Reference Specification

for

Internet Protocol (IP) and Circuit Switched (CSN) Networks Interconnection

IDA RS IP-CSN INTC
Issue 1, 1 June 2002

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Contents

NOTICE
This Guide is subject to review and revision.

1. Scope

1.1 This Reference Specification defines the requirements for IP telephony[1] to interwork with PSTN telephony (section 3), and provides a brief description on the generic gateway architecture for interworking (section 4).

1.2 The Reference Specification also outlines the technical requirements for connection of VoIP[2] terminal equipment to IP networks[3] (section 5).

1.3 Technical requirements for the following items are understudy:

a) Interworking between H.323 and SIP networks

b) SIP for mobile networks

1.4 References

ITU-T Rec. Y.140 (11/2000)	Global Information Infrastructure (GII): Reference points for interconnection framework
ITU-T Rec. Y.1001 (11/2000)	IP Framework – A framework for convergence of Telecommunications Network and IP Network technologies
ITU-T Rec. G.177 (09/99)	Transmission planning for voiceband services over hybrid Internet/PSTN connections
ITU-T Rec. H.323 (11/2000)	Packet-based Multimedia Communications Systems
ITU-T Rec. H.225.0 (11/2000)	Call signalling protocols and media stream packetization for packet-based multimedia communications systems
ITU-T Rec. H.245 (07/2001)	Control protocol for multimedia communication
ITU-T Rec. H.235 (11/2000)	Security and encryption for H-series (H.323 and other H.245 based) multimedia terminals
ITU-T Rec. H.248 (06/2000)	Gateway control protocol
ITU-T Rec. E.161	Arrangement of digits, letters and symbols on telephones and

1 "IP telephony" is used as a generic term for the conveyance of voice, fax and related services, partially or wholly over packet-switched IP-based networks. IP telephony is used interchangeably with "VoIP" (Voice over Internet Protocol). Also, "Internet telephony" is used to refer to IP telephony or VoIP conveyed partially or wholly over the Internet.

2 Refer to notes given in 1 for VoIP.

3 Refer to the notes given in Figure 1 of this Specification.

(02/2001)	other devices that can be used for gaining access to a telephone network
ITU-T Rec. Q.23 (1993)	Technical features of push-button telephone sets
ITU-T Rec. E.164 (05/97)	The international public telecommunication numbering plan
ITU-T Rec. G.114 (02/96)	One-way transmission time
ITU-T Rec. G.711 (1993)	Pulse code modulation (PCM) of voice frequencies
ITU-T Rec. P.310 (05/2000)	Transmission characteristics for telephone band (300-3400 Hz) digital telephones
ITU-T Rec. T.120 (07/96)	Data protocols for multimedia conferencing
ETSI TS 101 471 v.1.1.1 (2000-12)	TIPHON; Signalling for calls between H.323 terminals and terminals in a Switched-Circuit Network (SCN) Phase III: Scenarios 1, 2, 3 and 4
ETSI TS 101 520 v.1.1.1 (2000-09)	TIPHON; Implementation Conformance Statement (ICS) proforma for the support of packet based multimedia communication systems; Support of ITU-T Recommendation H.323
TIA/EIA/IS 811 July 2000	Telecommunications – Telephone Terminal Equipment – Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones
IEC 60950: 1999	International Electrotechnical Commission – Safety of Information Technology Equipment
IDA TS EMC March 2000	EMC Requirements for Telecommunication Equipment
Draft IDA Guide for GII (under discussion)	Guide for Global Information Infrastructure (GII) Standards (Title of Guide is subject to approval)

1.5 Abbreviations

DHCP	Dynamic Host Configuration Protocol (IETF RFC2131)
DSL	Digital Subscriber Line
DTMF	Dual-Tone Multi-Frequency
IPv4	Internal Protocol, version 4 (IETF RFC791)

ISDN	Integrated Services Digital Network
MEGACO/H.248	Media Gateway Control Protocol (IETF RFC2885) ITU-T has approved ITU-T Rec. H.248, which is identical to IETF RFC2885.
PPP	Point-to-Point Protocol (IETF RFC1661)
PSTN	Public Switched Telephone Network
RAS	Registration, Admission and Status
RSVP	Resource Reservation Protocol (IETF RFC2205)
RTP	Transport Protocol for Real-time Applications (IETF RFC1889)
RTCP	Real-time Transport Control Protocol
SIP	Session Initiation Protocol (IETF RFC2327)
SMDS	Switched Multi-megabit Data Service
SS7	ITU-T Signalling System Number 7
TCP	Transmission Control Protocol (IETF RFC793)
UDP	User Datagram Protocol (IETF RFC768)

2. Background

2.1 With rapid growth of the Internet, the ITU-T standards development takes on new emphasis. There is a need to define interconnection and interoperability between IP-based services[4] and telecommunications services, and to specify for the real time IP-based multimedia services performance parameters, which ought to be similar to the speed, capacity, ease of use, reliability and integrity of the public telephone networks. Besides interconnection and interoperability issues, other network issues and solutions associated with IP technology such as numbering, signalling protocols and security are proposed.

2.2 The ITU-T standards development relating to the Internet and IP-based networks[5] is related closely with the development of the Global Information Infrastructure (GII). The GII has been adopted as the platform for facilitating the development, implementation and interoperability of existing and future information services and applications within and across telecommunications, information technology and consumer electronics and content provider industries. GII principles, requirements, framework and scenarios are reviewed in the wider context of convergence between the telecommunications industry, the information technology industry and the entertainment/consumer electronics industry, under the "IDA Guide for the GII Standards".

[4] An IP-based service is defined as the functions, facilities and capacities implemented and executed above IP network services. It utilizes the IP transfer capability offered by a network provider.

An IP network service is defined as a data transmission service in which the data passed across the interface between the user and the provider is transferred in the form of IP packets (datagrams). IP network service includes the service provided by using the IP transfer capabilities.

IP transfer capability is defined as the set of network capabilities provided by the IP layer. It may be characterised by the traffic contract as well as performance attributes supported by control and management functions of the underlying protocol layers. An example of IP transfer capability is the basic best effort IP packet delivery.

[5] IP-based network is a network in which Internet Protocol (IP) is used as one of the layer 3 protocols.

3. Telephony Inter-working

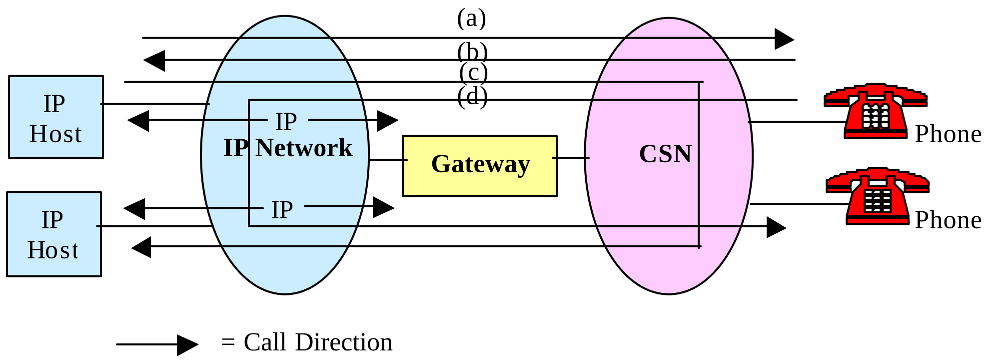
3.1 VoIP and Telephony Services

3.1.1 Four scenarios are used to illustrate the interworking of PSTN services (e.g. a telephone call) with IP-based services through the World Wide Web. Examples of such services are Click-to-Dial[6], Click-to-Fax[7] and Voice access to content[8]. Each of these scenarios requires at least one usage of the gateway.

a) A voice call from an IP terminal connected to an IP network to a PSTN phone
b) A voice call from a PSTN phone to an IP terminal connected to an IP network
c) A voice call from an IP terminal connected to an IP network to another IP terminal connected to an IP network via the PSTN
d) A voice call from a PSTN phone to another PSTN phone via an IP network

3.1.2 These four scenarios are described in the ITU-T framework for convergence of telecommunications network and IP network technologies (ITU-T Rec. Y.1001). The focus of ITU-T work is on interworking that is achieved at the service level via an intermediate gateway for the provision of an end-to-end voice service. There is no recommendation for an end-to-end protocol, and the IP stack is terminated at the gateway as shown in Figure 1.

[6] A Web user can initiate a PSTN call by clicking a button during a Web session. The call can either be outgoing or incoming because a user while browsing through a catalogue may click a button inviting a call from the sales office.
[7] A Web user can request a fax (and subsequently receive one) by clicking a button during a Web session.
[8] A Web user can have access to the Web content by telephone. The content is converted to speech and transmitted to the user on a telephone line.



Note: IP Network[9] – A network in which IP is used as a network layer protocol. CSN – Circuit Switched Network in which a fixed bandwidth channel is established for and dedicated to the duration of a communication session. The PSTN is an example of CSN where a circuit is established for the duration of a telephone call.

Figure 1 (based on Figure 12/Y.1001 and Figure 1/G.177):
Telephony service – basic inter-working architecture

3.2 Inter-working functions

3.2.1 Call connection procedure

IP host refers to an IP terminal equipment,[10] which either supports the ITU-T Rec. H.323, the IETF SIP or the MEGACO/H.248, connected to the IP network via a direct connection (e.g. Ethernet, Token Ring, etc.) or a dial-up connection (e.g. modem and PPP link). In scenarios (a) and (b) of Figure 1, functions are required for converting the call connection procedure of the IP network to that of the CSN and vice versa, and for establishing the end-to-end connection between the IP host and PSTN phone. These functions may be implemented in adapters attached to the IP host and/or in the gateway. In scenarios (c) and (d) of Figure 1, these functions may not be required because the call control protocol may be transmitted transparently across the IP network or the PSTN/ISDN CSN.

[9] The term "IP Network" is distinct from the term "Internet". Many IP networks exist, each operated by different owners. IP networks may be globally public (i.e. the Internet), totally private (i.e. with no open structure and without gateways to the Internet and other private IP networks) or combinations of public and private networks (e.g. a privately run IP network with gateways and access to the Internet, but not necessarily vice versa).

[10] An IP terminal equipment can be either dedicated (e.g. a telephone set) or general in purpose (e.g. a computer running an application that performs the functionality defined in either ITU-T Rec. H.323, IETF SIP or MEGACO/H.248 from the point of view of speech transmission).

3.2.2 Numbering and addressing

In scenario (a) of Figure 1, the address resolution functions are required to convert the IP network address (translate to ITU-T Rec. E.164 addresses) to that required by the CSN for designating the called terminal, and vice versa in scenario (b) of Figure 1.

3.2.3 Voice coding methods

In each scenario, functions are required for converting the voice encoding methods used for voice services over an IP network to those used for analogue voice over the PSTN or to those used for voice encoding over ISDN and digital trunk sections of PSTN, and vice versa. For performance to be on par with the PSTN/ISDN and the digital wireless networks, the IP terminal equipment shall support the ITU-T G.711 (μ-law and A-law) audio codec. Other codecs such as the ITU-T G.723 and the ITU-T G.729 may be supported.

3.2.4 Quality of Service (QoS) for IP services

Specifications for various classes of IP service, characterised by different qualities of service will be required in order to select IP services suitable for specific types of application. Voice and other real-time sensitive applications are examples of applications which have specific QoS requirements. The specifications will include values for transmission delay and packet loss and other relevant QoS parameters.

3.2.5 Factors influencing end-to-end speech quality of VoIP services

a) <u>Speech coder performance</u>
Generally, tandeming of speech codecs will bring about degradation in speech quality. The objective therefore is to remove tandem processing by ensuring that all systems (IP/PSTN gateway and wireless terminals) use the same speech coder.

b) <u>Transmission errors and packet loss</u>
While errors in transmission will be dealt with by higher level protocols, packet loss due to transmission delay will have to be compromised by a trade off between long delay and dropped packets.

c) <u>Loudness ratings</u>
If telephone handsets are used, they should meet the requirements of ITU-T Rec. P.310.

d) <u>Delay and echo</u>
One-way delays should not exceed 300 ms (refer to ITU-T Rec. G.114, Annex B for effects of long transmission delays on the subscriber). Also, echo control is required in the hybrid Internet/PSTN connections. All terminals should meet the Weighted Terminal Coupling Loss (TCL_W) objective of 45 dB, as specified for digital telephones in ITU-T Rec. P.310. Gateway should provide echo cancellation. At minimum, echo cancellers deployed in the gateway should meet requirements for digital network echo cancellers given in ITU-T Rec. G.168.

e) <u>Temporal (syllable) clipping</u>
To maintain good speech quality, clipping of speech segments $\geq$ 64 ms should always be avoided, and clipped segments < 64 ms should be kept below 0.2 percent of active speech (ITU-T Rec. G.116).

f) <u>Idle channel noise</u>
Background idle channel noise if present should be less than –68 dBm0p (ITU-T Rec. G.106).

g) <u>Bandwidth</u>
To maintain good speech quality ands intelligibility, a minimum pass-band of 300 – 3400 Hz (3 dB points) should be delivered.

h) <u>Stability loss</u>
For VoIP systems interfacing digitally to the PSTN, a minimum loss of 6 dB is recommended between the digital input and output paths of the VoIP system at the access port of the terminal.

3.2.6 Security

Specification for the provision of security in the IP network will be required. All H.323-based terminals shall adopt the procedures given in ITU-T Rec. H.235 for user authentication and message integrity. Media stream encryption is optional. However, if implemented, the procedures defined in H.323 Annex J/6.3 shall be supported.

4. Inter-working Gateway Architecture

4.1 The generic architecture of a gateway contains four functional components but may not be treated as a single entity depending on the scale and the nature of the multi-vendor equipment. The four components are:

a) Media Gateway (MG);

b) Media Gateway Controller (MGC);

c) Signalling Gateway (SG); and

d) Intelligent Databases (ID).

4.2 The MG will handle the real-time operations like terminating the circuit switched network (CSN) facilities (trunks, loops), handling of media streams and delivery of packets to the IP network. It performs these functions in the reverse order for media streams flowing from the IP network to the CSN. The MG will interface with the CSN as per PSTN/ISDN requirements and interface with the IP network using RTP for transmission.

4.3 The MGC controls the operation of the MG function. Interconnection between the MGC and the MG will be over an IP network. The MGC interfaces to the IP network for the purposes of conveying call control information to peers on the IP network using either SIP or protocols defined in the ITU-T Rec. H.323. An MGC-MG control protocol is required and one candidate for this is defined in ITU-T Rec. H.248 or the IETF equivalent, the MGCP (MEGACO/H.248).

4.4 The SG will handle any necessary interactions with the SS7 network. The SG is a signalling agent that receives/sends CSN native signalling at the border between the IP network and telecommunications network. In particular the SG function may relay, translate or terminate SS7 signalling in an SS7-Internet Gateway.

4.5 The ID will be used to acquire information required, for example, for credit card usage, toll-free services, directory services, etc.

5. VoIP Terminal Equipment Requirements

5.1 Common Characteristics of VoIP Terminal Equipment

The VoIP terminal equipment shall support the following common characteristics:

a) The VoIP terminal equipment shall have an audio codec capable of encoding and decoding speech according to ITU-T Rec. G.711 and capable of transmitting and receiving A-law and μ-law. It may support other codecs (ITU-T Rec. G.723 and G.729).

b) The VoIP terminal equipment shall minimally provide an alphanumeric keypad function with 12 pushbuttons following Option A and the standard 4x3 array given in ITU-T Rec. 161. It shall have the option to generate DTMF tones according to frequencies allocated to digits and symbols of the pushbutton set as shown in ITU-T Rec. Q.23.

c) If the VoIP terminal equipment has a handset, it shall comply with the requirements of ITU-T Rec. P.310.

d) The VoIP terminal equipment shall include an Ethernet connection conforming to the IEEE Std. 802.3, 10BASE-T and/or 100BASE-T interface standards.

e) The VoIP terminal equipment shall support UDP and RTP.

f) The VoIP terminal equipment shall support the DHCP.

g) The VoIP terminal equipment shall support IPv4. However, it may implement the IPv6. If that's the case, it shall also implement the Transition Mechanisms for IPv6 Hosts and Routers (RFC1933) for backward compatibility with the IPv4.

h) The VoIP terminal equipment shall be designed according to the safety requirements of the IEC 60950 and the electromagnetic compatibility requirements defined in IDA TS EMC.

5.2 VoIP Terminal Equipment using H.323

If the VoIP terminal equipment is an H.323-based terminal, besides supporting the common characteristics listed in section 5.1, it shall also be defined by the following characteristics:

a) The video codec is optional. However, if capability is provided, it shall comply with requirements given in ITU-T Rec. H.323.

b) Data channels are optional. However, ITU-T Rec. T.120 is the default basis for data interoperability between the H.323-based terminal and other H.323, H.324, H.320 or H.310 terminals.

c) It shall have the H.245 control channel to carry end-to-end control messages for the proper operation of the H.323-based terminals.

d) It shall have the RAS channel that uses H.225.0 messages to perform registration, admission, change to bandwidth, status and disengage procedures.

e) It shall have the call signalling channel that uses H.225.0 call signalling (implementing the Q.931 messages as described in H.225.0/§ 7.3) to establish connection between two H.323-based terminals.

f) It shall support the transmission of Keypad facility Information Element in the Information message for call establishment and call-related information as described in H.225.0/§ 7.3.6.

g) It shall support logical channels according to format defined in ITU-T Rec. H.225.0.

h) It shall support the protocol stack for the H.225.0 call signalling and the H.245 control protocols as shown in Figure 2 (Figure 1/H.225.0).

i) As an H.323-based terminal, it supports IPv4, and uses reliable transport (TCP) for the H.245 control channel, the T.120 data channels and the call signalling channel. However, it uses unreliable transport (UDP) with RTP/RTCP for handling audio and video streaming, and the RAS channel.

j) It may support quality of service using RSVP protocol as defined in H.323/Appendix II.

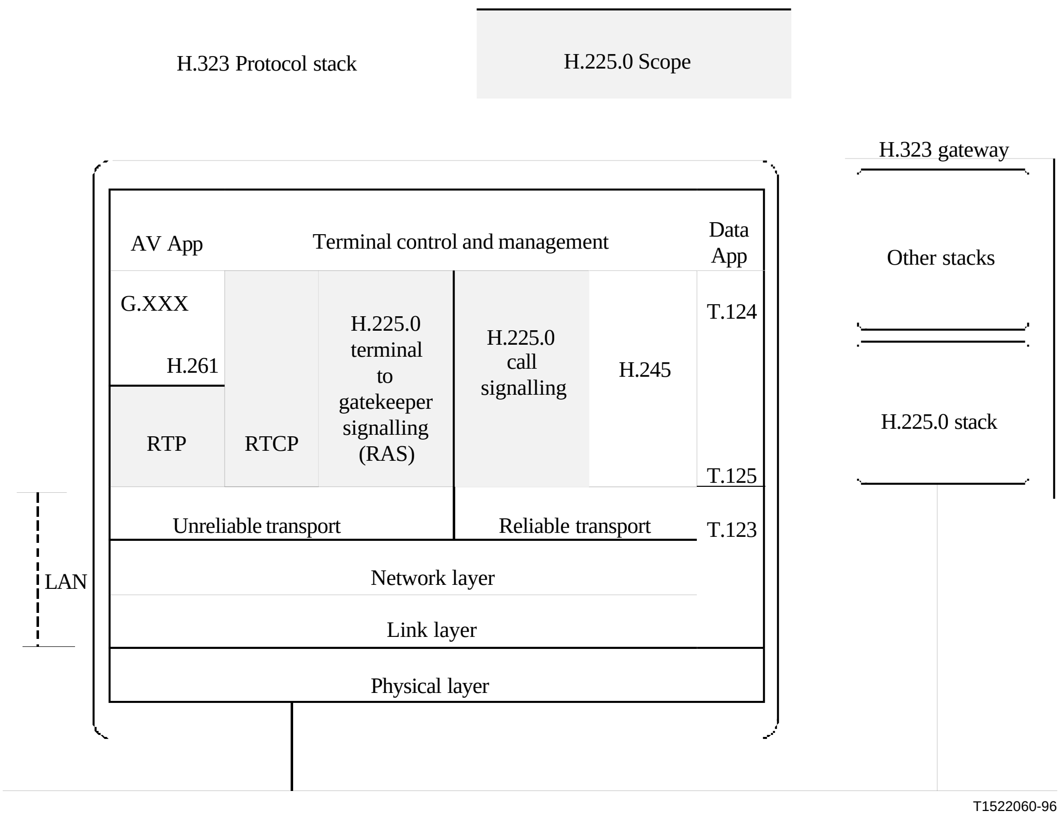


Figure 2 (Figure 1/H.225.0): **H.225.0 Scope**

5.3 VoIP Terminal Equipment using MEGACO/H.248

If the VoIP terminal equipment is an MEGACO/H.248 based terminal, besides supporting the common characteristics listed in section 5.1, it shall also support IETF MEGACO IP Phone Media Gateway standard.

5.4 VoIP Terminal Equipment using SIP

If the VoIP terminal equipment is SIP based, besides supporting the common characteristics listed in section 5.1, it shall support the functionality for a Redirectional Capable Client, and may support the functionality for an Authentication Capable Client, as defined in § A.1 of the IETF Session Initiation Protocol (RFC2543).