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SERIES Y: GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT-GENERATION NETWORKS

Internet protocol aspects – Interworking

TDM-IP interworking – User plane interworking

ITU-T Recommendation Y.1453

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ITU-T Recommendation Y.1453

TDM-IP interworking – User plane interworking

Summary

This Recommendation addresses required functions for network interworking between TDM networks up to DS3 or E3 rates and IP networks, in order to transport TDM traffic over IP networks. This Recommendation addresses user plane interworking mechanisms, connection multiplexing and procedures. These interworking mechanisms must ensure that TDM timing, signalling, voice quality, and alarm integrity be maintained. Details of the interworking model and required interworking functions are described. This Recommendation may not be suitable for use by Recognized Operating Agencies due to possible degradation of network synchronization performance as compared with native TDM transport.

Source

ITU-T Recommendation Y.1453 was approved on 29 March 2006 by ITU-T Study Group 13 (2005-2008) under the ITU-T Recommendation A.8 procedure.

Keywords

Interworking, IP, network interworking, TDM, UDP, user plane.

FOREWORD

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CONTENTS

Introduction

There is a need to define network interworking wherein traffic from conventional synchronous or plesiochronous networks (hereafter denoted TDM networks) is transported over IP networks. Such interworking must ensure that TDM timing, signalling, voice quality, and alarm integrity be maintained.

ITU-T Recommendation Y.1453

TDM-IP interworking – User plane interworking

1 Scope

This Recommendation addresses required functions for network interworking between TDM networks up to and including DS3/E3 rates and IP networks, in order to transport TDM traffic over IP networks. Transport of higher rate TDM services (such as SDH) over IP networks is beyond the scope of this Recommendation. This Recommendation addresses user plane interworking mechanisms, connection multiplexing and related procedures. These interworking mechanisms must ensure that TDM timing, signalling, telephony voice quality, and alarm integrity be maintained. Details of the interworking model and required interworking functions are described. This Recommendation may not be suitable for use by recognized operating agencies (ROAs) [1] due to possible degradation of network synchronization performance as compared with native TDM transport.

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; 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. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation F.110 (1996), *Operational provisions for the maritime mobile service.*

[2] ITU-T Recommendation Y.1411 (2003), *ATM-MPLS network interworking – Cell mode user plane interworking.*

[3] ITU-T Recommendation Y.1413 (2004), *TDM-MPLS network interworking – User plane interworking*.

[4] ITU-T Recommendation G.809 (2003), *Functional architecture of connectionless layer networks.*

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

[6] ITU-T Recommendation G.705 (2000), *Characteristics of plesiochronous digital hierarchy (PDH) equipment functional blocks.*

[7] ITU-T Recommendation G.114 (2003), *One-way transmission time.*

[8] ITU-T Recommendation G.826 (2002), *End-to-end error performance parameters and objectives for international, constant bit-rate digital paths and connections.*

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

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

[11] IETF RFC 791 (1981), *Internet Protocol – DARPA Internet Program – Protocol Specification.*

[12] IETF RFC 2460 (1998), *Internet Protocol, Version 6 (IPv6) Specification.*

[13] IETF RFC 768 (1980), *User Datagram Protocol.*

[14] ITU-T Recommendation G.703 (2001), *Physical/electrical characteristics of hierarchical digital interfaces.*

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

[16] 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.*

[17] ITU-T Recommendation I.231.1 (1988), *Circuit-mode bearer service categories – Circuit-mode 64 kbit/s unrestricted, 8 kHz structured bearer service.*

[18] ANSI T1.107 (2002), *Digital Hierarchy – Formats Specifications.*

[19] 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.*

[20] ITU-T Recommendation G.704 (1998), *Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44 736 kbit/s hierarchical levels.*

[21] ITU-T Recommendation Q.700 (1993), *Introduction to CCITT Signalling System No. 7.*

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

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

[24] IETF RFC 3550 (2003), *RTP: A Transport Protocol for Real-Time Applications.*

[25] ITU-T Recommendation Y.1540 (2002), *Internet protocol data communication service – IP packet transfer and availability performance parameters.*

[26] IETF RFC 2474 (1998), *Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers.*

[27] IETF RFC 3246 (2002), *An Expedited Forwarding PHB (Per-Hop Behavior).*

[28] IETF RFC 2210 (1997), *The Use of RSVP with IETF Integrated Services.*

[29] IETF RFC 2212 (1997), *Specification of Guaranteed Quality of Service.*

[30] ATM Forum af-vtoa-0078.000 (1997), *Circuit Emulation Service 2.0.*

[31] ITU-T Recommendation G.802 (1988), *Interworking between networks based on different digital hierarchies and speech encoding laws.*

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

[33] ITU-T Recommendation G.827 (2003), *Availability performance parameters and objectives for end-to-end international constant bit-rate digital paths.*

[34] ITU-T Recommendation G.1020 (2003), *Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks.*

[35] ITU-T Recommendation P.800 (1996), *Methods for subjective determination of transmission quality*.

[36] ITU-T Recommendation P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.

[37] ITU-T Recommendation I.231.5 (1988), *Circuit-mode bearer service categories: Circuit-mode 2 × 64 kbit/s unrestricted, 8 kHz structured bearer service*.

[38] ITU-T Recommendation I.231.10 (1992), *Circuit-mode bearer service categories: Circuit-mode multiple-rate unrestricted 8 kHz structured bearer service*.

[39] ITU-T Recommendation I.231.6 (1996), *Circuit-mode bearer service categories: Circuit-mode 384 kbit/s unrestricted, 8 kHz structured bearer service*.

3 Definitions

This Recommendation uses or defines the following terms:

3.1 interworking: See ITU-T Rec. Y.1411 [2].

3.2 interworking flow: A pair of G.809 flows [4] capable of simultaneously transferring information in opposite directions over an IP network for the purpose of transporting TDM traffic.

3.3 interworking function (IWF): See ITU-T Rec. Y.1411.

3.4 ingress IWF: The point where the continuous TDM stream is segmented and encapsulated into IP packets (TDM-to-IP direction).

3.5 egress IWF: The point where TDM segments are de-encapsulated from IP packets and reassembled into a continuous TDM stream (IP-to-TDM direction).

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AAL	ATM Adaptation Layer
AIS	Alarm Indication Signal
AP	Access Point
ATM	Asynchronous Transfer Mode
CAS	Channel Associated Signalling
CCS	Common Channel Signalling
CES	Circuit Emulation Service
CP	Connection Point
CSI	Convergence Sublayer Indication
CSRC	Contributing Source
dAIS	AIS defect
Diffserv	Differentiated Services
dLOA	Loss of Alignment defect
dLOS	Loss of Signal defect
DSn	Digital Signal level n
EF PHB	Expedited Forwarding Per Hop Behaviour
En	Electrical Interface Signal, level n
FAS	Frame Alignment Signal
FCS	Frame Check Sequence

GS	Guaranteed Service
HDLC	High level Data Link Control
Intserv	Integrated Services
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IWF	InterWorking Function
LOF	Loss of Frame Synchronization
LOS	Loss of Signal
MPLS	Multi-Protocol Label Switch
MTU	Maximum Transport Unit
OAM	Operation, Administration and Maintenance
PDU	Protocol Data Unit
PDV	Packet Delay Variation
PLC	Packet Loss Concealment
PM	Performance Monitoring
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
PT	Payload Type
QoS	Quality of Service
ROA	Recognized Operating Agency
RDI	Remote Defect Indication
RFC	Request for Comments
RTP	Real Time Protocol
SAR	Segmentation And Reassembly
SRTS	Synchronous Residual Time Stamp
SSRC	Synchronization Source
TDM	Time Division Multiplex
TFP	Termination Flow Point
UDP	User Datagram Protocol

5 Conventions

This Recommendation uses traditional terminology for digital signals at the various levels of the G.702 [5] rate hierarchy. In particular, the first level digital signal of rate 2048 kbit/s (P12 in G.705 [6] terminology) is designated E1, and the third level signal of rate 34 368 kbit/s derived from it (P31), E3. Similarly, the first level signal of rate 1544 kbit/s (P11) is designated DS1, its second level derivative of rate 6312 kbit/s (P21), DS2, and its third level derivative at rate 44 736 kbit/s (P32), DS3. DS0 has the signal rate of 64 kbit/s.

6 TDM-IP interworking

This Recommendation defines interworking with TDM services up to and including DS3 or E3 rates. Transport of higher rate TDM services (such as SDH) over IP networks is beyond the scope of this Recommendation.

TDM services are conventionally transported over networks operating in circuit-switched mode.

A TDM client requires that its server layer constrains accuracy, ordering, and temporal impairments within defined bounds. For a connectionless server layer, the severity of these impairments may increase non-linearly with server layer loading.

As the server layer loading may not be known in advance, and may vary over time, the layering of a TDM client over an IP server presents a significant challenge for equipment manufacturers and service providers to conform to ITU-T Recommendations relating to TDM performance. In particular, due to packet loss, delay and delay variation, the end-to-end delay [7], error [8], timing [9] and [10] performance will in general be degraded as compared with those experienced over a native TDM infrastructure. IP network performance is discussed in Appendix II.

Therefore, users of an implementation of this Recommendation should be aware that it may not be possible to predict or guarantee performance.

Figure 6-1 provides a general network architecture for TDM-IP network interworking where TDM networks are interconnected through an IP network [11] and [12]. Note that the path through the IP network will change over time as a result of IP routing protocols.

For the TDM-to-IP direction, the continuous TDM stream is segmented and encapsulated into UDP/IP [13] packets by the interworking function (IWF). For the IP-to-TDM direction, the TDM segments are extracted from the UDP/IP packets and the continuous TDM stream is reassembled.

Figure 6-2 depicts the network functional architecture of TDM-IP interworking using the diagrammatic techniques of ITU-T Rec. G.809 [4]. Examples for specific scenarios are given in Appendix III/Y.1413 [3].

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

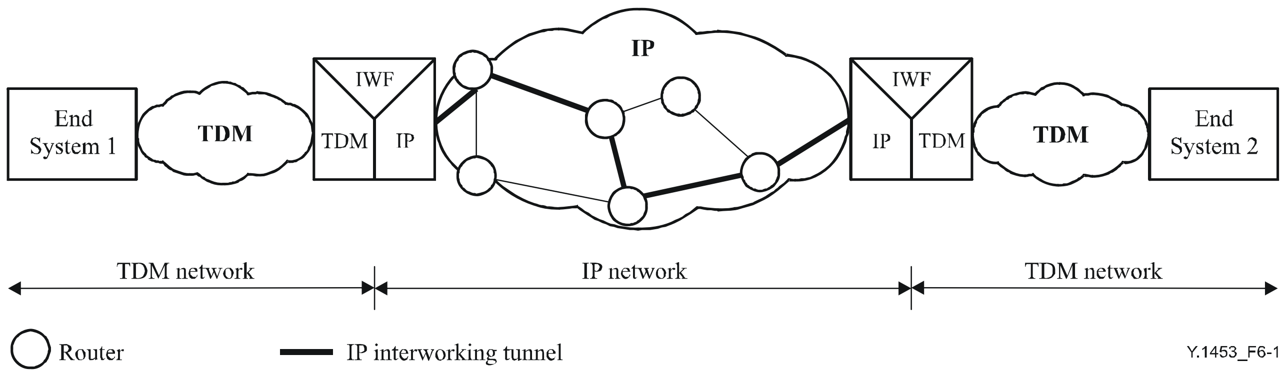

Figure 6-1/Y.1453 – Reference architecture for TDM-IP network interworking

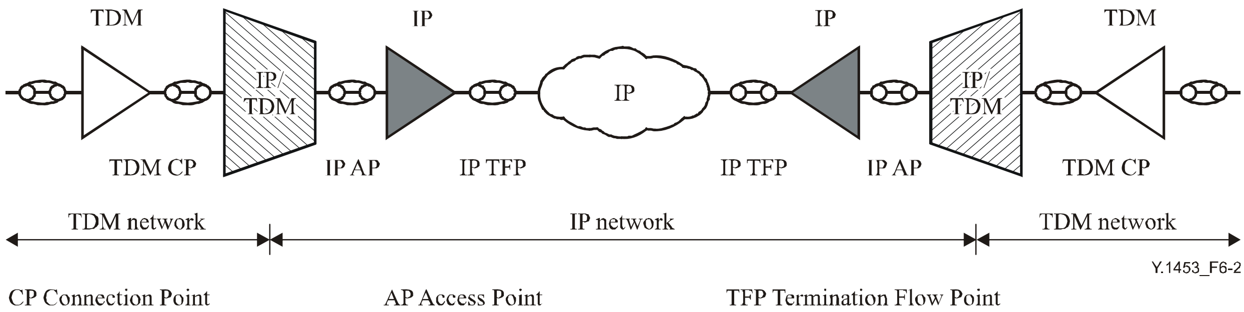

Figure 6-2/Y.1453 – Functional architecture of TDM-IP network interworking depicted according to the diagrammatic conventions of G.809

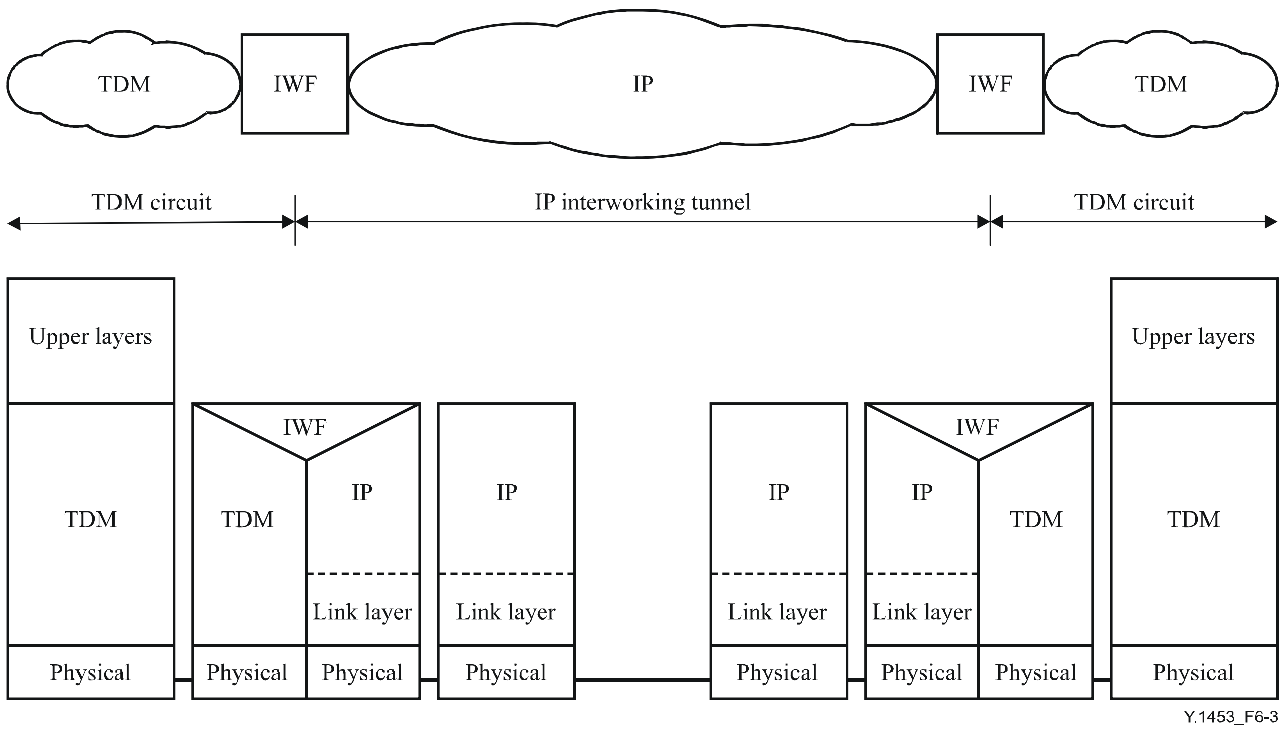

Figure 6-3/Y.1453 – Network reference model and protocol layers for TDM-IP user plane interworking

7 General requirements

7.1 User plane requirements

For transfer of TDM streams over IP networks, the following capabilities are required:

a) The ability to transport multiple TDM streams between two IWFs.

b) Support for bidirectional flows with symmetric bandwidth and binding to the duplex TDM.

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

1) DS1 at 1544 kbit/s as defined in ITU-T Rec. G.703 [14];
2) E1 at 2048 kbit/s as defined in ITU-T Rec. G.703;
3) DS2 at rate 6312 kbit/s as defined in ITU-T Rec. G.703;
4) Synchronous serial data as defined in ITU-T Recs V.36 [15] and V.37 [16];

5) N × 64 k (i.e., for N = 1 as defined in ITU-T Rec. I.231.1 [17], for N = 2 as defined in ITU-T Rec. I.231.5 [37], for N = 3 as defined in ITU-T Rec. I.231.10 [38] and for N = 6 as defined in ITU-T Rec. I.231.6 [39]);
6) DS3 at 44 736 kbit/s as defined in ANSI T1.107 [18];
7) E3 at 34 368 kbit/s as defined in ITU-T Rec. G.751 [19].

d) The ability to transport the following structured TDM types:
1) DS1 as defined in ITU-T Rec. G.704 [20];
2) Fractional DS1 carrying N timeslots with N from 1 to 23 as defined in ANSI T1.107;
3) E1 as defined in ITU-T Rec. G.704;
4) Fractional E1 carrying N timeslots with N from 1 to 30 as defined in ITU-T Rec. G.704;
5) Multiple synchronous DS0s;
6) DS2 defined in ITU-T Rec. G.704.

e) The ability to transport the structured TDM types of items d 1, 2, 3, 4, 6 with Channel Associated Signalling (CAS), as defined in ANSI T1.107 and ITU-T Rec. G.704.

f) The ability to transport trunk- or facility-associated Common Channel Signalling (CCS), e.g., as defined in ITU-T Recs Q.700 [21] and Q.931 [22].

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 a [9] or [10] traffic interface.

i) The ability to interwork with existing Circuit Emulation Services (CES) carried in MPLS [3] or ATM networks [23].

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

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

l) The ability to operate over arbitrary IP networks, but to exploit QoS features of IP networks if they are present.

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 maximum transport unit (MTU).

7.2 Management plane aspects

For transfer of TDM services over IP networks, the following capabilities shall be provisioned:

a) UDP source and destination port values for both directions.

b) The type of TDM traffic as per 7.1 c) and d).

c) For serial data (7.1 c) 4): the bit rate.

d) For N × 64 k (7.1 c) 4): the value of N.

e) For fractional E1 or DS1 (7.1 d) 2 or 4): the value of N.

f) The payload format (see clause 9).

g) For structure-agnostic transport: the number of payload octets per IP packet.

h) For unstructured DS1: if DS1-octet aligned mode is used.

i) For structure-locked encapsulation: the number of frames per IP packet.

j) For structure-indicated encapsulation:
   1) the number of 48-octet PDUs per packet;
   2) AAL 1 mode: unstructured, structured or structured with CAS.

k) Indication of whether RTP [24] is used.

l) If RTP is used:
   1) whether the timestamp is determined from a common clock;
   2) the common clock frequency divided by 8 kHz;
   3) the payload type (PT);
   4) the SSRC value.

7.3 Fault management aspects

The interworking function shall support transfer of defect information between IP and TDM networks, as depicted in Figure 7-1. In particular, local TDM defects [6], such as loss of signal or loss of synchronization, shall be signalled from ingress to egress IWF; and IP anomalies [25], such as packet misordering and loss, shall be detected by the egress IWF.

The interworking function shall transfer TDM defect indications through the IP network by setting appropriate flags in the common interworking indicators. Definition of TDM defect states and criteria for entering and exiting those states are defined in ITU-T Rec. G.705 [6]. 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., dLOS, dLOA or dAIS). In addition, if applicable, an appropriate alarm shall be sent to the management layer.

The egress IWF detects IP anomalies by monitoring the timely arrival of packets and by the sequence number in the common interworking indicators. Regardless of anomalies, the egress IWF shall ensure synchronization integrity of its local TDM interface. The egress IWF shall maintain statistical record of anomalies, and when the density of anomalies is such as to comprise a defect, it shall inform the ingress IWF of the defect and shall send the appropriate alarm to the management layer.

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

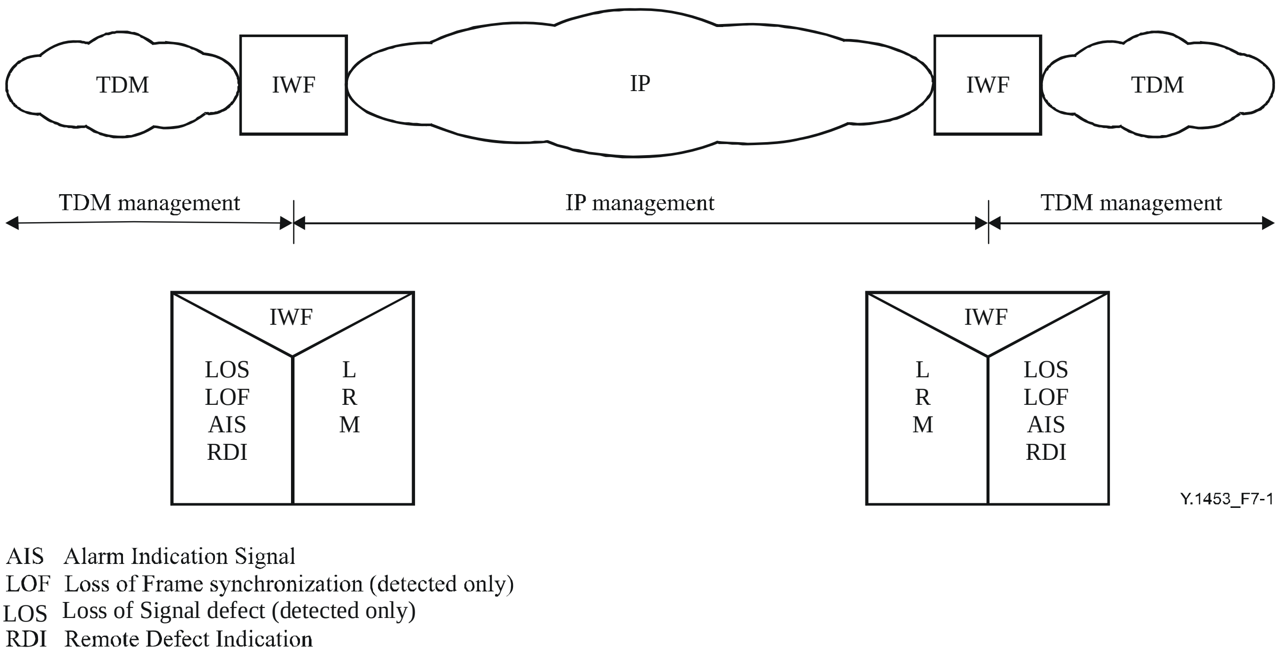

Figure 7-1/Y.1453 – Functional representation of TDM-IP fault management

7.4 Traffic management aspects

The IP flow shall be capable of providing the required QoS for all TDM connections, and must be capable of meeting the aggregate bandwidth requirements of all TDM connections transported.

If the IP network is Diffserv enabled according to RFC 2474 [26], then expedited forwarding per hop behaviour (EF PHB) per RFC 3246 [27] with appropriate traffic conditioning shall be used in order to provide a low latency and minimal jitter service. It is suggested that the IP network be somewhat over provisioned.

If the IP network is Intserv enabled according to RFC 2210 [28], then guaranteed service (GS) per RFC 2212 [29] with a bandwidth reservation greater than that of the aggregate TDM traffic shall be used in order to guarantee sufficient bandwidth and bounded delay.

The expected delay introduced by the network should be measured prior to traffic flow, to estimate latency. This measurement may only be meaningful when the service provider manages the IP network load.

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 IP network, the bandwidth presently being consumed by interworking flows and other clients of the IP network, and the bandwidth requested. When sufficient bandwidth is available, the request may be granted. When bandwidth is insufficient, the TDM connection request must be denied.

8 Functional group considerations for TDM-IP network interworking

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

IP
UDP
Optional timing information
Common interworking indicators
TDM Payload

NOTE – Bit 8 is the most significant bit.

Figure 8-1/Y.1453 – TDM-IP interworking functional groups

8.1 IP

This field is the standard IPv4 [11] or IPv6 [12] header.

8.2 UDP

Since it may be required to transport multiple emulated TDM streams between two IP addresses, a method of labelling TDM-IP flows is required. Only manual provisioning of this label is considered in this Recommendation. The label may be placed in the UDP source port field, or the UDP destination port field per RFC 768 [13]. When the source port field is used, the destination port field may contain an identifier indicating that the packet contains TDM data.

8.3 Common interworking indicators

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

<table>
<tr><th colspan="8">Bit</th></tr>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td colspan="8">Control</td></tr>
<tr><td colspan="2">FRAG</td><td colspan="6">Length</td></tr>
<tr><td colspan="8">Sequence number (2 octets)</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 8-2/Y.1453 – Common interworking indicators

8.3.1 Control field

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

<table>
<tr><th colspan="8">Bit</th></tr>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td colspan="4">Reserved</td><td>L</td><td>R</td><td colspan="2">M</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 8-3/Y.1453 – Control field

The reserved field shall be set to zero.

The L, R and M fields provide a means of transfer of TDM defect indications between IWFs. Their use should be in accordance with principles of the appropriate G series of ITU-T Recommendations with regard to operation, administration and maintenance (OAM).

L Local TDM failure: The L bit being set (i.e., L = 1) 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 then the L bit shall be cleared.

R Remote Receive failure: The R bit being set (i.e., R = 1) indicates that the source of the packet is not receiving packets from the IP network. Thus the setting of the R bit indicates failure of the opposite direction. This indication can be used to signal IP 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 9.2.1. The field is used as follows:

FRAG

0 0 Indicates that the entire (unfragmented) 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 an IP packet is transported over Ethernet, 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 indicates the size of the IP packet payload in octets, and its value is the sum of:

a) size of the Common interworking indicators (4 octets);

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 misordering.

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-IP direction):

- The sequence number should be set to a random value for the first IP packet transmitted over the interworking flow.
- For each subsequent IP 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 reordered, if possible. The mechanism for detecting packet loss is implementation specific.

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

- The egress IWF maintains an expected sequence number.
- The first packet received from the IP 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 RFC 3550 [24].

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

8.5 TDM payload

The format of the TDM payload is detailed in clause 9.

8.6 Summary of encapsulation format

This clause provides two encapsulation formats, one when an RTP header is absent (see Figure 8-4) and the other when an RTP header is present (see Figure 8-5).

Bit 8	7	6	5	4	3	2	1	Octets
IP version				IHL				1
IP TOS								2
Total length								3-4
Identification								5-6
Flags			Fragment offset					7
								8
Time to Live (TTL)								9
Protocol								10
IP header checksum								11-12
Source IP address								13-16
Destination IP address								17-20
Source UDP port number								21-22
Destination UDP port number								23-24
UDP Length								25-26
UDP checksum								27-28
Reserved				L	R	M		29
FRAG		Length						30
Sequence number								31-32
Adapted payload								33-n

NOTE – Bit 8 is the most significant bit.

Figure 8-4/Y.1453 – Encapsulation format without the use of RTP

The first twenty octets are the IP header; octets 21 through 28 are the UDP header. Octets 29 through 32 are the common interworking indicators.

Descriptions of the fields are as follows:

IP version, octet 1, bits 8 through 5

Indicates IP version number, e.g., for IPv4, IP Version = 4.

IHL, octet 1, bits 4 through 1

Indicates the length (in 32-bit words) of the IP header, e.g., IHL = 5.

IP TOS, octet 2

Indicates the IP type of service.

Total length, octets 3 and 4

Indicates the length (in octets) of header and IP payload.

Identification, octets 5 and 6

IP fragmentation identification field [11].

Flags, octet 7, bits 8 through 6

Indicates IP control flags and shall be set to Flags = 010 to avoid fragmentation.

Fragment Offset, octet 7, bits 5 through 1 and octet 8

Indicates where in the datagram the fragment belongs and is not used.

Time to Live, octet 9

Indicate the IP TTL field. Datagrams with zero in this field are to be discarded.

Protocol, octet 10

Indicates protocol type and shall be set to 0x11 (i.e., hexadecimal 11) to signify UDP.

IP Header Checksum, octets 11 and 12

Indicates the checksum for the IP header.

Source IP Address, octets 13 through 16

Indicates the source IP address.

Destination IP Address, octets 17 through 20

Indicates the destination IP address.

Source Port Number, octets 21 and 22, and
Destination Port Number, octets 23 and 24

Either of these fields may be used to uniquely identify the specific TDM stream being transported. The UDP flow shall be manually configured.

When the source port is used to identify the TDM stream, the destination port number may be used to identify the UDP packet as conforming to this Recommendation.

When used as a TDM stream identifier, the UDP port number shall be chosen from the range of dynamically allocated UDP ports numbers (49 152 through 65 535).

The choice of whether the source port field or destination port field is used as TDM stream identifier is implementation dependent, but the choice shall be agreed upon by ingress and egress IWFs.

UDP Length, octets 25 and 26

Indicates the length in octets of UDP header and UDP payload.

UDP Checksum, octets 27 and 28

Indicates the checksum of UDP/IP header and payload. If not computed it must be set to zero.

Reserved, octet 29, bits 8 to 5

Indicates a reserved field which shall be set to zero.

L, R, and M octet 29, bits 4 through 1

See 8.3.1.

FRAG, octet 30, bits 8 and 7

See 8.3.2.

Length, octet 30, bits 6 through 1

See 8.3.3.

Sequence number, octets 31 and 32

See 8.3.4.

When the RTP header appears, the packet format is as depicted in Figure 8-5.

Bit

8	7	6	5	4	3	2	1	Octets
IP version				IHL				1
IP TOS								2
Total length								3-4
Identification								5-6
Flags			Fragment offset					7
								8
Time to Live (TTL)								9
Protocol								10
IP header checksum								11-12
Source IP address								13-16
Destination IP address								17-20
Source UDP port number								21-22
Destination UDP port number								23-24
UDP Length								25-26
UDP checksum								27-28
RTV		P	X	CC				29
Mark	PT							30
RTP sequence number								31-32
RTP timestamp								33-36
SSRC identifier								37-40
Reserved				R	L	M		41
FRAG		Length						42
Sequence number								43-44
Adapted payload								45-n

NOTE – Bit 8 is the most significant bit.

Figure 8-5/Y.1453 – Encapsulation format with the use of RTP

Descriptions of fields not described above are as follows:

The fields of RTP header shall be used as follows:

- **RTV** (version) is always set to 2.
- **P** (padding), X (header extension), CC (CSRC count) and Mark (marker) are always set to 0. Accordingly, RTP header extensions, padding and contributing synchronization sources are never used.
- **PT** (payload type) is used as follows:
  a) A PT value shall be allocated from the range of dynamic values for each direction of the interworking flow.
  b) The ingress IWF shall set the PT field in the RTP header to the allocated value.
- The **RTP sequence number** shall be equal to the sequence number in the common interworking indicators.
- **RTP timestamps** are used for carrying timing information over the network:
  a) Their values are generated in accordance with the rules established in RFC 3550 [24].
  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/Y.1413 [3].

- The SSRC (synchronization source) identifier field in the RTP header may be used for detection of misconnections.

9 Payload formats

Clause 9.1 specifies the payload format for structure-agnostic transport, while clause 9.2 defines two payload formats for structure-aware transport. Subclause 9.2.1 specifies the structure-locked encapsulation, and subclause 9.2.2 specifies structure-indicated encapsulation based on AAL type 1, as defined in ITU-T Rec. I.363.1 [23] and ATM Forum CES 2.0 [30].

9.1 Structure-agnostic transport

Structure-agnostic transport completely disregards any TDM structure, in particular the structure imposed by standard TDM framing of ITU-T Rec. G.704 [20].

The payload format for structure-agnostic transport supports all the TDM services of 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 preconfigured;
- 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 III.

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

Whenever a packet is lost, or received too late for playout, or is received with the L bit set, then the egress IWF shall generate the appropriate amount of AIS towards its TDM interface.

NOTE – The use of AAL type 1 as described in 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 DS1 payload format

DS1 circuits may be delivered to the ingress IWF padded to an integer number of octets, as described in Annex B/G.802 [31]. In this format, the payload consists of an integer number of 25-octet 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

NOTE – Bit 8 is the most significant bit.

Figure 9-1/Y.1453 – Octet-aligned DS1 payload format

9.2 Structure-aware transport

Structure-aware transport maintains correct operation of the remote TDM interface by regenerating frame alignment signal (FAS) at egress, and preserves integrity of the TDM structure by structure-locking or structure-indication.

Whenever a packet is lost, or received too late for playout, or is received with the L bit set, then the egress IWF shall generate the appropriate amount of filler data in order to maintain TDM timing and FAS. While the insertion of arbitrary filler data may be sufficient to maintain the TDM timing, this may lead to reduced perceived quality of telephony voice channels contained in the TDM. Depending on the expected percentage of packet loss, packet loss concealment (PLC) mechanisms may need to be employed.

The payload formats for structure-aware transport support all the TDM services of 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. 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 [20] are sent to the TDM network.

For services listed in 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 microseconds (125 μs).

For services listed in 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 FRAG field in the Common interworking indicators, as described in 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, as identified in Table 1/G.704 [20], 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.

<table>
<tr><td>Frame</td><td colspan="8">Bit</td></tr>
<tr><td></td><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td></tr>
<tr><td rowspan="4">1</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
<tr><td rowspan="4">2</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
<tr><td>…</td><td colspan="8">…</td></tr>
<tr><td rowspan="4">M</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
</table>

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/Y.1453 – Payload format for structure-locked encapsulation without CAS (the IP packet does not carry a signalling substructure)

<table>
<tr><td>Frame</td><td colspan="8">Bit</td></tr>
<tr><td></td><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td></tr>
<tr><td rowspan="4">1</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
<tr><td rowspan="4">2</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
<tr><td>…</td><td colspan="8">…</td></tr>
<tr><td rowspan="4">M</td><td colspan="8">Bits belonging to timeslot 1</td></tr>
<tr><td colspan="8">Bits belonging to timeslot 2</td></tr>
<tr><td colspan="8">…</td></tr>
<tr><td colspan="8">Bits belonging to timeslot N</td></tr>
<tr><td rowspan="3">Signalling substructure</td><td colspan="4">Signalling bits for timeslot 1</td><td colspan="4">Signalling bits for timeslot 2</td></tr>
<tr><td colspan="4">Signalling bits for timeslot 3</td><td colspan="4">…</td></tr>
<tr><td colspan="4">Signalling bits for timeslot N</td><td colspan="4">Padding
(see Note 3)</td></tr>
</table>

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/Y.1453 – Payload formats for structure-locked encapsulation with CAS (the IP 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 ITU-T Rec. I.363.1 [23] and ATM Forum CES 2.0 [30], to form 48-octet AAL Type 1 SAR PDUs, as described in 2.4.2/I.363.1.

The packet payload consists of one or more PDUs, as depicted in Figures 9-4 and 9-5. The number of PDUs per packet:

- shall be preconfigured;
- 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 IV.

<table>
<tr><th colspan="8">Bit</th></tr>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td colspan="8">Common interworking indicators</td></tr>
<tr><td colspan="8">PDU payload (48 octets)</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 9-4/Y.1453 – Structure-indicated encapsulation with a single PDU per packet

<table>
<tr><th colspan="8">Bit</th></tr>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td colspan="8">Common interworking indicators</td></tr>
<tr><td colspan="8">PDU payload (48 octets)</td></tr>
<tr><td colspan="8">PDU payload (48 octets)</td></tr>
<tr><td colspan="8"></td></tr>
<tr><td colspan="8">PDU payload (48 octets)</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 9-5/Y.1453 – 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 this Recommendation.

For structure-agnostic transport, AAL type 1 provides no inherent advantage as compared to the method of 9.1; 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 AAL 1-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 convergence sublayer indication (CSI) [23] bit that signifies the appearance of a structure pointer for structured data transfer, and may be used for clock recovery (see clause 10).

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, ATM Forum CES 2.0 [30] defines two modes, structured and structured with CAS. Structured AAL type 1 carries octet-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 octet-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 9.2.1).

10 Timing aspects

TDM networks distribute timing information in order to maintain the required performance level. Since IP networks have no inherent timing distribution mechanism, other methods of timing distribution or recovery must be provided. Such methods are beyond the scope of this Recommendation.

11 Packet loss aspects

Some degree of packet loss cannot be avoided in an IP network, hence some packet 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-IP 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 AIS or filler data towards the End System in order to retain TDM timing. When CAS is employed, care should be taken by structure-aware mechanisms to maintain signalling state.

Structure-agnostic transport cannot identify structure overhead, and so transports it transparently in the TDM segments. Hence, filler data will in general introduce an incorrect FAS. It may be 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 [8].

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

For TDM carrying telephony channels, the insertion of filler data will lead to reduced perceived audio quality. Depending on the expected percentage of packet loss, packet loss concealment (PLC) mechanisms may need to be employed. PLC mechanisms are beyond the scope of this Recommendation.

12 Support of CAS and CCS

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

Handling of CAS and CCS 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

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

Structure-agnostic transport of 9.1 cannot 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 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 9.2.2 may append such a CAS substructure, or may rely on multi-frame alignment to safeguard CAS bits.

12.2 Support of CCS

CCS may be carried in one or more timeslots of the TDM signal as an asynchronous message flow, frequently as high level data link control (HLDC) frames.

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

13 Security considerations

Security aspects have not been addressed in this Recommendation.

Appendix I

Optional processing of HDLC-based CCS signals

The HDLC mode may be utilized in conjunction with Structure-Aware TDM transport to efficiently transport trunk associated HDLC-based CCS, such as SS7 [21] and ISDN PRI signalling [22]. 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-IP interworking shall transparently pass all HDLC data and control messages over a separate interworking flow.

At ingress the sender monitors flags until a frame is detected. The contents of the frame are collected and the frame check sequence (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 (as per 2.6/Q.921 [32]) has been performed. At egress, zero insertion is performed, the FCS is recalculated, and a valid HDLC frame reconstituted.

Appendix II

IP network performance metrics

This appendix discusses impairments to the emulated TDM service caused by errors within the IP network. It primarily addresses the relationships between the performance parameters of the underlying IP network and service impairment metrics for TDM services, namely errored seconds and severely errored seconds as defined in ITU-T Rec. G.826 [8], and availability ratio as defined in ITU-T Rec. G.827 [33]. In addition, specific performance measures for voice channels are discussed.

II.1 Errors in the IP network that impact TDM service

Whenever the IWF needs to generate AIS or filler data due to not having true TDM data available, TDM performance metrics will be impacted. This may occur due to three distinguishable IP network degradations, namely:

1) packet loss in the IP network;

2) packets being discarded due to detected errors; and

3) packets discarded due to jitter buffer overflow/underflow.

These three degradations are quantifiable using metrics defined for packet switched networks in other ITU-T Recommendations, such as ITU-T Rec. G.1020 [34].

ITU-T Rec. G.1020 defines packet network and terminal performance parameters that reflect the perceived quality of the speech and other voiceband applications. It is largely focused on quality impairments resulting from delay variation and packet loss which are peculiar to IP and other packet-based technologies, and that do not appear in traditional TDM networks. Although TDM services are not directly addressed by ITU-T Rec. G.1020, some of the metrics defined therein are applicable to voice channels carried by TDM.

II.1.1 Packet loss ratio

IP packet loss ratio is defined in ITU-T Rec. Y.1540 [25]. Each lost packet leads to a burst of bit errors in the reconstructed TDM stream.

II.1.2 Packet delay variation

IP packet delay variation (PDV) is defined in ITU-T Rec. Y.1540. 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.

II.1.3 Packet error ratio

IP packet error ratio is defined in ITU-T Rec. Y.1540. Bit errors induced in the IP 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.

II.1.4 Overall packet loss

Overall packet loss ratio is defined in ITU-T Rec. G.1020 [34]. 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. ITU-T Rec. G.1020 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.

II.2 Relationship to TDM service impairment metrics

ITU-T Rec. G.826 [8] 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 IP network.

II.2.1 Errored seconds ratio

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

If the majority of IP 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 I/G.1020.

II.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. ITU-T Rec. G.826 specifies end-to-end objectives for the percentage of seconds that may be severely errored.

If the majority of IP 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.

II.3 Availability requirements

The "unavailable state", as defined in ITU-T Rec. G.826, 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 IP network is defined in ITU-T Rec. Y.1540 [25] and can be correlated with the definition of TDM availability.

II.4 Voice quality requirements

Depending upon the packet loss rate of the underlying IP network, TDM carried over IP networks may not conform to the error objectives of ITU-T Rec. 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 ITU-T Recs P.800 [35] and P.862 [36], and the delay requirements set forth in ITU-T Rec. G.114 [7].

It is generally recognized that most applications will have acceptable performance when the one-way delay is lower than 150 ms, assuming adequate echo control is provided (higher delays may be 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 ITU-T Rec. P.800 [35] and objective measures in ITU-T Rec. P.862 [36]. TDM-IP interworking must ensure that perceptual voice quality be similar to that of the PSTN even in the presence of a reasonable overall packet loss ratio.

Appendix III

Suggested payload sizes for structure-agnostic transport

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

- synchronous serial data – 64 bytes.
- E1 – 256 bytes.
- DS1 – 192 bytes.
- E3 and DS3 – 1024 bytes.
- Payload sizes that are multiples of 47 bytes may be used in conjunction with unstructured ATM-CES [30].

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 IV

Suggested number of AAL 1 SAR PDUs per packet

The number of PDUs per IP packet is preconfigured and typically chosen taking into account latency and bandwidth constraints. Using a single PDU reduces latency to a minimum, but incurs the highest overhead. Suggested values are between 1 and 8 PDUs per packet for E1 and DS1 circuits, and between 5 and 15 PDUs per packet for E3 and DS3 circuits.

Using eight or more PDUs per packet invalidates the use of the AAL 1 sequence number mechanism, and hence complicates interworking with ATM-based CES systems.

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