SIP and Application Internetworking

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Abstract

Session Initiation Protocol (SIP) is an application level signaling protocol used in Internet telephony architecture. Though its primary role is to initiate sessions, SIP is wide used for signaling. This article discusses internetworking between SIP and different networks as well as the application internetworking.

1 Introduction

A significant feature of current Internet is provision of telephony, called Internet telephony or IP telephony, also known as voice over IP (VoIP). It handles transmitting voice information over packet networks instead of traditional PSTN. There are two competing protocols used for signaling and control for Internet telephony. One is ITU-T H.323, the other is Session Initiation Protocol (SIP) developed by IETF. The former embraces the more traditional circuit-switched approach on ISDN, the latter favors the more lightweight Internet approach based on HTTP. Therefore, SIP is gaining increasing popularity as well as due to its flexibility of internetworking.

SIP combines the features of the Advanced Intelligent Network from the telecom world for fixed and mobile telephony, with Internet features for email, Web, transactions, and entertainment [1]. This paper discusses both signaling and application internetworking of SIP with different networks. Firstly it involves the architectures, mechanisms or protocols when internetworking with PSTN and mobile networks as well as extending the Internet with the intelligent network capabilities. Then applications internetworking of SIP are briefly introduced via the examples of Real-time Fax and video conferencing. All of study base on IETF's SIP RFC, Internet drafts and papers published in IEEE journals or conferences.

1.1 Abbreviations

3A Authentication, Authorization and Accounting
3GPP 3rd Generation Partnership Project
BCSM Basic Call State Model

BSP Billing Service Provider
CCF Call Control Function
CS1 Capability Set number one
DMIF Delivery Multimedia Integration Framework
IETF Internet Engineering Task Force
IN Intelligent Network
ISDN Integrated Services Digital Network
ISUP ISDN User Part
ITU-T International Telecommunication Union
MIME Multipurpose Internet Mail Extensions
MG Media Gateway
MGC Media Gateway Controller
MTP Message Transfer Protocol
PSTN Public Switched Telephone Network
QoS Quality of Service
RFC Request For Comments
RTCP RTP Control Protocol
RTP Real-Time Transport Protocol
SCF Service Control Function
SCP Service Control Point
SDP Session Description Protocol
SG Signaling Gateway
SIP Session Initiation Protocol
SS7 Signaling System No.7
SSF Service Switching Function
SSP Service Switching Point
UA User Agent
UAC User Agent Client
UAS User Agent Server
URI Uniform Resource Indicator
VoIP Voice over IP

1.2 Article Structure

In the rest of this article, section 2 is overview of SIP, which involves the SIP components, SIP-enabled network architecture, methods and common functions. Then we focus on SIP's various signaling inter- networking with PSTN, IN and Mobile IP in section 3. Inside, firstly subsection 3.1 gives the PSTN-SIP gateway architecture and present briefly how to map functions and messages between SIP and ISUP. Sequentially, subsection 3.2 introduces the extended SIP proxy server with IN service function, as well the integration of IN call model and SIP call model. Last part of section 3 is about mobile IP background and SIP support for four modes of application-layer mobility. And then section 4 describes two valuable applications internetworking based on SIP, and billing issue is talked briefly as the final point. In the end, we conclude in section 5.

2 Background of SIP

2.1 Basic Architecture

A SIP-enabled IP communication network is composed of *SIP servers* and *SIP endpoints (User Agents)* [2].

User Agents (UA)
UA may be user devices or gateways to other networks. Since SIP is a client-server protocol, each UA contains both a User Agent Client (UAC) and a User Agent Server (UAS). The former initiates requests, and the latter generates responses to received requests. This differs from other architectures, such as Web browsing.

SIP server
As intermediary devices, they assist UAs in establishment and other functions. The fundamental types of them are shown in Figure 1.

- **SIP proxy** forwards or proxies the request from a UA or proxy to another location.
- **Redirect server** receives a request from a UA or proxy and returns a redirection response, indicating where the request should be retried.
- **Registrar server** receives SIP registration requests and updates the UA's info into a location server or other database.
- **Location server** is a database that keeps records of registered user identities. UAs generally interact with it indirectly through the proxy or other server.

In addition, there are other advanced servers such as:

- **Presence Server** It is to make the most of users' presence information in any type of Internet or telecom application.
- **Location server** It is an SIP/SIMPLE application framework designed for the development and delivery of real-time communication applications and service [3].

The distinction between SIP server types is logical only, not physical.

2.2 Signaling

The typical SIP transaction is the request/response model also depicted in Figure 1. SIP messages originating at a UA traverse one or more proxy servers and then reach one or more UAs. The detailed process is described with the steps 1-12 in [4]. And steps i-iv in Figure 1 illustrates the SIP registration.

Each of these steps may use some SIP method. The base SIP methods such as INVITE, ACK and BYE are indicated in Figure 2. Moreover,

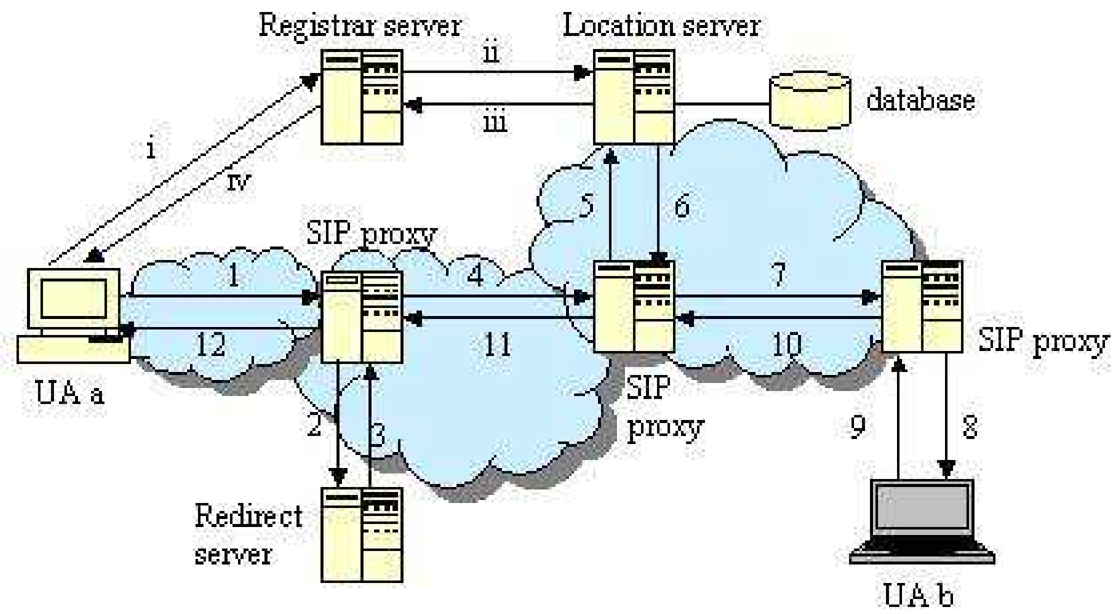

Figure 1: SIP Architecture and Operations

- CANCEL is used to cancel the pending session.
- REGISTER is used to register the UA with the location service.
- OPTIONS is used to query options and capabilities of the remote peer.

Nowadays there are other methods for extension to SIP. And new ones are continually being proposed to add additional functionality to the protocol.

2.3 Functions

Session-related Functions
SIP's principal role is to set up sessions or associations between two or more Internet end systems.

- Session Setup is a three-way handshake using INVITE and ACK methods.
- Media Negotiation allows UAs to decide among one or more media types proposed using Session Description Protocol (SDP).
- Session Modification can be done via re-INVITE, another INVITE/200/ ACK sequence after this session was established already.
- Session Termination uses BYE (end-to-end) method to terminate the session, which successfully established using the INVITE/200/ACK exchange.

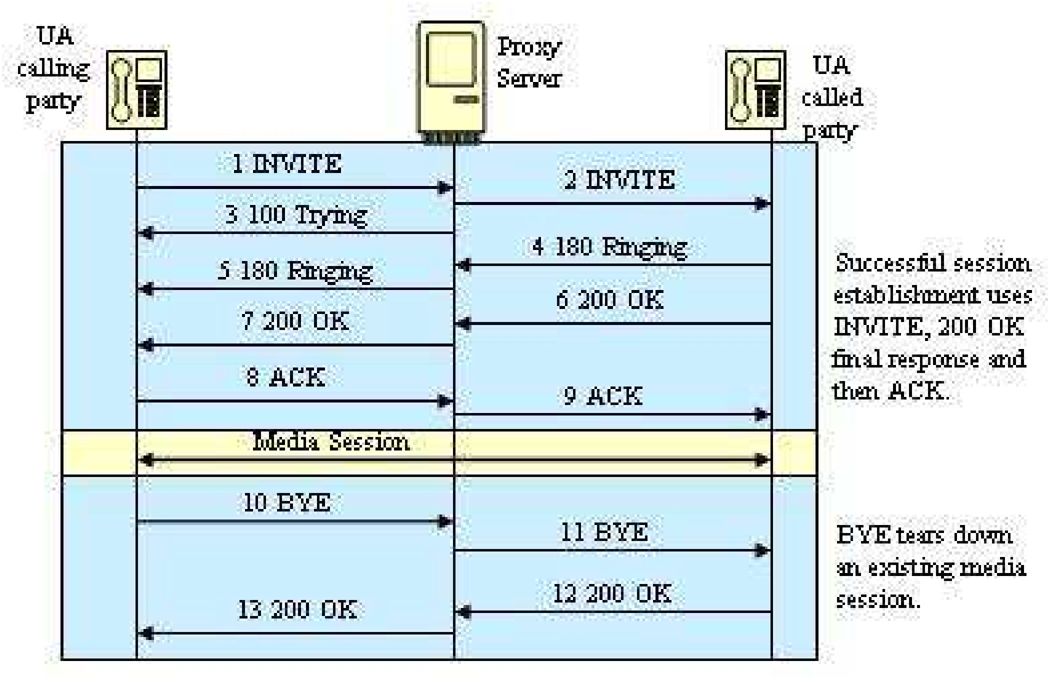

Figure 2: Message Flow Chart for Session Establishment and Terminate [1]

- Session Cancellation ends the call before the call setup is completed and established. Either UA or proxy server can start CANCEL (hop-by-hop) method to cancel.
- Call Control uses a controller or the REFER method to implement the third-party control based on the SIP peer-to-peer communication and end-to- end control.
- Quality of Service (QoS) Setup is the setting up of a QoS for connection with more complex methods exchanges and extensions to SIP.

Non-Session-related Functions
Functions in this subsection include Address Resolution (from Uniform Resource Indicator (URI) to a username at an IP address), Mobility (similar registration function as mobile phones), Message Transport, Event Subscription and notification, Security (authentication with public key cryptography using PGP format) and Extensibility [5].

2.4 Integration and Applicability

Internet telephony can only grow beyond a packetized replacement of PSTN if it integrates fully with other Internet services and protocols [5]. SIP can integrate with other common Internet services in a variety of ways. Though its applicability is limited, SIP can be used widely when internetworking introduced in the following sections.

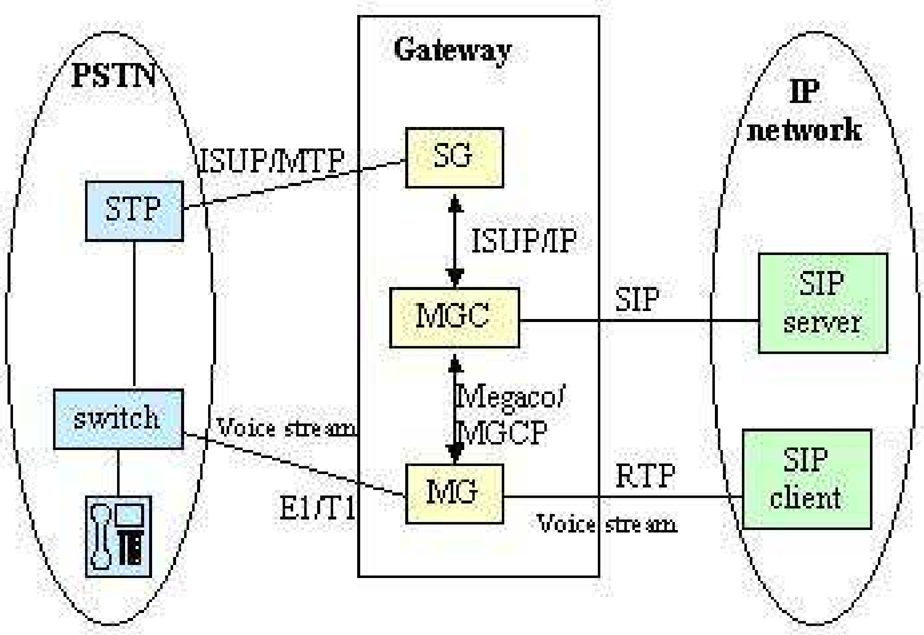

Figure 3: SIP-PSTN Gateway

3 Signaling Internetworking

3.1 Internetworking with PSTN

PSTN-SIP Gateway Architecture

In the traditional circuit-switched telephony PSTN, all functions such as call establishment, billing, routing and information exchange are all integrated in the Signaling System No.7 (SS7) ISDN User Part (ISUP) signaling protocol. But no translation function for SS7 signaling messages is provided for SIP. Therefore, the SIP-PSTN gateway architecture launches for connecting the PSTN with the Internet [6].

The gateway in the Figure 3 consists of three functional components: a signaling gateway (SG), a media gateway controller (MGC) and a media gateway (MG).

- SG routes all ISUP messages forward the MG. Meanwhile, Message Transport Protocol (MTP) as the lower layer in SS7 is replaced by IP, and ISUP as the upper layer is encapsulated into TCP/IP headers. Another task is to translate the dialed number into an IP address before the call is traversed.
- MGC converts the format of signaling from native one in PSTN to that used in IP network, control the MG by introducing Megaco/MGCP and performs 3A (authentication, authorization and accounting).
- MG provides the mapping and transcodeing between media in two kinds of network.

As an important part of the gateway, MGC has logical interfaces to both networks, so that it can be used as the bridge for calls to transit forwards the other side. The media conversion and transport are not of the major concern here, since it is manipulated to meet the needs dictated by signaling.

Issues in signaling internetworking
a. Mapping Function SIP-ISUP of MGC
MGC supports the mapping between ISUP and SIP. In other words, MGC speaks ISUP to PSTN and speaks SIP to IP network. It encapsulates the ISUP information coming from PSTN in the SIP message, and then forwards on. The MGC might package both SDP and ISUP elements into the same SIP message by using the Multipurpose Internet Mail Extensions (MIME) multipart format. Thus, the transparency of ISUP is ensured as well as identity of data transmission.

b. Mapping Messages between SIP and ISUP
Certain information should be translated from an SS7 ISUP message to an SIP message in order to allow SIP elements to make appropriate routing decisions [6]. This focuses on mapping between ISUP parameter and SIP header.

3.2 Internetworking with traditional Intelligent Network

The ease integration of telephony and data applications is the important factor of VoIP. So the VoIP network is necessary to provide the Intelligent Network (IN) capabilities. This means that the VoIP architecture need support a signaling infrastructure which possesses the same capabilities and features as the SS7 and provides the service features of Capability Set number x (CS-x) defined by IN.

Architecture
a. Simple Overview of IN Functional Architecture
The IN is based on the principle that the service-specific software is separated from the basic call processing and is run on a Service Control Point (SCP), while the switching node is Service Switching Points (SSP). SCP implements Service Control Function (SCF) that contains the service logic and commands the Call Control Function (CCF) controls the call and the connection [7].

b. Integration of two call models SIP-IN
Because the SIP server has not the IN call model, the SIP protocol state machine includes the set of an INVITE message, a BYE message, informational response codes, an acceptance of the invitation, an acknowledge for the acceptance, and the condition if the invitation was accepted. The number of states in a SIP server is much less than those in the IN. Therefore, we defines the SIP server support IN CCF/SSF functions by interfacing the SIP server with an IN controller as the Figure 4 shows. Hence, the SIP server can perform the call acceptance and final delivery, while the services may be accessed through the IN controller using an IN call model [7]. The call is treated by both of call models working in synchrony.

Mapping from SIP call to BCSM
The IN is based on a basic call state model (BCSM) that describes the different stages of a telephony call. BCSM consists of the Originating call model (O-BCSM) and the Terminating call model (T-BCSM). The integration requires analyzing which IN BCSM states have the meaning in a SIP based service context, and how they could be added to

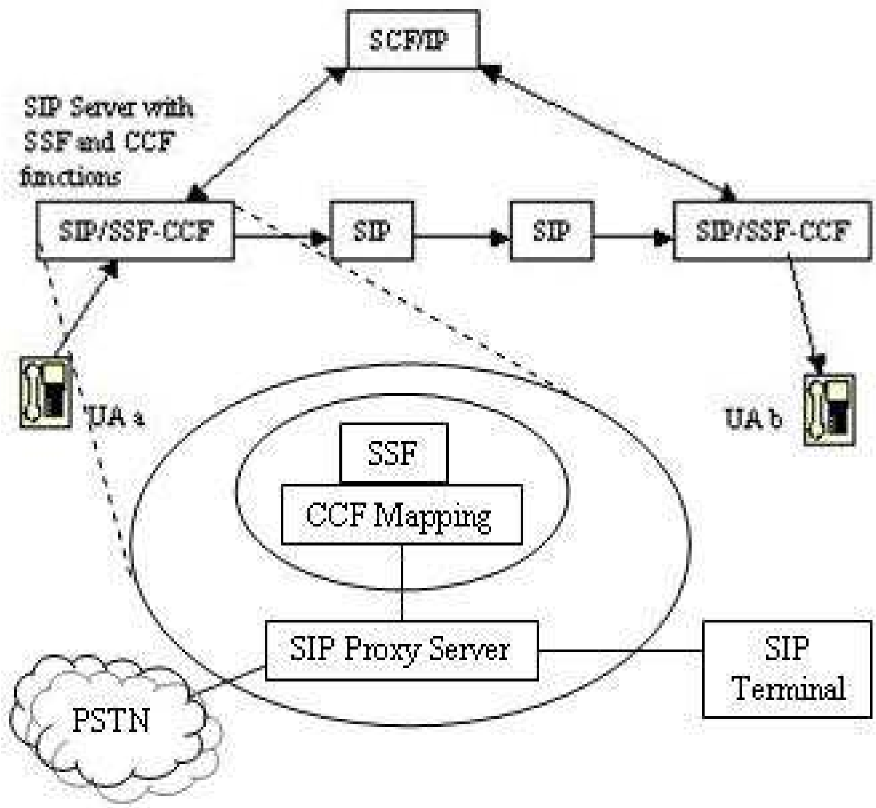

Figure 4: Architecture of IN Capable SIP Networks

SIP call model and understood in the extension of the IN call control model [8]. The detailed mapping procedure is explained in the Internet draft [9].

3.3 Mobile IP telephony

Mobile IP background

a. IP Addresses

Commonly the IP address usually represents the location of the computer. To solve this problem for mobility, two IP addresses are used in mobile equipment: one constant address for the endpoint identifier, and one temporary address known as care-of-address giving the location of the mobile.

b. Mobility Agents

There are two kinds of mobility agents: the home agent and the foreign agent, which keep track of the mobile's location. Each mobile firstly registers in the home agent in its home network, and is administrated by the foreign agent when visiting an the foreign network. In the latter case, the mobile has to obtain the care-of-address and that foreign agent cooperates with the home agent to deliver the datagram to the mobile.

c. Routing Schemes

There are two routing schemes for mobile IP. The first one is tunneling, that is, the home agent attracts all packets sent to the mobile and encapsulates them using IP-within-IP encapsulation, finally sends the resulting packets to the care-of-address of the mobile. Another scheme is route optimization. It enables to send packets directly to the care-of -address of the mobile, depending on an updated mobility binding provided to all peer communicating with the mobile node. Route optimization can offer smoother handoff than tunneling [10].

SIP Support for Four Modes of Mobility

a. Terminal Mobility

"Terminal mobility allows a device to move between IP subnets, while continuing to be reachable for incoming requests and maintaining sessions across subnet changes". SIP location management server with additional software is used for the location management of terminal mobility. In the location server, the user name, terminal identifier and terminal location are stored. Terminal mobility in SIP involves three stages: pre-call, mid- call and to recover from network. In addition, Hierarchical registration and Handover performance, RTP, TCP-based application and streaming multimedia applications are sub-issues to be discussed for terminal mobility in [11].

b. Session mobility

Session mobility allows a user to maintain a media session when changing terminals. In order to implement the session mobility using SIP, the primary end system A configures the other end system B, which is to receive and send the media stream. And then A conveys its IP addresses and ports to B using a new INVITE request. Two solutions can be used as configuration mechanisms. One is the third-party call control, the other is the REFER mechanism, depicted more in [11].

c. Personal mobility

Personal mobility allows addressing a single user located at different terminals by the same logical address [11]. In the other words, one address to many potential terminals mapping and many addresses to one terminal mapping are allowed.

SIP forking proxies [2] are used to make the device choice transparent to third parties, then a user can be reached at any of devices via the same name. Moreover, the registrars may recognize which different devices belong to the same person by using a number of heuristics such as chen.zhang@cc.hut.fi, cxzhang@cc.hut.fi, and doris.zhang@cc.hut.fi, which are all part of the same logical entity.

d. Service mobility

Service mobility allows the user to maintain access to their services when moving or changing devices and network service providers. Service mobility in SIP requires the SIP application to register with the registrar periodically (one hour) or whenever the network address changes. And the registration conveys parameter information to the registrar including the current network address, properties of the device and user configuration elements. The UA also uploads its timestamped version of the configuration information. The server updates its own version or returns a more recent copy in the registration response [11].

SIP development with 3GPP

The 3rd Generation Partnership Project, responsible for the standardization of the 3rd generation mobile networks has selected SIP as the call control protocol for 3G IP-based wireless networks [12]. Corresponding, some important issues are coming to be solved, for example, how SIP traverses the network addresses between IPv4 and Ipv6, and protocol translation.

4 Examples of Application Internetworking

4.1 Fax Transmission over IP Using SIP

Fax transmission is the important application in the business world. It is worth studying the transmission of fax traffic over IP networks using T.38 facsimile in conjunction with SIP [13].

The inter-working of T.38 real-time fax and SIP documented in the Internet drafts suggests two possible configurations of Internet telephony gateways. One is that supports only fax communication; the other does both voice and fax communication. The latter is more popular and general. The typical call flow between such two gateways includes three phases to transmission:

- SIP exchanges deal with the setup of a regular telephone call
- INVITE-OK-ACK exchange, fax parameter negotiation takes place; after parameters are decided on, fax data transmission starts.
- SIP message exchange for call teardown after fax transmission is over.

4.2 Internetworking between SIP and MPEG-4 for IP Video Conferencing

As we know, 3GPP selected SIP as the signaling protocol, and defined signaling flows for the IP multimedia call control based on SIP and SDP. In the other hand, MPEG-4 standards grow rapidly and target abroad range of low-bit rates multimedia applications from the classical streaming video and TV broadcasting to highly interactive applications. In order to achieve universal IP connectivity and seamless IP multimedia services, the standardized inter-working between MPEG-4 and SIP is required [14].

The core component of MPEG-4 multimedia framework is the Delivery Multimedia Integration Framework (DMIF). It is in charge of content location independent procedures for establishing and controlling MPEG-4 audiovisual sessions and of accessing individual media channels over RTP/UDP/IP. For inter-working, DMIF-SIP session signaling translators are defined in [15] as well as call sequence mapping and address translation.

4.3 SIP Billing

Application related to SIP internetworking invokes the topic of SIP billing [16]. Since most SIP session services take place over IP between individuals, the caller usually causes costs (resources, money e.g.) on the caller's side. However, currently Internet providers charge either a flat rate or by volume. It is fair that both sides share the cost only when they have an agreement. Hence, we are willing to mimic PSTN style billing with SIP, which enables callees to charge their callers.

A callee, who wishes to use a SIP billing service, accepts only trans -actions that have passed one of Billing Service Provider (BSP)'s proxies. BSP's proxy authenticates the callers with SIP mechanisms. Among the collected data, the call duration can be used for client billing. Or the call time and type can be used to find SIP sessions matching IP flow information cllected by the callee's access router. The callee may charge his caller using this data when he pays for traffic volume. The latter case resembles today's PSTN billing. To determine the price, pricing information must be included in the SIP message. For this purpose, we extend the SIP message with Price Tag to contain pricing information in the response and with Budget Unit in the INVITE message.

When billing need cross provider boundaries (domain), an authentication hierarchy is required–agreements between caller and callee's BSP and between both sides's BSP. The agreement consists of authentication data and bank account details.

During the above operations, security is critically considered all along, because the billing service is a trusted third party that collects data about SIP transactions, and is a component to prevent repudiation, too. Security for SIP billing involves integrity of pricing information header and authentication by BSP's proxy at least. The broad sense of security for SIP is addressed in [17].

5 Conclusion

As the Internet telephony becomes widely used, SIP as the session signal -ing protocol develops itself on internetworking with various networks, standards and protocols for broader range of applications and additional functions. Section 3 focuses on SIP internetworking with PSTN, IN and mobile IP networks. The MGC in ISUP/SIP gateway architecture is an important element for signaling conversion between PSTN and SIP. When describing the simplified architecture of IN capable SIP networks, the mapping of SIP call states and IN BCSM is briefly mentioned wihle more details can be found in the Internet draft. Mobile IP is a huge topic, so we focus on the SIP-based mobility in terms of application-layer mobility modes including terminal, personal and service mobility. Although the applicability of SIP is limited, the application internetworking with other protocols is wider and wider, especially when 3GPP and telecommuni -cation on multimedia act the vital roles today. In future, SIP will become the key packet switched signaling protocol that replaces most circuit switch ones in the world of mobile networks.

SIP application internetworking is taken with two popular examples: one is fax over IP with SIP that will be commonly used in business life; the other is inter-working with MPEF-4 DMIF for IP video conferencing. They are not explained deeply in technique but paid attention on the usage.

Discussed as the last point, SIP billing is a complex, but realistic topic relevant to applications and services, which involves billing schemes, extension of SIP messages and security issues.

IETF has made several Internet drafts that extend SIP from different points of view. They involve both the branches of SIP researching in this paper and those outsides our studying here such as security, QoS and various applications using SIP.

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