NATIONAL STANDARDS

FOR

SIP

No. SD/ SIP- 01/01 SEP. 2008

DEPARTMENT OF TELECOMMUNICATIONS
TELECOMMUNICATION ENGINEERING CENTRE

NATIONAL STANDARDS FOR SIP

No. SD/ SIP- 01/01 SEP. 2008

HISTORY SHEET

S. No.	Title	GR No.	Remarks
1.	National Standards for SIP	SD/SIP -01/01 SEP. 2008	Issue 01

NATIONAL STANDRDS FOR SIP

No. SD/ SIP-01/01 SIP. 2008

CONTENTS

List of Documents

1. TEC GR/Standards

S.NO.	Title	Document number
1.	National Standards for SIGTRAN	SD/SGT-01
2.	National CCS7 standards for MTP and ISUP	S/CCS-02
3.	National SCCP standards for Large Digital Switching Systems	S/CCS-03
4.	National Transaction Capabilities Application Part (TCAP) Standards	SD/CCS-05
5.	Intelligent Network Application Protocol (INAP) National Standards	SD/INP-01
6.	Universal Mobile Telecommunication System – Core Network (Release 4)	GR/UCN-01
7.	IP Multimedia Sub-System	GR/IMS-01

2. IETF / 3GPP /Other Standards

S.No.	Title	Document No.
1	SIP: Session Initiation Protocol	IETF RFC 3261
2	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)	IETF RFC 3262
3	Session Initiation Protocol (SIP): Locating SIP Servers.	IETF RFC 3263
4	Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)	IETF RFC 3389
5	SDP : Session Description Protocol	IETF RFC 2327
6	Domain names – Concepts and Facilities	IETF RFC 1034
7	A DNS RR for specifying the location of services (DNS SRV)	IETF RFC 2782
8	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals	IETF RFC 2833
9	The Naming Authority Pointer (NAPTR) DNS Resource Record	IETF RFC 2915
10	E.164 number and DNS	IETF RFC 2916
11	Real Time Streaming Protocol (RTSP)	IETF RFC 2326
12	User Datagram Protocol (UDP)	IETF RFC 768
13	Real Time Transport Protocol (RTP,RTCP)	IETF RFC 3550 &3551
14	Transmission Control Protocol (TCP)	IETF RFC 793
15	Stream Control Transmission Protocol (SCTP)	IETF RFC 2960
16	E.164 to Uniform Resource Identifier (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)	IETF RFC 3761
17	ISUP to SIP Mapping	IETF RFC 3398
18	The Session Initiation Protocol (SIP) UPDATE Method	IETF RFC 3311
19	Multipurpose Internet Mail Extensions (MIME) Part Two: Media Type	IETF RFC 2046
20	MIME media types for ISUP and QSIG Objects	IETF RFC 3204
21	URLs (Uniform Resource Locator) for Telephone Calls	IETF RFC 2806
22	User Requirements for the Session Initiation Protocol (SIP) in support of Deaf, Hard of Hearing and Speech-impaired Individuals	IETF RFC 3351
23	Session Initiation Protocol (SIP) – Specific Event Notification	IETF RFC 3265

24	Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part (03/2004)	ITU –T Q.1912.5
25	IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3	3GPP TS24.229
26	Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks	3GPP TS 29.163

CHAPTER-1

Scope

1.0 Scope

1.1 This document specifies the National Standards for Session Initiation Protocol (SIP) to be used in Indian Telecom network.

1.2 SIP is defined in IETF RFC 3261 and is an application layer control protocol for multimedia communication over an IP network.

1.3 The document covers the architecture, general SIP functionality and its inter working with NGN as well as PSTN/PLMN networks. Various timers supported and simple call flow is also mentioned.

1.4 For all ITU-T Recommendations and IETF standard/RFC referred in this document, the latest release shall be applicable.

1.5 For all TEC documents referred in this document, the latest issue with all associated Amendments, Addendum and Corrigendum shall be applicable.

CHAPTER-2

Overview and Architecture

2.1 Introduction

Session Initiation Protocol (SIP) is an application layer signalling protocol for creating, modifying and terminating sessions with one or more participants over IP network. These sessions include telephone calls, multimedia conferences, instant messaging etc. using audio, video and data. SIP can invite both persons and services (such as a media storage service) to participate in a session and can also be used to set up calls between PSTN/PLMN subscribers using gateways. SIP is defined in IETF RFC 3261.

2.2 Overview

SIP defines telephone numbers as URLs (Uniform Resource Locators), so that web pages can contain them. These addresses take the form of user@host, similar to e-mail addresses. The user part, which is left of the “@” sign, may be user name or a telephone number and host part, which is right of the “@” sign, is a domain name or IP address of Network.

SIP provides the capabilities to:

- Determine the location of the target end point.
- Determine the media capabilities of the target end point via Session Description Protocol (SDP).
- Determine the availability of the target end point.
- Establish a session between the originating and target end point if the call can be completed.
- Handle the transfer and termination of calls. SIP supports the transfer of calls from one end point to another.
- SIP-I is used for transportation of CCS7 signaling information over IP between two softswitches.

2.3 **SIP Architecture**

Implementation of Session Initiation Protocol (SIP) in NGN environment is shown in Figure – 1

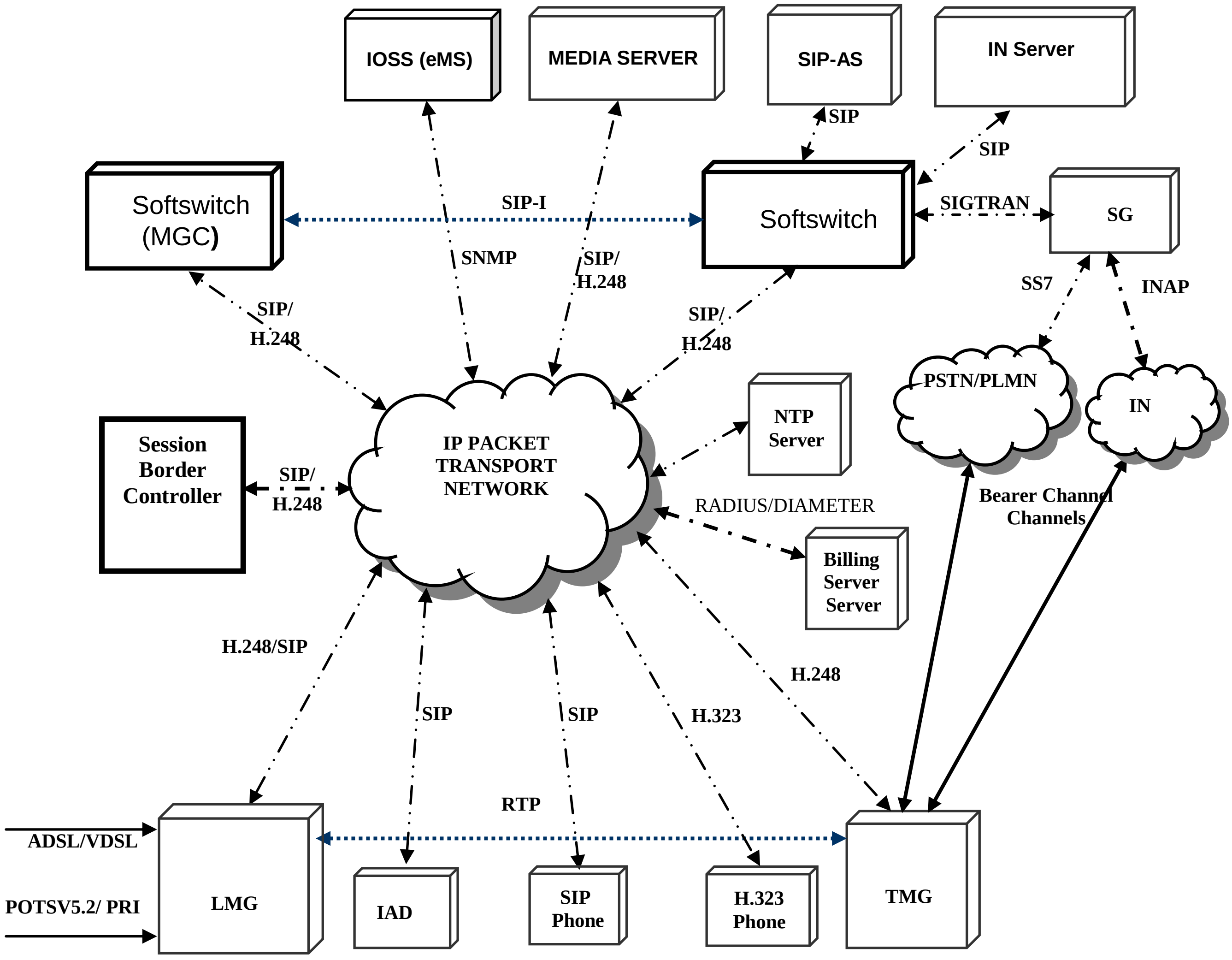
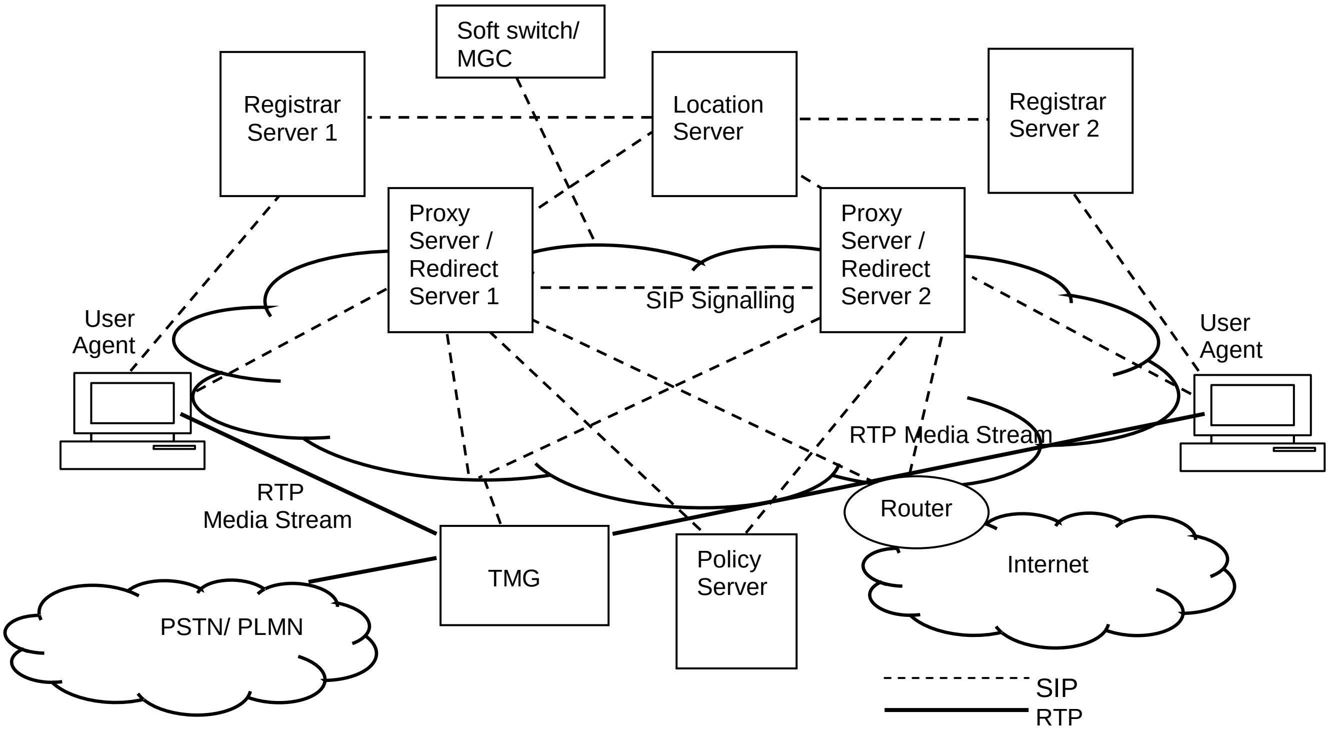


Figure 1: Implementation of SIP in NGN Environment

Following protocols are used between different Network Elements (NEs) :

Softswitch to Softswitch	-	SIP-I
Softswitch to SG	-	Sigtran
Softswitch to SIP-AS	-	SIP
Softswitch to IN Server	-	SIP
IN Platform to SG	-	INAP
LMG to TMG	-	RTP

SIP is a peer to peer protocol. Peers in a session are called User Agents which are basic network entities namely clients and servers. A client is an application program that sends SIP requests. A server is an application program that accepts SIP requests and sends back responses to the requests. Figure – 2 shows the architecture of a SIP network.



TMG - Trunk Media Gateway
LMG - LINE Media Gateway

Figure2: Architecture of SIP Network

2.3.1 **User Agent (UA)**

User Agents are SIP network terminals like SIP phones and SIP Gateways. UA contains application programs that send SIP requests and responses to initiate and receive calls over a SIP network. A user agent can function as a user agent client (UAC) as well as user agent server (UAS).

2.3.2 **Proxy Server**

A Proxy Server acts as the initial point of contact for all SIP requests. It acts as a server and a client for the purpose of making requests on behalf of other clients. Unlike User Agents, Proxy Servers do not initiate SIP requests on their own.

2.3.3 **Redirect Server**

A Redirect Server accepts a SIP request from a client for the called party destination address. In response to the request it finds out the address where the called party has logged on and sends back the required address to the client. It is typically co-located with a Proxy Server.

2.3.4 Registrar Server

A Registrar Server accepts SIP REGISTER requests from a user, indicating that he is available at a particular address in the network. A Registrar Server is typically co-located with a proxy or redirect server and may offer location services**.**

2.3.5 Location Server

A Location Server is used by a SIP redirect or proxy server to obtain information about a called party's possible location. The location server can also be an entity outside the SIP network.

2.3.6 Policy Server

The Policy Server is designed to use Common Open Policy Service to provide Quality of Service (QoS), bandwidth reservation for calls or call segments that are transmitted over the network.

2.4 SIP Message Syntax

SIP Message syntax is text-based. These messages are either requests from a client to a server and vice versa or responses from a server to a client.

CHAPTER-3

SIP Functionality

3.1 General Functionality

SIP shall comply with the following documents as mentioned in Table -1

Table -1

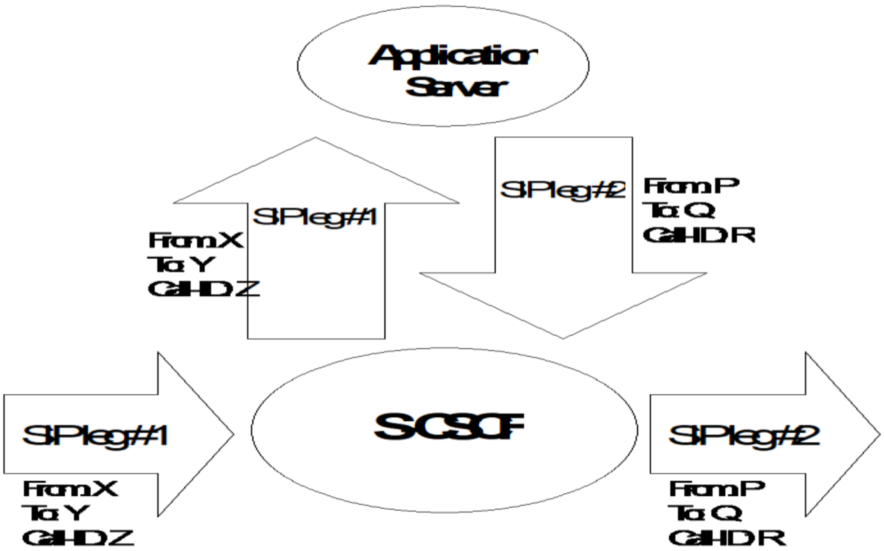
S.No.	Document No.	Title	Mandatory/ Optional
1	IETF RFC 3261	SIP: Session Initiation Protocol	Mandatory
2	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)	Mandatory
3	IETF RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers.	Mandatory
4	IETF RFC 3389	Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)	Mandatory
5	IETF RFC 2327	SDP : Session Description Protocol	Mandatory
6	IETF RFC 1034	Domain names – Concepts and Facilities	Mandatory
7	IETF RFC 2782	A DNS RR for specifying the location of services (DNS SRV)	Mandatory
8	IETF RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals	Mandatory
9	IETF RFC 2915	The Naming Authority Pointer (NAPTR) DNS Resource Record	Mandatory
10	IETF RFC 2916	E.164 number and DNS	Mandatory
11	IETF RFC 2326	Real Time Streaming Protocol (RTSP)	Optional
12	IETF RFC 768	User Datagram Protocol (UDP)	Mandatory
13	IETF RFC 3550 &3551	Real Time Transport Protocol (RTP,RTCP)	Mandatory
14	IETF RFC 793	Transmission Control Protocol (TCP)	Optional
15	IETF RFC 2960	Stream Control Transmission Protocol (SCTP)	Optional

16	IETF RFC 3761	E.164 to Uniform Resource Identifier (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)	Mandatory
17	IETF RFC 3398	ISUP to SIP Mapping	Mandatory
18	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method	Mandatory
19	IETF RFC 2046	Multipurpose Internet Mail Extensions (MIME) Part Two: Media Type	Mandatory
20	IETF RFC 3204	MIME media types for ISUP and QSIG Objects	Mandatory
21	IETF RFC 2806	URLs (Uniform Resource Locator) for Telephone Calls	Mandatory
22	IETF RFC 3351	User Requirements for the Session Initiation Protocol (SIP) in support of Deaf, Hard of Hearing and Speech-impaired Individuals	Optional
23	IETF RFC 3265	Session Initiation Protocol (SIP) – Specific Event Notification	Mandatory
24	ITU –T Q.1912.5	Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part (03/2004)	Mandatory
25	3GPP TS24.229	IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3	Mandatory
26	3GPP TS 29.163	Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks	Mandatory

3.2 SIP shall support the following Call Models

3.2.1 **Serving - Call Session Control Function (S-CSCF) and Call Session – Application Server (CS-AS) Interworking**

SIP shall support the high-level call model for interworking between the S-CSCF and the CS-AS as shown in Figure - 3. In the figure the “SIP leg” is equivalent with “SIP dialog” or “SIP session” and is sometimes also referred to as “call leg”.



S – CSCF - Serving - Call Session Control Function

CS-AS - Call Session – Application Server

Figure –3: CS-AS performing 3rd party call control (B2BUA)

The CS-AS shall apply the half call model defined by also if the originating and terminating user reside on the same physical CS-AS host.

For normal sessions, the AS shall behave as a Back-to-Back User Agent (B2BUA) compatible with the 3GPP specifications, which means that all incoming session invitations from the S-CSCF are terminated in the AS, while for every incoming session, a new session is initiated by the AS on the outgoing side (provided that the call is allowed).

The S-CSCF shall behave as a stateful proxy. Although the AS shall act as a B2BUA and create a new call-leg on the outgoing side, the CSCF shall be able to correlate the two call-legs by passing call state information in the Route header from the first call-leg via the AS to the second call-leg. 3GPP, i.e. the CS-AS treats originating and terminating requests separately,

3.2.2 Session Controller (SC) Call Model

SIP shall support Session Controller Call Model for SIP-to-SIP interworking. The Session Controller (SC) shall act as a Back-to-Back User Agent (B2BUA), which means that all incoming session invitations are terminated in the Session Controller, and then for every outgoing session, a new session shall be initiated by the SC on the outgoing side.

3.2.3 Media Gateway Controller (MGC) Call Model

SIP shall support the MGC Call Model for SIP-to-PSTN/PLMN and PSTN/PLMN-to-SIP call set-up. MGC shall act as a User Agent Server for SIP-to-PSTN/PLMN call set-up, and User Agent Client for PSTN/PLMN-to-SIP call set-up.

3.3 At least the following SIP methods shall be supported

3.3.1 REGISTER

The REGISTER method shall be used to create a binding between the user's SIP address and the current UE contact address (IP address & port).

3.3.2 INVITE

The INVITE method shall be used to invite a remote user to a new SIP session. It shall also be used to modify existing sessions, in which case it is referred to as a Re-INVITE.

3.3.3 ACK

The ACK method shall be used to acknowledge a final SIP response to an INVITE request.

3.3.4 BYE

The BYE method shall be used to terminate a SIP session. It can be sent by either the caller or the called party.

3.3.5 CANCEL

The CANCEL method shall be used to terminate an ongoing SIP session invitation. It shall not affect a completed request or existing calls.

3.3.6 MESSAGE

The MESSAGE method shall be used to send an instant message.

3.3.7 SUBSCRIBE

The SUBSCRIBE method shall be used to request presence information updates about a user. It shall also be used for interaction with other services, e.g. the Centrex service 'Busy Lamp Field'.

3.3.8 PUBLISH

The PUBLISH method shall be used to send presence information about/from a user to their Presence Server.

3.3.9 NOTIFY

The NOTIFY method shall be used by the Presence Server to send presence information about a user to other users. It shall also be used for some other services, e.g. Message Waiting Indication.

3.3.10 INFO

The INFO method shall be used for sending application level information along the SIP signalling path.

3.3.11 UPDATE

The UPDATE method shall be used to allow a UA to update parameters of a session (such as the set of media streams and their codecs).It shall have no impact on the state of a dialog.

3.3.12 OPTIONS

The OPTIONS method shall be used to allow a UA to query another UA or a proxy server as to its capabilities. This shall allow a client to discover information about the supported methods, content types, extensions, codecs, etc. without "ringing" the other party.

3.3.13 REFER

The REFER method shall be used to indicate that the recipient (identified by the Request-URI) should contact a third party using the contact information provided in the request.

3.3.14 PRACK

The PRACK method shall be used to enable the sending of reliable provisional responses.

3.4 SIP Message Routing

SIP shall support at least following SIP message headers for routing of system SIP messages. SIP headers are a set of parameters that could be assigned specific values inside a SIP message. They convey information about the SIP request or response.

3.4.1 Request URI

The Request-URI shall be used to indicate the user (or service/proxy) to which this request is being addressed and shall be used for message routing unless a Route header field is included in the received message.

The Request-URI shall contain a public SIP address, a TEL URI (or SIP address containing E.164 number and “user=phone” parameter), a B2BUA node address or a UE IP contact address. The phone number format may be a public E.164 number or a private number (e.g. extension number within a corporate numbering plan).

3.4.2 Route

The Route header field shall be used to force routing for a request through the listed set of proxies, i.e. the presence of a Route header overrides the Request-URI.

Within the system, the Route header shall be used for:

(i) Routing of originating SIP requests from the UE through the originating SC and P-CSCF to the S-CSCF

(ii) Routing of terminating SIP requests from the I-CSCF through the S-CSCF, P-CSCF and SC to the UE

(iii) Originating and terminating AS invocation, i.e. routing of SIP requests from the S-CSCF to the AS and back to the S-CSCF again.

3.4.3 Via

The Via header field shall be used to indicate the path taken by the request so far and indicates the path that should be followed in routing responses. If the proxy wishes to stay in the signalling path for responses, it inserts (adds) its own address in the Via header before proxying the request forwards.

3.4.4 Record-Route

The Record-Route header shall be inserted to INVITE by proxy to ensure that it will receive all subsequent requests and responses belonging to the same session.

3.4.5 Contact

The Contact header field shall be used to tell the terminating proxy where to send future requests when initiated inside the existing session.

The B2BUAs AS, SC and MGC all insert their own node address in the Contact header field when initiating an outgoing SIP request.

3.4.6 From

From header shall be used to convey the calling party information except in privacy call. In a non – privacy call, it may have the name and URL (Uniform Resource Locator) of the calling party.

3.4.7 To

To header shall be used to convey the called party information. This information remains unchanged even when the call is redirected.

3.4.8 Call - ID

Call – ID header shall be used, which contains a globally unique value for identifying the call the request is associated.

3.4.9 C Seq

C Seq header shall be used to indicate the current sequence number for the request. Subsequent requests must have a monotonically increasing number. Besides the number, the method name must be present.

3.4.10 Content – Length

Content – Length header shall be used to indicate the length of the body that follows the headers. When the length is zero, the body is absent and Content – Type header is not present.

3.4.11 Content – Type

Content – Type header shall be used to indicate the type of message body.

3.4.12 Session - Expires

Session – Expires shall be used to define the time interval for session heartbeat timer.

3.4.13 Supported

Supported header shall be used to indicate what extensions are supported.

3.5 Transport Protocol

Support of User Datagram Protocol (UDP) as the transport protocol for SIP is mandatory. Transmission Control Protocol (TCP) and Stream Control Transmission Protocol (SCTP) are optional for SIP transport.

3.6 Timer Values

Following Timer Values as per IETF RFC 3261 given below in Table -2 shall be supported.

Table - 2

Timer	Value	Meaning
T1	500ms default	RTT Estimate
T2	4s	The maximum retransmit interval for non-INVITE requests and INVITE responses
T4	5s	Maximum duration a message will remain in the network
Timer A	initially T1	INVITE request retransmit interval, for UDP only
Timer B	64*T1	INVITE transaction timeout timer
Timer C	>3 min	proxy INVITE transaction timeout
Timer D	> 32s for UDP 0s for TCP/SCTP	Wait time for response retransmits
Timer E	initially T1	non-INVITE request retransmit interval, UDP only
Timer F	64*T1	non-INVITE transaction timeout timer
Timer G	initially T1	INVITE response retransmit interval
Timer H	64*T1	Wait time for ACK receipt
Timer I	T4 for UDP 0s for TCP/SCTP	Wait time for ACK retransmits
Timer J	64*T1 for UDP 0s for TCP/SCTP	Wait time for non-Invite request retransmits
Timer K	T4 for UDP 0s for TCP/SCTP	Wait time for response retransmits

CHAPTER-4

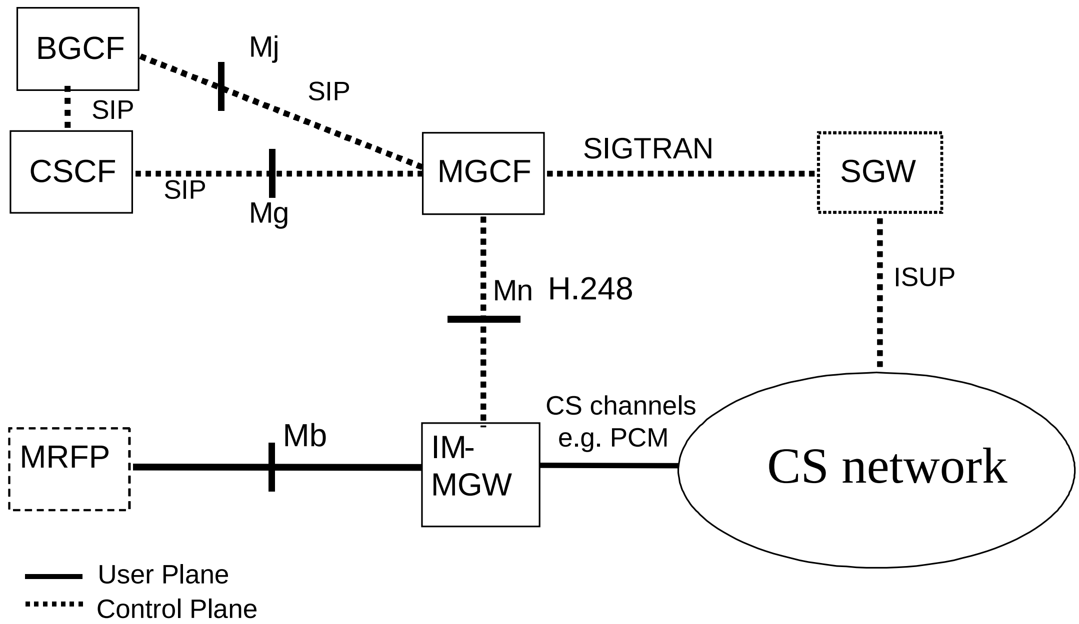
Inter working

4.1 **PSTN/PLMN Inteworking**

Session Initiation Protocol (SIP) shall provide the signaling conversion between the signaling protocols used in support inter working with PSTN/PLMN for Integrated Services Digital Network (ISDN) User Part (ISUP) of CCS7 to support basic bearer services and supplementary services for voice and non voice applications. The Media Gateway Controller (MGC) shall implement the Media Gateway Controller Function (MGCF) for PSTN/PLMN nterworking.

4.2 The PSTN/PLMN inter working function shall the PSTN/PLMN networks and the session control protocols used in the IP Multimedia subsystem networks – as well as the conversion between the circuit switched PSTN/PLMN bearer circuits and the packet switched VoIP media streams used in IMS networks.

4.3 Figure – 4 shows the ISUP-SIP Inter working for VoIP. The MGC shall provide the SIP based Mg and Mj reference points for call signaling and the H.248 based Mn reference point for controlling the media resources in the MGW [3GPP TS 29.163]. The signaling protocol supported towards the PSTN/PLMN network is ISUP.



BGCF - Breakout Gateway Control Function
CSCF - Call Session Control Function
MGCF - Media Gateway Control Function
SGW - Signaling Gateway
CS - Circuit Switched
IM-MGW - IP Multi Media – Media Gateway
MRFP - Multimedia Resource Function Processor

Figure – 4 ISUP-SIP Inter working

4.4 The ISUP and SIP inter working and Bearer Independent Call Control (BICC) and SIP inter working support in MGC shall be based on the ITU-T recommendation Q.1912.5.

4.5 The Q.1912.5 profile B and C operations shall be supported. The profile B defines the ISUP and SIP interworking with normal SIP protocol. The profile C defines the ISUP transparency over SIP by adding the MIME encapsulated ISUP messages in SIP messages.

4.6 The following traffic cases shall be supported:

(i) ISUP to SIP call

(ii) ISUP to SIP call, call establishment with ENUM

(iii) ISUP to SIP call, with call redirection with SIP 30x response

(iv) ISUP to SIP T38 fax call

(v) SIP to ISUP call

(vi) SIP to ISUP T38 fax call

(vii) ISUP and SIP inter working, call modification initiated by SIP (re-INVITE)

(viii) ISUP and SIP inter working, call Hold/Retrieval

(ix) ISUP and SIP inter working, emergency call

(x) ISUP and SIP inter working, call release

(xi) ISUP to ISUP by trunking the call over IP network using SIP

4.7 SIP shall also support the following when ISUP is transported over the IP network using SIGTRAN as per TEC document **“**National Standards for SIGTRAN” (SD/SGT-01):

4.7.1 ISUP Supplementary Services as described in Chapter – 4 (ISUP Supplementary Services) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.7.2 ISUP capabilities relating to Basic call and Supplementary Services as described in Clause 5.1 (Functional Description of ISUP) of Chapter – 5 (ISDN User Part) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.7.3 Messages, parameters and parameter information used by ISUP as described in Clause 5.2 (Messages and Parameters) of Chapter – 5 (ISDN User Part) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.7.4 Formats and codes of ISUP messages and the parameters required to support basic bearer services and the supplementary services as described in Clause 5.3 (Format and Codes) of Chapter – 5 (ISDN User Part) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.7.5 The ISUP signaling procedures for setting up and clearing down of national and international ISDN connections as per Clause 5.4 (Signaling Procedures) of Chapter – 5 (ISDN User Part) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.7.6 The Charging procedures for CCS7 as per Chapter – 6 (Charging Procedure) of TEC document No. S/CCS-02 (National CCS 7 Standards for MTP and ISUP).

4.8 SIP shall also inter work with IN Network by supporting messages as per TEC documents National SCCP standards for Large Digital Switching Systems (S/CCS-03), National TCAP Standards (SD/CCS-05) and Intelligent Network Application Protocol (INAP) National Standards (SD/INP-01)

4.9 SIP shall inter work with Universal Mobile Telecommunication System – Core Network by supporting TEC document Universal Mobile Telecommunication System – Core Network (Release 4) (GR/UCN-01).

4.10 SIP shall also inter work with IP Multimedia Sub-System (IMS) by supporting TEC document IP Multimedia Sub-System (GR/IMS-01).

4.11 **SIP – ISUP Mapping**

SIP shall support mapping of following ISUP parameters (With specific Cause code /event code) into specific SIP response codes as specified in Table-3. ISUP messages are mapped into SIP messages and vice versa by the MGC. For mapping of remaining ISUP parameters as mentioned in clause 4.7 of this document solution will be provided by Vendors.

Table -3

A. Mapping of ISUP messages into SIP INVITE responses			
S.No.	**ISUP Message**	**SIP Response Code**	**Remarks**
1.	ACM	180 Ringing 183 Session Progress	When ACM BCI (Backward Call Indicator) is 'Subscriber free'. For other values of ACM BCI.
2.	CPG	180 Ringing 183 Session Progress	For CPG Event 'Alerting' For other values of CPG Event.
3.	ANM	200 O.K.	
4.	CON	200 O.K	
5.	REL	4xx/5xx/6xx	
	1 unallocated number	404 Not found	
	2 no route to network	404 Not found	
	3 no route to destination	404 Not found	
	17 user busy	486 Busy here	
	18 no user responding	408 Request Time Out	
	19 no answer from the user	480 Temporarily unavailable	

	20 subscriber absent	480 Temporarily unavailable	
	21 call rejected	403 Forbidden	
	22 number changed (w/o diagnostic)	410 Gone	
	23 redirection to new destination	410 Gone	
	26 non-selected user clearing	404 Not Found	
	27 destination out of order	502 Bad Gateway	
	28 address incomplete	484 address incomplete	
	29 facility rejected	501 Not implemented	
	31 normal unspecified	480 Temporarily unavailable	
	34 no circuit available	503 Service unavailable	
	38 network out of order	503 Service unavailable	
	41 temporary failure	503 Service unavailable	
	42 switching equipment congestion	503 Service unavailable	
	47 resource unavailable	503 Service unavailable	
	55 incoming calls barred within CUG	403 Forbidden	
	57 bearer capability not authorized	403 Forbidden	
	58 bearer capability not presently available	503 Service unavailable	
	65 bearer capability not implemented	488 Not Acceptable Here	
	70 only restricted digital avail	488 Not Acceptable Here	
	79 service or option not implemented	501 Not implemented	
	87 user not member of CUG	403 Forbidden	

	88 incompatible destination	503 Service unavailable	
	102 recovery of timer expiry	504 Gateway timeout	
	111 protocol error	500 Server internal error	
	127 interworking unspecified	500 Server internal error	
	Other cause code	500 Server internal error	
B. Mapping of ISUP messages into SIP requests			
S.No.	**ISUP Message**	**SIP Request Method**	**Remarks**
1.	IAM	INVITE	
2.	REL	BYE	
3.	REL	CANCEL	(if no final response for the initial INVITE request has been received)
C. Mapping of SIP requests into ISUP messages			
S.No.	**SIP Request Method**	**ISUP Message**	**ISUP cause code**
1.	INVITE	IAM	N/A
2.	CANCEL	REL	16 “Normal clearing”
3.	BYE	REL	16 “Normal clearing”
D. Mapping of SIP INVITE responses into ISUP messages			
S.No.	**SIP response code**	**ISUP Message**	**Remarks**
1.	180/181/182/183	ACM	
2.	180	CPG (If ACM has been sent)	CPG Event Information “Alerting”
3.	181/182/183	CPG (If ACM has been sent)	CPG Event Information “Progress”
4.	200 OK	ANM	
5.	200 OK	CON	(if no ACM/CPG message(s) have been sent earlier during the session establishment)
6.	4xx/5xx/6xx	REL	AS mentioned below
	400 Bad Request	41 Temporary Failure	
	401 Unauthorized	21 Call rejected	
	402 Payment required	21 Call rejected	
	403 Forbidden	21 Call rejected	
	404 Not found	1 Unallocated number	

	405 Method not allowed	63 Service or option unavailable	
	406 Not acceptable	79 Service/option not implemented	
	407 Proxy authentication required	21 Call rejected	
	408 Request timeout	102 Recovery on timer expiry	
	410 Gone	22 Number changed (w/o diagnostic)	
	413 Request Entity too long	127 Interworking unspecified	
	414 Request-URI too long	127 Interworking unspecified	
	415 Unsupported media type	79 Service/option not implemented	
	416 Unsupported URI Scheme	127 Interworking unspecified	
	420 Bad extension	127 Interworking unspecified	
	421 Extension Required	127 Interworking unspecified	
	423 Interval Too Brief	127 Interworking unspecified	
	480 Temporarily unavailable	18 No user responding	
	481 Call/Transaction Does not exist	41 Temporary Failure	
	482 Loop Detected	25 Exchange - routing error	
	483 Too many hops	25 Exchange - routing error	
	484 Address incomplete	28 Invalid Number Format	
	485 Ambiguous	1 Unallocated number	
	486 Busy here	17 User busy	
	487 Request Terminated	31 Normal unspecified	(no mapping if request was cancelled)
	488 Not Acceptable	31 Normal unspecified	

	here		
	500 Server internal error	41 Temporary failure	
	501 Not implemented	79 Not implemented, unspecified	
	502 Bad gateway	38 Network out of order	
	503 Service unavailable	41 Temporary failure	
	504 Server time-out	102 Recovery on timer expiry	
	505 Version Not Supported	127 Interworking unspecified	
	513 Message Too Large	127 Interworking unspecified	
	600 Busy everywhere	17 User busy	
	603 Decline	21 Call rejected	
	604 Does not exist anywhere	1 Unallocated number	
	606 Not acceptable	31 Normal unspecified	
	Other response codes	31 Normal Unspecified	

4.12 SIP response codes

On Inter working with PSTN/PLMN, SIP shall support at least following response code classes and SIP response codes as mentioned in Table – 4. These response codes will be generated by MGC.

Table – 4

S.No.	Response Code Class and Meaning	Response Code	Response Code Meaning
1	1xx Provisional	100	**Trying**. This response is sent to an INVITE request whenever there is not immediate response available.
2		180	**Ringing.** This response is sent when MGC is alerting the endpoint.
3		183	**Session Progress**. This response is sent when audio has to be sent before the phones are connected.
4	2xx Successful	200	**OK.** This response indicates that User B has accepted the call initiated by User A.
5	3xx Redirection	300	Multiple Choices.
6		301	Moved Permanently
7		302	Moved Temporarily
8		305	Use Proxy
9		380	Alternate Service
10	4xx Request Failure	400	**Bad Request**. This response is sent when the request could not be understood due to malformed syntax.
11		404	**Not Found**. This response is sent when the called number can not be found in the database.
12		408	**Request Time out**. This response is sent when the request could not be resolved within a suitable amount of time.
13		413	**Request Entities Too Large**. This response is sent when the request could not be resolved because of the request's entity body being larger than expected.
14		414	**Request URI Too Long**. This response is sent when the request URI is longer than expected.
15		415	**Unsupported Media Type**. This response is sent when the request's body is in a format that is not supported.
16		416	**Unsupported URI Scheme**. This

			response is sent when the request can not be processed because of the scheme of the URI in the Request-URI being unknown to the server.
17		420	**Bad Extension**. This response is sent when a SIP request with a Require header containing unsupported extension tag is received.
18		421	**Extension Required**. This response is sent when a SIP request not indicating support for a SIP extension, within a Require or Supported header, required to process the request is received.
19		423	**Interval Too Brief**. This response is sent when the request is rejected because the expiration time of the resource refreshed by the request is too short.
20		480	**Temporarily not available.** This response is sent when the phone is temporary unavailable to receive call.
21		481	**Call Leg / Transaction Does Not Exist.** This response is sent when a SIP request is received for an unrecognized session.
22		483	**Too Many Hops.** This response is sent when an INVITE request is received containing a Max-Forward header with a zero value.
23		486	**Busy Here.** This response is sent when the called party is busy.
		487	**Request Cancelled.** This response is sent for an INVITE request if a CANCEL request is received for the INVITE request and no final response has yet been generated for the INVITE request.
24		488	**Not Acceptable Here.** This response is sent when a request body contains session description that is not acceptable by the receiving MGC.
25		491	**Request Pending.** This response is sent when the request is received by a UAS that had a pending request within the same dialog.
26		493	**Undecipherable.** This response is sent when the request received by a UAS

			contained an encrypted MIME body for which the recipient does not possess or will not provide an appropriate decryption key.
27	5xx Server Error Responses	500	**Internal Server Error.** This response is sent when there is some kind of error at the UAS side.
28		501	**Not Implemented.** This response is sent when a SIP request is received with an unsupported SIP method.
29		503	**Service Unavailable.** This response is sent when the server is temporary unable to process the request.
30		504	**Server Timeout.** This response is sent for an INVITE/re-INVITE request if MGC is not able to generate another response within a specific time.
31		505	**Version Not Supported.** This response is sent when a SIP request is received with a version number other than ”2.0” in the request line.
32	6xx Global Failures	603	Decline
33		604	Does not exist anywhere
34		606	Not Acceptable

4.13 **SIP Call to PSTN/PLMN Subscriber**

Connections to the PSTN/PLMN are made through SIP based PSTN/PLMN gateway controller (e.g. MGC) that translates SIP messages into PSTN/PLMN message formats. A simple call flow from SIP phone to PSTN/PLMN subscriber along with relevant message conversion as per IETF RFC 3398 is shown in Figure – 5 and described briefly below.

User 1 is a user of SIP phone or any other SIP enabled device. User 2 is a PSTN/PLMN subscriber. User 1 dials the E.164 number of User 2 to reach it. User 1 can use either his SIP address or SIP telephone number in the INVITE message. Following steps are involved in the call setup.

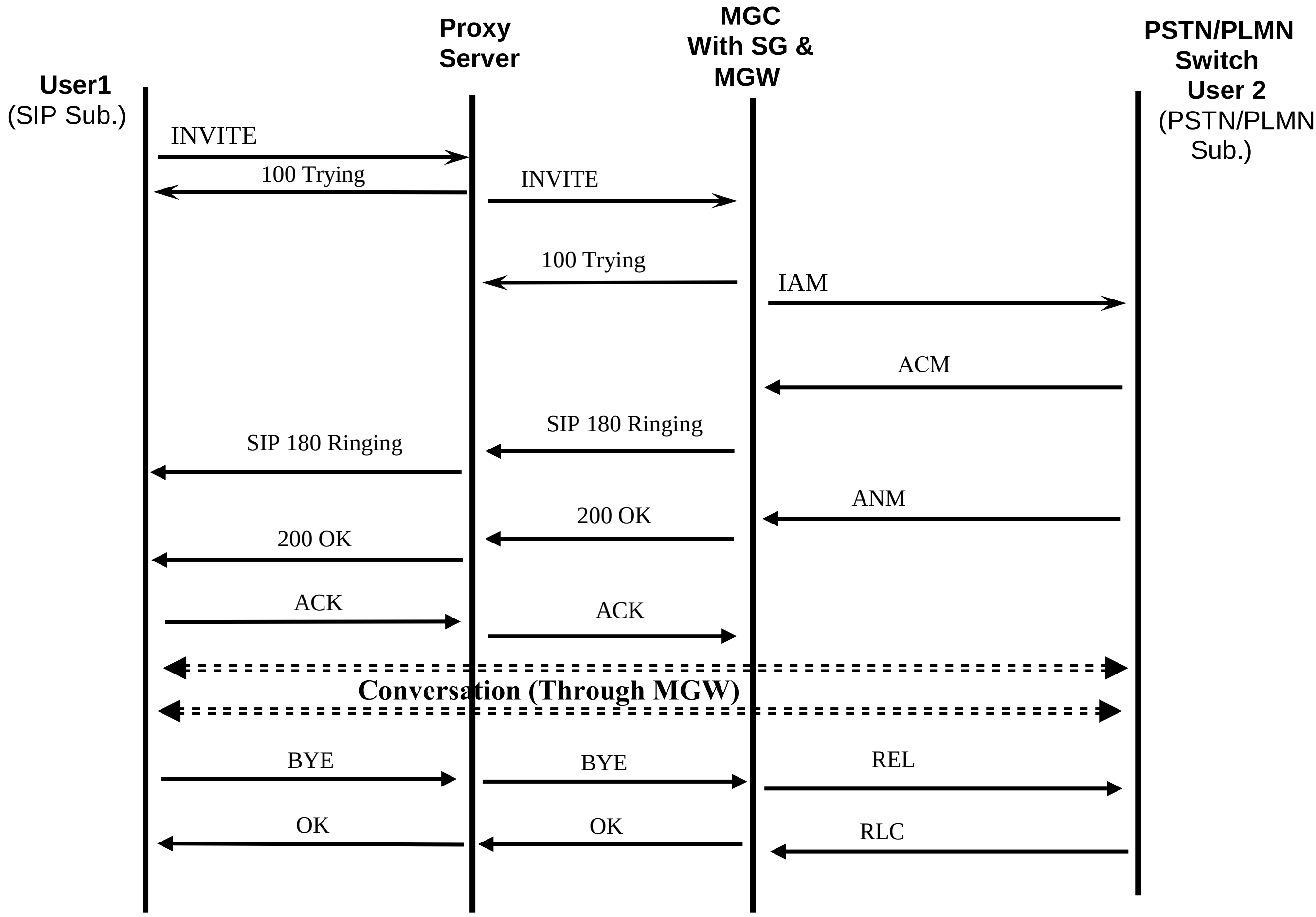


Figure 5: SIP Call to PSTN/PLMN Subscriber

i). User1 places the call to a Proxy Server, which is used to route the call. On receiving INVITE message, Proxy Server sends a ‘100 Trying’ response message to User1, indicating that it is trying to establish the call.

ii). Proxy Server uses a Location Server to determine the PSTN/PLMN MGW IP address for terminating the call.

iii). The Location Server returns PSTN/PLMN MGW IP address to the Proxy Server.

iv). Proxy Server forwards the call to the MGC by sending SIP INVITE message.

v). MGC returns ‘100 Trying’ response to the proxy server, indicating that it is trying to establish the call. At the same time it sends ISUP IAM (Initial Address Message) to the PSTN/PLMN switch. The calling and the called numbers are included in the IAM message.

vi). After receiving all the digits, the PSTN/PLMN switch replies with ISUP ACM (Address Complete message). This signals MGC that the switch has received all the digits and is processing the call.

vii). MGC sends ‘180 Ringing’ response to User1 via Proxy Server, indicating that ring is going to User2.

viii). When User2 answers the call, the PSTN/PLMN switch sends an ISUP ANM (Answer Message) to MGC.

ix). MGC responds with SIP ‘200 OK’ message to Proxy Server to indicate that the called subscriber has answered the call. In response to ‘200 OK’ message, SIP acknowledgement response, ACK, is sent to confirm that a final response to an INVITE request as been received.

x). A bi-directional communication path is established between the calling and the called subscriber via MGW.

xi). To end the call by User1, a BYE message is sent to Proxy Server. Proxy Server sends the BYE message to the MGC. MGC replies back with ‘OK’ message, acknowledging the receipt of call release request.

xii). MGC releases the RTP communication path that was being used for communication with User1 and signals the call release to the PSTN/PLMN switch by sending an ISUP REL (Call Release Message). The PSTN/PLMN Switch releases the call and replies with ISUP RLC (Release Complete Message).

Call from PSTN/PLMN to SIP phone will have similar steps in opposite direction.

Abbreviations

3GPP	3^{rd} Generation Partnership Project
AS	Application Server
B2BUA	Back-to-Back User Agent
BGCF	Breakout Gateway Control Function
CN	Core Network
CS	Circuit Switched
CSCF	Call Session Control Function
DNS	Domain Name System
DOT	Department of Telecommunication
GR	Generic Requirements
I-CSCF	Interrogating-CSCF
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
INAP	Intelligent Network Application Protocol
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	Integrated Services Digital Network User Part
ITU	International Telecommunication Union
LMG	Line Media Gateway
MGCF	Media Gateway Control Function
MGC	Media Gateway Controller
NGN	Next Generation Network
PC	Personal Computer
P-CSCF	Proxy-CSCF
PGW	Public Switched Telephone Network (PSTN) Gateway
PLMN	Public Land Mobile Network
POTS	Plain Old Telephone System
PSTN	Public Switched Telephone Network
RFC	Request For Comment
RTP	Real-time Transport Protocol

RTSP	Real-time Streaming Protocol
S-CSCF	Serving-CSCF
SC	Session Controller
SCCP	Signaling Connection Control Part
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGW	Signaling Gateway
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TCAP	Transaction Capabilities Application Part
TEC	Telecom Engineering Centre
TMG	Trunk Media Gateway
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
URL	Uniform Resource Locator