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SERIES Y: GLOBAL INFORMATION INFRASTRUCTURE AND INTERNET PROTOCOL ASPECTS

Internet protocol aspects – Architecture, access, network capabilities and resource management

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Service requirements and architecture for voice services over Multi-Protocol Label Switching

ITU-T Recommendation Y.1261

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ITU-T Recommendation Y.1261

Service requirements and architecture for voice services over Multi-Protocol Label Switching

Summary

This Recommendation describes service requirements and architecture for transport of Voice Services over MPLS networks.

Voice Services over MPLS networks include a family of protocols that may be supported over MPLS, including Voice over IP (as specified by IETF), Voice over MPLS (as specified by MPLS Forum) and I.366.2 Voice Trunking Format (based on ITU-T Rec. I.366.2 adapted for MPLS).

Possible approaches for transporting Voice Services over MPLS are provided in Appendix II.

Detailed protocol specifications are subject of other Recommendations.

Source

ITU-T Recommendation Y.1261 was prepared by ITU-T Study Group 13 (2001-2004) and approved under the WTSA Resolution 1 procedure on 14 December 2002.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation Y.1261

Service requirements and architecture for voice services over Multi-Protocol Label Switching

1 Scope

The scope of this Recommendation is to describe services requirements and architecture for transport of Voice Services over MPLS networks.

Voice Services over MPLS networks include a family of protocols that may be supported over MPLS, including Voice over IP (as specified by IETF), Voice over MPLS (as specified by MPLS Forum) and I.366.2 Voice Trunking Format (based on ITU-T Rec. I.366.2 adapted for MPLS).

An overview of several possible approaches that have been proposed as the basis for the family of protocols for transporting Voice services over MPLS is provided in Appendix II. Nothing in this Recommendation precludes the progression of all these protocols for the transport of Voice Services over MPLS.

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.

2.1 Normative references

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

[2] 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*.

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

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

[5] ITU-T Recommendation I.361 (1999), *B-ISDN ATM layer specification*.

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

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

[8] ITU-T Recommendation I.363.5 (1996), *B-ISDN Adaptation Layer specification: Type 5 AAL*.

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

[10] ITU-T Recommendation I.610 (1999), *B-ISDN operation and maintenance principles and functions*.

[11] ITU-T Recommendation Q.922 (1992), *ISDN data link layer specification for frame mode bearer services*.

[12] ETSI TS 126 071 V3.0.1 (2000-02), *Universal Mobile Telecommunications System (UMTS); AMR Speech Codec; General description*.

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

[14] IETF RFC 3032 (2001), *MPLS Label Stack encoding*.

[15] IETF RFC 3036 (2001), *LDP Specification*.

[16] IETF RFC 3270 (2002) *Multi-Protocol Label Switching (MPLS). Support of Differentiated Services*.

2.2 Informative references

[17] ITU-T Recommendation Q.1950 (2002), *Bearer independent call bearer control protocol*.

[18] IETF RFC (1996), *RTP: A Transport Protocol For Real-Time applications*.

[19] IETF RFC 1890 (1996), *RTP Profile for Audio and Video Conferences with Minimal Control*.

[20] FRF.11.1 (1999), *Voice over Frame Relay Implementation Agreement*.

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

3 Definitions

This Recommendation defines the following terms:

3.1 voice services over MPLS: The term encompasses the set of technical approaches used to transport encoded voice over MPLS. This terminology is independent of the method by which MPLS frames are transported.

3.2 active voice: A sampled audio interval that has been determined to contain speech as opposed to silence. The classification is made by a Voice Activity Detection algorithm. It enables discontinuous transmission, whereby the bit rate of the signal is reduced during silent intervals.

3.3 audio: In the context of this Recommendation, audio is used as a general term for signals of the audible medium, including voice and voiceband data.

3.4 channel-associated signalling: Bits dedicated for connection control across a 1544 kbit/s or 2048 kbit/s interface that carries 64 kbit/s channels. Procedures are based on the state of up to four signalling bits (A, B, C, D) that are allocated per channel per multiframe.

3.5 circuit mode data: A continuous stream of digital information at 64 kbit/s or 2 × 64 kbit/s having an 8 kHz structure.

3.6 dialled digits: Multifrequency audio tones typically used for inter-register signalling of addresses during call set-up or for end-to-end device control during an established call. Depending on the system, codes are defined for the digits 0-9 of a telephone keypad and other auxiliary signals.

3.7 facsimile demodulation/remodulation: The process of detecting facsimile traffic, extracting digital information from the incoming analogue modulated signal, transporting this across a trunk in packet formats, and reproducing the facsimile control and image information by remodulation at the other end.

3.8 silence insertion descriptor: A compressed representation of the audio background noise that can be sent during silent intervals. SIDs may not be continuous and may only be sent when there is a change in noise characteristics. Playing out received SIDs is known as Comfort Noise Generation.

4 Abbreviations

This Recommendation uses the following abbreviations:

AAL ATM Adaptation Layer

AMR Adaptive Multi-Rate

ATM Asynchronous Transfer Mode

BICC Bearer Independent Call Control

CAS Channel-Associated Signalling

CPCS Common Part Convergence Sublayer

DTMF Dual-Tone Multi-Frequency

GW Gateway

IP Internet Protocol

ISDN Integrated Services Digital Network

LER Label Edge Router

LSP Label Switched Path

LSR Label Switching Router

MG Media Gateway

MGC Media Gateway Controller

MPLS Multi-Protocol Label Switching

MSC Mobile Switching Center

MTU Maximum Transmission Unit

NT Network Termination

PSTN Public Switched Telephone Network

RAN Radio Access Network

RTP Real-Time Transport Protocol

SSCS Service-Specific Convergence Sublayer

TDM Time Division Multiplexing

UDP User Datagram Protocol

5 Architecture

MPLS is being introduced into IP networks to support Traffic Engineering and various applications. One of the motivations for Voice Services over MPLS is to take advantage of the MPLS network capabilities to improve voice services by:

- using label-switched-paths as a bearer capability for encoded voice thereby providing appropriate QoS capability;
- providing efficient transport mechanism.

5.1 Voice Services over MPLS logical service reference model

A logical service reference model for Voice Services over MPLS consists of two main components:

1) a high-capacity MPLS core network consisting of MPLS nodes (LSR) with support of MPLS control protocols;

2) Gateway Devices used at the edge of the MPLS network to perform interworking between a variety of technologies such as ATM, frame relay, TDM, IP, PSTN, ISDN, etc. and the MPLS network.

Figure 1 shows the Voice Services over MPLS logical service reference model including control plane, transport plane, and management plane.

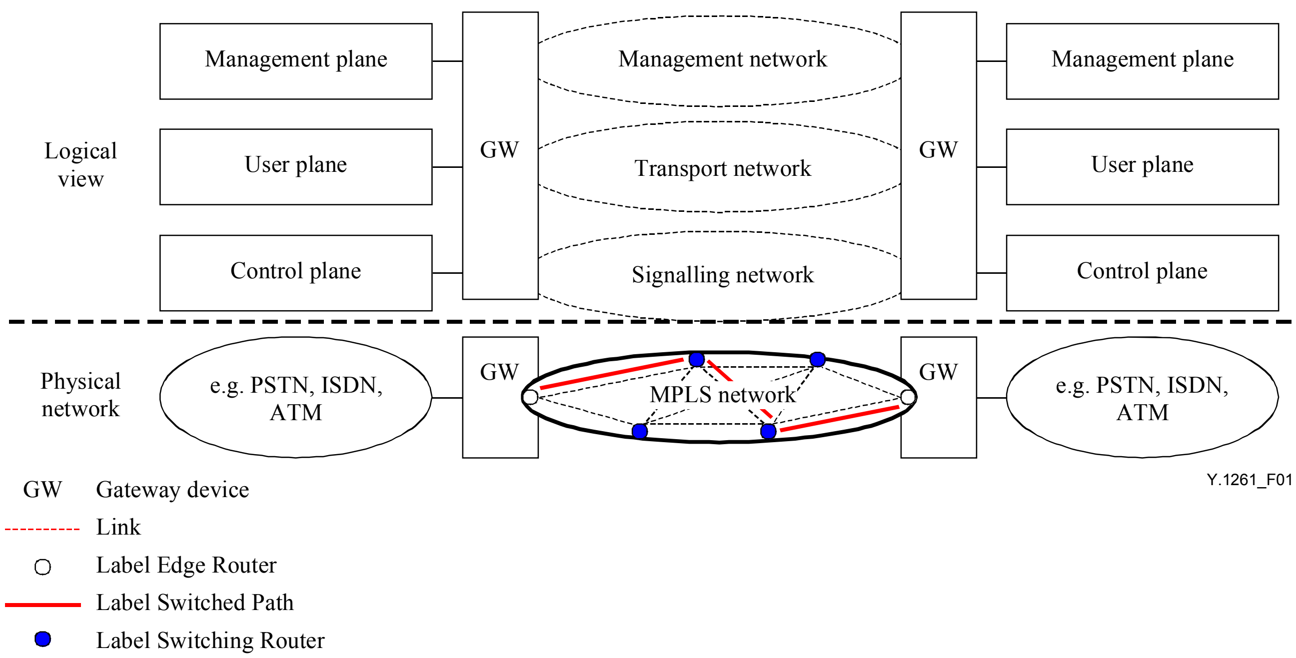


Figure 1/Y.1261 – Voice Services over MPLS logical service reference model

5.2 Voice services over MPLS Reference Architecture

Figure 2 identifies the Reference Architecture for Voice Services over MPLS. The MPLS network contains a number of Gateway devices (GW), Label Switching Routers (LSR), and Label Switched Paths (LSP) [13], [14], [15]. An example LSP is shown as a solid line in the figure. Gateways may be directly connected to each other or indirectly connected through a number of LSRs.

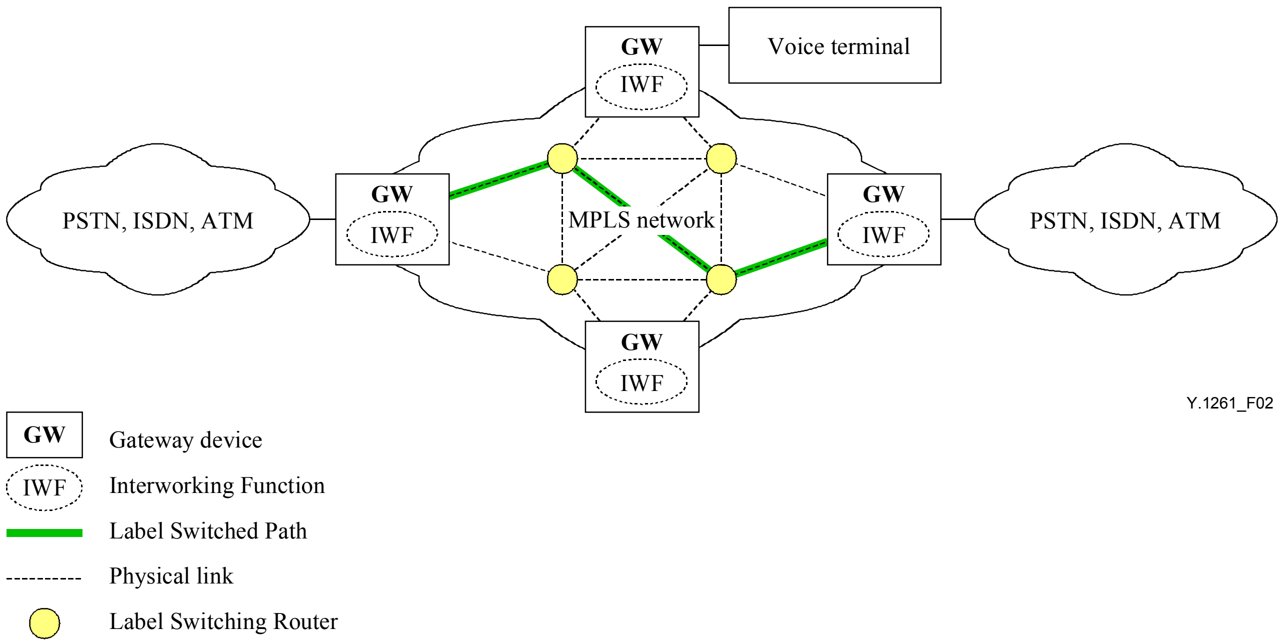


Figure 2/Y.1261 – Voice services over MPLS Reference Architecture

Gateway devices include LER functionality as well as the interworking functions.

This architecture should be capable of supporting different LSP configurations (e.g., E-LSP, L-LSP [16], hierarchical LSP, etc.) to convey voice payloads in an MPLS environment.

NOTE – Voice Services over MPLS can be deployed in both core and access networks. Network architecture functions and definitions for the usage of Voice Services over MPLS in access networks are for further study.

6 Requirements for Voice Services over MPLS

The following subclauses provide a list of requirements for Voice Services over MPLS. The relative priority of these requirements is Service Provider specific.

6.1 General service requirements

The following provides a list of general service requirements:

1) The architecture should be capable of supporting carrier-grade networks (e.g., support of a large number of voice calls as well as a rapid rate of voice call requests, etc.) in accordance with existing Recommendations for voice networks.
2) If a solution makes use of mechanisms which apply only in restricted environments, a clear statement about the restricted applicability must be included.
3) Multiple applications may be supported (e.g., facsimile, video, etc.)

Table 1 below summarises the categories and the corresponding delivered services which have been identified in the "Voice Services over MPLS" family.

Table 1/Y.1261 – Categories for Voice Services over MPLS

Category	Services delivered
Audio	Digitized Voice
	Alarm
	Silence information descriptor
	Dialled digits
	Channel-associated signalling
	Facsimile – voiceband data
	Facsimile remodulation/demodulation
	Circuit mode data for 64 kbit/s or 2×64 kbit/s
Multirate	Circuit mode data for $N \times 64$kbit/s, $2 < N \leq 31$
	Alarm
Video	Video
	Alarm
NOTE – Either one category or any combination of categories may be supported.	

6.2 User plane requirements

The following provides a list of requirements for the user plane:

1) Multiplexing encoded voice samples from different calls into a single MPLS frame should be supported.
2) The bandwidth should not be used by the voice application when it is not needed – when the talk is silent or when the call is completed.
3) The user plane protocol should be efficient in terms of bandwidth.
4) The capability to implement echo cancellation functionality to reduce echo should be supported.
5) QoS/SLA (jitter, loss, bandwidth, delay) commitments for the voice service must be supported.
6) Uncompressed (i.e., G.711 64 kbit/s, both A-law and μ-law) and compressed voice (G.726 16/24/32/40 kbit/s, G.729 8 kbit/s, G.723.1 5.3/6.3 kbit/s) should be supported ([1], [2], [3], [4]).
7) Standard encoding schemes should be supported.
8) Specific families of related encoding algorithms (e.g., AMR [12]) should experience minimum disruption in audio when the algorithm rate is changed during the call.
9) Silence suppression, i.e., non-transfer of MPLS subframes during silent intervals should be supported. Generic Silence Insertion Descriptor for voice encodings that do not contain this capability should be supported.
10) DTMF transport should be supported.
11) Voiceband data through modem detection should be supported.
12) Transport of Channel-Associated Signalling bits should be supported.

6.3 User plane requirements for combined audio and video service

The following provides a list of user plane requirements for the combined audio and video service:

1) Multimedia applications may be supported.

2) If multimedia applications are supported, standard video schemes should be supported (e.g., CelB, JPEG, H261, MPV, MP2T, and H263).

3) One video call may be mapped to one LSP. Mapping more than one video call to single LSP is for further study.

4) Higher resolution timer (e.g., 90 kHz for IP video) and indication of end of frame should be supported.

5) Variable length and large payloads up to MTU size minus the MPLS overhead of the underlying link layer for MPLS should be supported.

6.4 Control plane requirements

The following provides a list of requirements for the control plane:

1) LSPs that are used for Voice Services over MPLS may be either established on demand or via management procedures.

2) Voice call channel parameters (e.g., bandwidth, QoS, etc.) applicable to both single and multiplexed channels in an LSP may be either statically or dynamically provisioned.

6.5 Management plane requirements

The management plane should provide the following functions:

a) Resource management (e.g., bandwidth, label, addresses, etc.)

b) Parameter management (e.g., priority of traffic, sampling and framing period of voice)

c) Monitoring and maintenance:
   - connection identification;
   - connection state monitoring;
   - fault detection and notification;
   - alarm capabilities.

7 Voice Services over MPLS interworking requirements

7.1 Voice Services over MPLS Interworking requirements with Voice over IP

1) ITU-T Rec. G.711 must be supported without negotiation.

2) Video and combined audio and video services may be supported.

3) Tones and DTMF digits should be supported.

7.2 Voice Services over MPLS interworking requirements with Voice over ATM

1) Narrowband services defined in ITU-T Rec. I.366.2 should be supported.

2) Interworking of operation and maintenance capabilities defined in ITU-T Rec. I.610 [10] should be supported.

3) Mapping of ATM traffic and QoS parameters to MPLS LSPs should be supported.

7.3 Voice Services over MPLS interworking requirements with Voice over Frame Relay

Support is for further study.

7.4 Voice Services over MPLS interworking requirements with PSTN/ISDN

1) Interworking with PSTN/ISDN should be supported.

7.5 Voice Services over MPLS Interworking requirements with mobile access networks

Support is for further study.

Appendix I

Voice over Packet Protocol architectures

In recent years, voice over packet protocols were developed for different packet networks. They include: Voice over Frame Relay, Voice over ATM using AAL type2 as described in ITU-T Rec. I.366.2 [9] and Voice over Internet Protocol. Each of these packet voice schemes developed a transport protocol for providing end-to-end network transport for voice applications.

These Voice over Packet protocols provide functionalities similar to Voice Services over MPLS in the User plane protocol.

I.1 Voice over Frame Relay

The Frame Relay Forum developed the Voice over Frame Relay implementation agreement [20]. Transport of uncompressed and compressed voice is provided with a generalized frame format that supports multiplexing of subchannels on a single frame relay connection. Support for the unique needs of the different voice compression algorithms is accommodated with algorithm-specific "transfer syntax" definitions. These definitions establish algorithm specific frame formats and procedures.

Transport of supporting information for voice communication, such as signalling indications (e.g., ABCD bits), dialled digits, and facsimile data, is also provided through the use of transfer syntax definitions specific to the information being sent.

I.1.1 FR reference model and service description

In the FR reference model (Figure I.1), a Voice Frame Relay Access Device can exchange voice information over a FR connection with another Voice Frame Relay Access Device. It could also use the Voice over Frame Relay protocol stack on a connection to a transparent channel bank into a private network (middle right) or to a PBX (bottom right).

A block diagram for the Voice over Frame Relay service is provided in Figure I.2 and identifies the information provided to users of the Voice over Frame Relay service.

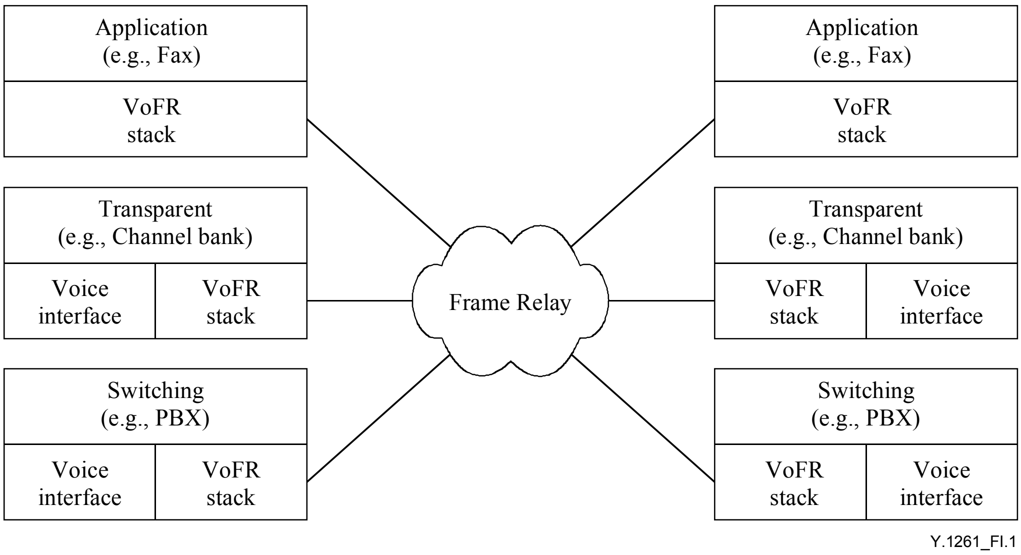


Figure I.1/Y.1261 – Voice over Frame Relay network reference model

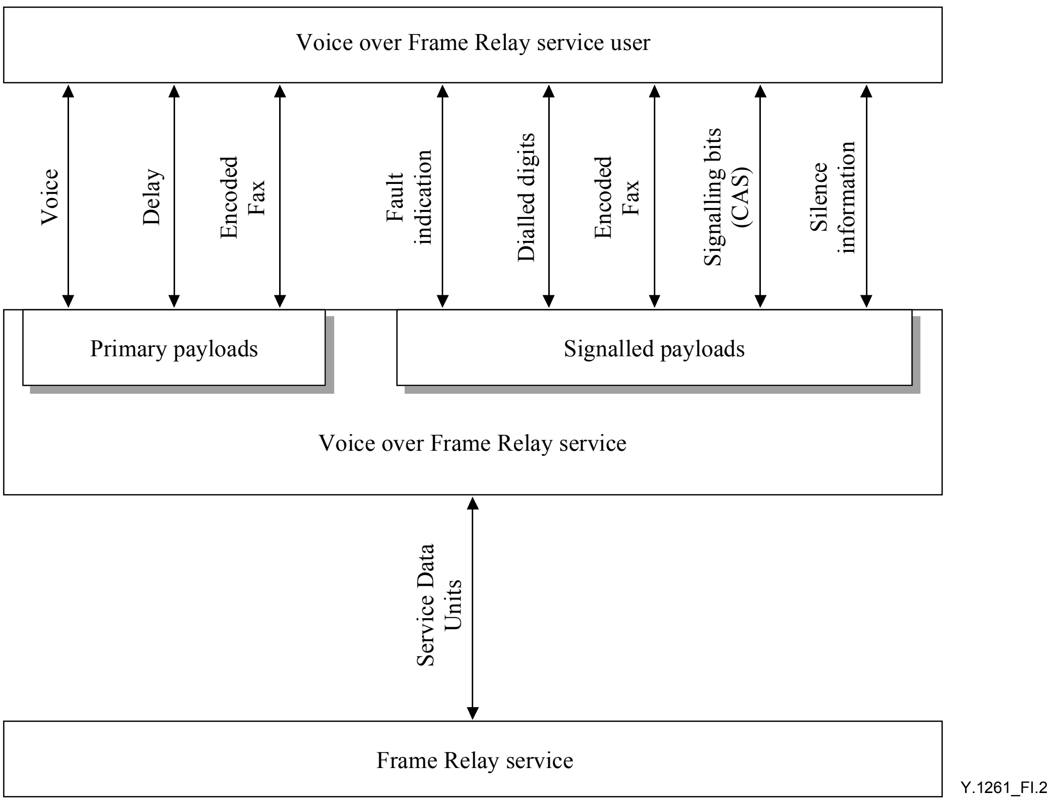


Figure I.2/Y.1261 – Voice over Frame Relay service block diagram

I.1.2 Multiplexing

The Voice over Frame Relay service supports multiple voice and data channels on a single frame relay data link connection. The Voice over Frame Relay service delivers frames on each sub-channel in the order they were sent.

Voice and data payloads are multiplexed within a voice over frame relay data link connection by encapsulation within the FR frame. Each payload is packaged as a sub-frame within a frame's information field. Sub-frames may be combined within a single frame to increase processing and transport efficiencies. Each sub-frame contains a header and payload. The sub-frame header identifies the voice/data sub-channel and, when required, payload type and length.

I.1.3 Voice over Frame Relay user plane protocol stack

In the case of Voice over Frame Relay, the user plane protocol stack consists of: *Physical layer/ Q.922 Annex A/Voice over Frame Relay* as shown in Figure I.3 [11]. The higher layer (i.e., Voice over Frame Relay) is providing the voice protocol. The Voice over Frame Relay protocol is the User plane protocol in this case and provides peer-to-peer layer communications in a Frame Relay network.

Voice
Voice over Frame relay
Q.922 Annex A
Physical layer

Figure I.3/Y.1261 – Voice over Frame Relay protocol stack

I.2 Voice over ATM as described in AAL type 2 Service Specific Convergence Sublayer for trunking [9]

Figure I.4 shows an example of narrow-band trunking supported in AAL type 2 Service Specific Convergence Sublayer for trunking [9].

The Service-Specific Convergence Sublayer (SSCS) carries the information content of one narrow-band call over each AAL type 2 connection – with the appropriate bearer capability. Secondary messaging, such as frame mode data, dialled digits, channel-associated signalling bits, and alarms may be interleaved on the same AAL type 2 connection.

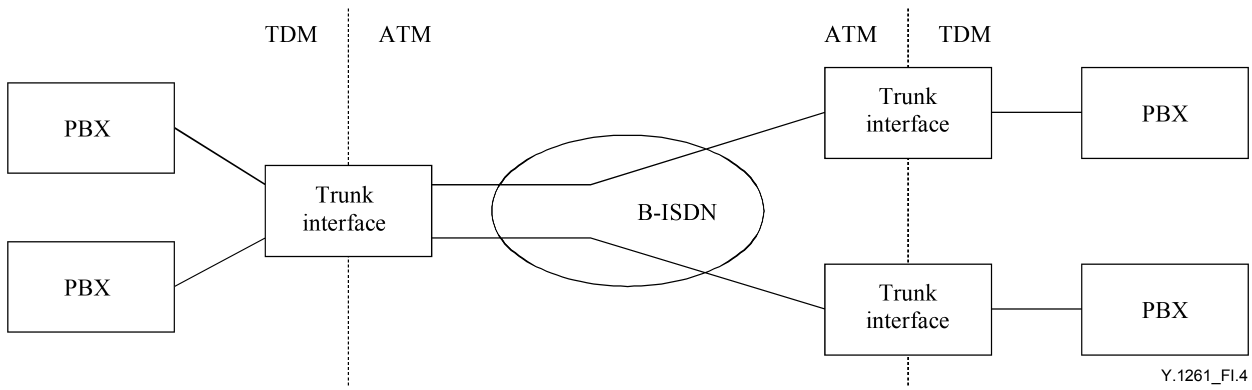


Figure I.4/Y.1261 – Deployment example of narrow-band trunking

At each end of a trunk, the User coordinates the operations of the SSCS. The services offered by the SSCS are delivered through Service Access Points (SAPs).

The audio, circuit mode data, and facsimile demodulation/remodulation services represent primary information streams of the audio service. Only one of these streams can be transported on an AAL type 2 connection at a given time.

The dialled digits service is a secondary information stream. It could be transported simultaneously with one of the primary streams, but it is anticipated that the primary stream will be made idle during the transport of dialled digits. The frame mode data, channel-associated signalling, and alarms services are secondary information streams that can be transported simultaneously with one of the primary information streams.

I.2.1 User plane protocol stack for Voice over ATM as described in AAL type 2 Service-Specific Convergence Sublayer for trunking [9]

For voice over ATM using AAL Type 2 as described in ITU-T Rec. I.366.2, the user plane protocol stack consists of: *physical layer/ATM layer/ (AAL 2 common part + AAL 2 service specific part)/Voice.*

The service-specific part of AAL sublayer performs different functions depending on the service. I.366.2 defines how to transport audio service information between trunking interfaces.

Voice
AAL 2 Service-Specific Part (I.366.2)
AAL 2 Common Part (I.366.2)
ATM
Physical Layer

Figure I.5/Y.1261 – User plane protocol stack for Voice over ATM as described in AAL type 2 Service-Specific Convergence Sublayer for trunking [9]

I.3 Voice over IP

The user plane protocol for a Voice over IP stack is shown in Figure I.6.

Voice
RTP
UDP
IP
Link Layer
Physical Layer

Figure I.6/Y.1261 – Voice over IP protocol stack

The RTP protocol provides end-to-end network transport functions suitable for audio application. Those functions include payload type identification, sequence numbering, and time stamping.

RTP provides the information required by a particular application and will often be integrated into the application processing rather than being implemented as a separate layer. RTP along with application profiles for audio provides the transport services for voice application. The profile specifications define a set of payload type codes and their mapping to payload formats.

The RTP header contains timing information and a sequence number that allows the receiver to reconstruct the sample produced by the source.

Appendix II

Alternate solutions for Voice Services over MPLS

MPLS provides a link layer independent transport framework and uses existing IP mechanisms for addressing and routing of traffic. There are several possibilities to provide Voice Services over MPLS.

It is important to realize that MPLS networks will evolve from existing networks; existing services and capabilities will have to be supported and interwork with the MPLS networks.

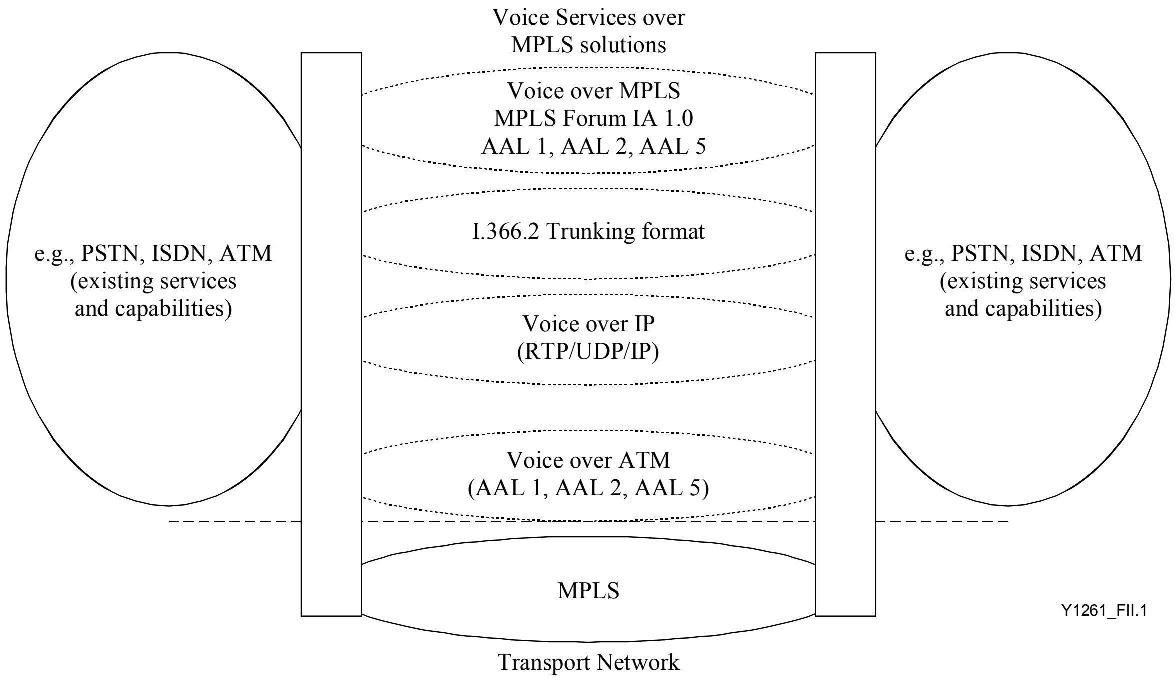


Figure II.1/Y.1261 – Alternate solutions for Voice Services over MPLS

II.1 Voice over ATM

The coded voice over an ATM connection [5] can be transported using:

a) Voice over AAL 1;

b) Voice over AAL 2;

c) Voice over AAL 5.

This approach uses respectively ATM AAL type 1 [6], AAL type 2 [7] [9], or AAL type 5 [8].

Using this approach over an MPLS network, ATM/MPLS network interworking or AAL/MPLS network interworking can be used. ATM cells or AAL frames are received at the ATM/MPLS network gateway.

NOTE – ATM/MPLS and AAL/MPLS interworking specifications are currently under development.

Voice
AAL 1 or AAL 2 or AAL 5
ATM
MPLS
Link layer
Physical layer

Figure II.2/Y.1261 – Voice over ATM protocol stack

II.2 Voice over IP

MPLS will be used to transport IP frames. Voice over IP is transparent to MPLS. RFC 1889 [18], and RFC 1890 [15] are used by this approach.

Voice
RTP
UDP
IP
MPLS
Link layer
Physical layer

Figure II.3/Y.1261 – Voice over IP protocol stack

II.3 I.366.2 Voice Trunking format over MPLS

This approach describes a mechanism using the same format of AAL type 2 CPCS packet encapsulated within MPLS frames. It uses variable length packets with are formatted as described in ITU-T Rec. I.366.2, but carried by MPLS frames and not ATM cells.

In this approach, the segmentation and reassembly (SAR) functionality as well as the start field are not required. Similarly, the AAL-CU timer and related functionality are not required.

Voice
I.366.2 Trunking format over MPLS
MPLS
Link layer
Physical layer

Figure II.4/Y.1261 – I.366.2 Voice Trunking format over MPLS protocol stack

II.4 Voice over MPLS

This approach is based on the MPLS Forum Implementation Agreement 1.0 [21]. In this case, the protocol stack consists of voice samples directly encapsulated in the MPLS frame.

Voice
Voice over MPLS MPLS Forum IA 1.0
MPLS
Link layer
Physical layer

Figure II.5/Y.1261 – Voice over MPLS protocol stack

II.5 Summary of different solutions for Voice Services over MPLS

Figure II.6 summarizes a number of protocol stacks to support Voice Services over MPLS (those described in Appendix II).

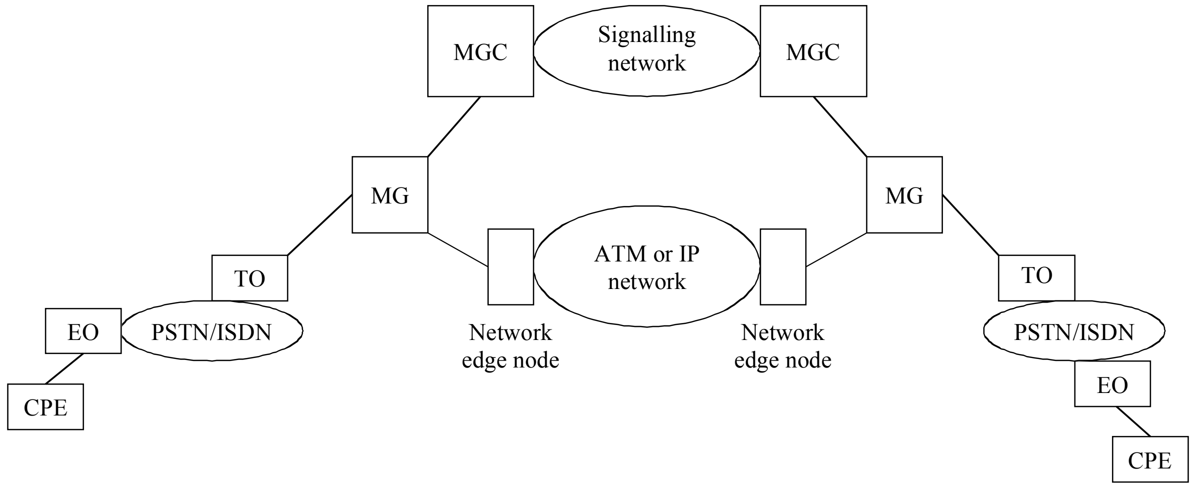
<table>
<tr><td colspan="6">Voice</td></tr>
<tr><td colspan="2" rowspan="2"></td><td>RTP</td><td rowspan="2">AAL 1
(I.363.1)</td><td rowspan="2">AAL 2
(I.363.2, I.366.2)</td><td rowspan="2">AAL 5
(I.363.5)</td></tr>
<tr><td>UDP</td></tr>
<tr><td>Voice over MPLS
MPLS Forum IA 1.0</td><td>I.366.2 Voice trunking format over MPLS</td><td>IP</td><td colspan="3">ATM</td></tr>
<tr><td colspan="6">MPLS layer</td></tr>
<tr><td colspan="6">Link layer</td></tr>
<tr><td colspan="6">Physical layer</td></tr>
</table>

Figure II.6/Y.1261 – Various protocol stacks for support of Voice Services over MPLS

Appendix III

BICC architecture

An instance of the Bearer Independent Call Control (BICC) architecture is shown in Figure III.1 [17]. It consists of ATM or IP backbone networks and Media gateways (MG) connected to network edge nodes. The role of a MG is to map TDM time slots to ATM cells or IP packets and vice versa. MGs can be either line (local loop) or trunk MGs. Media gateway controllers (MGC) perform call control functions and interact with one or more MGs under their control.



CPE Customer Premises Equipment
EO End Office
TO Trunk Office

Figure III.1/Y.1261 – An instance of BICC architecture

When MGs are connected to an ATM network, AAL type 1 or AAL type 2 may be used to transport voice services. When MGs are connected to an IP network, RTP over UDP and IP is used. MGs interface to the PSTN with TDM 64 kbit/s channels.

So far BICC supports ATM networks with AAL type 1 and type 2 and IP networks with RTP/UDP/IP for voice transport.

Current BICC capabilities will remain for some time when the network evolves to the multi-service MPLS network.

Detailed BICC architecture and its evolution for MPLS network control is out of scope of this Recommendation.

Appendix IV

Examples of Voice Services over MPLS deployment scenarios

LSPs that are used for Voice Services over MPLS could be either established on demand or via management procedures. Depending on the specific application, the mapping of voice call information streams (e.g., time slots on a primary rate interface) to multiplexed streams could be either static or dynamic. Voice calls could even be switched at an external interface to one of several outgoing LSPs, based on an analysis of the destination address.

These are possible applications of Voice Services over MPLS and are provided just as deployment examples.

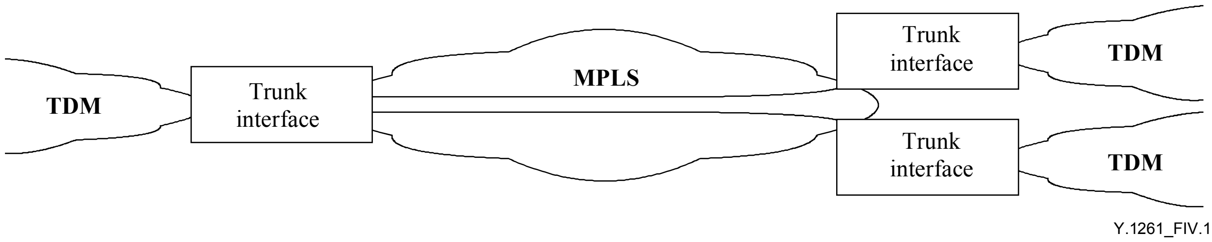


Figure IV.1/Y.1261 – Deployment example of Voice Services over MPLS for trunking

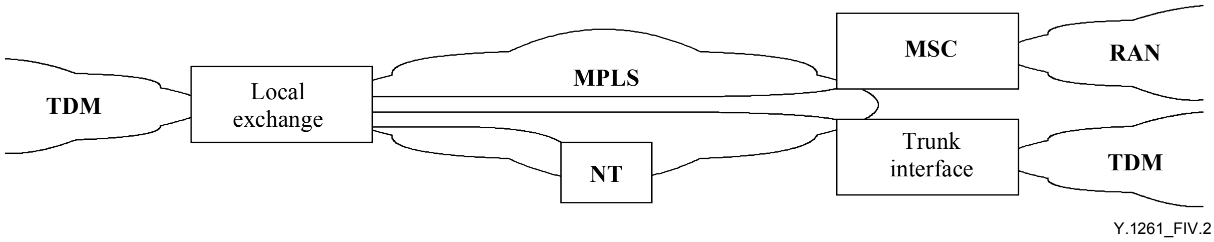


Figure IV.2/Y.1261 – Deployment example of Voice Services over MPLS for trunking, for mobile access, and for fixed line access

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