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

Internet protocol aspects – Interworking

Voice trunking over IP networks

ITU-T Recommendation Y.1452

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ITU-T Recommendation Y.1452

Voice trunking over IP networks

Summary

This Recommendation addresses required functions for voice trunking over IP networks. It specifies the required protocols, the interactions between these protocols and audio channel mechanisms, the operation of the IWF, and the mechanisms for transporting over point-to-point or complex IP networks. This Recommendation may not be suitable for use by recognized operating agencies (ROA).

Source

ITU-T Recommendation Y.1452 was approved on 1 March 2006 by ITU-T Study Group 13 (2005-2008) under the ITU-T Recommendation A.8 procedure.

Keywords

Interworking, IP, network interworking, UDP, user plane, voice services, voice trunking, VoIP.

FOREWORD

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CONTENTS

Introduction

This Recommendation specifies the required functions and procedures necessary for support of narrow-band voice services by IP networks. Narrow-band voice services include digital audio streams, telephony call progress tones, facsimile, and optionally, circuit mode data. Details of the encapsulation of encoded audio streams are specified.

ITU-T Recommendation Y.1452

Voice trunking over IP networks

1 Scope

This Recommendation specifies the required functions and procedures necessary for support of narrow-band voice services by IP networks.

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

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

This Recommendation may not be suitable for use by recognized operating agencies (ROA).

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.1411 (2003), *ATM-MPLS network interworking – Cell mode user plane interworking*.

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

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

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

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

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

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

[8] 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)*.

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

[10] ITU-T Recommendation G.722.1 (2005), *Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss*.

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

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

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

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

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

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

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

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

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

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

[21] ITU-T Recommendation T.30 (2005), *Procedures for document facsimile transmission in the general switched telephone network*.

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

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

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

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

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

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

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

[29] IETF RFC 791 (1981), *Internet Protocol*.

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

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

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

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

[34] ATM Forum specification af-vmoa-0145.001 (2003), *Loop Emulation Service Using AAL 2 Rev 1*.

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

[36] ITU-T Recommendation I.366.1 (1998), *Segmentation and Reassembly Service Specific Convergence Sublayer for the AAL type 2*.

[37] IETF RFC 2508 (1999), *Compressing IP/UDP/RTP Headers for Low-Speed Serial Links*.

[38] IETF RFC 2507 (1999), *IP Header Compression*.

[39] IETF RFC 3095 (2001), *Robust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed*.

[40] ITU-T Recommendation X.800 (1991), *Security architecture for Open Systems Interconnection for CCITT Applications*.

3 Definitions

This Recommendation uses or defines the following terms:

3.1 interworking: See ITU-T Rec. Y.1411 [1].

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

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

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

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms.

AAL 2	ATM Adaptation Layer type 2
AMR	Adaptive Multi-Rate
CAC	Connection Admission Control
CAS	Channel Associated Signalling
CCS	Common Channel Signalling
CID	Channel Identifier
CLI	Calling Line Identification
COT	COnTinuity signal
CPS	Common Part Sub-layer (for AAL)
CPT	Call Progress Tone
Diffserv	Differentiated Services
DTMF	Dual Tone Multi-Frequency
EF PHB	Expedited Forwarding Per Hop Behaviour
GS	Guaranteed Service
HEC	Header Error Control
Intserv	Integrated Services
IP	Internet Protocol
ISDN	Integrated Services Digital Network
IWF	InterWorking Function
LES	Loop Emulation Service
MTU	Maximum Transport Unit
OAM	Operation, Administration and Maintenance
PDU	Protocol Data Unit
PLC	Packet Loss Concealment

PSTN	Public Switched Telephone Network
QoS	Quality of Service
RFC	Request for Comments
ROA	Recognized Operating Agency
ROHC	Robust Header Compression
SSCS	Service-Specific Convergence Sublayer (of AAL)
TDM	Time Division Multiplex
TFP	Termination Flow Point
TTL	Time To Live
UDP	User Datagram Protocol
UUI	User-to-User Information
VoDSL	Voice over DSL
VoIP	Voice over IP
VToIP	Voice Trunking over IP

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, facsimile and modem transmissions, etc.) and 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 Time Division Multiplex (TDM) link carrying multiple voice-grade channels or an IP link carrying multiple VoIP streams. In this Recommendation, the term "voice trunking" signifies transport of multiple voice channels using a single IP flow. "VoIP trunking" refers to the special case of transport of multiple VoIP packets using a single IP flow.

6 Voice trunking over IP networks

The following figures present the Reference Architecture for voice trunking over IP (VToIP). The functionality described in this Recommendation is implemented in IWFs, which receive multiple voice channels, generally from TDM end-systems, from the PSTN/ISDN, or from VoIP connections. The IWFs multiplex the voice channels, and forward them over IP networks. Figure 6-1 depicts voice services originating and terminating in a TDM end-system. This same situation is shown using G.809 [2] diagrammatic techniques in Figure 6-2. Figure 6-3 depicts the case of VoIP trunking.

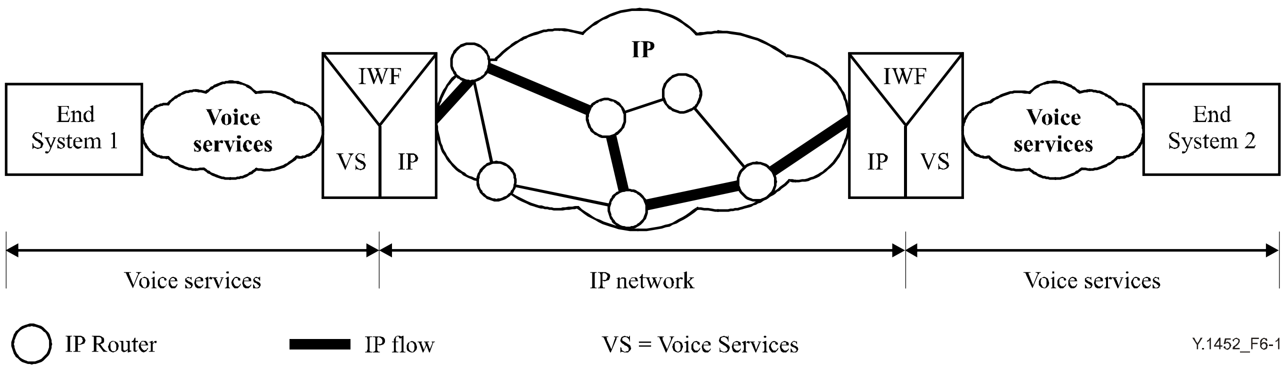

Figure 6-1/Y.1452 – Reference architecture for voice trunking over IP

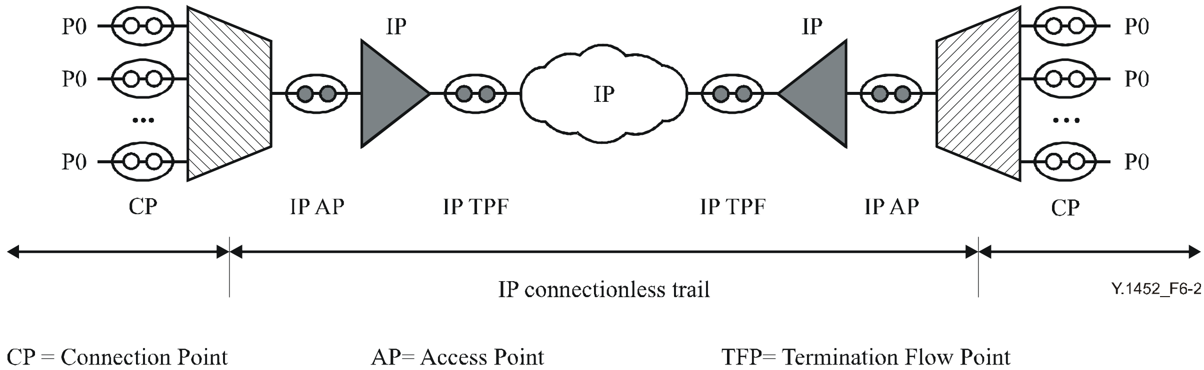

Figure 6-2/Y.1452 – Functional architecture for multiplexing P0 channels over IP networks

NOTE – Figure 6-2 represents one direction of the bidirectional voice session.

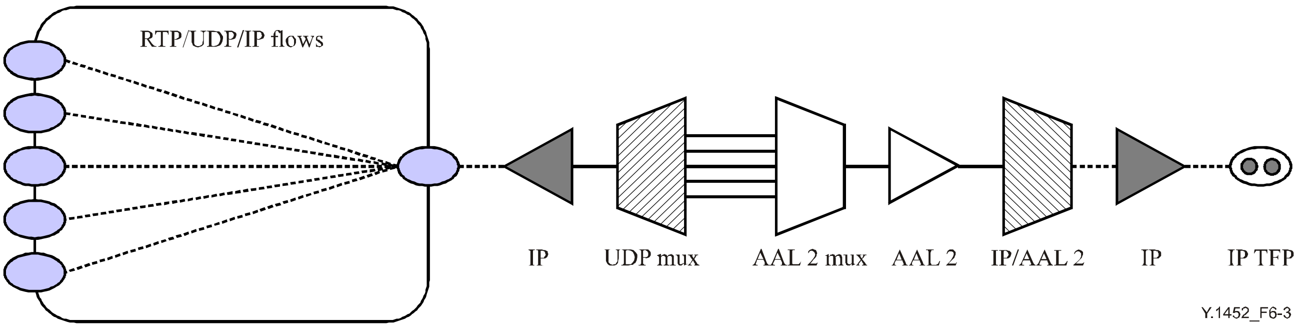

Figure 6-3/Y.1452 – Functional architecture for multiplexing VoIP streams over IP networks

7 General requirements

7.1 User plane requirements

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

a) The ability to encapsulate data from a telephony grade channel in an IP packet.

b) The ability to transport telephony grade audio encoded according to ITU-T Recs G.711 [3], G.723.1 [4], G.726 [5], G.727 [6], G.729 [7] and adaptive multi-rate (AMR) [8] coders.

c) Optional ability to encapsulate wideband speech.

d) Optional ability to transport wideband speech encoded according to ITU-T Recs G.722 [9], G.722.1 [10] and G.722.2 [11] coders.

e) The ability to encapsulate multiple VoIP streams in a single IP packet.

f) The ability to reliably detect packet loss, in order to support packet loss concealment (PLC) by suitable PLC algorithms, such as Appendix I/G.711 [12].

g) The ability to transfer subscriber signalling, such as dual tone multi-frequency (DTMF) according to ITU-T Recs Q.23 [13] and Q.24 [14], call progress tones (CPT) according to ITU-T Rec. E.180/Q.35 [15], and calling line identification (CLI) [16] either in the audio stream, or by suitable relay.

h) The ability to transfer inter-office signalling systems R1 according to ITU-T Rec. Q.310-332 [17], R2 according to ITU-T Rec. Q.400-490 [18], and continuity signal (COT) as defined in ITU-T Rec. Q.724 [19], either in the audio stream, or by suitable relay.

i) The ability to acquire, encapsulate and transfer channel associated signalling (CAS) bits.

j) Optional support for transport of 64 kbit/s clear-channel data, in particular for common channel signalling (CCS).

k) The ability to transfer standards-based facsimile (ITU-T Recs T.4 [20], T.30 [21], V.17 [22], and V.29 [23]), telephone mode text (ITU-T Rec. V.18 [24]), and voice-grade modem signals (V series modems), either in the audio stream (when remote timing allows), or by suitable relay.

l) Optional support for interworking with ATM-based AAL type 2 services, in particular IMT-2000 cellular systems, loop emulation service (LES) and voice over DSL (VoDSL).

m) The ability to exploit the entire maximum transport unit (MTU).

7.2 Management plane aspects

For transfer of voice services, the following need to be provisioned:

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

b) Interface type (analogue, TDM or VoIP).

c) Voice channel parameters (e.g., bandwidth, frame duration).

d) Audio encoding method (e.g., G.711, G.723.1, G.726, G.727, G.729, AMR, G.722, G.722.1, G.722.2) and encoding-dependent parameters.

7.3 Fault management aspects

As individual voice channels do not carry defect indications, there are no fault management aspects.

7.4 Traffic management aspects

The IP network shall be capable of providing the required QoS for all voice channels, and shall be capable of meeting the aggregate bandwidth requirements of all voice channels transported.

If the IP network is Diffserv enabled according to RFC 2474 [25], then Expedited Forwarding Per Hop Behaviour (EF PHB) per RFC 3246 [26] 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 [27], then Guaranteed Service (GS) per RFC 2212 [28] with the appropriate bandwidth reservation shall be used in order to provide a bandwidth guarantee equal to or greater than that of the aggregate voice traffic.

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

7.5 Connection admission control for the IWF

When bandwidth guarantees can be provided, then the IWF should provide connection admission control (CAC). The admission decision shall be based on the total available bandwidth, the bandwidth presently being used, and the bandwidth requested. When sufficient bandwidth is available the request may be granted. When bandwidth is insufficient the connection request shall be denied.

7.6 Congestion control

When a network becomes congested, the traditional recourse is either to reroute the service in order to avoid the congested links, or to withdraw the service. In many cases, the former option is not available due to the provider of the service lacking the required control over the underlying IP network. The remaining option, that of service removal, is usually an unacceptable alternative for voice trunking applications. Not only does service removal impact a large number of users, its impression on users accustomed to high availability would be highly detrimental to the service's (and the service provider's) image.

In many cases, the voice services flow can statistically co-exist with other traffic, and congestion is due solely to temporary load peaks.

Since voice trunking will usually be combined with signal processing such as silence suppression and voice compression, the trunk may consume less bandwidth, and hence its effect on neighbouring flows is minimized. Moreover, when congestion is detected, there are several bandwidth conservation options available that facilitate further reduction. For example, speech compression could be enabled or a stronger compression can be chosen. Hence, these would most likely help to avoid withdrawal of the voice trunking service. The additional compression may indeed be perceivable to users, but would undoubtedly be considered much more acceptable than service outages. Once the congestion clears the original service emulation characteristics can be restored.

8 Functional group considerations for VToIP network interworking

Figure 8-1 provides an illustration of functional grouping for VToIP.

IP
UDP
Common interworking indicators
VToIP Payload

Figure 8-1/Y.1452 – VToIP functional groups

8.1 IP

This field is the standard IPv4 [29] or IPv6 [30] header.

8.2 UDP

Since it may be required to transport multiple flows between two IP addresses, a method of labelling VToIP 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 [31]. When the source port field is used, the destination port field may contain an identifier indicating that the packet contains voice trunking 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 length field, and a sequence number field, as depicted in Figure 8-2.

<table>
<tr><td colspan="8" align="center">Bit</td></tr>
<tr><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 colspan="8" align="center">Control</td></tr>
<tr><td colspan="2" align="center">Reserved</td><td colspan="6" align="center">Length</td></tr>
<tr><td colspan="8" align="center">Sequence number (2 octets)</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 8-2/Y.1452 – Common interworking indicators

8.3.1 Control field

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

<table>
<tr><td colspan="8" align="center">Bit</td></tr>
<tr><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 colspan="4" align="center">Reserved</td><td align="center">L</td><td colspan="3" align="center">Reserved</td></tr>
</table>

NOTE – Bit 8 is the most significant bit.

Figure 8-3/Y.1452 – Control field

The reserved fields shall be set to zero.

The L field provides a means of transparent transfer of defect indications between IWFs, when the voice channels are derived from a TDM interface. Their use should be in accordance with principles of the appropriate G-series of Recommendations with regard to operation, administration and maintenance (OAM).

L (Local failure): The L bit being set indicates that the ingress IWF has detected or has been informed of a defect impacting the input 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 fault is rectified the L bit shall be cleared.

8.3.2 Length field

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

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

a) size of the Common interworking indicators; and

b) size of the payload;

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

8.3.3 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.

8.3.3.1 Setting the sequence numbers

The following procedures apply at the ingress IWF (voice channel-to-IP direction):

- The sequence number should be set to a random value for the first IP packet transmitted on the interworking flow.
- For each subsequent IP 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 packets. Misordered packets should be re-ordered if possible. The mechanism by which a packet is considered lost is implementation specific.

The following procedures apply at the egress IWF (IP-to-voice channel 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.

9 Payload format

Voice trunking over IP payloads consist of one or more variable-length AAL type 2 common part sublayer (CPS) packets, as described in ITU-T Rec. I.363.2 [32]. Each AAL type 2 CPS packet contains 3 bytes of CPS header and between 1 and 64 bytes of channel payload. If interworking with ATM-based AAL type 2 systems is required [33], the channel payload size may be restricted to be smaller than 64 octets (typically 45 or 44 octets). An IP packet may be constructed by inserting CPS packets corresponding to all active voice channels, by appending CPS packets ready at a certain time, or by any other means.

The channel payload may consist of raw voice frames or VoIP packets, the latter being described in clause 11 below.

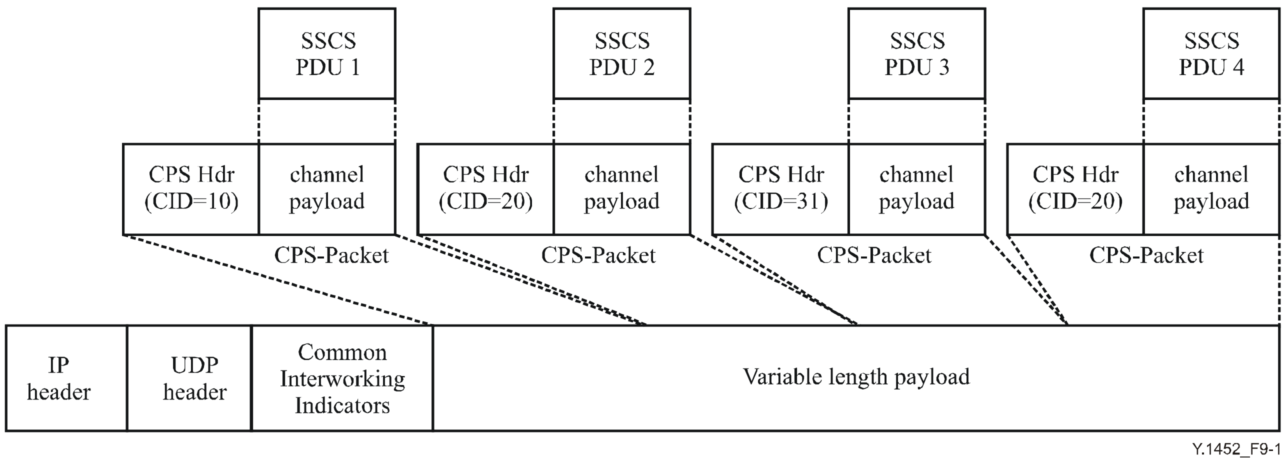

Figure 9-1/Y.1452 – Packing of CPS packets into an IP packet

Each IP packet will consist of an UDP/IP header, Common interworking indicators, and one or more complete CPS packets, as shown in Figure 9-1. The maximum number of CPS packets per IP packet is determined by the MTU of the IP network. A single IP 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 channel identifier (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 voice over DSL (VoDSL) is required, then CID = 8 through CID = 15 are used for special purposes as specified in af-vmoa-0145.001 [34]. The same CID value may appear multiple times in a single IP packet. When this occurs, order shall be maintained.

10 Encapsulation format

The complete VToIP packet structure is as depicted in Figure 10-1.

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	Reserved			29
FRAG		Length						30
Sequence number								31-32
Adapted payload								33-n

NOTE – Bit 8 is the most significant bit.

Figure 10-1/Y.1452 – Encapsulation format

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.

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

Indicates the IP fragmentation identification field per RFC 791 [29].

Flags, Octet 7, bits 8 through 6

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

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

Indicates where in the datagram the fragment belongs. This field is not used in this Recommendation.

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 source of the stream being transported. The UDP flow shall be manually configured.

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

When used as a stream identifier, the UDP port number shall be chosen from the range of dynamically allocated UDP ports numbers (49152 through 65535).

The choice of whether the source port field or destination port field is used as 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 adapted payload.

UDP Checksum, Octets 27 and 28

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

Reserved, Octet 29, bits 8 through 5 and bits 3 through 1

Indicate reserved fields, both of which shall be set to zero.

L, Octet 29, bit 4

See 8.3.1.

FRAG, Octet 30, bits 8 and 7

Indicates fragmentation and is set to "00" to indicate there is no fragmentation.

Length, Octet 30, bits 6 through 1

See 8.3.2.

Sequence number, Octets 31 and 32

See 8.3.3.

11 Aggregating VoIP streams

In certain applications, point-to-point transport of multiple VoIP streams is required. VoIP is carried using RTP, according to RFC 3550 [35]. Running multiple conventional VoIP streams in parallel is highly inefficient, as the RTP/UDP/IP header overhead may exceed the size of the voice payload. Combining several consecutive payloads of each channel into a single packet incurs additional latency, since in this case IP header compression schemes cannot function over IP networks.

Similar to the standard voice trunking application, one can combine the contents of multiple VoIP streams into a single packet, thus incurring only a single header overhead for large numbers of channels. See Figure 11-1.

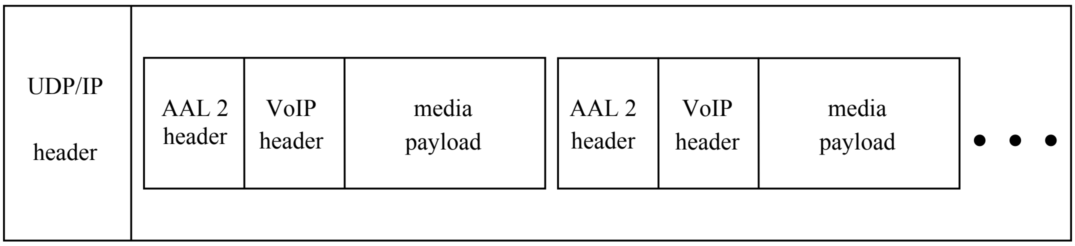

Figure 11-1/Y.1452 – Encapsulation format for VoIP trunking

In certain cases (e.g., 10 milliseconds of uncompressed G.711 voice) the size of the VoIP header plus media payload may exceed the 64-octet maximum size of a CPS packet. In such a case, the user-to-user information (UUI) field in the AAL 2 header is used to indicate fragmentation, as discussed in ITU-T Rec. I.366.1 [36].

In order to further conserve bandwidth, compressed header mechanisms may also be utilized. Compressed RTP as described in RFC 2508 [37] requires a link layer capable of providing an indication of four special packet formats in addition to uncompressed IPv4 and IPv6 formats. Other header compression schemes such as IP Header Compression as described in RFC 2507 [38] and robust header compression (ROHC) described in RFC 3095 [39] expand the scheme to other IP packet types, but require additional discrimination. These indications are conveyed via the UUI field in the AAL 2 CPS header, according to Table 11-1. All values not listed in the figure are reserved.

Table 11-1/Y.1452 – Use of UUI field for VoIP trunking

UUI value	Meaning
0	No header – pure voice channel payload, final packet
1	Uncompressed header, final packet
2	RFC 2507 compressed TCP
3	RFC 2507 compressed TCP_NODELTA
4	RFC 2507 compressed NON_TCP
5	RFC 2508 compressed RTP with 8-bit CID
6	RFC 2508 compressed RTP with 16-bit CID
7	RFC 2508 compressed UDP with 8-bit CID
8	RFC 2508 compressed UDP with 16-bit CID
9	RFC 2507/2508 context state packet
10	RFC 3095 ROHC compression
27	Non-terminal packet

NOTE – Although some compressed header protocols have their own CID fields (8-bit or 16-bit), these should not be confused with the 8-bit AAL type 2 CID.

12 Security considerations

This Recommendation does not recommend the invocation of any of the security services identified in ITU-T Rec. X.800 [40].

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