INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

# Series Q

## Supplement 49

(03/2004)

SERIES Q: SWITCHING AND SIGNALLING

# Technical Report TRQ.2840: Signalling requirements to support IP telephony

ITU-T Q-series Recommendations – Supplement 49

ITU-T Q-SERIES RECOMMENDATIONS

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

## Supplement 49 to ITU-T Q-series Recommendations

# Technical Report TRQ.2840: Signalling requirements to support IP telephony


**Summary**

This Supplement to the Q series of ITU-T Recommendations consists of a Technical Report that specifies the signalling requirements to support IP telephony. IP telephony is defined as a service enabling the exchange of voice information primarily in the form of packets using Internet protocols (IP), while Internet telephony is defined as a particular application of the Internet, such that Internet telephony falls outside the scope of this Supplement. Network configurations are classified into four types (phone-to-phone; IP phone-to-phone; phone-to-IP phone; IP phone-to-IP phone) and the characteristics of each configuration are described using a general example. The network capabilities needed to support IP telephony and interworking with legacy telephone networks are identified in this Supplement. Finally, the signalling requirements and control protocols needed to support IP telephony services in public networks are identified.



**Source**

Supplement 49 to ITU-T Q-series Recommendations was agreed on 12 March 2004 by ITU-T Study Group 11 (2001-2004).

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

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# Supplement 49 to ITU-T Q-series Recommendations

## Technical Report TRQ.2840: Signalling requirements to support IP telephony

## 1 Scope

This Supplement identifies the network capabilities that must be satisfied to enable the support of IP telephony as well as interworking with legacy telephony networks. It identifies the relevant network configurations, the network functions required to support the IP telephony feature, and the protocol stacks.

## 2 Introduction

The traffic characteristics of IP-based applications, and voiceband data communications generally, are significantly different from those of traditional speech communications upon which the engineering of the Public Switched Telephone Network (PSTN) has historically been based. Transaction-oriented traffic is handled inefficiently by the traditional PSTN, with the "call" establishment time being at least of the same, and often a greater, order of magnitude than the duration of the transactions themselves. This has particularly significance for "e-commerce" applications. WTSA-2000 created Question 7/11 to investigate approaches to signalling that efficiently directs these new traffic demands to (an) appropriately engineered network(s), while minimizing the potential for service degradation experienced by PSTN users.

Furthermore, at WTPF-2001 (World Telecommunication Policy Forum, 2001), four opinions were adopted regarding the global introduction of IP telephony services. One of these opinions expressed a need to study the technical issues associated with interworking and the coexistence of the PSTN and IP-based networks to provide telephony services.

This Supplement completes the study instigated by WTPF-2001 and explains the signalling needed to support IP telephony, provides example network configurations and identifies the capabilities that a network must have to support IP telephony.

## 3 References


[1] ITU-T Recommendations Q.761 to Q.764 (1999) – *Specifications of Signalling System No. 7 ISDN User Part (ISUP)*.

[2] ITU-T Recommendations Q.1902.1 to Q.1902.4 (2001) – *Specifications of the Bearer Independent Call Control (BICC) protocol*.

[3] IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.

[4] IETF RFC 2327 (1998), *SDP: Session Description Protocol*.

[5] IETF RFC 3262 (2002), *Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*.

[6] IETF RFC 3323 (2002), *A Privacy Mechanism for the Session Initiation Protocol (SIP)*.

[7] IETF RFC 3325 (2002), *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*.

[8] ITU-T Recommendation E.106 (2003), *International Emergency Preference Scheme (IEPS) for disaster relief operations*.

[9] ITU-T Recommendation E.164 (1997), *The international public telecommunication numbering plan*.

[10] ITU-T Recommendation E.370 (2001), *Service principles when public circuit-switched international telecommunication networks interwork with IP-based networks*.

[11] ITU-T Recommendation G.107 (2003), *The E-Model, a computational model for use in transmission planning*.

[12] ITU-T Recommendation H.248 (2000), *Gateway control protocol*.

[13] ITU-T Recommendation H.323 (2003), *Packet-based multimedia communications systems*.

[14] ITU-T Recommendations H.450.1 to H.450.12, *Specifications for the support of supplementary services in H.323*.

[15] ITU-T Recommendation I.250 (1988), *Definition of supplementary services*.

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

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

## 4 Abbreviations

DNS Domain Name System

ENUM tElephone NUmber Mapping

ETS Emergency Telecommunications Service

IEPS International Emergency Preference Scheme

NAT Network Address Translation

PSTN Public Switched Telephone Network

STP Signalling Transfer Point (SS No. 7 Signalling network)

## 5 Terms and definitions

**5.1 Media Gateway (MG)**: A media gateway converts the media provided by one type of network to the format required by another type of network. For example, an MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network). This gateway could be capable of processing audio, video and T.120 alone or in any combination, and would be capable of full-duplex media translations. The MG may also play audio/video messages and perform other IVR functions, or provide media conferencing. In the context of this Supplement, the term "media gateway" refers to a voice gateway.

**5.2 Media Gateway Controller (MGC)**: Controller that controls the parts of the call state that pertain to connection control for the media channels within a media gateway.

**5.3 Call Agent (CA)**: Function that controls the provision of services to users.

**5.4 tElephone NUmber Mapping (ENUM)**: Protocols for mapping telephone numbers to IP phone identifiers (i.e., E.164 numbers to URIs).

**5.5 IP network**: An IP network is a network that uses IP technologies to transport information. It may be a Private IP network, or a Carrier's network.

**5.6 phone**: Phone refers to a PSTN terminal.

**5.7 IP phone**: IP phone refers to a terminal (e.g., dedicated voice terminal or multipurpose personal computer) that is connected directly (e.g., through an Ethernet interface or an xDSL line) to an IP network.

**5.8 IP telephony**: IP telephony is a service that enables the exchange of voice information, primarily in the form of packets, using IP protocols.

**5.9 Internet telephony**: The combination of the term 'Internet' with the term 'telephony' is regarded as a specific use of the Internet, rather than a service. The Internet offers many capabilities to users, including the ability to carry bidirectional speech in real time or near real time. This is considered to be an intrinsic capability of the Internet and not a telecommunication service.

NOTE – Internet telephony is a particular application of the Internet and, therefore, falls outside of the scope of this Supplement.

## 6 Network configurations for using IP telephony

In this clause, the network configuration, charging and numbering plans needed to support IP telephony are introduced as a general example. This Supplement discusses the following four network configurations.

Configuration A: phone-to-phone communication;

Configuration B-1: IP phone-to-phone communication;

Configuration B-2: phone-to-IP phone communication;

Configuration C: IP phone-to-IP phone communication.

The remaining subclauses describe the characteristics of each configuration.

### 6.1 Configuration A: Phone-to-phone communication (with IP transit network)

This configuration (see Figure 6-1) uses the PSTN to originate and terminate a call (using the switching function of an existing PSTN), and converts speech into IP packets in the transit network.

In the IWF (such as MG, MGC, SG functions) between the PSTN and IP network at the originating and terminating sides, control signalling (ISUP – H.323/SIP conversion) and information signalling (64 kbit/s bearer – IP packet conversion) are converted. In the IP network, a call is controlled by the H.323/SIP protocol. A user dials a phone number to identify the terminating phone terminal and also, in some cases, additional information (e.g., through the use of prefix dialling) to select an IP transit network.

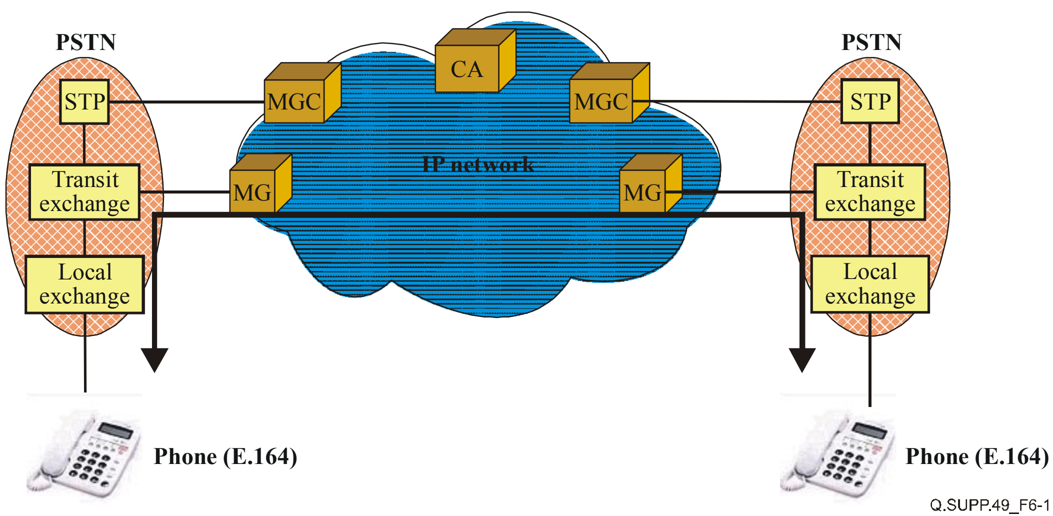


**Figure 6-1 – Network configuration A (phone-to-phone communication)**

## 6.2 Configuration B-1: IP phone-to-phone communication

In this configuration (see Figure 6-2), the originating network is an IP network and the terminating network is a PSTN.

In the IWF (such as MGC, MG, SG functions) between a PSTN and an IP network at the terminating side, the signalling protocol (ISUP – H.323/SIP conversion) and the user information (64-kbit/s bearer – IP packet conversion) are converted. In the IP network, a call is controlled by the H.323/SIP protocol. The originating IP phone user dials a phone number to identify the terminating phone terminal.

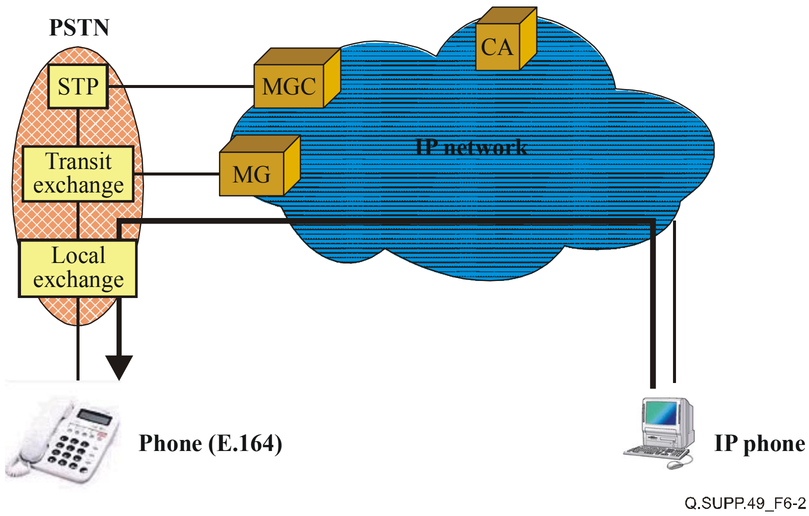


**Figure 6-2 – Network configuration B-1 (IP phone-to-phone communication)**

## 6.3 Configuration B-2: Phone-to-IP phone communication

In this configuration (see Figure 6-3), the originating network is a PSTN and the terminating network is an IP network.

In the IWF (such as MGC, MG, SG functions) between a PSTN and IP network at the originating side, the signalling protocol (ISUP – H.323/SIP conversion) and the information (64-kbit/s bearer – IP packet conversion) are converted. In the IP network, a call is controlled by the H.323/SIP protocol. The originating phone user dials a phone number to identify the terminating IP phone terminal.

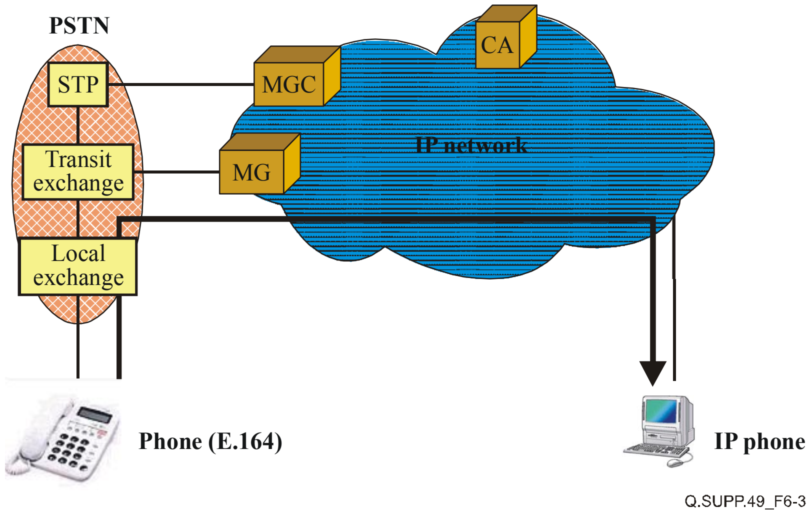


**Figure 6-3 – Network configuration B-2 (phone-to-IP phone communication)**

## 6.4 Configuration C: IP phone-to-IP phone communication

In this configuration (see Figure 6-4), all networks are IP. Calls are controlled by H.323/SIP signalling in the IP network. The terminating IP phone user is identified by an ID (e.g., a sequence of alphanumeric characters). The network operator assigns an ID to each user as they are registered. In addition to IDs, the IP phones can also have phone numbers which can be used to dialling (in the call control level IDs are used).

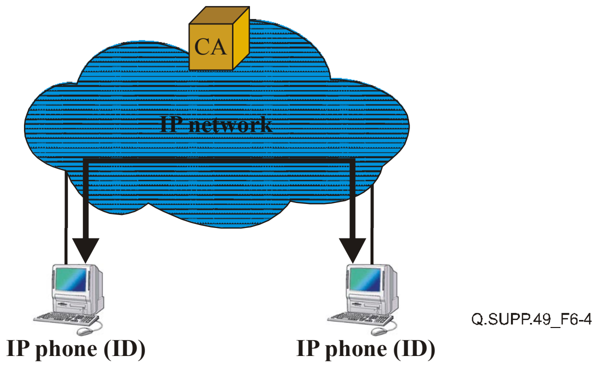


**Figure 6-4 – Network configuration C (IP phone-to-IP phone communication)**

# 7 Network capabilities to support IP telephony

This clause identifies the required functions and the related control protocols.

## 7.1 Service aspects

### 7.1.1 International IEPS/ETS

In the event of a natural or man-made disaster, the international communication networks may become congested. It is a requirement of ITU-T Rec. E.106 that international emergency agencies and relief organizations authorized by the home government be able to communicate. The user authentication and authorization of IEPS subscribers should be automated to support such emergency situations. To facilitate this service requirement, it is recommended that the protocols be amended to support the priority handling of IEPS calls across the international network. The signalling requirements and protocols used to support IEPS for IP telephony are being studied by the ITU-T and IETF.

NOTE – The signalling requirements for the ETS and the IEPS are being studied by the relevant standards organizations. The status of the studies being performed by the relevant standards organizations is shown in Appendix II.

### 7.1.2 Support of emergencies

Among those calls invoked from an IP network (Configurations B-1, C), emergency calls (911 in America, 999 in the United Kingdom, 112 in Europe) are not supported in many cases, so that an alternative means is required, such as calling from the PSTN instead of the IP network. In the future, to popularize IP telephony, emergency call services must be supported. In some countries, it is already a regulatory requirement to support the emergency call, also in cases where the call is invoked from an IP network. The systems used to respond to emergency calls and emergency services will vary within each country depending on the implementations. The following are examples of the issues that must be solved to enable the support of emergency calls:

– Appropriate routing to the nearest emergency call centre within the calling area when using the common specific number for a given country.
– The method used to realize the signalling used to recognize the priority level of an emergency call when interworking the PSTN network and IP network.
– The QoS requirements for an emergency call.

For calls invoked from the PSTN (Configurations A, B-2), emergency calls are supported by the features of the local exchange and the issues mentioned above are not an issue with the current telephone network technology.

NOTE – The procedure used to support emergency calls in a national network depends on the given country's regulations.

### 7.1.3 Support of location-dependent services

For those calls invoked from an IP network (Configurations B-1, C), with routing to the appropriate destination depending on the calling area, it is necessary to consider the support of services for selecting the destination according to the local calling area (e.g., services that connect to an announcement centre in the local calling area to provide local weather forecast information). As a concrete solution, a means of fixing the destination using location information for the access router that accommodates the calling user must be devised.

For calls invoked from the PSTN (Configurations A, B-2), support of location-dependent services is provided by the features of the local exchange. The issues mentioned above are not an issue with the current telephone network technology.

### 7.1.4 Support of supplementary services

To ensure the popularization of IP telephony, supplementary services (call hold, call transfer and so on) must be supported in addition to the basic call services. Regarding some of the supplementary services that are supported by the PSTN, their H.323 counterparts are defined in the H.450.x series of ITU-T Recommendations, while services for SIP are defined within IETF. For phone-to-phone (Configuration A), the supplementary services are defined in the I.251.x series of ITU-T Recommendations and it is felt that it would be better to support the same supplementary services for IP telephony. The supplementary services that are supported by each Recommendation are shown in Appendix I.

NOTE 1 – The scenario for migrating services from PSTN to IP telephony must be discussed and the step-by-step development method may be adapted to service developments. This issue is one of the ongoing study items covered by ITU-T.

NOTE 2 – The signalling requirements for supporting IP telephony over an IP cable network must be taken into account.

## 7.2 Charging considerations

Signalling requirements related to charging require further study.

## 7.3 Quality of Service considerations

Regarding the quality of IP telephony, the following issues require consideration.

– For evaluating call quality, the objective evaluation method is defined in ITU-T Rec. G.107 (R value), ITU-T Rec. P.862 (PESQ) and the subjective evaluation method is defined in ITU-T Rec. P.800 (MOS value). Currently, ITU-T SG 12 is developing a standard for assessing voice transmission quality based on IP protocol information, ETSI TIPHON, while the Telecommunication Industry Association (TIA) in the United States as well as the Japan's Ministry of Public Management, Home Affairs, Posts and Telecommunications have adopted the R value as the index of the quality class (the QoS category) for IP telephony. Currently, there is no unified evaluation method and no consensus for mapping between the quality class and service level. Instead, this is dependent on the regulations in each country.

– The Diffserv and the MPLS that are defined in the IETF are regarded as being effective means of distinguishing between data traffic that constitutes non-real-time communication and IP telephony that constitutes real-time communication over the IP network, while controlling the quality of the traffic to assure each service.

– When a call that causes the capacity set for IP telephony to be exceeded is made over the IP network, quality degradation of the calls that are in progress is assumed by the newly established call. To assure the quality of the current calls, there is a need for a method of controlling the number of connections made at any one time. (A call that would cause the set capacity to be exceeded should be rejected by the CA control mechanism.)

– To protect a network from burst traffic such as that which arises in the event of a disaster or upon the occurrence of some major event, and to assure the quality as much as possible, appropriate congestion control is necessary by ensuring cooperation between the IP network and the PSTN.

– These features of the congestion control must be able to cooperate with the traffic control mechanism of the PSTN, and studies must be done to determine how to realize a control mechanism for calls originating in the IP network and incoming calls from the PSTN or outgoing calls to the PSTN.

NOTE 1 – The current description in this clause includes the status of the studies of the relevant standards organizations from the viewpoint of background information.

NOTE 2 – The signalling requirements for IP-QoS are being studied by the relevant standards organizations. The statuses of the studies being performed by the relevant standards organizations are shown in Appendix II.

## 7.4 User identification and addressing considerations

There are several possibilities related to the building of a numbering space for IP telephony. The following cases are examples of these kinds of addressing and numbering aspects:

– A call terminating to a phone (Configurations A, B-1) can connect to the destination provided that the originating phone, or IP phone, specifies the existing E.164 number of the destination phone terminal. For a call made from a phone to an IP phone (Configuration B-2), the E.164 number has to be assigned to the terminating IP phone because the E.164 number is specified by the calling phone. It is possible to make a connection by using the IP address without using the E.164 number in the case of an IP phone-to-IP phone connection (Configuration C). But, when the E.164 number is assigned to an IP phone, a call established by dialling an E.164 number is also possible.

– The URIs within particular IP domains could be of a different format (e.g., SIP, SIPS or TEL URI), such that translation at the IP domain borders should be taken into account.

– When an E.164 number is assigned to a terminal within an IP network, ENUM can be used. This is being studied as a means of enabling translation from the E.164 number into the IP terminal address. The ENUM is defined in IETF RFC 2916 "E.164 number and DNS". An overview is shown in Figure 7-1.

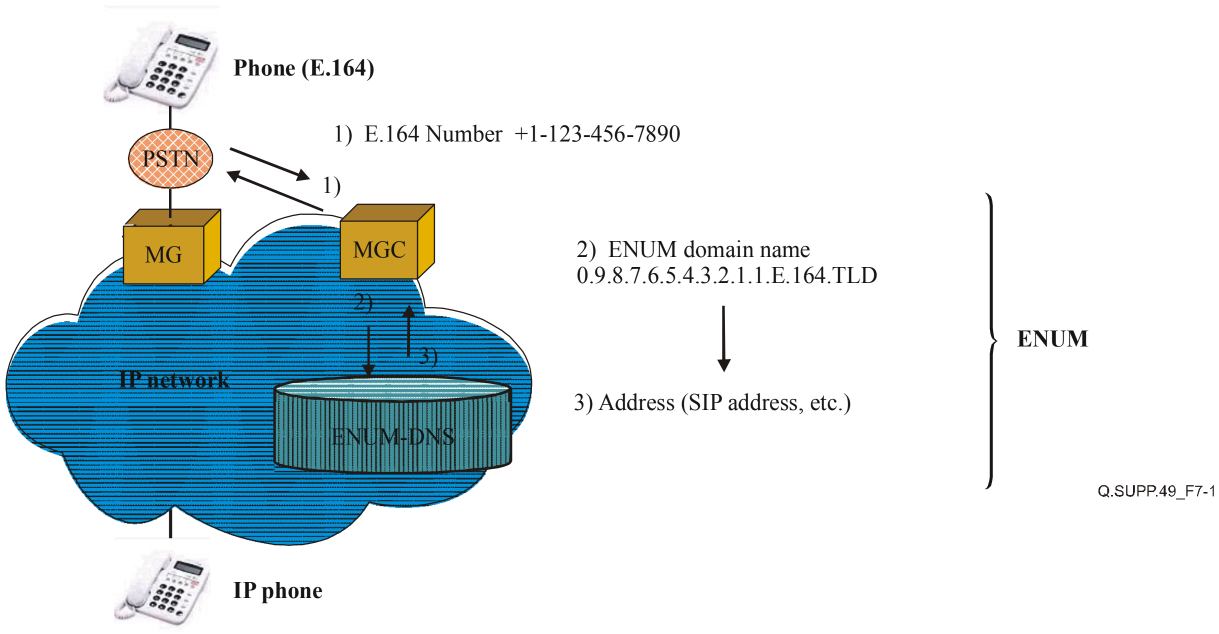


**Figure 7-1 – Overview of ENUM**

NOTE – Further study is required to enable the support of ENUM as a translation mechanism between the E.164 and IP terminal address such that different environments can be handled.

– There are several possible means of translating E.164 numbers into IP terminal addresses to support connectivity from ISDN/PSTN to an IP terminal. One possibility is the use of existing databases (like RADIUS), in which the IP terminal address and E.164 number of the IP user are already registered.

## 7.5 Security considerations

The security of IP telephony faces the following common issue, regardless of the network configuration (Configurations A, B-1, B-2, C).

A mechanism is necessary to protect the user and system from bugging, camouflage, illicit access to the IP network, cracking, and cyber terrorism. The ITU-T recommends the use of authentication and encryption in H.235 for communication with H.323. A similar means of authentication and encryption is also recommended for SIP.

## 8 Network capabilities to support IP telephony interworking between the PSTN and the IP network

There are the following two network-to-network interfaces (NNI), according to the network configuration.

1) NNI between IP-IP networks (Configurations A, B-1, B-2, C).

2) NNI between PSTN-IP networks (Configurations A, B-1, B-2).

The network configurations for IP phone-to-phone and phone-to-IP phone (Configurations B-1, B-2) are shown below as an example.

In the case of Figure 8-1, NNI between IP-IP networks, when the protocol (H.323, SIP, and so on) being used by the IP network differs between carriers, a study of the protocol conversion feature is necessary. Also, in the future, it will be necessary to study the agreement on the quality assurance and quality evaluation between the different IP network carriers.

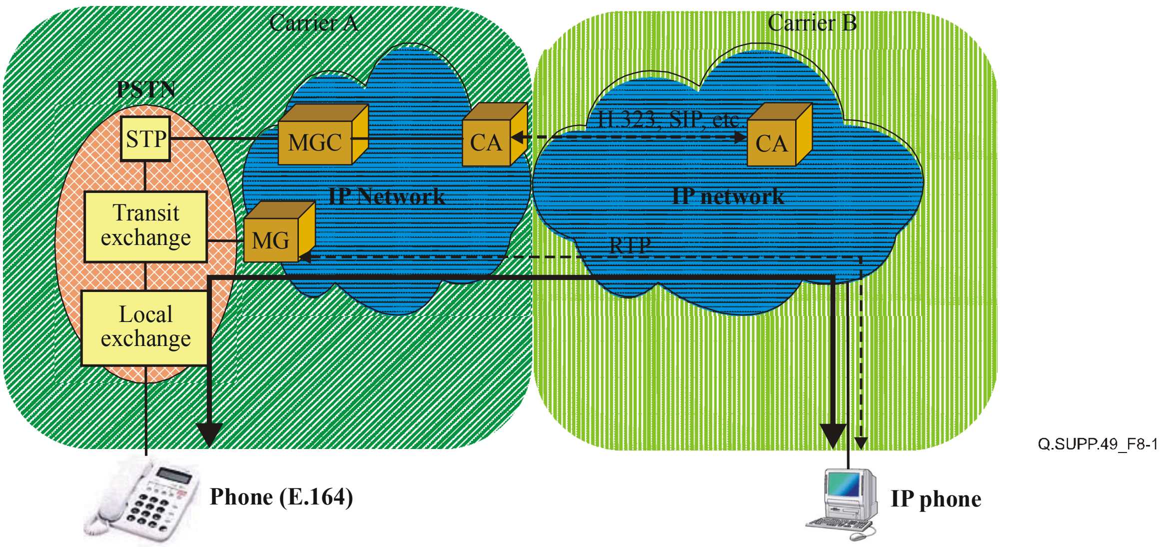


**Figure 8-1 – Network configuration connecting carriers (NNI between IP-IP network)**

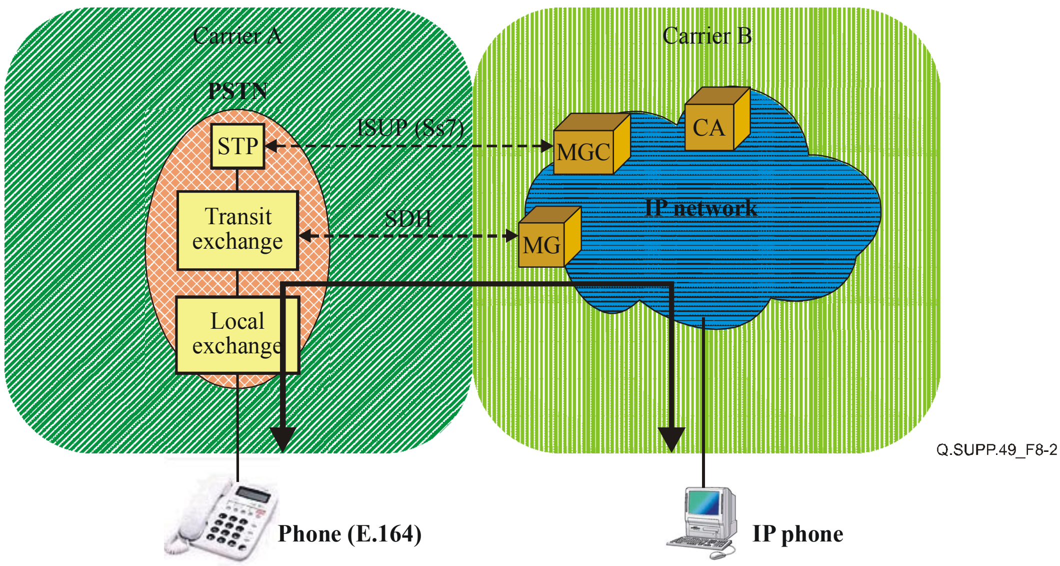


**Figure 8-2 – Network configuration connecting carriers (NNI between PSTN-IP network)**

NOTE – A study of the signalling requirement for the interworking between the PSTN and the IP network must be done by ITU-T.

## 9 Network Address Translation (NAT) and firewalls

The technology that is key to an IP network has, to date, proven to be a challenge to IP telephony (Configurations B-1, B-2, C) and incorporates Network Address Translation (NAT) and firewalls.

NAT is a protocol device that translates a global IP address to a private IP address. When using H.323 and SIP with a terminal that has been assigned a private address, the private address is included in the message for H.323 and SIP but the private IP address contained in the message is not translated to a global IP address by the NAT. As a result, the private IP address is notified to the terminating terminal as the originating terminal IP address and the IP packets sent to the originating terminal from the terminating terminal fail to reach their destination. Private IP addresses are not used by the popular IPv6, but are used by the current IPv4 network. Some solutions have been proposed, namely, the NAT traversal feature with Universal Plug and Play (UPnP) and Simple Traversal of UDP over NATs (STUN) methods.

A firewall has a feature that either passes or blocks IP packets according to the TCP/UDP port number, but the port number of the real-time transfer protocol (RTP) for voice communication with H.323 is different in each negotiation. Therefore, the firewall cannot specify a port setting. As a result, the H.323 and SIP protocols cannot traverse a firewall.

NOTE – The status of the studies being performed by the relevant standards organizations is shown in Appendix II.

## 10 Control protocols for support of IP telephony

This clause describes the protocol stacks used for the call control and media transport in IP telephony.

Call Control protocols: SIP (IETF), H.323 (ITU-T), BICC (ITU-T)

Media Gateway Control protocols: H.248 (ITU-T)/Megaco (IETF)

Signalling transport protocols: UDP (IETF), TCP (IETF) and SCTP (IETF) including the specified adaptation layers.

Media Transport Protocols: RTP/RTCP (IETF) over UDP (IETF)

An example of the adaptation of each protocol for configurations B-1 and B-2 is shown in Figure 10-1, while a protocol stack is shown in Figure 10-2.

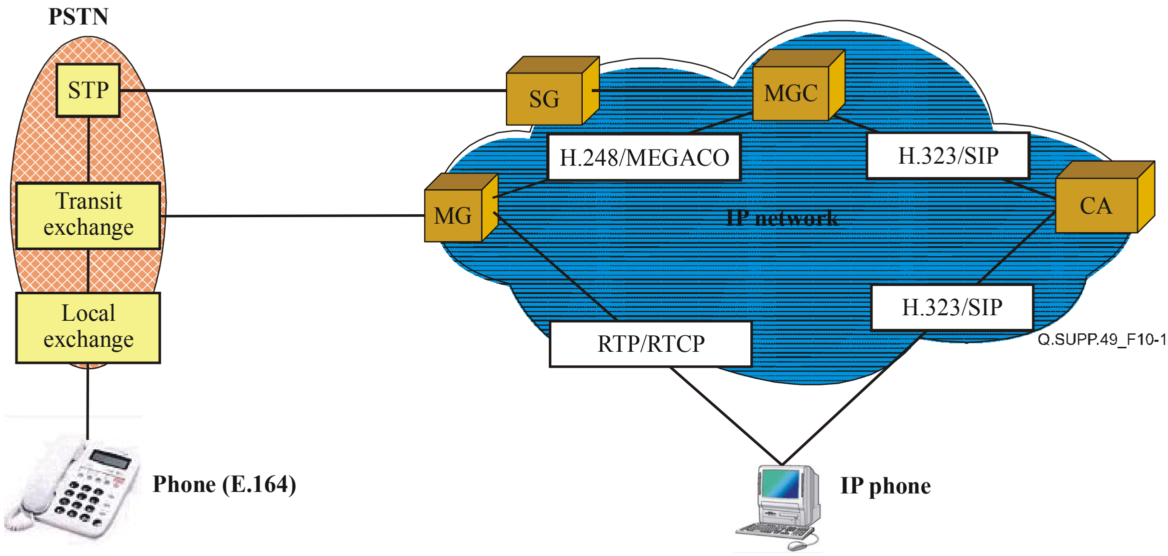


**Figure 10-1 – Example of protocol adoption**

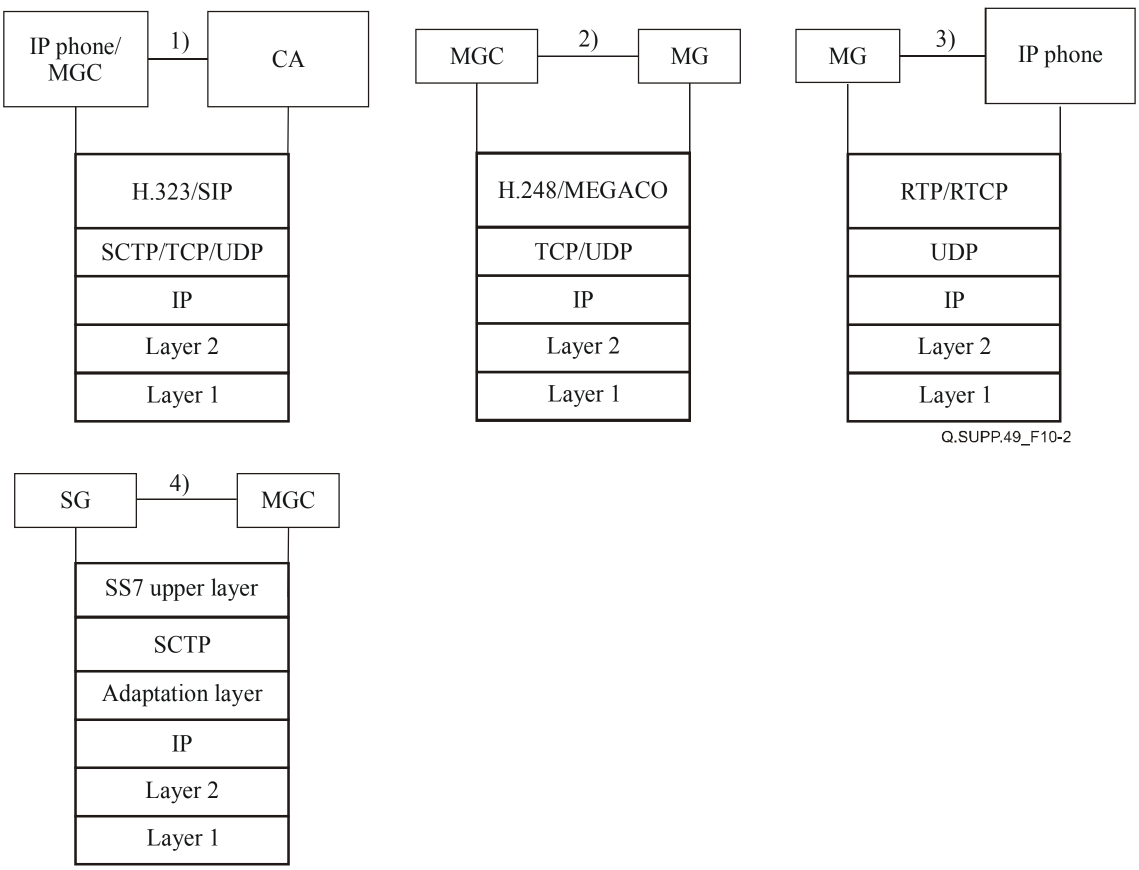


**Figure 10-2 – Protocol stack (example for configurations B-1 and B-2)**

NOTE – Figures 10-1 and 10-2 are examples of protocol adoption. These figures are intended to act as a starting point for further discussion regarding the protocols for supporting IP telephony. The various protocols should be considered and added to these figures.

Applicability of the J-series Recommendations in support of IP telephony is for further study.

# Appendix I

## Supplementary services and the corresponding ITU-T Recommendations that define them

### Table I.1 – Supplementary services defined for PSTN

Supplementary services defined in the I.251, I.252, I.253, I.254, I.255, I.256 and I.257 series of ITU-T Recommendations:

| | |
|---|---|
| I.251 | *Number Identification supplementary services* |
| I.251.1 | Direct-Dialling-In (DDI) |
| I.251.2 | Multiple Subscriber Number (MSN) |
| I.251.3 | Calling Line Identification Presentation (CLIP) |
| I.251.4 | Calling Line Identification Restriction (CLIR) |
| I.251.5 | Connected Line Identification Presentation (COLP) |
| I.251.6 | Connected Line Identification Restriction (COLR) |
| I.251.7 | Malicious Call Identification (MCI) |
| I.251.8 | Sub-addressing supplementary service (SUB) |
| I.252 | *Call Offering supplementary services* |
| I.252.1 | Call Transfer (CT) |
| I.252.2 | Call Forwarding Busy (CFB) |
| I.252.3 | Call Forwarding No Reply (CFNR) |
| I.252.4 | Call Forwarding Unconditional (CFU) |
| I.252.5 | Call Deflection (CD) |
| I.252.6 | Line Hunting (LH) |
| I.253 | *Call Completion supplementary services* |
| I.253.1 | Call Waiting (CW) |
| I.253.2 | Call Hold (HOLD) |
| I.253.3 | Completion of calls to busy subscribers (CCBS) |
| I.254 | *Multiparty supplementary services* |
| I.254.1 | Conference Calling (CONF) |
| I.254.2 | Three-Party Service (3PTY) |
| I.255 | *Community of Interest supplementary services* |
| I.255.1 | Closed User Group (CUG) |
| I.255.2 | Support of Private Numbering Plans (PNP) |
| I.256 | *Charging supplementary services* |
| I.256.1 | Credit card calling (CRED) |
| I.256.2 | Advice of charge (AOC) |
| I.256.3 | Reverse charging (REV) |
| I.257 | *Additional Information Transfer supplementary services* |
| I.257.1 | User-to-User Signalling (UUS) |

**Table I.2 – Supplementary services defined for IP telephony**

Supplementary services defined for H.323 in the H.450 series of ITU-T Recommendations:

| | |
|---|---|
| H.450.1 | *Generic functional protocol for the support of supplementary services in H.323.* |
| H.450.2 | *Call transfer supplementary service for H.323.* |
| H.450.3 | *Call diversion supplementary service for H.323.* |
| H.450.4 | *Call hold supplementary service for H.323.* |
| H.450.5 | *Call park and call pickup supplementary services for H.323.* |
| H.450.6 | *Call waiting supplementary service for H.323.* |
| H.450.7 | *Message waiting indication supplementary service for H.323.* |
| H.450.8 | *Name identification supplementary service for H.323.* |
| H.450.9 | *Call completion supplementary services for H.323.* |
| H.450.10 | *Call offering supplementary services for H.323.* |
| H.450.11 | *Call intrusion supplementary service for H.323.* |
| H.450.12 | *Common Information Additional Network Feature for H.323.* |

Supplementary services defined for SIP (draft-ietf-sipping-service-examples-02.txt):

Call Hold – Consultation Hold, Music On Hold

Unattended Transfer, Attended Transfer

Call Forwarding Unconditional, Call Forwarding – Busy, Call Forwarding – No Answer

3-way Conference – Third Party is Added, 3-way Conference – Third Party Joins

Single-Line Extension

Find-Me

Call Management (Incoming Call Screening), Call Management (Outgoing Call Screening)

Call Park

Call Pickup

Automatic Redial

Click to Dial

# Appendix II

## Standard development related to IP telephony

**Table II.1 – Status of studies being undertaken by international standards organizations**

| Clause | Item | Content | Organization | Recommendations /drafts |
|---|---|---|---|---|
| 7.1 Service aspects | International IEPS/ETS | IEPS | ITU-T SG 2 E.106 (IEPS) | E.106 |
| | | IEMS | ITU-T SG 16 | F.706 |
| | | Signalling requirements for IEPS/ETS | ITU-T SG 11 | Note 5 |
| | | Protocols for IEPS/ETS | ITU-T SG 11 | Q.761-764, Q.767, Q.1902.1-1902.4 Q.2761-2764 |
| | | IEPS in the Internet | IETF IEPREP-WG | NOTE 3 – For the latest draft document, see the IEPREP website |
| | | IEPS/ETS, etc. | ITU-T SG 4, 9, 12, 13, 17, SSG ETSI TM Forum | – |
| | Support of emergency calls | Fire, police, ambulance | Note 1 | Note 1 |
| | Support of location-dependent services | Weather forecast information | Note 1 | Note 1 |
| | Support of supplementary services | Various services | ITU-T SG 16 | H.450.1-12 |
| | | | IETF SIPPING-WG | NOTE 4 – For the latest draft document, see the SIPPING-WG website |
| 7.2 Charging aspects | – | – | SG 2, SG 3 | – |

**Table II.1 – Status of studies being undertaken by international standards organizations**

| Clause | Item | Content | Organization | Recommendations /drafts |
|---|---|---|---|---|
| 7.3 QoS | Signalling requirements and protocols for QoS | Diffserv | IETF (Diffserv-WG) | RFC 2474 |
| | | MPLS | ITU-T SG 11 | Q.2920 Q. Suppl. 46 – TRQ 2830 (Note 5) |
| | | | ITU-T SG 13 | Y.1411, Y.1412 |
| | | | IETF (MPLS-WG) | RFC 3031 |
| | | IP-QoS | ITU-T SG 11 | Note 5 |
| | Speech quality | R value | ITU-T SG 12 | G.107 |
| | | PESQ | | P.862 |
| | | PSQM | | P.861 |
| | | MOS value | | P.800 |
| | Connection quality (Connection delay, connection release delay, lost call rate) | Connection quality | ITU-T SG 2<br>ITU-T SG 13 | E.671 (Note 5) |
| | Stable quality failure rate | Stable quality | ITU-T SG 13 | Y.1540, Y.1541 |
| | Transmission quality | Transmission quality | ITU-T SG 13 | Y.1541 |
| | Etc. | QoS for the IP telephony in the cable network | ITU-T SG 9 | – |
| 7.4 User identification and addressing | Addressing | ENUM | IETF ENUM-WG | RFC 2916 |
| | | | ITU-T SG 2 | – |
| | Routing | Routing protocols | IETF IPTEL-WG | – |
| 7.5 Security | Authentication and encryption | H.235 | ITU-T SG 16 | H.235 |
| | | SIP | IETF SIP-WG | RFC 3261 |
| 8 PSTN interworking | – | – | – | – |
| 9 NAT and firewall | NAT | STUN | IETF-MIDCOM-WG | RFC 3489 |
| | | UPNP NAT Traversal | UPNP Forum | – |
| | FW | STUN | IETF-MIDCOM-WG | RFC 3489 |

**Table II.1 – Status of studies being undertaken by international standards organizations**

<table>
<tr><th>Clause</th><th>Item</th><th>Content</th><th>Organization</th><th>Recommendations /drafts</th></tr>
<tr><td rowspan="8">10 Protocol</td><td rowspan="4">Call control</td><td>H.323</td><td>ITU-T SG 16</td><td>H.323</td></tr>
<tr><td>SIP</td><td>IETF SIP-WG</td><td>RFC 3261</td></tr>
<tr><td rowspan="2">H.248/MEGACO</td><td>IETF MEGACO-WG</td><td>RFC 3015</td></tr>
<tr><td>ITU-T SG 16</td><td>H.248</td></tr>
<tr><td>Media control</td><td>RTP/RTCP</td><td>IETF MMUSIC-WG</td><td>RFC 1889</td></tr>
<tr><td rowspan="3">Interwork</td><td rowspan="3">SIP-ISUP inter-working</td><td>IETF SIPPING-WG</td><td>RFC 3398<br>NOTE 4 – For the latest draft document, see URL of SIPPING-WG</td></tr>
<tr><td>ITU-T SG 11</td><td>Q. Suppl. 45 – TRQ.2815</td></tr>
<tr><td>ITU-T SG 11</td><td>Q.1912.5</td></tr>
<tr><td rowspan="2">others</td><td>NGN</td><td>–</td><td>ITU-T SG 13<br>ITU-T SG 11</td><td>–</td></tr>
<tr><td>ETSI</td><td>–</td><td>ETSI/Project TIPHON</td><td>Note 2</td></tr>
<tr><td colspan="5">NOTE 1 – Varies with the national implementation.<br>NOTE 2 –<br>• ETSI TS 101 878 V1.1.1: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Service Capability Definition; Service Capabilities for a simple call" – available at http://docbox.etsi.org/TIPHON/TIPHON/07-drafts/wg1/_Published/ts_101878v111p.doc.<br>• ETSI TS 101 315 V1.1.1: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Functional entities, information flow and reference point definitions; Guidelines for the application of TIPHON functional architecture to inter-domain services" available at http://docbox.etsi.org/TIPHON/TIPHON/07-drafts/wg2/_Published/ts_101315v111p.zip.<br>NOTE 3 – http://www.ietf.org/html.charters/ieprep-charter.html.<br>NOTE 4 – http://www.ietf.org/html.charters/sipping-charter.html.<br>NOTE 5 – This subject is being studied currently by ITU-T and the reader is advised to look for future publications.</td></tr>
</table>

# Appendix III

## J-series Recommendations in support of IP telephony

### Table III.1 – J-series Recommendations in support of IP telephony

| | |
|---|---|
| J.160 | *Architectural framework for the delivery of time-critical services over cable television networks using cable modems.* |
| J.161 | *Audio codec requirements for the provision of bidirectional audio service over cable television networks using cable modems.* |
| J.162 | *Network call signalling protocol for the delivery of time-critical services over cable television networks using cable modems.* |
| J.163 | *Dynamic quality of service for the provision of real-time services over cable television networks using cable modems.* |
| J.164 | *Event message requirements for the support of real-time services over cable television networks using cable modems.* |
| J.165 | *IPCablecom Internet signalling transport protocol (ISTP).* |
| J.166 | *IPCablecom Management Information base (MIB) framework.* |
| J.167 | *Media Terminal Adapter (MTA) device provisioning requirements for the delivery of real-time services over cable television networks using cable modems.* |
| J.168 | *IPCablecom Media Terminal Adapter (MTA) MIB requirements.* |
| J.169 | *IPCablecom network call signalling (NCS) MIB requirements.* |
| J.170 | *IPCablecom security specification.* |
| J.171 | *IPCablecom Trunking Gateway Control Protocol (TGCP).* |
| J.172 | *IPCablecom management event mechanism.* |
| J.173 | *IPCablecom embedded MTA primary line support.* |
| J.174 | *IPCablecom interdomain quality of service.* |
| J.175 | *Audio server protocol.* |
| J.176 | *IPCablecom management event mechanism MIB.* |
| J.177 | *IPCablecom CMS subscriber provisioning specification.* |
| J.178 | *IPCablecom CMS to CMS signalling.* |

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