INTERNATIONAL TELECOMMUNICATION UNION

STUDY GROUP 13

TELECOMMUNICATION STANDARDIZATION SECTOR

STUDY PERIOD 2001-2004

English only

Original: English

Question:	5/13	Geneva, 3-12 February 2004

TEMPORARY DOCUMENT

Source:	Editor
Title:	Draft Recommendation Y.1413 (formerly Y.tdmpls)

ITU-T Recommendation Y.1413

TDM-MPLS network interworking - User plane interworking

Summary

This ITU-T Recommendation addresses required functions for network interworking between TDM networks up to DS3 or E3 rates and MPLS networks. This Recommendation addresses user plane interworking mechanisms and procedures. Details of the interworking model and required interworking functions are described.

Keywords

TDM, MPLS, Interworking, Network, user plane

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Introduction

There is a need to define interworking between conventional synchronous or plesiochronous networks (hereafter denoted TDM networks) with MPLS networks. Such interworking must ensure that TDM timing, signalling, voice quality, and alarm integrity be maintained.

Recommendation Y.1413

TDM-MPLS network interworking–User plane interworking

1 Scope

This Recommendation focuses on required functions for network interworking between TDM and MPLS, specifically the user plane interworking mechanisms and procedures for transport. In particular it specifies a list of requirements, interworking scenarios and interworking encapsulation formats and semantics for TDM-MPLS network interworking. Since TDM connections are inherently point-to-point, this interworking defines a single connection between two IWFs. This Recommendation only addresses TDM rates up to and including T3 and E3.

2 References

The following ITU-T Recommendations and other references contain provisions, which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] ITU-T Recommendation G.705 (2000) Characteristics of plesichronous digital hierarchy (PDH) functional blocks

[2] ITU-T Recommendation G.702 (1988) - Digital hierarchy bit rates

[3] ITU-T Recommendation G.704 (1998): Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44736 kbit/s hierarchical levels

[4] ITU-T Recommendation G.751 (1988): Digital multiplex equipments operating at the third order bit rate of 34 368 kbit/s and the fourth order bit rate of 139 264 kbit/s and using positive justification

[5] ANSI T1.107 (1995): Digital Hierarchy - Format Specification

[6] ETSI GSM 08.51 (Version 8.0.0 1999) Digital cellular telecommunications system (Phase 2+); Base Station Controller - Base Transceiver Station (BSC - BTS) interface; General aspects

[7] ITU-T Recommendation Y.1411 (2003) ATM-MPLS Network Interworking – Cell Mode user Plane Interworking

[8] IETF RFC 3031 (2001): Multiprotocol label switching architecture

[9] ITU-T Recommendation G.805 (2000) General functional architecture of transport networks

[10] ITU-T Recommendation V.36 (1988) Modems for synchronous data transmission using 60-180 kHz group band circuits

[11] ITU-T Recommendation V.37 (1988) Synchronous data transmission at a data signalling rate higher than 72 kbit/s using 60-108 kHz group band circuits

[12] ITU-T Recommendation I.231.1 (1993) Circuit mode bearer service categories – circuit mode 64 kbit/s unrestricted, 8 kHz structured bearer service

[13] ITU-T Recommendation Q.700 (1993): Introduction to CCITT Signalling System No. 7

[14] ITU-T Recommendation Q.931 (1998): ISDN user-network interface layer 3 specification for basic call control

[15] ITU-T Recommendation G.823 (2000): The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy

[16] ITU-T Recommendation G.824 (2000): The control of jitter and wander within digital networks which are based on the 1544 kbit/s hierarchy

[17] ITU-T Recommendation Y.1711 (2002): OAM mechanism for MPLS networks

[18] IETF RFC 3270 (2002): MPLS Support of Differentiated Services

[19] IETF RFC 3209 (12/2001) RSVP-TE: Extensions to RSVP for LSP Tunnels

[20] IETF RFC 3032 (2001): MPLS label stack encoding

[21] IETF RFC 3550 (2003): RTP: A Transport Protocol for Real-Time Applications

[22] ITU-T Recommendation I.363.1 (1996) B-ISDN ATM Adaptation Layer specification: Type 1 AAL

[23] ATM Forum af-vtoa-0078.000 (1997) Circuit Emulation Service (CES) 2.0

[24] ITU-T Recommendation G.802 (1993) Interworking between networks based on different digital hierarchies and speech encoding laws

[25] ITU-T Recommendation I.363.2 (2000): B-ISDN ATM Adaptation Layer specification: Type 2 AAL

[26] ITU-T Recommendation I.366.2 (1999): AAL type 2 service specific convergence sub-layer for trunking

[27] ITU-T Recommendation G.826 (1999): Error performance parameters and objectives for international constant bit rate digital paths at or above the primary rate

[28] ITU-T Recommendation Q.921 (1997) ISDN user network interface – Data link layer specification

[29] ITU-T Recommendation G.703 (1998): Physical/electrical characteristics of hierarchical digital interfaces

[30] ITU-T Recommendation G.827 (2003) Availability parameters and objectives for path elements of international constant bit-rate digital paths at or above the primary rate

[31] ITU-T Recommendation G.1020 (2003) Performance parameter definitions for quality of speech and other voice band applications using IP networks

[32] ITU-T Recommendation P.562 (2000) Analysis and interpretation of INMD voice service measurements

[33] ITU-T Recommendation P.862 (2001) Perceptual evaluation of speech quality (PESQ)

[34] ITU-T Recommendation G.114 (2000): One-way Transmission Time

3 Definitions

This Recommendation defines the following terms:

TDM: A term that conventionally refers to the isochronous bit streams used in telephony networks; in particular those belonging to PDH (plesiochronous digital hierarchy) as described in ITU-T Recommendation G.705 [1]. The bit rates traditionally used in various regions of the world are detailed in G.702 [2].

Structured TDM: TDM with any level of structure imposed by a FA (Frame Alignment Signal), such as that defined in [3], [4], [5], or [6].

Unstructured TDM: A TDM bit stream with no structure imposed, so that all bits are available for user data.

Structure -Agnostic Transport: Transport of unstructured TDM, or of structured TDM when the structure is completely disregarded by the transport mechanism. Structure-agnostic transport maintains the precise bit sequence of data and any structure overhead that may be present. The encapsulation provides no mechanisms for the location or utilization of a FA.

Structure -Aware Transport: Transport of structured TDM taking at least some level of the structure into account. In structure-aware transport it is not required to carry all bits of the TDM bit-stream over the MPLS network; specifically, FAS may be stripped at ingress and regenerated at egress.

Structure -Locked Encapsulation: Encapsulation utilized for structure-aware TDM transport where TDM structure boundaries are indicated by packet payload boundaries.

Structure -Indicated Encapsulation: Encapsulation utilized for structure-aware TDM transport where TDM structure boundaries are indicated by pointers.

TDM Segment: Octets extracted from a continuous TDM stream. Each octet in the TDM segment is filled with bits starting from its most significant bit, and octets are placed in the segment in the order received.

Interworking: See Recommendation Y.1411 [7].

Interworking Function (IWF): See Recommendation Y.1411 [7].

Ingress IWF: The IWF where a continuous TDM stream is segmented and encapsulated into MPLS packets (TDM-to-MPLS direction).

Egress IWF: The IWF where the TDM segments are extracted from MPLS packets and reassembled into a continuous TDM stream (MPLS-to-TDM direction).

4 Abbreviations

This Recommendation uses the following abbreviations.

AAL ATM Adaptation Layer

AIS Alarm Indication Signal

AP Access Point

ATM Asynchronous Transfer Mode

CAS Channel Associated Signalling

CES Circuit Emulation Service

CCS Common Channel Signalling

CP Connection Point

EXP Experimental Bits

FAS Frame Alignment Signal

HDLC High level Data Link Control

IWF Interworking Function

LOF Loss of Frame Synchronization

LOS Loss of Signal

LSP Label Switched Path

LSR Label Switching Router

MPLS Multi-Protocol Label Switching

MTU Maximum Transport Unit

OAM Operation and Maintenance

PDB Per Domain Behaviour

PDU Protocol Data Unit

PHB Per Hop Behaviour

PM Performance Monitoring

PSC PHB Scheduling Class

QoS Quality of Service

RDI Remote Defect Indication

RFC Request for Comments

RTP Real Time Protocol

SAR Segmentation And Reassembly

TCP Termination Connection Point

TDM Time Division Multiplex

TTL Time to Live

5 Conventions

This recommendation uses traditional terminology for digital signals at the various levels of the G.702 rate hierarchy. In particular, the first level digital signal of rate 2048 kbit/s (P12 in G.705 terminology) is designated E1, and the third level signal of rate 34368 kbit/s derived from it (P31), E3. Similarly, the first level signal of rate 1544 kbit/s (P11) is designated T1, its second level derivative of rate 6312 kbits/s (P21), T2, and its third level derivative at rate 44736 kbits/s (P32), T3.

6 TDM-MPLS interworking

The Multi-protocol label switching (MPLS) technology [8] allows multiple services (such as IP, ATM, frame relay, and TDM) to be supported over a single networking infrastructure.

This Recommendation defines interworking with TDM services up to and including T3 or E3 rates. Interworking of MPLS with higher rate TDM services, such as SONET/SDH, are beyond the scope of this Recommendation.

Figure 6-1 provides a general network architecture for TDM – MPLS network interworking where TDM networks are interconnected through an MPLS network. For the TDM-to-MPLS direction, the continuous TDM stream is segmented and encapsulated into an MPLS packet by the interworking function (IWF). For the MPLS-to-TDM direction, the TDM segments are extracted from the MPLS packets and the continuous TDM stream is reassembled.

Figure 6-2 depicts the network functional architecture of TDM – MPLS interworking using the diagrammatic techniques of G.805 [9]. Examples for specific scenarios are given in Appendix III.

Figure 6-3 shows the network reference model and protocol layers for TDM-MPLS user plane interworking.

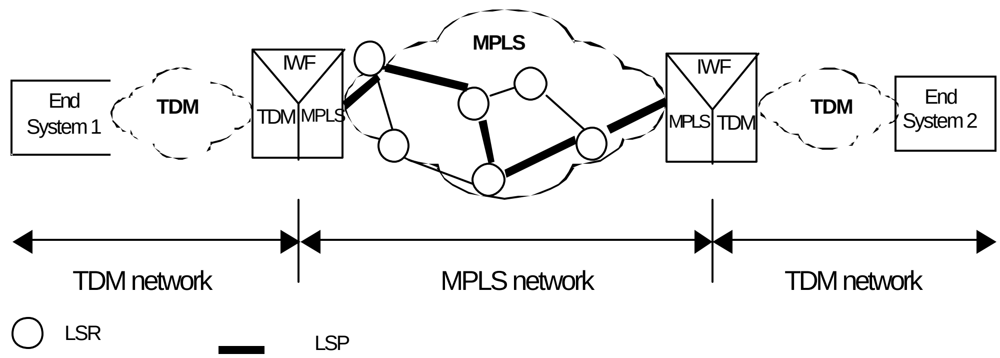


Figure 6-1: Reference architecture for TDM-MPLS network interworking

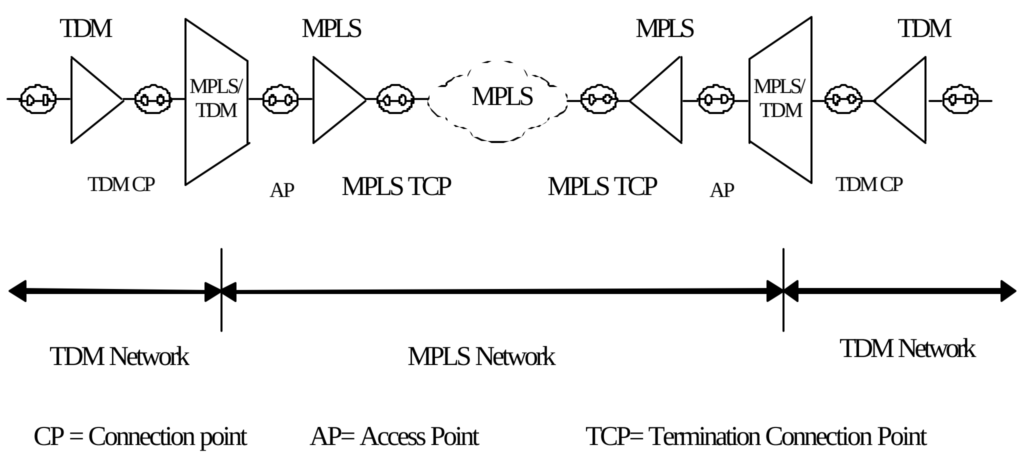


Figure 6-2: Functional architecture of TDM-MPLS interworking depicted according to the diagrammatic conventions of G.805

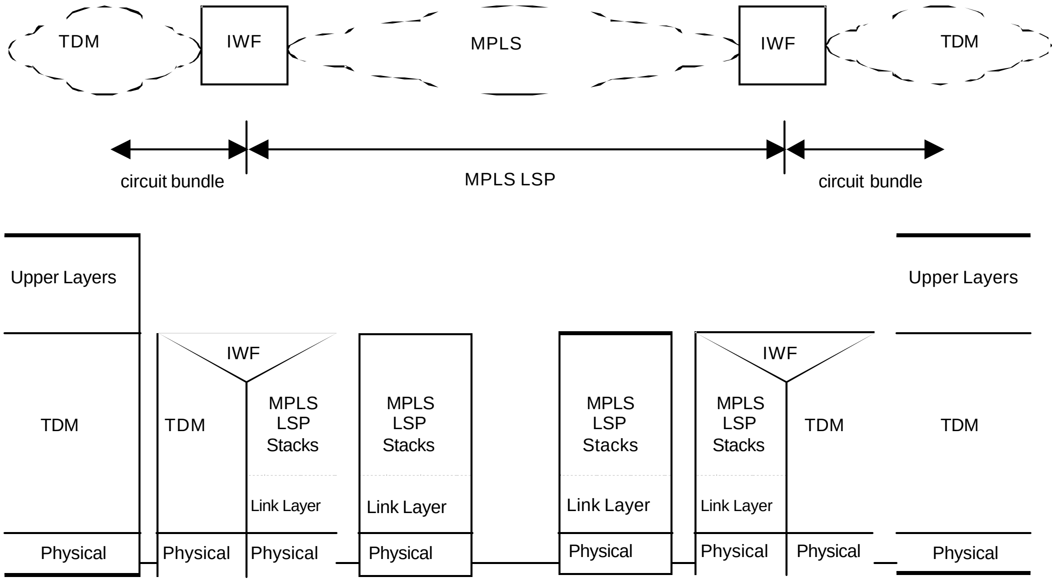


Figure 6-3: Network reference model and protocol layers for TDM-MPLS user plane interworking

7 General Requirements

7.1 User plane requirements

For transparent transfer of TDM in the user plane, the following capabilities are required:

a) The ability to transport multiple TDM streams in an interworking LSP.

b) Support for bi-directional connections with symmetric bandwidth and binding to the duplex TDM.

c) The ability to transport the following unstructured TDM types:

1. T1 at 1544 kbit/s
2. E1 at 2048 kbit/s
3. T2 at rate 6312 kbit/s
4. synchronous serial data as defined in V.36 [10] and V.37 [11]
5. N*64k (i.e. 64 kbit/s, 128 kbit/s, 192 kbit/s) data such as defined in I.231.1 [12]
6. T3 at 44736 kbit/s as defined in T1.107 [5]
7. E3 at 34368 kbit/s as defined in G.751. [4]

d) The ability to transport the following structured TDM types:

1. T1 as defined in G.704 [3]
2. fractional T1 carrying N timeslots with N from 1 to 23 as defined in T1.107
3. E1 as defined in G.704
4. fractional E1 carrying N timeslots with N from 1 to 30 as defined in G.704
5. multiple synchronous DS0s
6. T2 defined in G.704.

e) The ability to transport the structured TDM types of items d 1, 2, 3, 4, 6 with CAS signalling, as defined in T1.107 and G.704.

f) The ability to transport trunk-associated CCS signalling, e.g. as defined in Q.700 [13] and Q.931 [14].

g) The ability of the egress IWF to derive timing from an external clock signal, or to exploit a common clock source, or to recover TDM timing by adaptive means.

h) Conformance of timing recovery to the jitter and wander specifications of [15] and [16].

i) The ability to interwork with existing CES services.

j) The ability to reliably detect packet loss and misordering.

k) The ability to inject filler data to compensate for lost packets.

l) The ability to properly function with deployed MPLS switches that differentiate between IP and interworking LSP packets based on the initial four bits of the packet content.

m) The ability of the IWFs to maintain TDM frame synchronization (and multi-frame synchronization when applicable) for structure-aware transport.

n) The ability to set payload length to ensure that packet size does not exceed the path MTU.

7.2 Control plane aspects

For transparent transfer of TDM related services, the following are to be signalled or provisioned:

a) Setup and configuration of transport and interworking LSPs.

b) Request for two point-to-point connections with equal bandwidth, and the association of their interworking labels to create a bi-directional connection.

c) The number of payload octets per MPLS packet for a given interworking LSP.

d) The rate and type of TDM traffic.

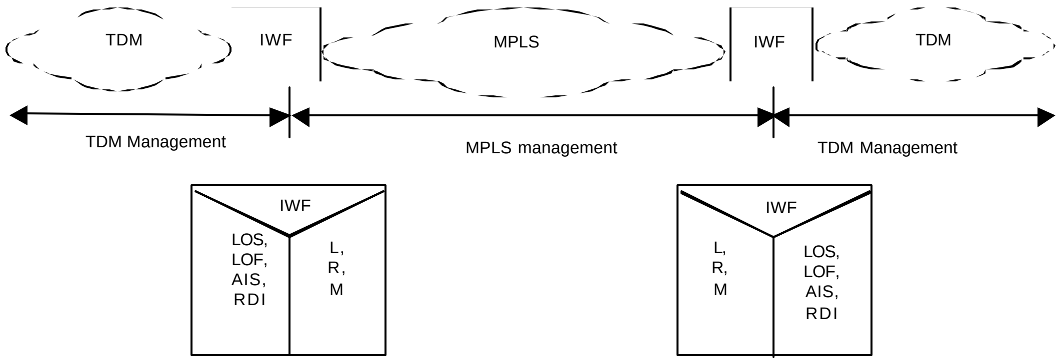
7.3 Fault Management Aspects

The interworking function shall support transfer of defect information between MPLS and TDM networks, as depicted in Figure 7-1. In particular, local TDM defects, such as loss of signal or loss of synchronization, shall be signalled from ingress to egress IWF; and MPLS defects, such as misordering or explicit MPLS defect indication, shall be signalled from egress to ingress IWF.

The interworking function shall transfer TDM defect indications through the MPLS network by setting appropriate flags in the common interworking indicators. The encoding need not be one-to-one, i.e., a single indicator of invalid TDM data may be used to indicate multiple TDM defects or indications (e.g. LOS, LOF or AIS). In addition, if applicable an appropriate alarm shall be sent to the management layer. Client server interactions between TDM and MPLS OAM [17] are for further study.

When the egress IWF detects remote MPLS defects, in addition to informing the ingress IWF of the defect and maintaining timing integrity of the local TDM interface, it shall send the appropriate alarm to the management layer.

The ability to distinguish between faults in the MPLS network and those in the remote TDM network shall be provided.



LOS= Loss of Signal defect (detected only)
LOF= Loss of Frame synchronization (detected only)
AIS = Alarm Indication Signal
RDI = Remote Defect Indication

Figure 7-1: Functional representation of TDM-MPLS fault management

7.4 Traffic Management aspects

The transport LSP shall be capable of providing the required QoS for all TDM connections. The transport LSP shall meet the aggregate bandwidth requirements of all TDM connections transported.

If the MPLS network is Diffserv enabled according to RFC 3270 [18], then EF-PHB (expedited forwarding) class based PDB shall be used, in order to provide a low latency and minimal jitter service. It is suggested that the transport LSP be somewhat over provisioned.

If the MPLS network is Intserv enabled according to RFC 3209 [19], then GS (Guaranteed Service) with the appropriate bandwidth reservation shall be used in order to provide a bandwidth guarantee equal to or greater than that of the aggregate TDM traffic.

The delay introduced by the MPLS network should be measured prior to traffic flow, to estimate latency.

7.5 Connection admission control for the IWF

When bandwidth guarantees can be provided, then the IWF should provide connection admission control. The admission decision shall be based on the total bandwidth allocation of the transport LSP, the bandwidth already allocated to existing interworking LSPs, and the bandwidth requested. When sufficient bandwidth is available the request may be granted. When bandwidth is insufficient, either the TDM connection request is denied or the IWF may request an increase in transport LSP bandwidth in order to admit the TDM connection.

8 Functional group considerations for TDM-MPLS Network interworking

Figure 8-1 provides an illustration of functional grouping for TDM-MPLS network interworking.

Transport label
Interworking label
Common interworking indicators
Optional Timing Information
TDM Payload

Figure 8-1: TDM-MPLS interworking functional groups

8.1 Transport label

Since LSPs are unidirectional while the TDM is inherently bidirectional, the association of two transport LSPs in opposite directions will be required. The LSPs may have different label values.

The 4-octet transport label identifies a LSP used to transport traffic between two IWFs. The transport label is a standard MPLS shim header [20] that is processed at each LSR. The S bit is cleared for this label, indicating that this is not the bottom of the label stack. The setting of the EXP and TTL fields of the transport label is outside the scope of this Recommendation.

8.2 Interworking label

Since LSPs are unidirectional while the TDM is inherently bidirectional, the association of two interworking LSPs in opposite directions will be required. The LSPs may have different label values.

The interworking function maintains context information that associates TDM connections with the interworking LSP.

The 4-octet interworking label uniquely identifies one interworking LSP carried inside a transport LSP. More than one interworking LSP may be supported by one transport LSP.

The interworking label is a standard MPLS shim header [20], with its S bit set, to indicate the bottom of the label stack. Since TDM-MPLS interworking is a strict point-to-point application, the TTL value should be set to 2. The setting of the EXP field of the interworking label is for further study.

8.3 Common interworking indicators

The functions in the Common interworking indicators are related to the interworking LSP and are independent of any specific service or encapsulation. In general the Common interworking indicators is comprised of a control field, a fragmentation field (FRG), a length field, and a sequence number field, as depicted in Figure 8-2.

Bit							
8	7	6	5	4	3	2	1
Control							
FRG		Length indicator					
Sequence number (2 octets)							

Figure 8-2: Common interworking indicators

8.3.1 Control field

The format of the control field is depicted in Figure 8-3.

Bit							
8	7	6	5	4	3	2	1
Reserved				L	R	M	

Figure 8-3: Control field

The reserved field shall be set to zero in order to facilitate correct operation with deployed MPLS switches that differentiate between IP and interworking LSP packets based on these four bits.

The L, R and M fields provide a means of transparent transfer of TDM defect indications between IWFs. Their use should be in accordance with principles of the appropriate G series of recommendations with regards to OAM.

L Local TDM failure: The L bit being set indicates that the ingress IWF has detected or has been informed of a TDM defect impacting the TDM data. When the L bit is set the contents of the packet may not be meaningful, and the payload may be suppressed in order to conserve bandwidth. Once set, if the TDM fault is rectified the L bit shall be cleared.

R Remote Receive failure: The R bit being set indicates that the source of the packet is not receiving packets from the MPLS network. Thus the setting of the R bit indicates failure of the opposite direction. This indication can be used to signal MPLS network congestion or other network related faults. The R bit shall be set after a preconfigured number of consecutive packets are not received, and shall be cleared once packets are once again received.

M Defect Modifier: Use of the M field is optional, and when used it supplements the meaning of the L bit.

When L is cleared (indicating valid TDM data) the M field is used as follows:

M

0 0 indicates no local defect modification.

0 1 reserved

1 0 reports receipt of RDI at the TDM input to the ingress IWF

1 1 reserved.

When L bit is set (indicating invalid TDM data) the M field is used as follows:

M

0 0 indicates a TDM defect that should trigger AIS generation at the far end.

0 1 indicates idle TDM data, which should not cause any alarm to be raised. If the payload has been suppressed then appropriate idle code should be generated at egress.

1 0 indicates corrupted but potentially recoverable TDM data. The use of this indication is for further study.

1 1 reserved.

8.3.2 Fragmentation field

This field is used for fragmenting multi-frame structures into multiple packets as described in sub-clause 9.2.1. The field is used as follows:

FRG

0 0 indicates that the entire (un-fragmented) multi-frame structure is carried in a single packet

0 1 indicates the packet carrying the first fragment

1 0 indicates the packet carrying the last fragment

1 1 indicates a packet carrying an intermediate fragment.

8.3.3 Length field

When the LSP path includes an Ethernet link, a minimum packet size of 64 octets is required. This may require padding to be applied to the interworking packet payload in order to reach this minimum packet size. The padding size can be determined from the length field so that the padding can be extracted at the egress.

The Length field provides, in octets, the size of the MPLS packet payload, and its value is the sum of:

a. size of the Common interworking indicators,
b. size of the optional timing information, and
c. size of the payload;

unless this sum equals or exceeds 64 octets, in which case the Length field shall be set to zero.

8.3.4 Sequence number field

The Sequence number field is a two-octet field that is used to detect lost packets and packet mis-ordering.

The sequence number space is a 16-bit, unsigned circular space, set and processed as defined below.

8.3.4.1 Setting the sequence numbers

The following procedures apply at the ingress IWF (TDM-to-MPLS direction):

- The sequence number should be set to a random value for the first MPLS packet transmitted on the interworking LSP.
- For each subsequent MPLS packet, the sequence number shall be incremented by 1, modulo 2^{16}.

8.3.4.2 Processing the sequence numbers

The purpose of the sequence number processing is to detect lost or misordered packets. The treatment of lost packets is discussed in clause 11. Misordered packets should be re-ordered if possible. The mechanism by which a packet is considered lost is implementation specific.

The following procedures apply at the egress IWF (MPLS-to-TDM direction):

- The egress IWF maintains an expected sequence number.
- The first packet received from the MPLS network is always considered to be the expected packet, and the expected sequence number is equated to its sequence number.
- If the sequence number equals or is greater (in the cyclic sense) than the expected number then the expected sequence number is set to the received number incremented by 1 modulo 2^{16}, otherwise the expected number is unchanged.

8.4 Optional Timing Information

Optional timing information may be carried using the RTP header defined in [21].

If used, the RTP header shall appear in each interworking packet immediately after the common interworking indicators field and immediately before the payload.

The fields of RTP header shall be used as following:

1. V (version) is always set to 2
2. P (padding), X (header extension), CC (CSRC count) and M (marker) are always set to 0. Accordingly, RTP header extensions, padding and contributing synchronization sources are never used.
3. PT (payload type) is used as following:

   a) A PT value shall be allocated from the range of dynamic values for each direction of the interworking LSP.

   b) The ingress IWF shall set the PT field in the RTP header to the allocated value.
4. The sequence number in the RTP header shall be equal to the sequence number in the common interworking indicators.
5. Timestamps are used for carrying timing information over the network, as explained in section 9:

   a) Their values are generated in accordance with the rules established in [21].

   b) The clock frequency used for generating timestamps should be an integer multiple of 8 kHz. Guidance for the proper selection of this clock frequency is given in Appendix V.
6. The SSRC (synchronization source) field in the RTP header may be used for detection of misconnections.

9 Payload Formats

Sub-clause 9.1 specifies the payload format for structure-agnostic transport, while sub-clause 8.2 defines two payload formats for structure-aware transport. Sub-clause 9.2.1 specifies the structure-locked encapsulation, and sub-clause 9.2.2 specifies structure-indicated encapsulation based on AAL type 1, as defined in [22] and [23].

9.1 Structure-Agnostic Transport

Structure-agnostic transport completely disregards any TDM structure, in particular the structure imposed by standard TDM framing [3].

The payload format for structure-agnostic transport supports all the TDM services of sub-clause 7.1 items c, d and e.

For structure-agnostic transport arbitrary fixed length TDM segments are used, with no byte or frame alignment implied. The number of octets in the TDM segment:

- shall be defined upon initialization,
- may be exchanged using a signalling protocol,
- shall be the same for both directions, and
- shall remain unchanged for the lifespan of the connection for valid TDM data.

Guidance for the proper selection of the number of octets per packet is given in Appendix VI.

When the L bit is set TDM-MPLS packets may omit invalid TDM payloads in order to conserve bandwidth.

NOTE: The use of AAL type 1 as described in sub-clause 9.2.2 below may also be used for structure-agnostic transport. Examples where this may be beneficial are when interworking with ATM-based circuit emulation systems, or when SRTS-based clock recovery is used.

9.1.1 Octet-Aligned T1 Payload Format

T1 circuits may be delivered to the ingress IWF padded to an integer number of bytes, as described in G.802 Annex B [24]. In this format, the payload consists of an integer number of 25-byte sub-frames, each sub-frame consisting of 193 bits of TDM data and 7 bits of padding as shown in Figure 9.1 below:

Bit 8	7	6	5	4	3	2	1
First octet of TDM data							
Second octet of TDM data … Twenty third octet of TDM data							
Twenty fourth octet of TDM data							
***	7 Padding bits						

***: Last bit of TDM data

Figure 9-1: Octet-Aligned T1 payload format

9.2 Structure-Aware transport

Structure-aware transport maintains correct operation of the remote TDM interface by removing structure overhead at ingress and regenerating it at egress, and preserves integrity of the TDM structure by structure-locking or structure indication.

The payload formats for structure-aware transport support all the TDM services of sub-clause 7.1 items d and e.

9.2.1 Structure-locked Encapsulation

All packets shall carry the same amount of TDM data in both directions of the interworking LSP. Thus, the time required to fill a packet with TDM data is always the same.

If the egress IWF substitutes filler data due to having received a packet with L bit set, it shall ensure that proper FAS bits are sent to the TDM network.

For services specified in sub-clause 7.1 item d the packet payload is comprised of an integer number of frames, and is aligned on the first octet of the first frame. If the packet payload is comprised of M frames, the packetization latency will be M times 125 µsec.

For services specified in sub-clause 7.1 item e the packet payload is comprised of an entire multi-frame. Alternatively, the multi-frame may be fragmented into an integer number of equal-sized fragments, where the first octet of each fragment is the first octet of a frame. Each fragment is placed into a separate packet and fragmentation is indicated by the FRG field in the Common interworking indicators, as described in sub-clause 8.3.2. The CAS signalling information shall be appended as a dedicated signalling substructure, as follows:

- the four CAS bits belonging to each consecutive timeslot are placed in the signalling substructure as depicted in Figure 9-3,
- the CAS bits A, B, C, and D are positioned from most significant to least significant bit of the nibble,
- if the number of timeslots is odd, a padding nibble shall be appended in order to maintain octet alignment,

- if the multi-frame structure is fragmented among several packets, the signalling substructure is always appended to the last fragment of the structure.

The resulting payload formats are shown in Figures 9-2 and 9-3 below.

frame	bit 8 7 6 5 4 3 2 1
1	Bits belonging to timeslot 1
	Bits belonging to timeslot 2
	… Bits belonging to timeslot N
2	Bits belonging to timeslot 1
	Bits belonging to timeslot 2
	…
	Bits belonging to timeslot N
…	…
M	Bits belonging to timeslot 1
	Bits belonging to timeslot 2
	… Bits belonging to timeslot N

Note 1: Bit 8 is the most significant bit

Note 2: The packet contains M TDM frames with N timeslots per frame.

Figure 9-2: Payload format for structure-locked encapsulation without CAS (the MPLS packet does not carry a signalling substructure)

frame	bit 8 7 6 5 4 3 2 1	
1	Bits belonging to timeslot 1	
	Bits belonging to timeslot 2	
	…	
	Bits belonging to timeslot N	
2	Bits belonging to timeslot 1	
	Bits belonging to timeslot 2	
	…	
	Bits belonging to timeslot N	
…	…	
M	Bits belonging to timeslot 1	
	Bits belonging to timeslot 2	
	…	
	Bits belonging to timeslot N	
Signalling substructure	Signalling bits for timeslot 1	Signalling bits for timeslot 2
	Signalling bits for timeslot 3	…
	Signalling bits for timeslot N	Padding (see note 2)

Note 1: Bit 8 is the most significant bit

Note 2: The packet contains M TDM frames with N timeslots per frame, plus the signalling substructure.

Note 3: if N is odd, four bits of padding are added.

Figure 9-3: Payload formats for structure-locked encapsulation with CAS (the MPLS packet carries the signalling substructure)

9.2.2 Structure-indicated encapsulation

For this encapsulation the TDM bit stream is adapted using AAL Type 1, as described in [22] and [23], to form 48-octet AAL Type 1 SAR PDUs, as described in sub-clause 2.4.2 of [22].

The packet payload consists of one or more PDUs, as depicted in Figures 8.2 and 8.3. The number of PDUs per packet:

- shall be defined upon initialization,

- may be exchanged using a signalling protocol,
- shall be the same for both directions, and
- shall remain unchanged for the lifespan of the connection.

Guidance for the selection of the number of PDUs per packet is given in Appendix VII.

Bit

8	7	6	5	4	3	2	1
Common interworking indicators							
PDU payload (48 octets)							

Figure 9-4: Structure-indicated encapsulation with a single PDU per packet

Bit

8	7	6	5	4	3	2	1
Common interworking indicators							
PDU payload (48 octets)							
PDU payload (48 octets)							
PDU payload (48 octets)							

Figure 9-5: Structure-indicated encapsulation with multiple PDUs per packet

AAL type 1 differentiates between unstructured and structured data transfer, which correspond to the structure-agnostic and structure-aware transport of the present recommendation.

For structure-agnostic transport, AAL type 1 provides no inherent advantage as compared to the method of section 8.1; however, there may be scenarios for which its use is desirable. For example, when it is necessary to interwork with an existing AAL type 1 ATM circuit emulation systems, or when clock recovery based on AAL1-specific mechanisms is favoured.

Each 48-octet SAR-PDU consists of a SAR-PDU header, and a SAR-PDU payload. The SAR-PDU header contains a CSI bit that signifies the appearance of a structure pointer for structured data transfer, and may be used for clock recovery (see clause 10 below).

For unstructured AAL type 1, the 48 octets in each sub-frame contain a single octet SAR-PDU header, and 47 octets (376 bits) of TDM data.

For structure-aware transport, [23] defines two modes, structured and structured with CAS. Structured AAL type 1 carries byte-aligned TDM and maintains TDM frame synchronization by embedding a pointer to the beginning of the next frame in the SAR-PDU header. Structured AAL type 1 with CAS carries byte aligned TDM and maintains TDM frame and multi-frame synchronization by embedding a pointer to the beginning of the next multi-frame; it furthermore contains a substructure including the CAS signalling bits (see sub-clause 9.2.1).

10 Timing Aspects

TDM networks are synchronous and hierarchically distribute precise timing in order to maintain the required error performance. MPLS networks, not having been designed for TDM transport, have no inherent timing distribution mechanism, and so some other method of timing distribution must be provided.

Four principal timing distribution scenarios may be identified, that differ in the availability and placement of timing sources. The selection of timing distribution mechanism may be made independently per TDM-MPLS interworking LSP.

10.1 Timing Distribution Scenarios

In this section we describe the four timing distribution scenarios.

10.1.1 Reference Clock Available at the TDM End Systems

Figure 10.1 depicts the scenario wherein the TDM end systems share a reference clock, distributed by means beyond the scope of this recommendation. Alternatively, primary reference clocks may be available at both sites, and due to their accuracy the two clocks may be considered identical.

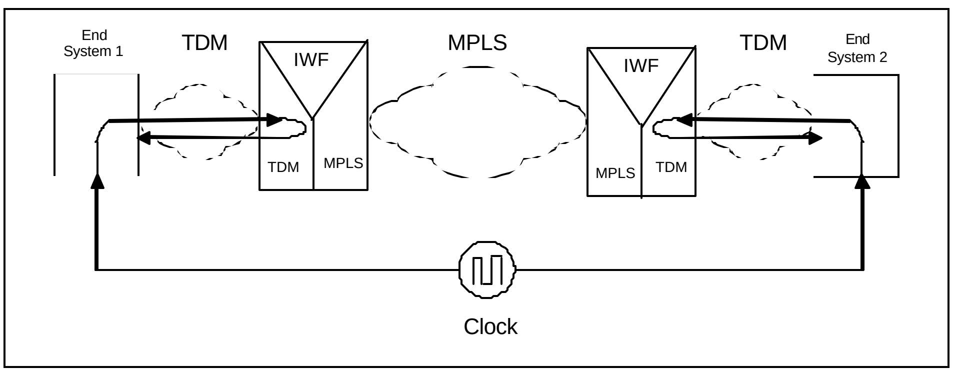


Figure 10-1: Reference clock available at end systems

In this scenario each end system uses the reference clock to generate the timing used to transmit TDM data towards the IWF. The IWFs slave their timing circuitry to this TDM input when transmitting TDM towards the end systems.

10.1.2 Reference Clock Available at the IWFs

Figure 10.2 depicts the scenario wherein the two IWFs share a reference clock, distributed by means beyond the scope of this recommendation. Each IWF uses the reference clock to generate the timing used to transmit TDM data towards the end system. The end systems slave their timing circuitry to this TDM input when transmitting TDM towards the IWF.

A scenario wherein one TDM network functions according to sub-clause 10.1.1 and the other according to the present sub-clause is also possible.

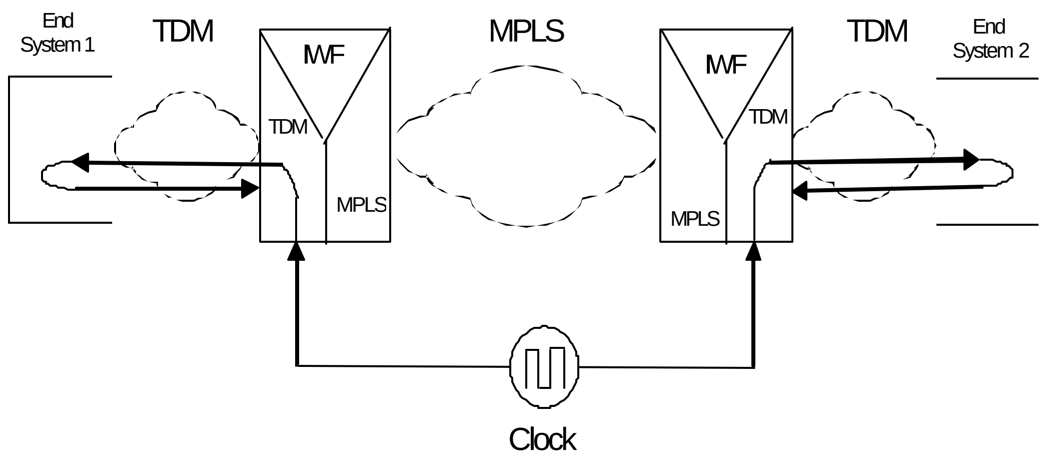


Figure 10.2: Reference clock available at IWFs

10.1.3 Common Clock Available at IWFs

Figure 10.3 depicts the scenario wherein one of the TDM end systems is required to slave its timing circuitry to that of the other, and the IWFs share a common clock independent of the TDM timing. In this case the relationship between the frequencies of the master TDM clock and the reference clock may be encoded in some manner, and transmitted across the packet network. The master frequency can then be regenerated at the remote IWF by modifying the common clock frequency based on the encoded relationship received.

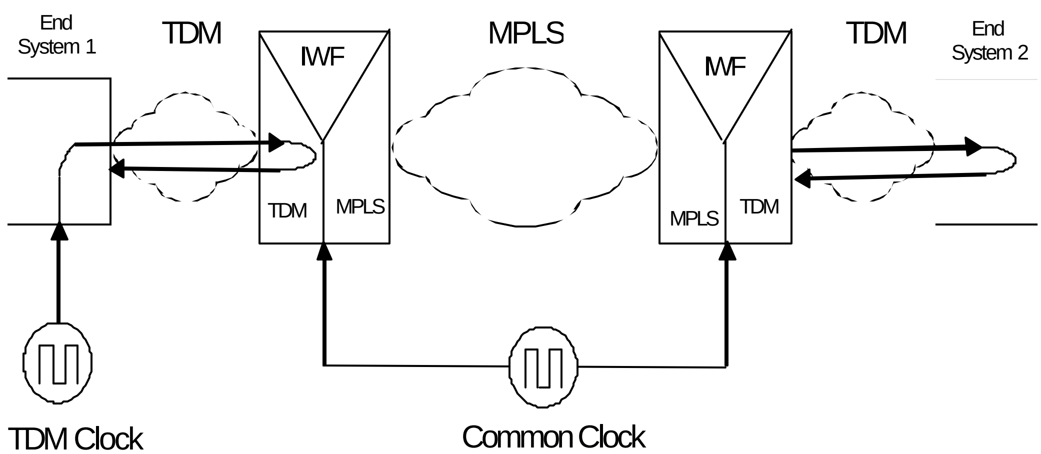


Figure 10.3: Common Clock available at IWFs

Two mechanisms for encoding the relationship between the TDM clock to be recovered and the common reference clock are well known. The SRTS mechanism described in [I.363.1] encodes the residual of the ratio between the reference and TDM frequencies, while RTP timestamps may be used to encode the difference between the master TDM clock and the common reference.

In a variation of this scenario both TDM end systems may have accurate, but independent, source clocks and both IWFs may independently derive their clocks based on the encoded relationship received.

10.1.4 Adaptive Clock Recovery

Figure 10.4 depicts the scenario wherein one of the TDM end systems is required to slave its timing circuitry to that of the other, and no common reference clock is available. In this case an adaptive clock recovery function must be used at the egress IWF. The adaptive clock recovery function utilizes only observable characteristics of the packets arriving through the MPLS network, such as the precise time of arrival of the packet to the IWF and the fill-level of the jitter buffer as a function of time. Due to the packet delay variation in the MPLS network, filtering processes that combat the statistical nature of the observable characteristics must be employed. Frequency Locked Loops (FLL) and Phase Locked Loops (PLL) are well suited for this task.

In a variation of this scenario both TDM end systems may have accurate, but independent, source clocks and both IWFs may utilize adaptive clock recovery.

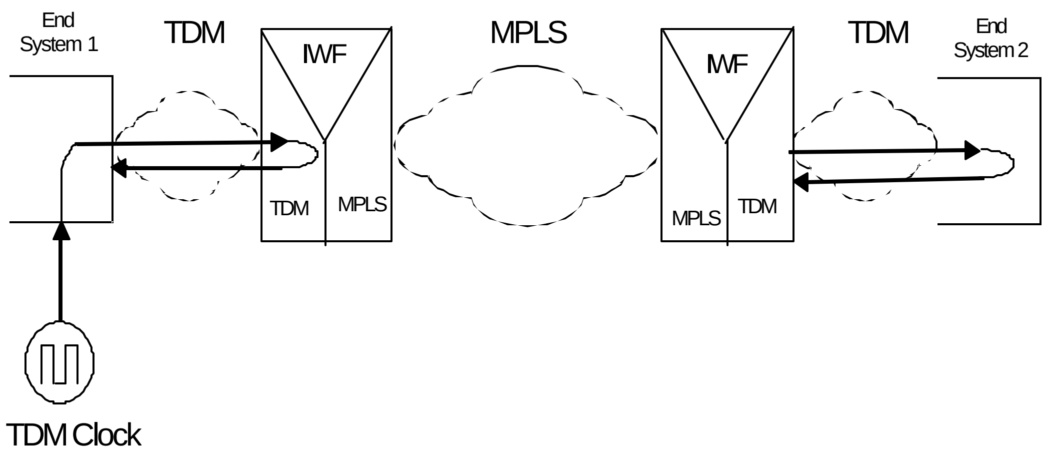


Figure 10.4: Adaptive clock recovery

11 Packet Loss Aspects

Some degree of packet loss can not be avoided in the MPLS network, hence some packet order integrity mechanism shall be provided. Malformed packets and out of order packets may also be considered as lost. Retransmission is not a viable option for TDM-MPLS interworking, and so appropriate action shall be taken to compensate for packet loss.

When loss of packets is detected the IWF shall insert the required amount of filler data towards the End System in order to retain TDM timing. When the CAS signalling is employed, care should be taken to maintain signalling state.

Structure-agnostic transport can not identify structure overhead, and so transports it transparently in the TDM segments. In the presence of packet loss it is possible to enhance FAS integrity by appropriately aligning the packet duration with the FAS period. However, the End System interface will still observe a corresponding amount of errored blocks [25].

For structure-aware transport structure overhead will be regenerated by the IWF As a consequence, presence of packet loss in the MPLS network will be completely hidden from the End System TDM interface.

For TDM carrying telephony channels the insertion of filler data may lead to reduced perceived audio quality.

12 Support of CAS and CCS Signalling

CAS or CCS telephony signalling may be employed over TDM networks, and these signals must be reliably transported over the MPLS network for the End Systems to properly function.

Handling of CAS and CCS signalling shall be transparent, i.e. the IWF should not need detailed understanding of the End System signalling protocols in order to properly transport this signalling.

12.1 Support of CAS Signalling

CAS is carried in the TDM frames as a sequence of bits that are uniquely associated with particular timeslots.

Structure-agnostic transport of sub-clause 9.1 can not identify CAS bits, and so transports them transparently in the TDM segments. Hence, in the presence of packet loss, it is not possible to ensure integrity of the CAS bits, and structure-agnostic transport relies on the End Systems to be able to withstand a certain interval of error condition.

The structure-locked method of sub-clause 9.2.1 ensures CAS integrity by appending to the packet an explicit CAS substructure, as depicted in Figure 9-3. The structure-indicated method of sub-clause 8.2.2 may also append such a CAS substructure, or may rely on multi-frame alignment to safeguard CAS bits.

12.2 Support of CCS Signalling

CCS may be carried in one or more timeslots of the TDM signal as an asynchronous message flow, frequently as HDLC frames.

Such channels may be idle for long periods. In such cases the HDLC mode defined in Appendix II may be employed.

13 Security considerations

There are no security related issues identified in this Recommendation.

Appendix I – Alternative methods for TDM-MPLS interworking

(This appendix does not form an integral part of this Recommendation)

In addition to the methods described in this Recommendation, other standardized protocols may be used for transport of TDM traffic across MPLS networks. The use of these protocols is described in this appendix.

I.1 Use of Recommendation Y.1411

Since TDM traffic can be carried over ATM circuit emulation services using AAL type 1, the protocols described in Y.1411 [7] may be used to indirectly transport TDM over interworking LSPs. In such a case the TDM is first converted into an AAL type 1 ATM flow according to [22] and [23], and thereafter this ATM flow is encapsulated as described in [7].

The N-to-one mode concatenates ATM cells including their cell headers, with the exception of the HEC. Hence, a valid and locally unique VPI/VCI must be allocated to the TDM bundle before this mode can be utilized.

While use of Y.1411 enables utilization of network devices designed for ATM-MPLS interworking and facilitates service interworking with existing ATM circuit emulation systems, it has higher overhead (an additional 4 bytes per 48 byte cell) and its use impedes exploitation of some features of the above intrinsic mode. For example, due to the separation of the TDM processing from the edge devices, access to timing related information may be lost, resulting in jitter and wander attenuation inferior to that obtainable via the adapted mode. Packet interpolation and TDM alarm handling may also suffer.

I.2 Use of AAL Type 2

For TDM carrying multiplexed telephony channels, transport of voice services by variable rate AAL2 ([25] and [26]) adaptation may be used to carry TDM over MPLS. In this case the TDM structure is decomposed into individual channels at ingress, and then regenerated at egress.

Using this method will be more bandwidth efficient when timeslots are dynamically allocated, or silence can be detected and suppressed, or speech compression and fax/modem-relay are employed. TDM error measures such as G.826 [27] are not meaningful in this case.

Appendix II – Optional processing of HDLC-based CCS signals

(This appendix does not form an integral part of this Recommendation)

The HDLC mode may be utilized in conjunction with structure-aware TDM transport to efficiently transport trunk associated HDLC-based CCS, such as SS7 [13] and ISDN PRI signalling [14]. This mechanism is not intended for general HDLC payloads, and only supports HDLC messages that are shorter than the maximum PDU size.

The HDLC mode should only be used when the majority of the bandwidth of the HDLC stream is occupied by idle flags. Otherwise the CCS channel should be treated as an ordinary timeslot.

The HDLC-MPLS interworking shall transparently pass all HDLC data and control messages over a separate interworking LSP.

At ingress the sender monitors flags until a frame is detected. The contents of the frame are collected and the FCS tested. If the FCS is incorrect the frame is discarded, otherwise the frame is sent after initial or final flags and FCS have been discarded and zero removal (per sub-clause 2.6 of [28]) has been performed. At egress, zero insertion is performed, the FCS is recalculated, and a valid HDLC frame reconstituted.

Appendix III – Examples of functional diagrams

(This appendix does not form an integral part of this Recommendation)

TDM-MPLS network interworking defines a G.805 client-server relationship between the TDM client and the MPLS server layer networks. The IWF is an adaptation function that accepts the client TDM characteristic information and processes it to enable its transfer over a trail in the server MPLS network. Hence the MPLS network forms a link connection supporting the TDM trail, providing a function that could also be filled by an SDH or ATM network.

Figure III.1 depicts the structure-agnostic transport of T1 (P11), E1 (P12), T3 (P32), or E3 (P31) signals over an MPLS network. The TDM originates at a G.703 [29] physical layer denoted Eq (q=11, 12, 31, 32), and is converted to a TDM bit stream by the Eq/Pqx adaptation function. After the TDM-MPLS IWF (adaptation function MPLS/Pqx) the MPLS packets enter the MPLS network at the MPLS trail termination function.

Figure III.2 depicts the structure-aware transport of T1 (P11) or E1 (P12) signals over an MPLS network. The TDM originates as before, but before the IWF it is converted to a P0N composite N*DS0 signal without level 1 overhead. At egress, the level 1 overhead must be regenerated.

Figure III.3 depicts an example of structure-agnostic transport of T1, E1, T3, or E3 signals that originated in VC11, VC12 or VC3 signals in an SDH network. Once converted to Pqx bit streams, the treatment is the same as in the first figure.

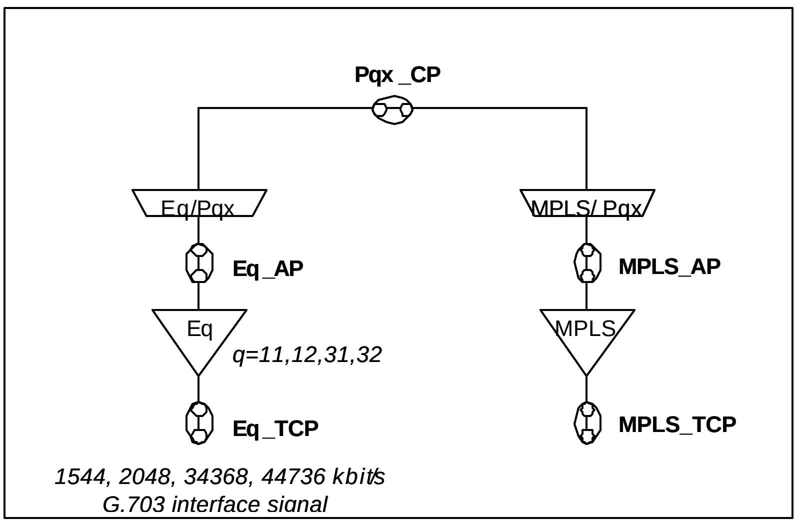
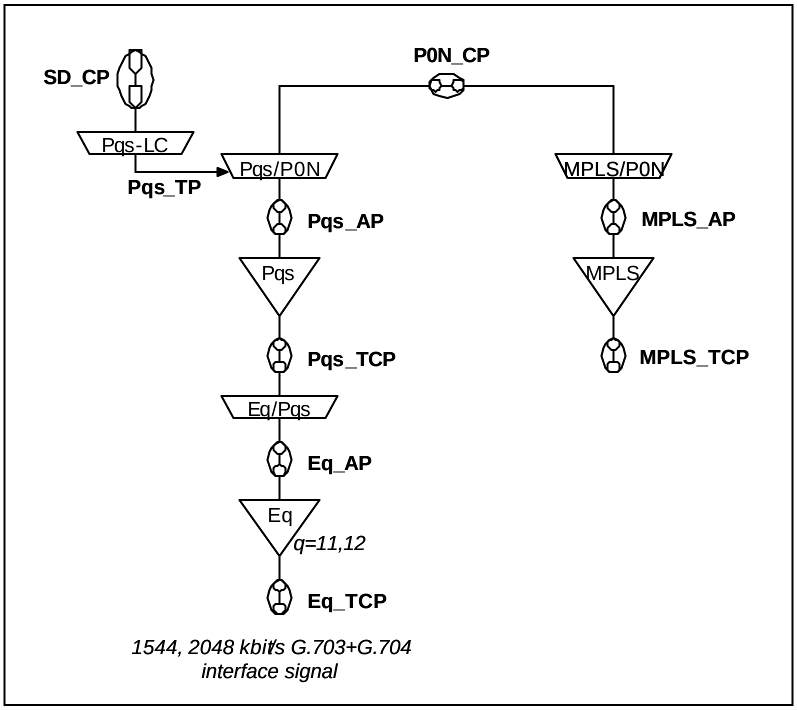


Figure III- 1 Structure-agnostic TDM over MPLS functional model

for 1544, 2048, 34368 and 44736 kbit/s PDH signals



NOTE: P0N represents the nx64 kbit/s signal (with or without CAS/CCS)

Figure III-2: Structure-aware TDM over MPLS functional model for 1544, 2048 kbit/s PDH signals

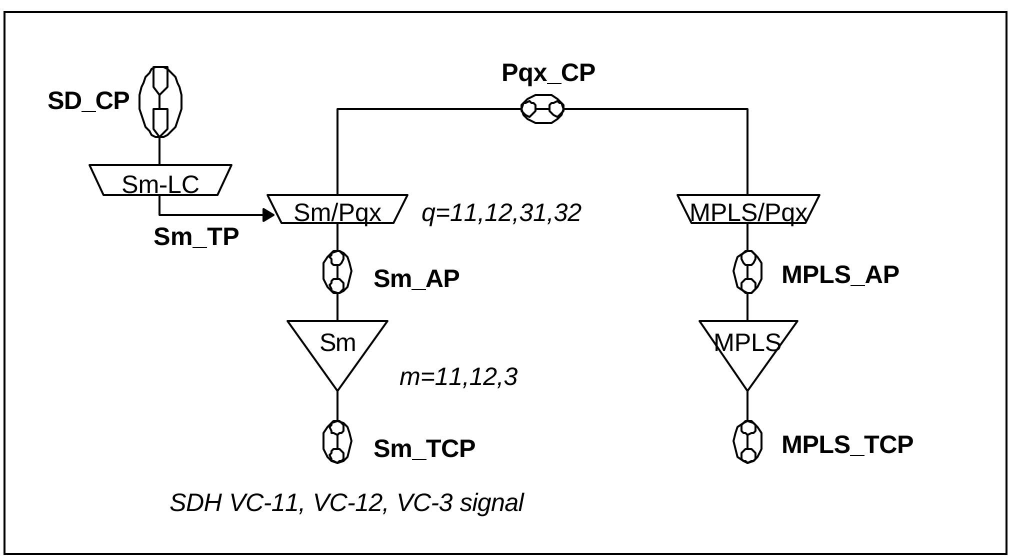


Figure III-3: Structure-agnostic TDM over MPLS functional model for 1544, 2048, 34368 and 44736 kbit/s PDH signals carried over SDH VC-m

Appendix IV – MPLS network performance metrics

(This appendix does not form an integral part of this Recommendation)

This appendix discusses impairments to the emulated TDM service caused by errors within the MPLS network. It primarily addresses the relationships between the performance parameters of the underlying MPLS network and service impairment metrics for TDM services, namely errored seconds and severely errored seconds as defined in G.826 [27], and availability ratio as defined in G.827 [30]. In addition, specific performance measures for voice channels are discussed.

IV.1 Errors in the MPLS Network that Impact TDM Service

There are three principal performance parameters for an MPLS network that impact the TDM service impairment metrics, namely Packet Loss Ratio, Packet Error Ratio and Packet Delay Variation. These metrics are defined in other ITU recommendations.

IV.1.1 Packet Loss ratio

Each lost packet will cause a burst of bit errors in the reconstructed TDM stream.

IV.1.2 Packet Delay Variation

Packet Delay Variation is best characterised using quintiles, so that a small number of packets will have delay variation outside the prescribed level. Since PDV is used to set the jitter buffer size, these packets may either arrive too late or too early to be accommodated. Such packets will be discarded and treated as lost, resulting once again in a burst of bit errors in the reconstructed TDM stream. In some implementations all misordered packets will be discarded and treated as lost.

IV.1.3 Packet Error Ratio

Bit errors induced in the MPLS network will normally be detected by a layer 2 error detection mechanism, causing the packet to be discarded. This leads to a burst of bit errors in the TDM stream. More rarely a packet containing bit errors may evade error detection and contribute directly to TDM bit errors.

IV.1.4 Overall Packet Loss

Each of the above errors (packet loss, packet error, and excess packet delay variation) may result in lost or discarded packets, causing a burst of bit errors in the TDM service. Recommendation G.1020 [31] defines a composite measure for these types of errors in an IP network, termed “overall packet loss”.

In order to maintain timing integrity, the egress IWF inserts the proper amount of filler data into the reconstructed TDM stream. The data to be inserted is implementation dependent.

IV.2 Relationship to TDM service impairment metrics

Recommendation G.826 [27] defines “errored seconds” and “severely errored seconds”, performance parameters related to the integrity of the data being transferred across the TDM circuit. The discussion below relates these TDM performance measures to the overall packet loss ratio in the MPLS network.

IV.2.1 Errored Seconds Ratio

An errored second is a one-second interval with one or more bit errors. G.826 specifies end-to-end objectives for the percentage of seconds that may be errored for each TDM type.

If the majority of MPLS packets lost or discarded are isolated events, then each individual packet lost or discarded may result in an errored second, and only an extremely small overall packet loss ratio is commensurate with G.826 constraints. If, on the other hand, most of the packet loss occurs in bursts, many consecutive loss events contribute to the same errored second, and a much higher packet loss ratio is allowed. Quantitative modelling of such behaviour can be performed using network models, such as those described in Appendix 1 of [31].

IV.2.2 Severely Errored Seconds Requirement

A severely errored second is defined as a one second period where 30% or more of the blocks of TDM data received are errored. G.826 specifies end-to-end objectives for the percentage of seconds that may be severely errored.

If the majority of MPLS packets lost or discarded are in bursts, and these bursts are of sufficient duration, then severely errored seconds may result in the reconstructed TDM stream. On the other hand, isolated loss events lead to low severely errored second ratios. Once again network modelling may shed light on the numerical relationship between packet loss and G.826 conformance.

IV.3 Availability Requirements

The “unavailable state”, as defined in [30], is entered at the start of a period of 10 consecutive severely errored seconds. The “available state” is resumed at the start of a period of 10 consecutive seconds, none of which are severely errored.

The availability of an MPLS network may be defined in a similar manner, with the onset of unavailability beginning at the start of a period of ten consecutive seconds in each of which the packet loss rate exceeds 15%. It can be seen that the definitions of MPLS and TDM availability are well correlated.

IV.4 Voice Quality Requirements

We have seen that depending upon the packet loss rate of the underlying MPLS network, TDM carried over MPLS networks may not conform to the error objectives of G.826.

However, voice traffic carried in TDM streams may still be able to meet standard voice quality objectives. Of particular importance are the reduction in voice quality as specified in P.562 [32] and P.826 [33], and the delay requirements set forth in G.114 [34].

G.114 specifies that one-way transmission times of up to 150 milliseconds are universally acceptable, assuming adequate echo control is provided (higher delays are acceptable in some cases). Network planning and configuration of jitter buffers must take this constraint into account.

Packet loss in voice traffic can cause gaps or artefacts that result in choppy, garbled or even unintelligible speech. Subjective measures of voice quality are given in [32] and objective measures in [33]. TDM-MPLS interworking must ensure that perceptual voice quality be similar to that of the GSTN even in the presence of a reasonable overall packet loss ratio.

Appendix V - Suggested common clock frequencies for RTP

(This appendix does not form an integral part of this Recommendation)

There are four principal criteria for selecting the frequency of a common reference clock:

1) the reference frequency should be readily derivable
2) the reference frequency must be a multiple of 8 kHz
3) the reference frequency should be high, but not so high as to incur frequent timestamp roll-over
4) the frequency should not be too close to an integer multiple of the service clock frequency.

Based on these criteria, the following frequencies are suggested.

For systems with access to a common SONET/SDH network, a frequency of 19.44 MHz (2430 * 8 kHz).

For systems with access to a common ATM network, 9.72MHz (1215*8 kHz) or 19.44 MHz (2430*8 kHz).

For systems using GPS, 8.184MHz (1023*8 kHz).

For systems connected by a single hop of 100 Mbps Ethernet where it is possible to lock the physical layer clock, 25MHz (3125*8 kHz).

For systems connected by a single hop of Gigabit Ethernet where it is possible to lock the physical layer clock, 10 MHz (1250*8 kHz).

Appendix VI - Suggested payload sizes for structure-agnostic transport

(This appendix does not form an integral part of this Recommendation)

Structure-agnostic transport implementations should be capable of supporting the following payload sizes:

- synchronous serial data - 64 bytes
- E1 - 256 bytes
- T1 - 192 bytes
- E3 and T3 - 1024 bytes.
- Payload sizes that are multiples of 47 bytes may be used in conjunction with unstructured ATM-CES [22],[23].

Any payload size that does not lead to packet fragmentation may be used after having been agreed upon by ingress and egress IWFs.

By choosing sizes that are multiples or integer divisors of FAS periods, one may increase the tolerance to packet loss.

Appendix VII Suggested number of AAL1 SAR PDUs per packet

(This appendix does not form an integral part of this Recommendation)

The number of PDUs per MPLS packet is pre-configured and typically chosen taking into account latency and bandwidth constraints. Using a single PDU reduces latency to a minimum, but incurs the highest overhead.

Using eight or more PDUs per packet invalidates the use of the AAL1 sequence number mechanism, and hence complicates interworking with ATM based CES systems.

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