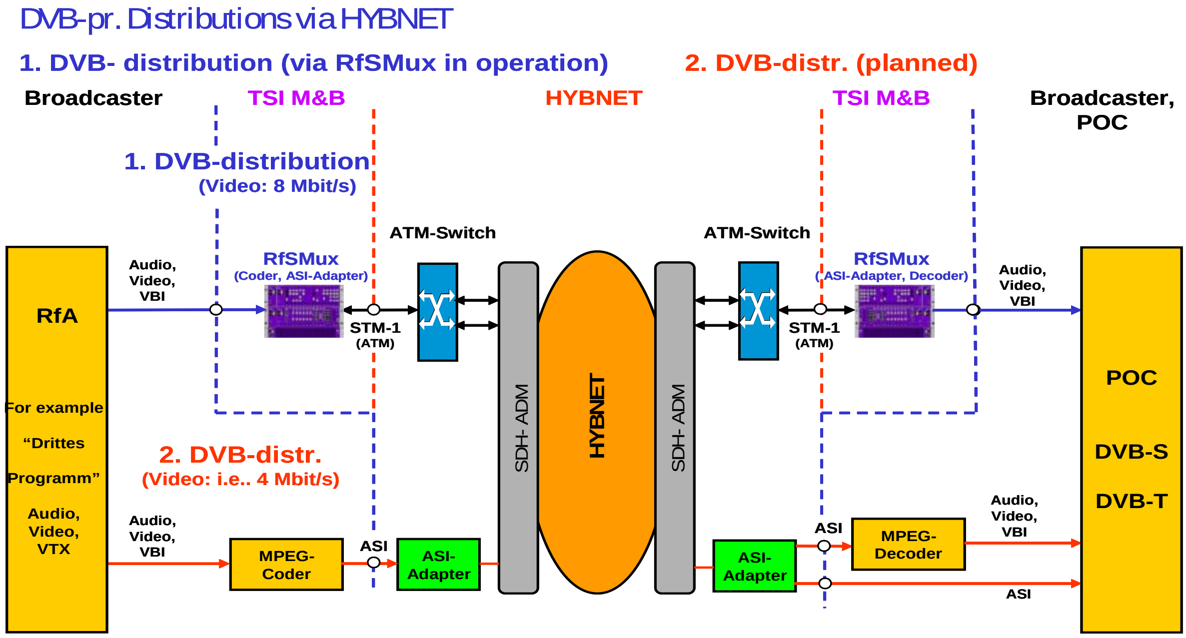


**Figure 41: DVB distribution migration**

## 4.2 EBU FINE

## General

The FiNE network has been developed by Eurovision to complement the Eurovision satellite network. Several points of presence are connected. They are strategic locations, points originating heavy traffic, or meeting points to connect to other networks.

The network carries compressed video (SDI or ASI interfaces are available), or data (E1 or Ethernet interfaces).

## Topology

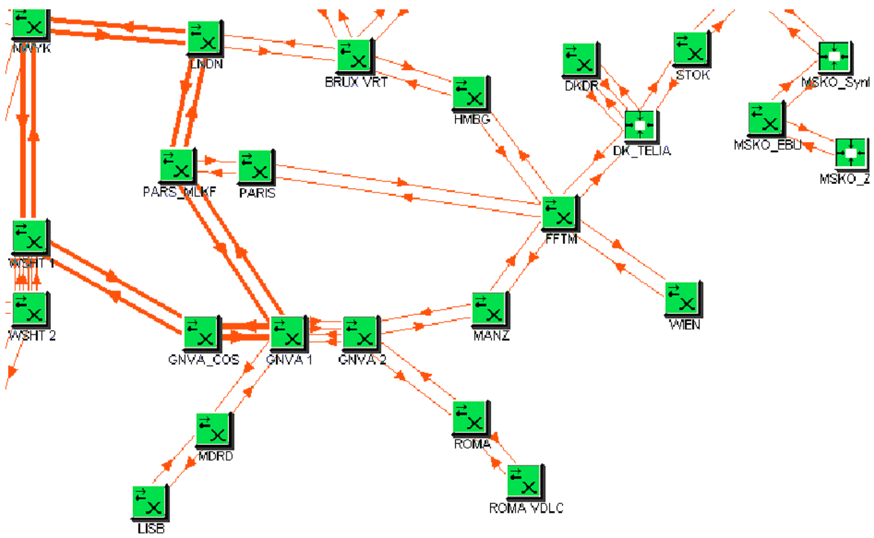


**Figure 42: FINE Topology**

The SDH links between the nodes are 155 Mbit/s (OC-3/STM1), or 622 Mbit/s (OC-12/STM4).

### Control

One part of the bandwidth is dedicated to network management. The nodes and the peripheral equipment (encoders and decoders) are controlled through a web interface.

Eurovision has developed software (D2F) that allows network control and monitoring and includes a database for the planning of transmissions.

## 4.3 Rai Way (WayNet)

The Rai transport network for TV and audio contribution and distribution services (WayNet) was developed in Italy and it is based on SDH (Synchronous Digital Hierarchy) radio links having a capacity of 1xSTM-1, 2xSTM-1 and 3xSTM-1.

The network protocol stack is currently DVB/SDH and switching is carried out at SDH level, via ADM (Add-Drop Multiplexer) and DXC (Digital Cross Connect) equipment.

### Network topology

The WayNet network (Figure 43) consists of about 120 stations along five main trunks, providing distribution and contribution for the TV and Radio Programmes, and services related to transmission of voice, data and technical management information.

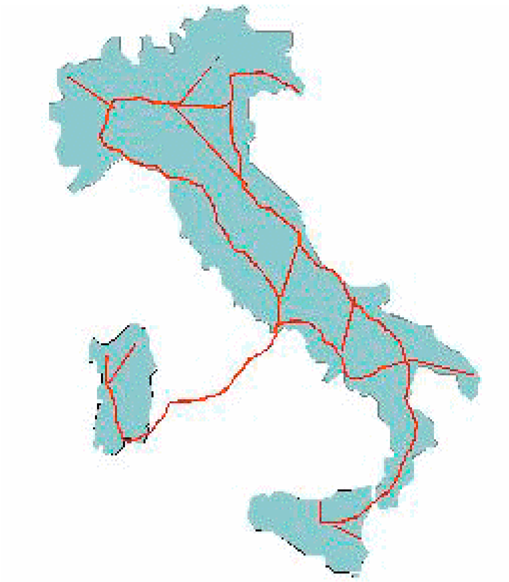

**Figure 43: WayNet Topology**

For distribution purposes, the network reaches all the main Transmission Sites, while the contribution network connects the 4 national Production Centres in Rome, Milan, Naples and Turin and the 20 Regional sites, and it makes about 60 insertion points available to connect Outside Broadcasting (OB) units to the contribution network.

The network provides a capacity of N x 155 Mbit/s (with N varying from 1 to 3) on the main bi-directional trunks. Each STM-1 stream can include up to three VC3 containers accessible via PDH 45 Mbit/s interfaces. Network adaptation from the Transport Stream format (video encoder output) into (unframed) PDH format includes error protection by Reed-Solomon RS(188,204) and interleaving with depth 12, according to a solution directly derived from the DVB approach for broadcasting systems.

The 45 Mbit/s network accesses can transport 1, 2 or 3 TV channels at 38, 19 or 12 Mbit/s respectively, multiplexed in a single MPEG-2 Transport Stream. For contribution services, typically 2 TV channels using the MPEG 4:2:2 profile at 19 Mbit/s are included in a unique TS, while, for distribution services, 3 TV channels coded in MPEG MP@ML at 12 Mbit/s are arranged.

Switching in the network nodes is carried out at VC3/SDH level, by ADM or DXC equipment. This causes some limitations in the network flexibility, as it is not possible to independently route video signals carried in the same VC3 to different destinations.

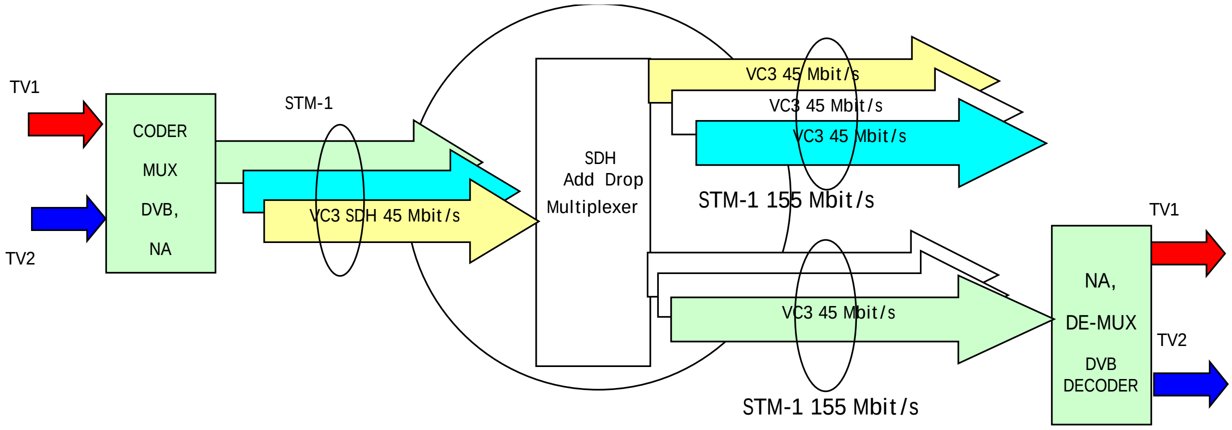


**Figure 44: DVB/SDH technology with ADM switching at VC3 level (45 Mbit/s)**

Planning, scheduling, routing and activation of the connections is carried out in RaiWay Central Network Management Centre in Rome. Connections may be planned in advance and booked (e.g. a week or a day before) or at the last minute (mainly for news events). The Integrated Network Management System (INMS) has been recently updated, reducing the need for human intervention and shortening the circuit delivery time.

In the years 2000 - 2003 the network transported about 60,000 - 65,000 contribution connections per year. The traffic is characterised by daily peaks (mainly concentrated around the News programmes), and weekly peaks during the weekends (mainly due to sports events).

In order to plan the future evolution of the network, the following elements have to be considered:

- a progressive increase of the total required network capacity (say 20% in five years);
- the transport of DTT MUXes, requiring the distribution of the national TV programmes from Rome to the 20 Regional Centres, and the distribution of the aggregate Bouquets with regional insertions from the Regional Centres to the main DTT transmitters;
- a possible increase in the future of file-transfer IP-based traffic (server-to-server) and a corresponding reduction of real-time video traffic.

The introduction of more flexible switching mechanisms, capable of routing any stream capacity instead of the current VC3 capacity, such as ATM, IP, DTM or NG-SDH, would contribute to solve the future capacity problems of the network.

In particular, NG-SDH would accommodate the current SDH backbone, gradually introducing the new switching technology:

- Contribution video streams can be routed independently;
- DTT distribution and 2 Mbit/s streams use only the needed portion of the bandwidth;
- Part of the bandwidth can be dedicated to IP services (Ethernet);
- VC3 switching can still be used, if needed

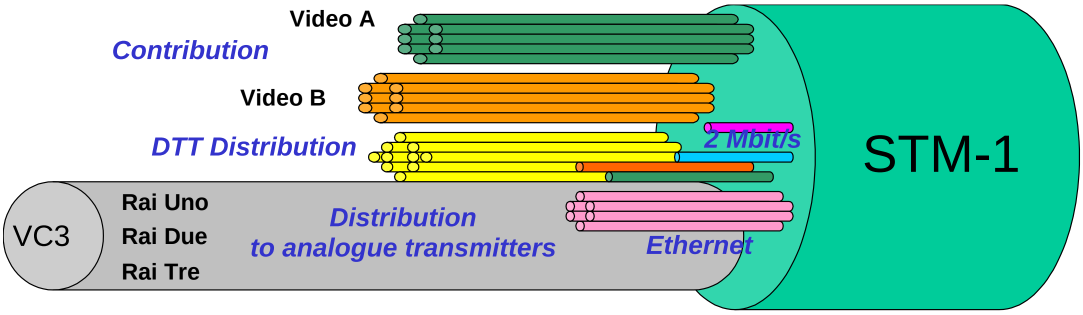


**Figure 45: Possible capacity allocation in a NG-SDH network using VCAT**

## 4.4 ORF L-Net

L-Net Digital was designed to meet the requirements of the next few years:

- A/V contribution
- A/V distribution
- high availability
- flexibility
- plan ahead functionality
- ease of scalability
- short delays
- linear contribution
- ease of use
- Interconnection of
    - 1 central site
    - 9 regional studios
    - 9 main transmitter stations
    - 1 radio production studio
- 2 TV programmes (one central programme can be interrupted through a local programme by the regional studio anytime)
- 4 radio programmes: one central programme an be interrupted through a local programme by the regional studio anytime)

The chosen solution:

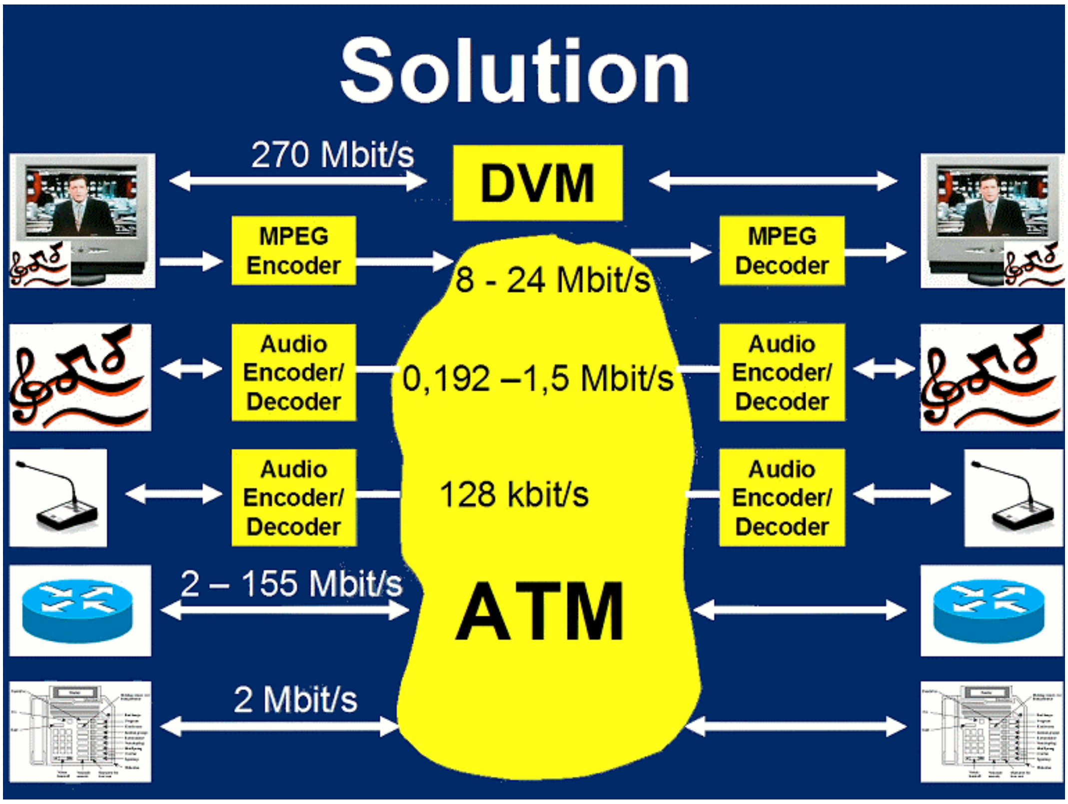


**Figure 46: L-Net Solution**

Sample of a partial hardware setup at one local site

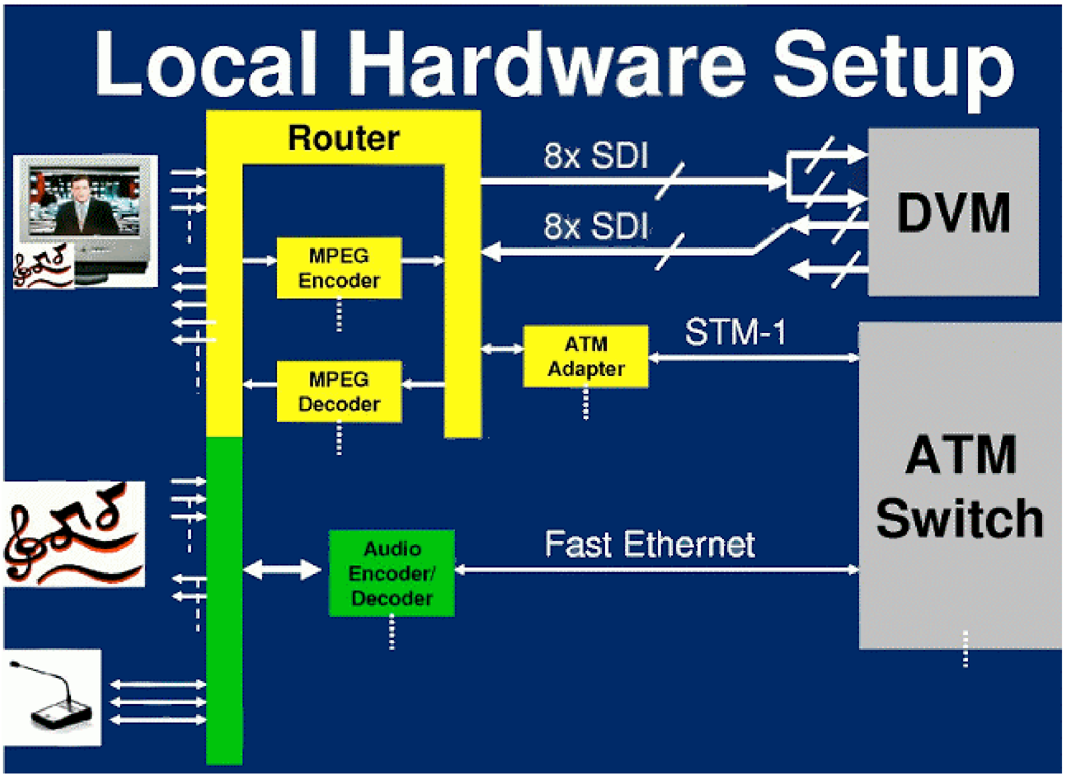


**Figure 47: Example**

Each local site is also connected with its main transmitter station.

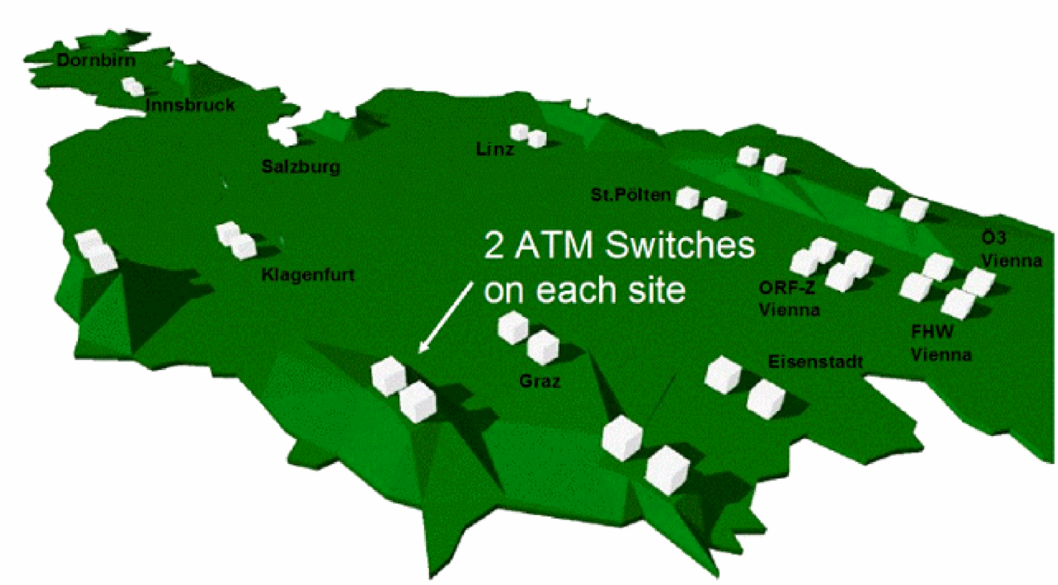


**Figure 48: ATM network (1)**

All sites are connected via a mesh network either by STM-1 fibre or radio links.

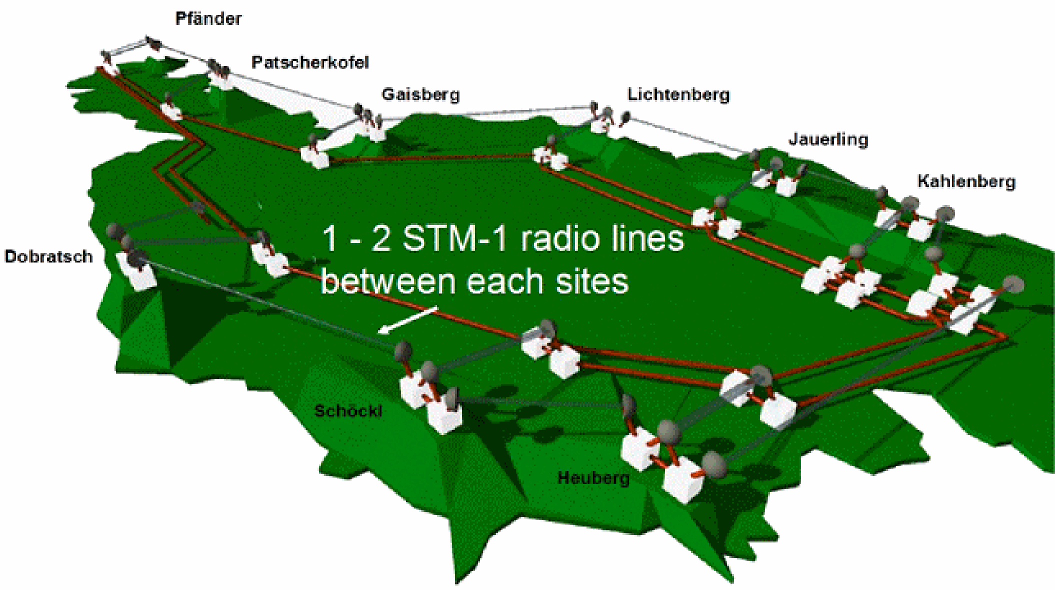


**Figure 49: ATM network (2)**

Figure 50 shows how a sample connection is built up through the network and how fault tolerance is achieved by the management system.

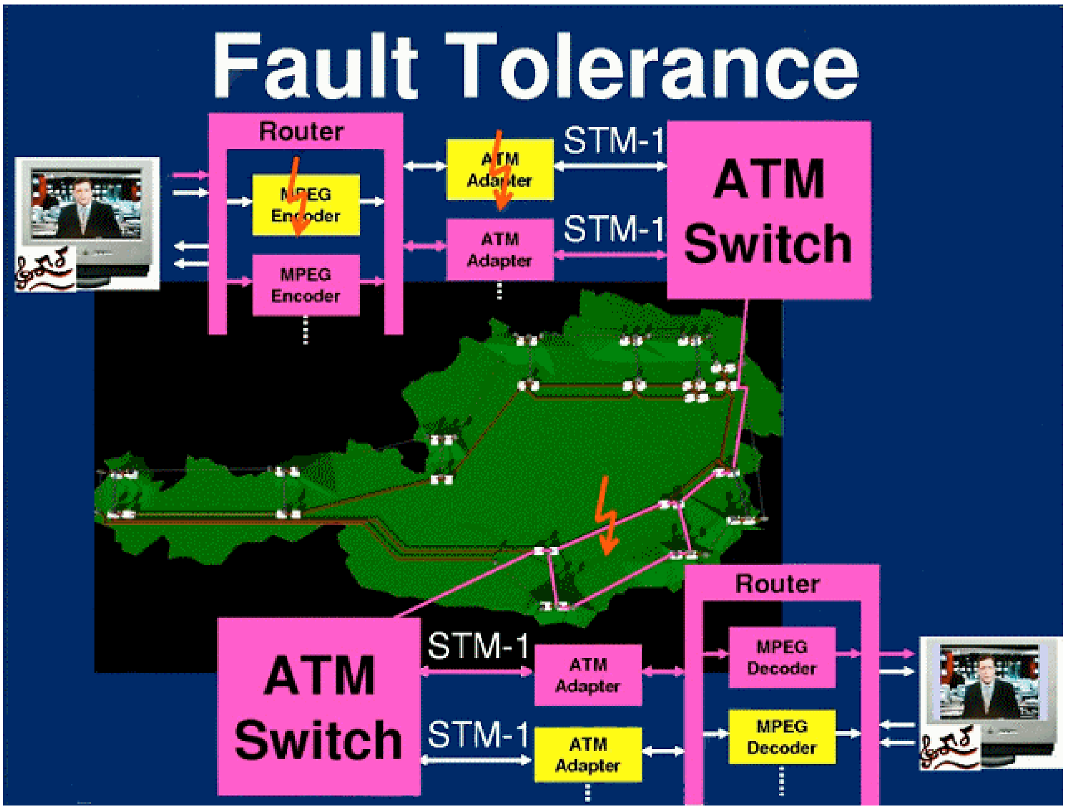


**Figure 50: L-Net Fault tolerance**

Nearly all event places in Austria can get connected with this network to transmit video, audio, intercom and LAN to one or more local studios or to the ORF Centre.

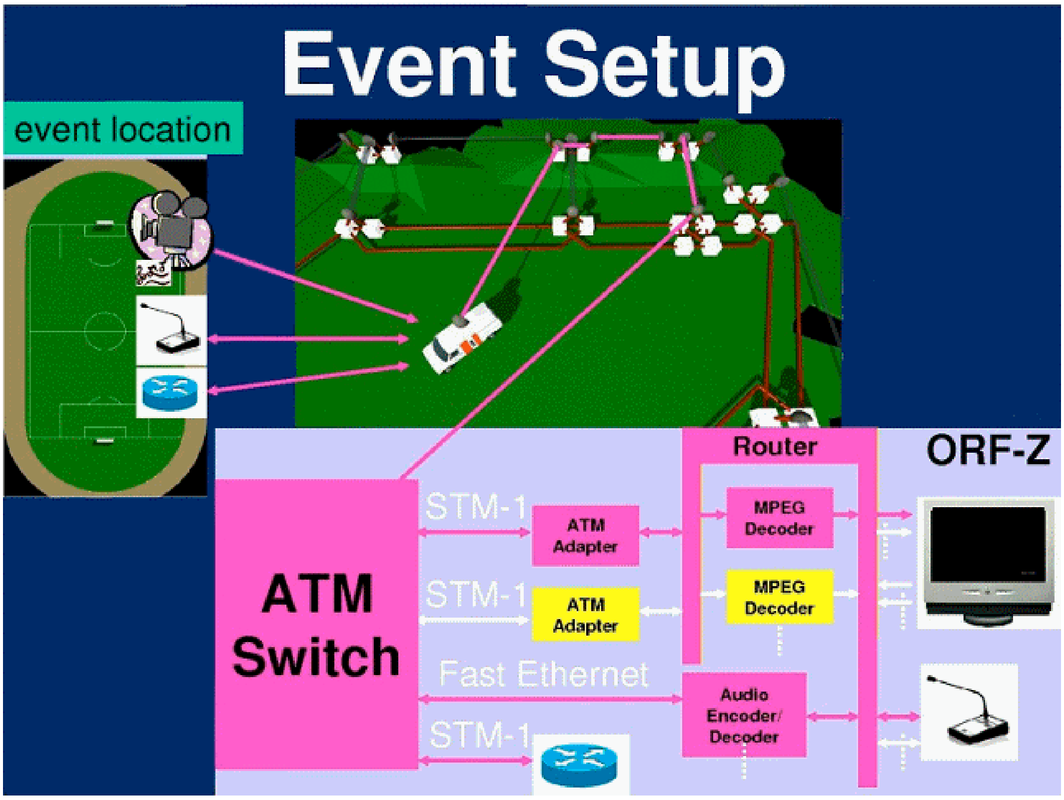


**Figure 51: Event setup**

The umbrella management makes this network easily useable for normal operators without ATM, MPEG and hardware setup knowledge.

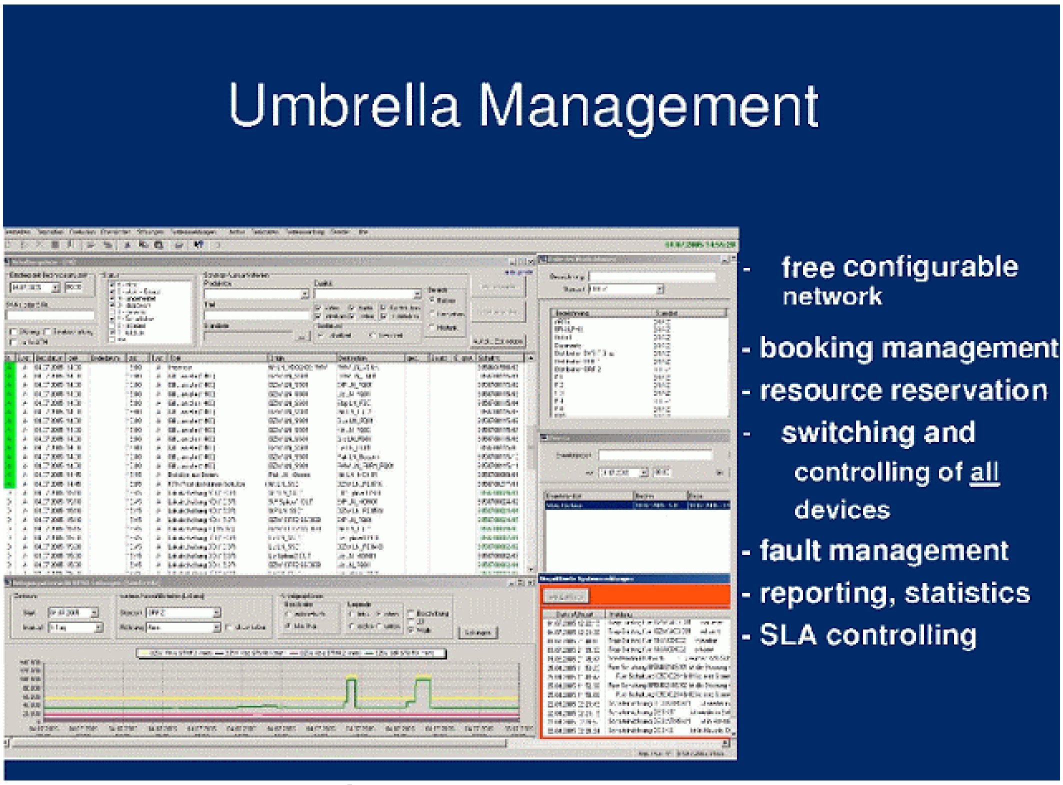


**Figure 52: L-Net Management**

## 4.5 VRT WAN

The VRT-WAN network consists of two interconnected networks (interconnection at the main office of the VRT):

- The fibre optical network in Brussels
- A SDH/PDH over microwave network in Flanders

### Brussels network

Figure 55 shows the fibre optic network in Brussels. It connects among others the following places:

- EU buildings: EU parliament, Council of Ministers (Justus Lipsius), European Commission (Berlaymont)
- Belgian and Flemish government buildings
- Broadcasters: ORF, ZDF, WDR, …
- IPC (International Press Centre) / EBU at IPC


**Figure 53 The Brussels VRT network**

The network is built with CWDM equipment (16 wavelengths that can be flexibly used in either direction). The fibre required for this network is leased (dark fibre). The CWDM equipment is installed and operated by the VRT. The network is mainly used to transport uncompressed video (SDI) but it also transports data (FE and GbE), intercom and suchlike. Each SDI signal is transported over one wavelength. Since the summer of 2006 the CWDM equipment has enabled the transportation of 4 x SDI (or ASI) on one wavelength.

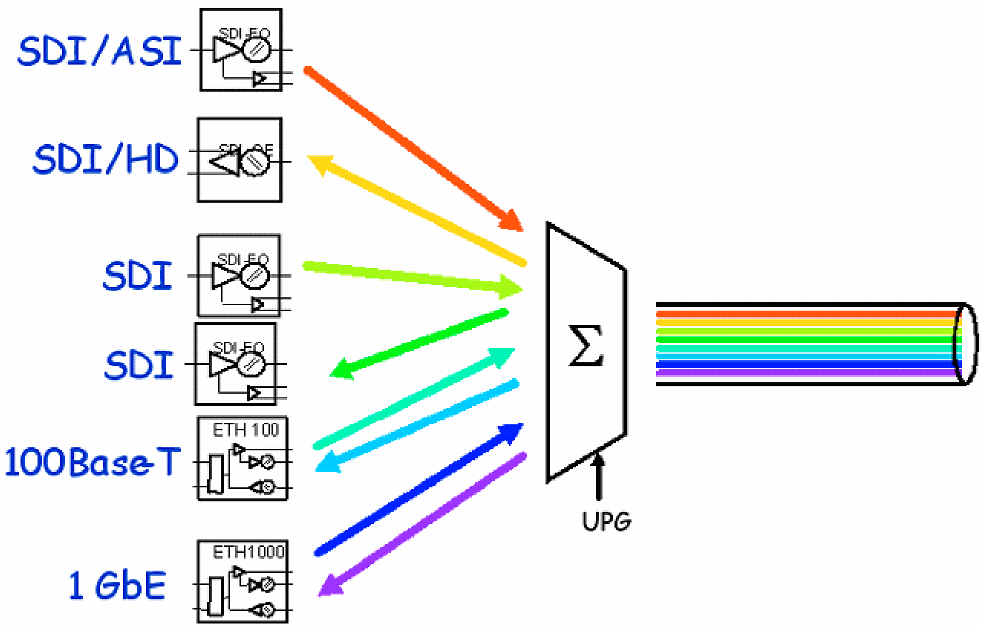


**Figure 54: CWDM technology**

### The Flemish network

The Flemish network consists of a SDH star network and a PDH ring network, both on microwave radio links. The SDH network has a capacity of 2 x STM-1 and is used to transport audio (E1), data (IP over E1) and video (compressed into E3 via ATM). The SDH network only connects the major transmitter sites, from where among others analogue radio and analogue and digital TV transmissions are done. The 8 x E1 PDH ring network connects the (smaller) DAB transmitter sites.

Since the beginning of 2006 some links in the Flemish network are also being deployed over (dark) fibre. The same CWDM equipment as in the Brussels network is used to transport video (PAL+ and DVB-ASI) and audio (AES/EBU embedded in SDI). Two wavelengths are used to transport bidirectional STM-4 NG-SDH signals. NG-SDH multiplexers are used for the transport of DAB (E1), data (IP over 100base-T). Switching of video signals is done with video matrices.

## 5. Conclusions

There is no general advice as to which technology to use. It depends on the broadcaster's needs and on the situation in the relevant country (such as the availability of dark fibre etc.). The descriptions and the assessment of the different technologies in each section should be a helpful tool for a broadcaster to decide which solution suits his needs best.

## 6. Abbreviations and Acronyms

| | |
|---|---|
| AAL | ATM Adaptation Layer |
| ADM | Add-Drop Multiplexer |
| ASI | Asynchronous Serial Interface |
| ATM | Asynchronous Transfer Mode |
| BGP | Border Gateway Protocol |
| C-band | Conventional band |
| CMIP | Common Management Information Protocol |
| CoS | Class of Service |
| CRC | Cyclic Redundancy Check |
| CR-LDP | Constraint-based Routed Label Distribution Protocol |
| CSI | Convergence Sublayer Indication |
| CSMA/CD | Carrier Sense Multiple Access-Collision Detection |
| CWDM | Coarse Wavelength Division Multiplexing |
| DVB | Digital Video Broadcasting |
| DWDM | Dense Wavelength Division Multiplexing |
| DXC | Digital Cross Connect |
| EBU | European Broadcasting Union |
| EDFA | Erbium Doped Fibre Amplifier |
| FE | Fast Ethernet (100 Mbit/s) |
| FEC | Forward Error Correction |
| *FEC* | Forward Equivalence Class (MPLS) |
| FIFO | First in, First out |
| FR | Frame Relay |
| FSC | Fibre Switch Capable |
| GbE | Giga bit Ethernet |

| | |
|---|---|
| Gbit/s | Giga bit per second |
| GFP | Generic Frame Procedure |
| GHz | Giga Hertz |
| GMPLS | Generalized Multi-Protocol Label Switching |
| IETF | Internet Engineering Task Force |
| IGMP | Internet Group Membership Protocol |
| ILM | Incoming Label Map |
| IP | Internet Protocol |
| IS-IS | Intermediate System to Intermediate System |
| ITU-T | International Telecommunication Union, Telecommunication standardisation sector |
| km | kilometre |
| LAN | Local Area Network |
| L-band | Long band |
| L2SC | Layer 2 Switch Capable |
| LDP | Label Distribution Protocol |
| LER | Label Edge Router |
| LMP | Link Management Protocol |
| LSC | Lambda Switch Capable |
| MAC | Medium Access Control |
| Mbit/s | Mega bit per second |
| MEMS | Micro-Electro Mechanical Systems |
| MOSPF | Multicast OSPF |
| MPIS | MultiProtocol Lambda (Wavelength) Switching |
| MPLS | MultiProtocol Label Switching |
| NHLFE | Next Hop Label Forwarding Entity |
| nm | nanometre |
| NOC | Network Operation Centre |
| OADM | Optical Add Drop Multiplexer |
| OCh | Optical Channel |
| ODU | Optical channel Data Unit |
| OH | OverHead |
| OMS | Optical Multiplex Section |
| OPU | Optical channel Payload Unit |
| OSPF | Open Shortest Path First |
| OTN | Optical Transport Network |
| OTS | Optical Transmission Section |
| OTU | Optical Transport Unit |
| OXC | Optical Cross Connect |
| PDH | Plesiochronous Digital Hierarchy |
| PIM | Protocol Independent Multicast |
| POH | Packet Overhead |
| POS | Packet over SONET/SDH |
| PPP | Point-to-Point |

| | |
|---|---|
| PSC | Packet Switch Capable |
| PSTN | Public Switched Telephone Network |
| PVC | Permanent Virtual Connection |
| QoS | Quality of Service |
| RFC | Requests For Comments |
| ROADM | Reconfigurable Optical Add Drop Multiplexer |
| RSVP | Resource ReSerVation Protocol |
| RTP | Real-time Protocol |
| SDH | Synchronous Digital Hierarchy |
| SDI | Serial Digital Interface |
| SONET | Synchronous Optical Network |
| SN | Sequence Number |
| SNMP | Simple Network Management Protocol |
| SNP | Sequence Number Protection |
| SRLG | Shared Risk Link Groups |
| SS7 | Signalling System No. 7 |
| STM | Synchronous Transport Module (or Mode) |
| TCP | Transmission Control Protocol |
| TDM | Time Division Multiplexing |
| TE | Traffic Engineering |
| THz | Tera Hertz |
| ToS | Type of Service |
| TTL | Time to Live |
| UDP | User Datagram Protocol |
| VC | Virtual Container |
| VCAT | Virtual Concatenation |
| VCI | Virtual Channel (Connection) Identifier |
| VPI | Virtual Path Identifier |
| VPN | Virtual Private Network |
| WAN | Wide Area Network |
| WDM | Wavelength Division Multiplexing |

# 7. References

[PROMPEG 1] Pro-MPEG Code of Practice #3 release 2: Transmission of Professional MPEG-2 Transport Streams over IP Networks, http://www.pro-mpeg.org/

[PROMPEG 2] Pro-MPEG Code of Practice #4 release 1: Transmission of High Bitrate Studio Streams over IP Networks http://www.pro-mpeg.org/

[Perkins] Colin Perkins: "RTP". ISBN 0-672-32249-8

[ETSI06]: ETSI TS 102 034 V1.2.1 (2006-09): Digital Video Broadcasting (DVB); Transport of MPEG-2 Based DVB Services over IP Based Networks