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

Internet protocol aspects – Interworking

Voice services – MPLS network interworking

ITU-T Recommendation Y.1414

ITU-T Y-SERIES RECOMMENDATIONS

GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT GENERATION NETWORKS

GLOBAL INFORMATION INFRASTRUCTURE	
General	Y.100–Y.199
Services, applications and middleware	Y.200–Y.299
Network aspects	Y.300–Y.399
Interfaces and protocols	Y.400–Y.499
Numbering, addressing and naming	Y.500–Y.599
Operation, administration and maintenance	Y.600–Y.699
Security	Y.700–Y.799
Performances	Y.800–Y.899
INTERNET PROTOCOL ASPECTS	
General	Y.1000–Y.1099
Services and applications	Y.1100–Y.1199
Architecture, access, network capabilities and resource management	Y.1200–Y.1299
Transport	Y.1300–Y.1399
Interworking	**Y.1400–Y.1499**
Quality of service and network performance	Y.1500–Y.1599
Signalling	Y.1600–Y.1699
Operation, administration and maintenance	Y.1700–Y.1799
Charging	Y.1800–Y.1899
NEXT GENERATION NETWORKS	
Frameworks and functional architecture models	Y.2000–Y.2099
Quality of Service and performance	Y.2100–Y.2199
Service aspects: Service capabilities and service architecture	Y.2200–Y.2249
Service aspects: Interoperability of services and networks in NGN	Y.2250–Y.2299
Numbering, naming and addressing	Y.2300–Y.2399
Network management	Y.2400–Y.2499
Network control architectures and protocols	Y.2500–Y.2599
Security	Y.2700–Y.2799
Generalized mobility	Y.2800–Y.2899

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation Y.1414

Voice services – MPLS network interworking

Summary

This Recommendation focuses on the required functions and procedures necessary for support of narrow-band voice services by MPLS networks. Details of the encapsulation of encoded audio streams in MPLS packets are specified.

Source

ITU-T Recommendation Y.1414 was approved on 29 July 2004 by ITU-T Study Group 13 (2001-2004) under the ITU-T Recommendation A.8 procedure.

Keywords

AAL type 2, Interworking, MPLS, Network, VoIP, Voice services, User Plane.

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CONTENTS

Introduction

This Recommendation focuses on the required functions and procedures necessary for support of narrow-band voice services by MPLS networks. Details of the encapsulation of encoded audio streams in MPLS packets are specified.

ITU-T Recommendation Y.1414

Voice services – MPLS network interworking

1 Scope

This Recommendation focuses on the required functions and procedures necessary for support of narrow-band voice services by MPLS networks. ITU-T Rec. Y.1261 [1] describes the service requirements and architecture for transport of voice services over MPLS networks.

Narrow-band voice services include encoded audio streams, telephony call progress tones, fax, and optionally circuit mode data.

This Recommendation specifies encapsulation of encoded audio in MPLS packets. Algorithms for encoding audio streams are beyond the scope of this Recommendation.

This Recommendation includes three modes:

- voice over IP over MPLS;
- voice over MPLS using AAL type 2 SSCS [2] and [3] for narrow-band services; and
- voice over MPLS using MPLS Forum IA 1.0 [4].

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 Y.1261 (2002), *Service requirements and architecture for voice services over Multi-Protocol Label Switching*.

[2] ITU-T Recommendation I.366.2 (2000), *AAL type 2 service specific convergence sublayer for narrow-band services*, plus Corrigendum 1 (2002).

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

[4] MPLS Forum Implementation Agreement MPLS Forum 1.0 (2001), *Voice Over MPLS – Bearer Transport*.

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

[6] ITU-T Recommendation G.805 (2000), *Generic functional architecture of transport networks*.

[7] ITU-T Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[8] ITU-T Recommendation G.723.1 (1996), *Speech coders: Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

[9] ITU-T Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.

[10] ITU-T Recommendation G.727 (1990), *5-, 4-, 3- and 2-bit/sample embedded adaptive differential pulse code modulation (ADPCM)*.

[11] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)*.

[12] ETSI EN 301 703 V7.0.2 (1999), *Digital Cellular Telecommunications System (Phase 2+) (GSM) Adaptive Multi-Rate (AMR); Speech processing functions; General description (GSM 06.71 version 7.0.2 Release 1998)*.

[13] ITU-T Recommendation G.722 (1988), *7 kHz audio-coding within 64 kbit/s*.

[14] ITU-T Recommendation G.722.1 (2000), *Coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss*.

[15] ITU-T Recommendation G.722.2 (2003), *Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)*.

[16] ITU-T Recommendation G.711 Appendix I (1999), *A high quality low-complexity algorithm for packet loss concealment with G.711*.

[17] ITU-T Recommendation Q.23 (1988), *Technical features of push-button telephone sets*.

[18] ITU-T Recommendation Q.24 (1988), *Multifrequency push-button signal reception*.

[19] ITU-T Recommendation E.180/Q.35 (1998), *Technical characteristics of tones for the telephone service*.

[20] ITU-T Recommendation I.251.3 (1992), *Number identification supplementary services: Calling Line Identification Presentation*.

[21] ITU-T Recommendation Q.310-Q.332 (1988), *Specifications of Signalling System R1*.

[22] ITU-T Recommendation Q.400-Q.490 (1988), *Specifications of Signalling System R2*.

[23] ITU-T Recommendation Q.724 (1988), *Telephone user part signalling procedures*, plus Amendment 1 (1993).

[24] ITU-T Recommendation T.4 (2003), *Standardization of Group 3 facsimile terminals for document transmission*.

[25] ITU-T Recommendation T.30 (2003), *Procedures for document facsimile transmission in the general switched telephone network*.

[26] ITU-T Recommendation V.17 (1991), *A 2-wire modem for facsimile applications with rates up to 14 400 bit/s*.

[27] ITU-T Recommendation V.29 (1988), *9600 bits per second modem standardized for use on point-to-point 4-wire leased telephone-type circuits*.

[28] ITU-T Recommendation V.18 (2000), *Operational and interworking requirements for DCEs operating in the text telephone mode*.

[29] IETF RFC 3032 (2001), *MPLS Label Stack Encoding*.

[30] IETF RFC 3031 (2001), *Multiprotocol Label Switching Architecture*.

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

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

[33] ATM Forum specification af-vmoa-0145.001 (2003), *Loop Emulation Service Using AAL2 Rev 1*.

[34] MPLS and Frame Relay Alliance 5.0.0 (2003), *I.366.2 Voice Trunking Format over MPLS, Implementation Agreement*.

3 Definitions

This Recommendation defines or uses the following terms:

3.1 interworking: See ITU-T Rec. Y.1411 [5].

3.2 interworking function (IWF): See ITU-T Rec. Y.1411 [5].

3.3 ingress IWF: The point where the voice services are encapsulated into an MPLS packet (Voice-to-MPLS direction).

3.4 egress IWF: The point where the voice services are de-encapsulated from an MPLS packet (MPLS-to-voice direction).

4 Abbreviations

This Recommendation uses the following abbreviations:

AAL2	ATM Adaptation Layer 2
CAS	Channel Associated Signalling
CCS	Common Channel Signalling
CID	Channel Identifier
CLI	Calling Line Identification
CPS	Common Part Sublayer
CPT	Call Progress Tone
CRC	Cyclic Redundancy Check
CU	Combined Use
DTMF	Dual Tone Multi Frequency
HEC	Header Error Control
IP	Internet Protocol
IWF	InterWorking Function
LSP	Label Switched Path
LSR	Label Switching Router
MPLS	Multi-Protocol Label Switching
MTU	Maximum Transport Unit
OAM	Operation And Maintenance
PDU	Protocol Data Unit
QoS	Quality of Service
RFC	Request for Comments
RTP	Real Time Protocol
SSCS	Service Specific Convergence Sublayer (of AAL)
STF	Start Field
TDM	Time Division Multiplexing
TTL	Time To Live
UDP	User Datagram Protocol

UUI	User-to-User Indication
VoIP	Voice over IP
VS	Voice Services

5 Conventions

In this Recommendation the term voice services is synonymous with narrow-band services, and includes 8 kHz digitized audio (carrying voice, telephony tones, fax and modem transmissions, etc). It optionally may include 16 kHz digitized audio ("wideband speech"), and 64 kbit/s data.

In this Recommendation voice services are considered without regard to the physical interface over which they are provided. In particular, while this physical interface may be a TDM link carrying multiple voice-grade channels, other cases are frequently encountered.

6 Voice services – MPLS network interworking

Figure 6-1 provides a general network architecture for support of narrow-band voice services by MPLS networks. For the voice services-to-MPLS direction, the continuous voice services data (stream or frame) is encapsulated into an MPLS packet by the interworking function (IWF). For the MPLS-to-voice services direction, the voice services data is extracted from the MPLS packets.

The MPLS network contains a number of Label Switching Routers (LSR), and Label Switched Paths (LSP). An example LSP is shown as a solid line in Figure 6-1.

Figure 6-2 depicts the network functional architecture of voice services-MPLS network interworking using the diagrammatic techniques of ITU-T Rec. G.805 [6].

Figure 6-3 shows the network reference model and protocol layers for user plane aspects of voice services-MPLS interworking.

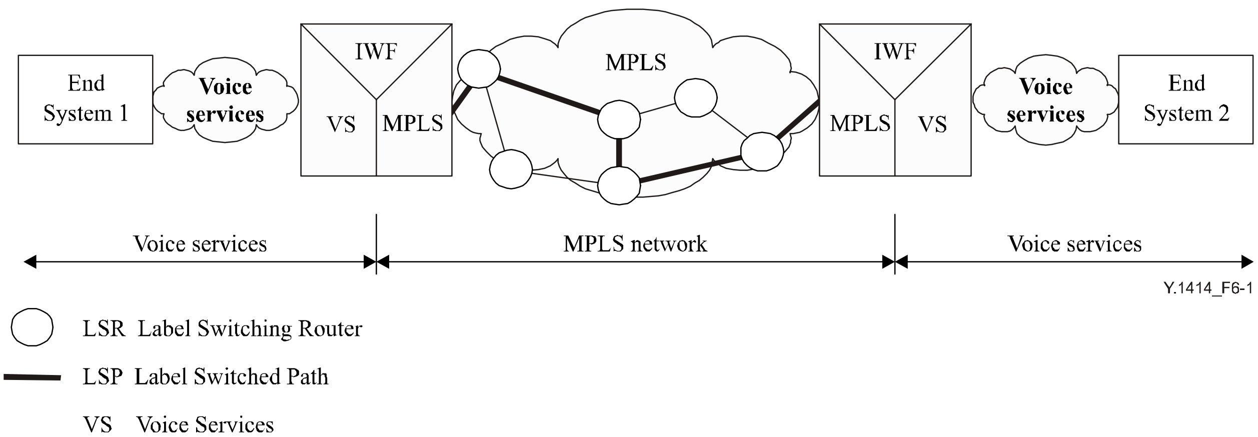


Figure 6-1/Y.1414 – Reference architecture for voice services-MPLS network interworking

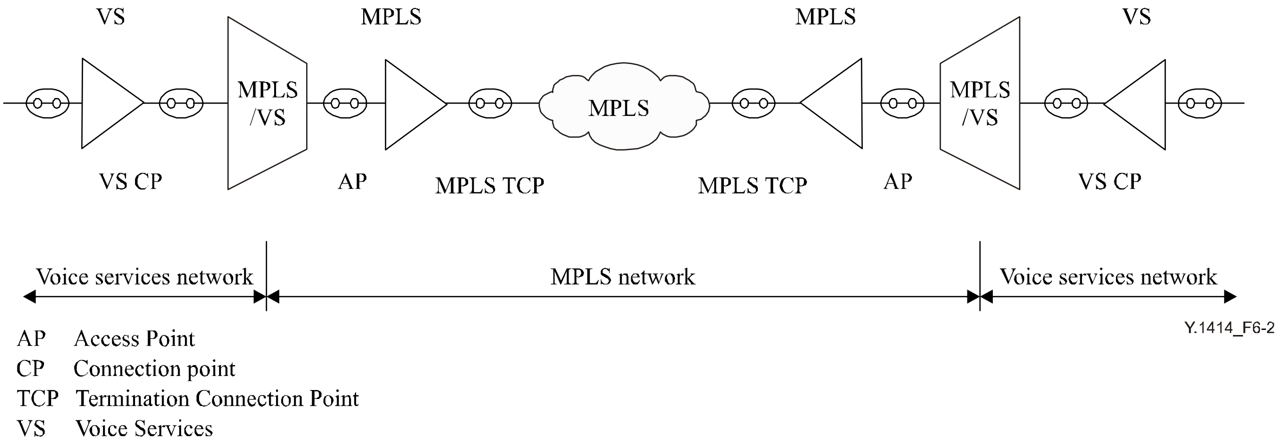


Figure 6-2/Y.1414 – Functional architecture of voice services-MPLS network interworking depicted according to the diagrammatic conventions of G.805

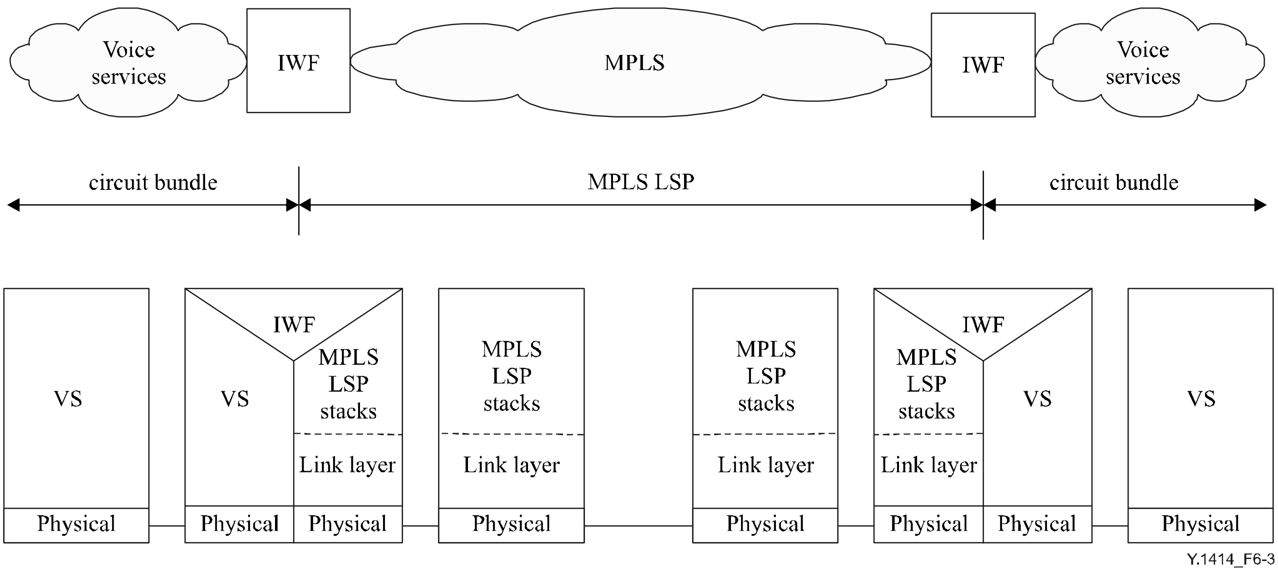


Figure 6-3/Y.1414 – Network reference model and protocol layers for voice services-MPLS user plane interworking

This Recommendation specifies voice services-MPLS network interworking for three of the possible solutions identified in Appendix II/Y.1261 [1]. The three solutions specified in this Recommendation are:

- voice over IP over MPLS;
- voice over MPLS using AAL type 2 SSCS for narrow-band services; and
- voice over MPLS using MPLS Forum IA 1.0.

7 General requirements

A set of service requirements for Voice Services over MPLS have been identified in ITU-T Rec. Y.1261 [1]. This clause identifies specific requirements for the support of narrow-band voice services by MPLS networks.

7.1 User plane requirements

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

- The ability to encapsulate data from a telephony grade channel in an MPLS packet.
- The ability to transport telephony grade audio according to ITU-T Recs G.711 [7], G.723.1 [8], G.726 [9], G.727 [10], G.729 [11] and Adaptive Multi-Rate (AMR) [12] coders.
- Optional ability to encapsulate wideband speech.
- Optional ability to transport wideband speech according to ITU-T Recs G.722 [13], G.722.1 [14] and G.722.2 [15] coders.
- The ability to reliably detect packet loss, in order to support the concealment of packet loss events by suitable PLC algorithms, such as G.711 Appendix I [16].
- The ability to transfer subscriber signalling, such as DTMF according to ITU-T Recs Q.23 [17] and Q.24 [18], call progress tones (CPT) according to ITU-T Rec. E.180/Q.35 [19], and calling line identification (CLI) [20] received either in the audio stream, or by suitable relay.
- The ability to transfer inter-office signalling systems R1 according to ITU-T Rec. Q.310-332 [21], R2 according to ITU-T Rec. Q.400-490 [22], and continuity message COT as defined in ITU-T Rec. Q.724 [23], received either in the audio stream, or by suitable relay.
- The ability to acquire, encapsulate and transfer CAS bits.
- Optional support for transport of 64 kbit/s clear-channel data, in particular for CCS.
- The ability to transfer standards-based fax (ITU-T Recs T.4 [24], T.30 [25], V.17 [26], and V.29 [27]), telephone mode text (ITU-T Rec. V.18 [28]), and voice-grade modem signals (V series modems), either in the audio stream (when remote timing allows), or by suitable relay.
- Optional support for interworking with ATM-based AAL type 2 services, in particular IMT-2000 cellular systems, Loop Emulation Service and VoDSL.
- The ability to exploit the entire MTU.

7.2 Control plane aspects

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

- Voice channel parameters (e.g., bandwidth, QoS, etc.).
- Interworking LSP label(s) between IWFs.
- Correlation of interworking labels for one or more bidirectional connections per interworking LSP. Mechanisms are to be defined.
- Indication of encapsulation modes as either "Voice over IP over MPLS", "Voice over MPLS using AAL type 2 SSCS for narrow-band services" or "Voice over MPLS using MPLS Forum IA 1.0".
- Association of each Interworking LSP label with a transport LSP label.

7.3 Management plane aspects

Interworking of defects between voice services and MPLS is out of scope of this Recommendation.

7.4 Traffic management aspects

The transport LSP shall be capable of providing the required QoS and aggregate bandwidth requirements for all voice services channels being transported. Each voice services channel may have different bandwidth requirements because of the choice of codec, and optional speech activity detection.

8 Functional group considerations for voice services-MPLS network interworking

The functional grouping is shown in Figure 8-1 and is specified in the following clauses.

MPLS transport label (4 octets)
Interworking label (4 octets) Note
Common Interworking Indicators (4 octets) Note
Payload

NOTE – These fields are not present in Voice over IP over MPLS mode.

Figure 8-1/Y.1414 – Voice services over MPLS interworking functional groups

8.1 MPLS transport label

Since LSPs are unidirectional while voice services (e.g., TDM) are inherently bidirectional, two transport LSPs in opposite directions are required.

The 4-octet transport label identifies an LSP used to transport traffic between two IWFs. The transport label is a standard MPLS shim header, as specified in IETF RFC 3032 [29], that is processed at each LSR. When the Interworking label is present, the S bit is set to "0" for the MPLS transport 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

This clause only applies in the case where there is an Interworking label. Since LSPs are unidirectional while voice services (e.g., TDM) are 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 voice service channels 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, as specified in IETF RFC 3032 [29], with its S bit set to 1, to indicate the bottom of the label stack. Since voice services-MPLS network 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 Common interworking indicators field is always present, except for the Voice over IP over MPLS mode. This 4-octet field consists of a one-octet control field, a one-octet length field and a two-octet sequence number field.

8.3.1 Control field

The first four bits of the control 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 remaining bits of the control field are reserved, and shall be set to zero at ingress and ignored at egress.

8.3.2 Length field

The one-octet length field is comprised of a two bits reserved field and six bits called length indicator which indicates the length of the payload. 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 indicator so that the padding can be removed at the egress.

The length field is used as follows: If the packet's length (defined as the length of the voice payload plus the length of the common interworking indicators field) is less than 64 octets, the length indicator is set to the packet's length in octets. Otherwise the length field is set to zero.

8.3.3 Sequence number field

The Sequence number field is a two-octet field that is used to detect lost or misordered MPLS packets. This field is always present when the Common interworking indicators are used, and its use is mandatory. The combination of the sequence number field and the UUI (User-to-User Indication) or counter fields uniquely identifies the audio payload.

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

8.3.3.1 Setting the sequence numbers

The following procedures apply at the ingress IWF (voice services 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}.

8.3.3.2 Processing the sequence numbers

The purpose of the sequence number processing is to detect lost or misordered MPLS packets. Misordered packets should be re-ordered if possible, and otherwise treated as lost. Packet loss shall trigger the packet loss concealment mechanism of the audio encoding (e.g., G.711 Appendix I).

If an egress IWF detects excessive packet loss, it should inform the management plane, enabling it to take the appropriate actions. The threshold for informing the management plane is preconfigured.

The following procedures apply at the egress IWF (MPLS to voice services 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.

9 Voice over IP over MPLS

The intent of Voice over IP over MPLS is to take standard VoIP, and encapsulate it in MPLS packets, using standard procedures for the transport of IP over MPLS. The encoding of the VoIP packet is not defined in this Recommendation. The VoIP over MPLS packet is constructed by prepending an MPLS label stack to a VoIP packet. Standard procedures for transport of IP over MPLS must be used.

The functional grouping for VoIP over MPLS is shown in Figure 9-1.

MPLS transport label (RFC 3031) [30]
IP
UDP (RFC 768) [31]
RTP (RFC 3550) [32]
Audio payload

Figure 9-1/Y.1414 – Functional grouping for VoIP over MPLS

10 Voice over MPLS using AAL type 2 SSCS for narrow-band services

This clause defines mechanisms for conveying voice services as described in ITU-T Rec. I.366.2 [2], using AAL type 2 Common Part Sublayer Packets (CPS-Packets) [3] directly over MPLS. CPS-Packets are encapsulated in the MPLS PDUs using the frame formats and procedures described in this clause.

The payload for this mode consists of one or more variable-length AAL type 2 PDUs. A similar mechanism is described in [34]. Each AAL type 2 PDU contains 3 bytes of overhead and between 1 and 64 bytes of payload. If interworking with ATM-based AAL type 2 systems is required, the payload size may be restricted to be smaller than 64 octets (typically 45 or 44 octets). A packet may be constructed by inserting PDUs corresponding to all active channels, by appending PDUs ready at a certain time, or by any other means.

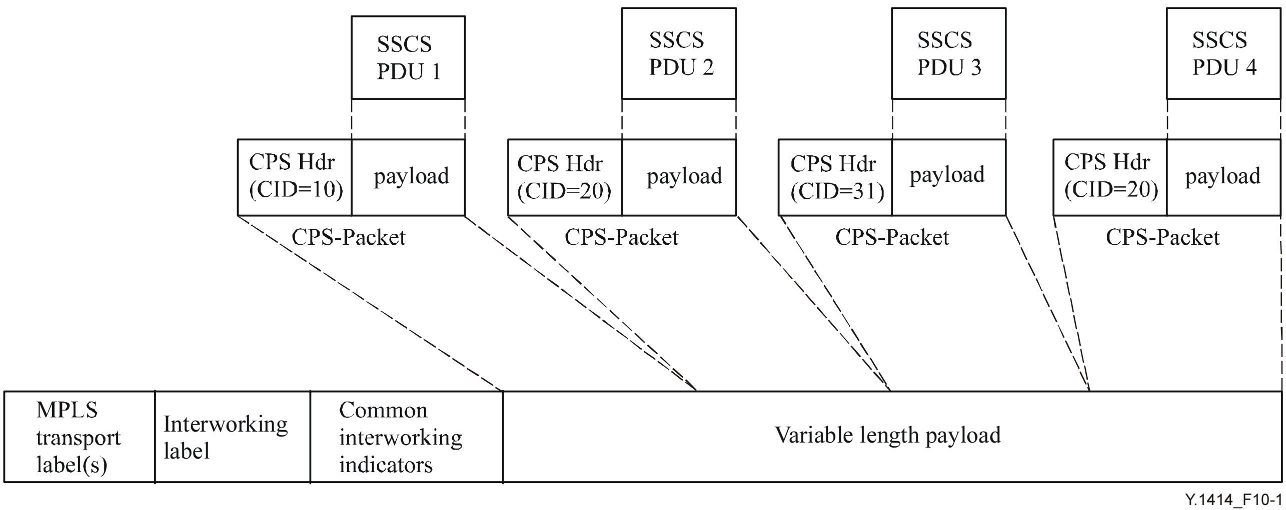


Figure 10-1/Y.1414 – Packing of SSCS-PDUs into MPLS with variable length PDU

Each MPLS packet will consist of an interworking label, Common interworking indicators, and one or more complete CPS packets, as shown in Figure 10-1. The maximum number of CPS packets per MPLS packet is determined by MPLS network limitations. A single MPLS Packet may contain any combination of Type 1 and Type 3 CPS packets.

The CPS-PDU Header Start Field (STF) is not used because there are no partial CPS packets.

The CID field of a CPS packet is 8 bits long. To ensure consistency with Table 4/I.363.2, CID value 0 is not used, and values 1-7 are reserved, thus limiting the number of AAL type 2 connections to 248. If interworking with VoDSL is required, then CID = 8 through CID = 15 are used for special purposes as specified in af-vmoa-0145.001 [33]. The same CID value may appear multiple times in a single MPLS packet. When this occurs, order shall be maintained.

NOTE – MPLS and Frame Relay Implementation Agreement IA 5.0.0 [34] defines a mechanism that is similar to that described in this clause.

10.1 Transport label

See 8.1.

10.2 Interworking label

See 8.2.

10.3 Common interworking indicators

See 8.3.

10.4 Payload

The payload consists of one or more CPS packets, as defined in ITU-T Rec. I.363.2 [3].

All mechanisms described in ITU-T Recs I.363.2 [3] and I.366.2 [2], may be used. In particular, CID encoding according to ITU-T Rec. I.363.2, encoding formats defined in ITU-T Rec. I.366.2, and transport of CAS and CCS signalling as described in ATM forum af-vmoa-0145.001 [33] may all be used. However, the overlap functionality defined in ITU-T Rec. I.363.2 is not required, the STF field is NOT used, and the AAL type 2 Timer_CU and related functionalities are by default not required, but may be used.

Computation of error detection codes, namely the HEC in the AAL2 PDU header and the CRC in the CAS packet, may be omitted, if an appropriate error detection mechanism is provided by MPLS. In such cases these fields shall be set to zero.

11 Voice over MPLS using MPLS Forum IA 1.0

This mode is based on MPLS Forum Implementation Agreement 1.0 [4] and its subsequent revisions. Appendix I summarizes the frame structure for this mode.

MPLS Forum IA 1.0 includes the definition of a header format, and supports various payload types including Audio, Dialled digits, Channel Associated Signalling and Silence Insertion Descriptor.

The format and procedures are based on ITU-T Rec. I.366.2 AAL type 2 SSCS for narrow-band services. However, some of the sub-frame header fields are larger than the corresponding fields in AAL type 2, larger payloads are allowed and profiles are not employed.

11.1 Transport label

See 8.1.

11.2 Interworking label

See 8.2.

11.3 Common interworking indicators

See 8.3.

11.4 Payload

The payload format is defined in MPLS Forum IA 1.0. The payload is encapsulated as described in clause 8.

11.4.1 Format of the primary sub-frame

The format is as specified in 5.2 of MPLS Forum IA 1.0 with the following modifications:

Channel Identifier (CID)

The CID value identifies the user of the channel. The voice services over MPLS channel is a bidirectional channel. The same value of channel identification shall be used for both directions.

The value "0" is not used for channel identification. The values "1" through "7" are reserved.

The values "8" through "255" are used to identify the users of the voice over MPLS.

Table 11-1/Y.1414 – Coding of the CID field

CID value	Use
0	Not used
1 2 3 through 7	Reserved for Layer Management peer-to-peer procedures Reserved for Signalling Reserved
8 through 255	Identification of voice over MPLS user entity.

NOTE – Future revisions of the MPLS Forum IA are expected to align with usage of CID in this clause.

11.4.2 Procedures

Procedures for processing voice over MPLS sub-frames are specified in MPLS Forum IA 1.0 and its annexes.

Appendix I

Primary and control sub-frames used in MPLS Forum 1.0

This appendix provides an informative overview of the Sub-frame types that have been defined in MPLS Forum implementation agreement in order to transport multiplexed voice calls and control information in an MPLS LSP. Multiple voice calls may be transported over an Interworking LSP. Further details may be obtained from [4] and clause 11.

I.1 Sub-frames

Two types of sub-frames are defined in [4], Primary and Control, and may be transmitted as required. Multiple primary sub-frames may be multiplexed within a single MPLS packet. The control sub-frames are not multiplexed and are sent separately; that is, only one control sub-frame at a time may be carried within an MPLS packet. Primary sub-frames and control sub-frames are not multiplexed together within a single MPLS packet.

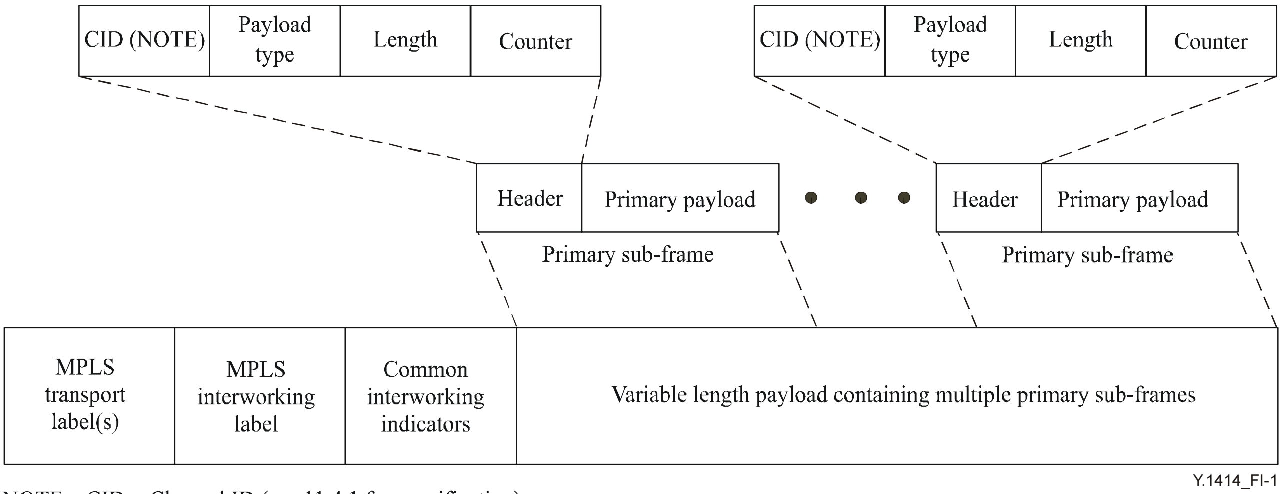
The maximum number of sub-frames per MPLS packet will be determined by the MPLS network limitations. When more than one sub-frame from a user connection is packed within an MPLS packet, their order should be maintained.

A primary payload contains the traffic that is fundamental to the operation of a connection identified by a Channel Identifier (CID). It includes encoded voice and silence information descriptor(s). Primary payloads are variable length sub-frames.

Control sub-frames may be sent to support the primary payload (e.g., dialled digits for a primary payload of encoded voice) and other control functions. These payloads are differentiated from the primary payload by a Payload Type value in the sub-frame header. A range of payload type values is assigned to primary payload and control payloads. Control sub-frames are fixed length and most of them are sent with a triple redundant transmission with a fixed interval between them. The CID and payload type fields are common to both primary and control payload formats.

I.2 Primary Sub-frame

A primary payload contains the traffic that is fundamental to the operation of a connection identified by a Channel Identifier (CID). It includes encoded voice and silence information descriptor(s). Primary payloads are variable length sub-frames. The MPLS packet structure allowing the multiplexing of primary sub-frames is shown in Figure I.1.



NOTE – CID = Channel ID (see 11.4.1 for specification).

Figure I.1/Y.1414 – LSP structure for multiplexing primary sub-frames of voice calls

The multiplexing structure in this mode consists of an MPLS Transport Label, an Interworking Label, Common Interworking Indicators, and a Payload. The Payload may contain one or more primary sub-frames [4] consisting of a 4-octet header and variable length primary payload.

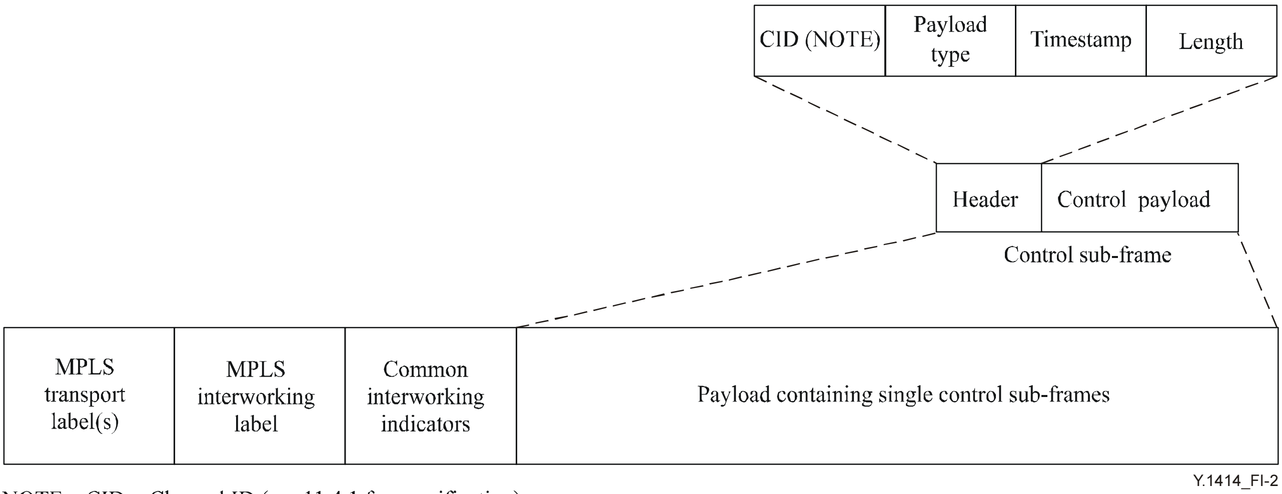
The Channel ID (CID) allows up to 248 voice over MPLS calls to be multiplexed within a single interworking LSP.

NOTE – A unique CID identifies each sub-frame but the primary sub-frames may be transmitted in any order whenever information for a channel is available. The same CID value may appear multiple times in a single MPLS packet.

I.3 Control sub-frame

Control sub-frames may be sent to support the primary payload (e.g., dialled digits for a primary payload of encoded voice) and other control functions. These payloads are differentiated from the primary payload by a Payload Type value in the sub-frame header. A range of payload type values is assigned to primary payload and control payloads. Control sub-frames are fixed length and most of them are sent with a triple redundant transmission with a fixed interval between them. The CID and payload type fields are common to both primary and control payload formats.

The control sub-frames that are currently defined are: Dialled digits and Channel associated signalling. The MPLS packet structure for control sub-frames of Voice over MPLS calls is shown in Figure I.2.



NOTE – CID = Channel ID (see 11.4.1 for specification).

Figure I.2/Y.1414 – LSP Structure for Control Sub-frame

I.4 Format of the primary sub-frame

To maintain word (32 bits) alignment, the payload information must be a multiple of 4 octets. If the payload is not a multiple of 4 octets, up to 3 PAD octets are included to make it word aligned.

A primary payload is either a sequence of encoded voice sub-frames or a single Silence Insertion Descriptor sub-frame.

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