Accessing Traditional Intelligent Services From SIP Network

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Abstract: Technical analysis about accessing IN services from SIP VoIP network is addressed, some key technical issues, such as call states mapping and Messages conversion schemes, are presented. A new functional entity for accessing IN services from VoIP network -- Intelligent Services Gateway (ISG) -- is introduced.

Key Words: IP Telephony, Intelligent Services, IN/IP Integration, SIP, Intelligent Service Gateway

1 Introduction

With the wider acceptance and deployment of IP telephony, both carriers and users expect that some value-added services also could be provided in Internet as it does in the traditional telecommunication networks. SIP [1] and H.323 [2] (proposed by IETF and ITU-T respectively) are the de facto standards in the IP telephony industry. But neither SIP nor H.323 could fully meet the requirements of supporting intelligent services in IP networks currently, so reuse of telecommunication intelligent network architecture and service capabilities is a better shortcut to match this purpose rapidly. [3-7]

This paper addresses some technical analysis about accessing IN services from SIP network. A new functional entity -- Intelligent Services Gateway (ISG) in the IN/IP integration architecture is proposed. ISG is a functional entity in the distributed intelligent service architecture and performs a services gateway function between two heterogeneous networks. The main advantage of using ISG is that the service capabilities of both networks are combined and utilized efficiently while the integrity of telecommunication IN architecture and Internet Telephony architecture is still kept.

The remainder of the paper is organized as follows. In Section 2, the SIP IN functional architecture is reviewed and a kind of Soft-Switch – Intelligent Service Gateway is presented. In Section 3, SIP protocol is brief described first, then a detailed analysis of the relationship between IN Basic Call State Model (BCSM) and SIP telephony call model is given. In Section 4, the function of the ISG entity and Intelligent Service Application Protocol are discussed, and a simple free-phone service example is described. Finally, a conclusion is given out in Section 5.

2 SIP IN Architecture

The generalized Internet intelligent network architecture can be illustrated as shown in Figure 1[8]. There are several ways to access the intelligent services from the Internet, they can be divided into two categories, the so-called script approach and the transparent approach. The main idea of the script approach is to run scripts directly on a call control element (CCE) for both simplicity and performance reasons. Currently, within the script approach there are two possibilities: Call Processing Language [9] and Common Gateway Interface (CGI)[10]. CPL is a kind of conditions-actions pair type language and its script is associated with particular telephony addresses normally. CPL scripts will be typically used to replace the user location functionality of a SIP proxy. The SIP CGI provides an interface and set of primitives for implementing services on SIP servers. CGI is more powerful but less efficient than CPL. On the other hand, the main idea of the transparent approach is to get some traditional IN services from PSTN transparently. Access to the SCP services may be accomplished with either TCAP/INAP messages through

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a Soft-Switch or via an API (JAIN [11] Parlay [12] or other APIs) through a transaction entity. The suggested architecture of generalized Internet intelligent services provides both the transparent support of the traditional IN services from SIP networks and the script based services logic.

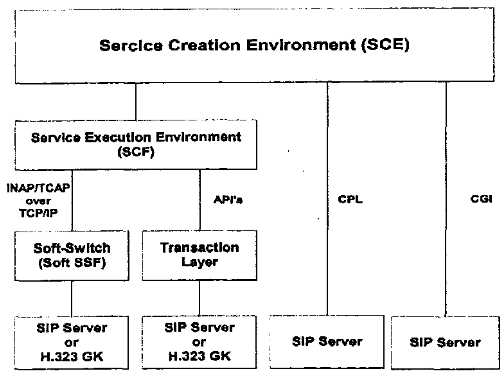


Figure1. Generalized Internet IN architecture (adapted from [8])

This paper addresses mainly the accessing of traditional IN services transparently from PSTN to SIP network users. There are several reasons to access traditional IN services from SIP telephony networks: firstly, VoIP network and PSTN will coexist for a long time, service inter-workings between heterogeneous networks is an inevasible trends; secondly, intelligent service requirements from the SIP telephony network have emerged and some of these intelligent services could obtained from traditional PSTN; thirdly, reuse of the IN services in the Internet can reduce the network investments and speed up significantly the intelligent services deployment procedure.

The proposed functional architecture of the transparent approach is shown in Figure 2 [13], the main idea behind it is to reuse the existing IN services architecture for SIP VoIP networks. There are some papers addressing the SIP proxy functional extension and call states mapping for accessing traditional IN services architecture [14-16]. Similar to the functional extension in H323 Gatekeeper, the addition of the Soft-SSF function to perform the switches simulation is a key technology. But all the proposals in these papers only cover the single VoIP Protocol. Considering that SIP and H.323 are the de fact standards in VoIP networks, we propose a universal functional entity – ISG to perform state mapping and parameters conversion of IN services. ISG is a functional entity that can perform CCF/SSF function in IP networks. It could be combined with a SIP Proxy or H.323 GateKeeper or be a stand-alone entity working together with Proxies/GateKeepers [17]. The main advantages of using this universal functional entity are that the service capabilities of both networks are combined and utilized efficiently while the integrity of telecommunication IN architecture and Internet architecture is kept. Both SIP and H323 users can access IN services through this universal intelligent services gateway.

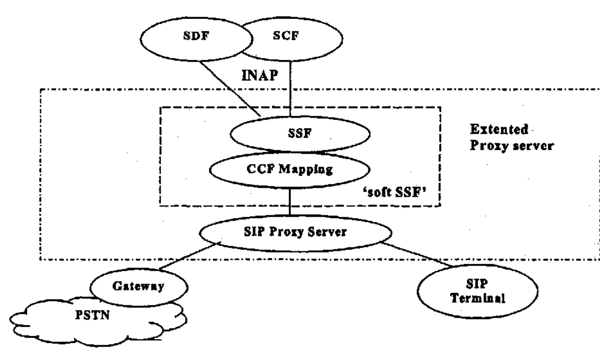


Figure2. Functional Architecture of IN/IP Integration for IN CS-4 to Support SIP Systems (From [13])

3 SIP Call Model supporting IN/IP Integration

3.1 SIP call messages

SIP, as proposed by IETF, is used in the IP telephony architecture as an alternative to ITU-T H.323 umbrella recommendations since it is simpler to implement. It also can be compared to ITU-T H.225 (Registration, Admission, Status (RAS)), and Q.931 messages. SIP message headers are defined in plain text and look similar to Email headers. The Session Description Protocol (SDP), which is used by SIP, can be compared to H.245 (channel open/close and terminal capability handling). Session descriptions have a list format containing information about the session.

SIP uses a client/server model similar to the Hypertext Transfer Protocol (HTTP). It is used in conjunction with other protocols such as SDP, RTP and RSVP. SIP could establish connections via TCP or UDP.

To initiate a session, a client sends an INVITE message to a server. An INVITE message typically contains a sufficient session description in SDP to establish communication. SIP Request and Response messages are listed below.

<u>SIP Request Messages</u>	<u>SIP Response Messages</u>
INVITE – invite client or server to establish a session	1xx Informational
ACK – a final confirmation following response to an INVITE message	2xx Success
BYE – close the session	3xx Redirection
CANCEL – cancel a pending request	4xx Client Error
OPTIONS – ask for info about capabilities before a session	5xx Server Error
REGISTER – inform a location server of the client's IP address	6xx Global Failure

SIP response messages use a 3 digits number. The first digit defines the category. The next two digits allow up to 100 variations, e.g. 200 OK means a successful invitation.

3.2 SIP Call States and procedures

In SIP based systems, an extended SIP proxy with call control intelligence is defined for inter-working with IN architecture. This intelligence will enable nominated SIP proxies to retain significant call control state. It should be able to synchronize the SIP Call State model with the IN Basic Call State model defined in ITU-T related standards. Normally, this SIP proxy is comprised of both a client and a server. It is required to analyze which IN BCSM states have the meaning in a SIP based service context, how could they be added to SIP call model and understood in the extension of the IN call control model.

According to the specifications of SIP [1], we can get the call states machines for SIP Client and SIP Server respectively [18]. Figure 3 shows a typical SIP phone call establishment and clearing sequence between two endpoints with a SIP Proxy transferred call signaling, together with the associated SIP Server/Client call control states.

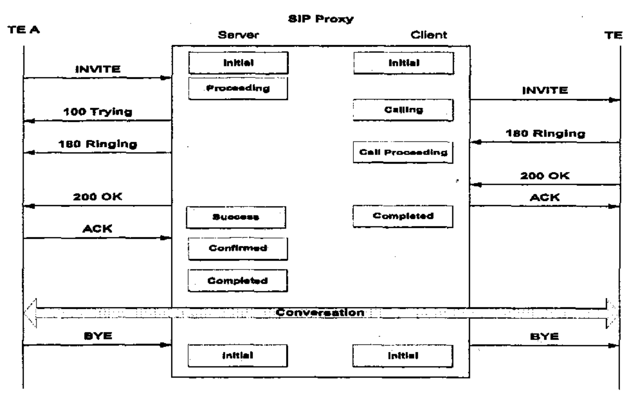


Figure 3. SIP Call Control Signaling and States

3.3 *Mapping of IN BCSM with SIP Call States*

In order to minimize the affection of both IN services architecture and SIP telephony architecture, the better call states integration method is to overlay IN BCSM model with the SIP call control states machine. In this approach, each intelligent service call control procedure is processed synchronously both by IN call model and IP telephony call model. Original IN services could be provided to Internet users by the coordinated work of both call models. Obviously, this overlaid solution could allow IN services deployment rapidly in Internet while the integrity of the original systems is still kept.

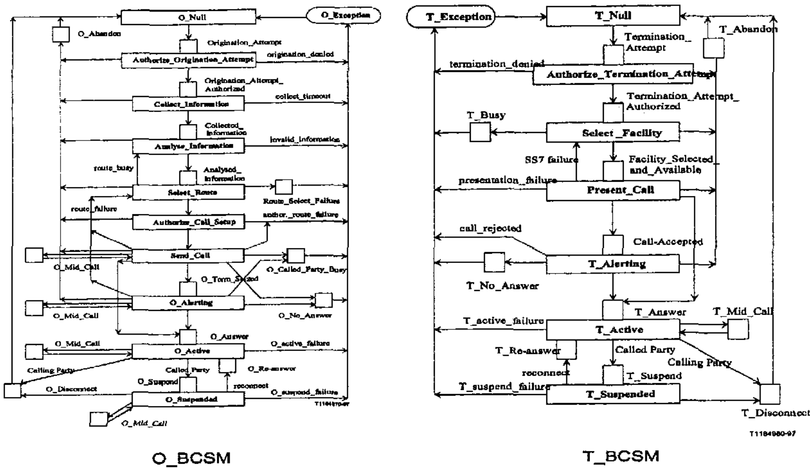


Figure 4. IN Basic Call State Model (BCSM) (from [7])

Although there are some differences between the circuit switched network and IP network, and current IN BCSM were designed for the former environment from the beginning, call states mapping is still possible between IN BCSM and IP telephony. Call states mapping between IN BCSM and H.323 systems is described in [7] and implemented in [17]. Based on IN O/T BCSM (Figure 4) and SIP telephony call

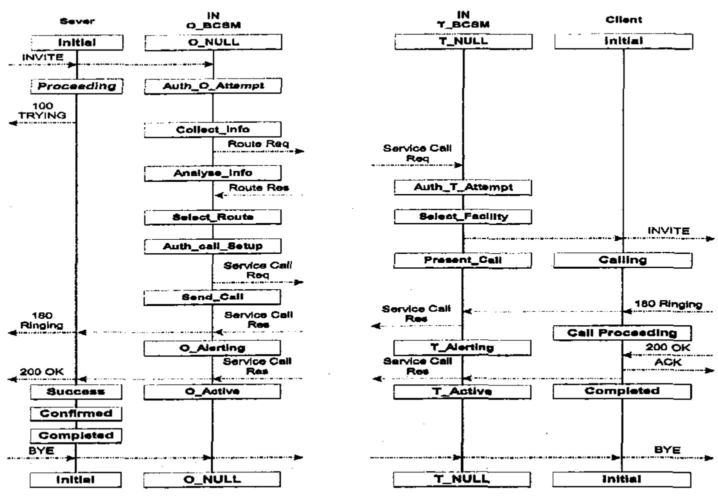


Figure 5. Mapping of SIP call states and IN BCSM

flows, we can also get the state mapping relationship between SIP call states and IN BCSM states as the Figure 5. From server side of SIP proxy point of view, its **Initial** state could be corresponded to the

O_NULL PIC (Point in Call) of the O_BCSM. Its **Processing** state could be corresponded to the **Authorize_Origination_Attempt, Collect_Information, Analyse_Information, Select_Route, Authorize_Call_Setup, Send_Call** and **O_Alerting** PICs. Its **Success, Confirmed** and **Complete** States could be corresponded to the **O_Active** PIC. From client side of SIP proxy point of view, its **Initial** state could be corresponded to the **T_NULL, Authorize_Termination_Attempt, Select_Facility** PICs of the T_BCSM. Its **Calling** state could be corresponded to the **Present_Call** PIC. Its **Call Proceeding** state could be corresponded to the **T_Alerting.** Its **Complete** State could be corresponded to the **T_Active** PIC. It should be noted that an interface operation between a SIP Proxy and a SCP is necessary to achieve inter-working of SIP with IN services. The detailed mapping procedure and call flows are described in [16].

4 Accessing Intelligent Service from SIP Network

Normally, extend the functionality of SIP proxy with intelligent services capabilities is the first idea of deployment intelligent services in SIP network. But this scheme will deeply impact the implementation and deployment of SIP systems. Considering that another popular IP telephony architecture -- H.323 system could also meet the same problem, extending the functionality of SIP proxy or H.323 GateKeeper separately maybe is not the most suitable solution. Based on this consideration, we have proposed a unified functional entity –Intelligent Service Gateway (ISG) to perform the states and messages mapping functions for both systems.

4.1 Intelligent Services Gateway (ISG)

Normally, gateway can provide services concatenation function to both connected networks. Each network could provide some new services, which is belonging to another network connected by gateway, to its users. There are some basic principles and methods to construct a gateway between the heterogeneous networks, Direct Concatenation, Interface Adaptation and Service Adaptation are the three different concatenation method discussed in [19]. Service Adaptation is the easiest way to build a gateway for inter-working between different protocol hierarchies in the heterogeneous network environment. This method can be used as a basic principle to construct our ISG.

ISG is a functional entity that provides telephony services concatenation functions between IP networks and Intelligent Network (IN). It implements mapping of IN service call model and the IP telephony native protocol state machine (SIP system or H.323 system). The FSM of ISG service call control entity and IN BCSM are integrated together according to the descriptions in Section 3.3. ISG also performs message and parameter conversion between different call control protocols. This conversion is performed at service level above the both protocol hierarchies. So most SIP proxies/H.323 GateKeeper, which have nothing to do with the IN services, need not to change their protocol architectures. Only those IP telephony call control entities that located in special positions and connected with PSTN IN service gateways need to extend some communication protocol entities for interaction of IN services call control messages with ISGs.

The ISG is the only entity performing IN/IP states mapping function and communicating with SCP in PSTN. Working with Service Control Gateway (SCG)/SCP, it can provide the IN services control logic by using an enhanced INAP/TCAP protocol over IP networks. Acting as an independent logic functional entity, ISG is a universal gateway to IP network, it can communicate both with SIP and H.323 IP telephony call control entities through a proprietary standard protocol over IP networks. This proprietary protocol can support the IN service requests from Internet users through IP telephony call control entities. This API based protocol is defined as Intelligent Services Application Protocol (ISAP).

4.2 Intelligent Services Application Protocol (ISAP)

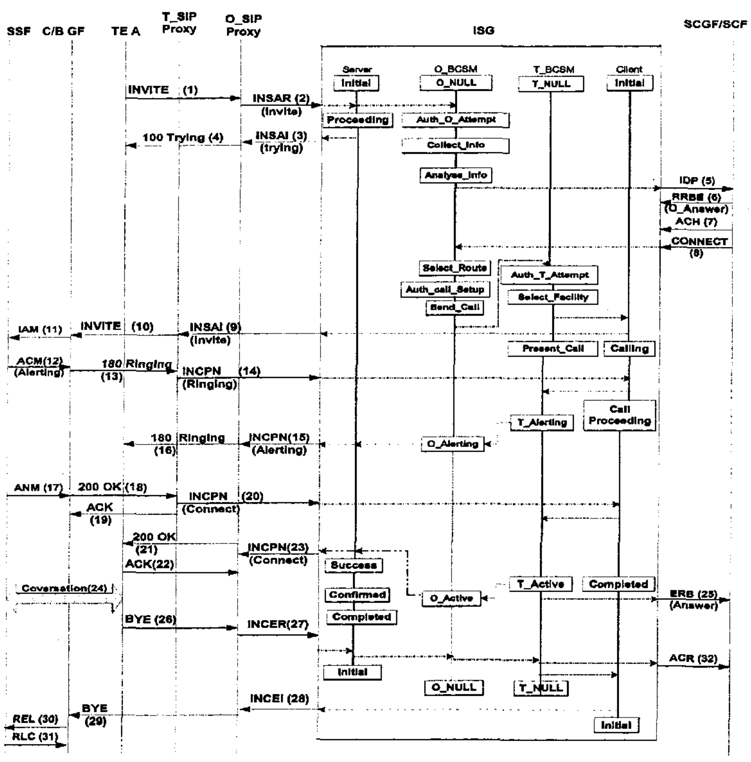
The ISAP must be able to carry sufficient call related data from the Soft SSP and deliver this necessary information to the SCG/SCP, so that the service logic decisions can be made. It must allow the SCG/SCP (in combination with ISG) to control intelligent VoIP calls and manipulate call information.

SIP describes the call control procedures of SIP Client/Server/Proxy while ITU-T H.323 umbrella protocols describe the call control procedures of GateKeeper. According to these specifications, we have defined the state machine of service call control model in ISAP. There are 21 messages defined in ISAP, the names and functions of these messages are listed in Table 1.

Table 1. ISAP messages

Name	Function	Name	Function
INSAR	Call setup report	INCPI	Call Progress indication
INSAI	Call setup indication	INCER	Call end report
INSRI	Call reject indication	INCEN	Call end notification
INOMR	IN messages report	INCEI	Call end indication
INOMN	IN messages notification	INERQ	ISE register request
INOMI	IN messages indication	INERF	ISE register confirm
INSFR	Call failure report	INERJ	ISE register reject
INFAI	IN facility indication	INEDQ	ISE disengage request
INFAN	IN facility notification	INEDF	ISE disengage confirm
INFAR	IN facility report	INEDJ	ISE disengage reject
INCPN	Call Progress Notification		

ISAP is used as a mediation protocol between IP telephony call control protocols (SIP protocol or H.323 series recommendations) and IN services control protocol (INAP). Not only the states mapping functions but also the message and parameter conversion functions is performed by it. Neither IP telephony call control protocols nor IN services control protocols are needed change too much for IN/IP integration. Here, we take Free-phone services as an example to describe how does the ISAP work in our IN/IP integration architecture. Figure 6 illustrates the message flow sequence and the state mapping between SIP and IN BCSM in ISG according to a normal call example in a simple Free-phone service.



Note: INAP Operations: IDP – Initial DP, RRBE – Request Report BCSM Event, Connect, ACH – Apply Charging, ACR – Apply Charging Report, ERB – Event Report BCSM

Figure 6. Free-phone services. Normal call between SIP and GSTN users

5 Conclusion

After introducing SIP IN architecture, some key issues of SIP call model supporting IN/IP integration are analyzed. An Intelligent Services Gateway (ISG) for accessing IN service from SIP VoIP network is proposed to meet users' increasing demands. The architecture integrity of both PSTN IN services and IP telephony service could be kept by introducing ISG. One of the IN's benchmark services – Free-phone service – is used to illustrate our design scheme.

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