

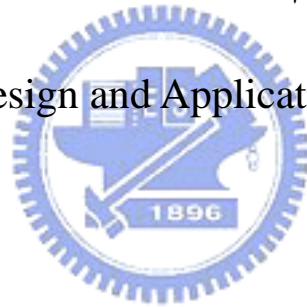
國立交通大學

資訊工程學系

碩士論文

W-CSCF 之設計及其應用

The Design and Applications of W-CSCF



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中華民國九十三年六月

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摘 要

近年來，第二代無線通訊系統已經在全世界發展成熟了，然而，行動使用者可藉由更多的服務來豐富他們的生活。因此，一個新的行動技術；第三代無線通訊系統被發展來提供更高階的服務。第三代通訊發展團體 (3rd Generation Partnership Project, 3GPP) 定義了一個網際網路通訊協定多媒體子系統 (IP Multimedia Subsystem, IMS) 來提供多媒體服務，包括語音、影像、即時雙人遊戲、多媒體視訊會議等服務。通話控制功能 (Call Session Control Function, CSCF) 是 IMS 中最重要的一個功能接點，有三種 CSCF 被定義：近端通話控制功能 (Proxy CSCF, P-CSCF)，詢問通話控制功能 (Interrogating CSCF, I-CSCF)，和 服務通話控制功能 (Serving CSCF, S-CSCF)。

在本論文中，我們提出一個無線區域網路之通話控制功能 (WLAN-based CSCF, W-CSCF)，整合 IMS 中的 I-CSCF 和 S-CSCF，使得無線網路的使用者也能使用到 IMS 系統所提供的服務。而無線網路使用者必須先透過無線區域網路之封包無線服務端點 (WLAN-based GPRS Support Node, WGSN) 才能連結到 W-CSCF。另外，我們實作及按通話 (Push-to-talk, PTT) 這個服務來驗證我們所設計的 W-CSCF。並且，我們提出一個方法使得發話者能夠事先得知受發端使用者目前的狀態，進而減少通話錯失的可能。

The Design and Applications of the W-CSCF

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ABSTRACT

In recent years, the second generation mobile communication system (2G) is becoming matured worldwide. However, the 2G mobile users may desire more services to facilitate their life. Therefore, a new mobile technology called the third generation (3G) mobile network has been developed to provide more value-added applications. The Third Generation Partnership Project (3GPP) defines the IP Multimedia Subsystem (IMS) to support multimedia services, such as voice telephony, video, real-time interactive games and multimedia conferencing. Call Session Control Function (CSCF) is the main function entity in the 3G IMS. Three kinds of CSCFs are defined: Proxy CSCF (P-CSCF), Interrogating CSCF (I-CSCF) and Serving CSCF (S-CSCF).

In this thesis, we propose a WLAN-based CSCF (W-CSCF) which integrates the I-CSCF and the S-CSCF in the IMS to enable WLAN users to access IMS services. The WLAN user can connect to a W-CSCF through a WLAN-based GPRS support node (WGSN). We have implemented Push-to-talk (PTT) service to demonstrate the design of W-CSCF. Moreover, a PTT user is constantly informed the status of the users in the PTT Group, to which the user is belonged.

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Chapter 1 Introduction

Telecommunication network consists of the traditional public switched telephone network (PSTN) and the mobile networks. The Global System for Mobile Communications (GSM) is the popular mobile technology for the second generation (2G) mobile system [1]. In recent years, GSM grows mature and the number of mobile telephony users grows rapidly, since people can use the mobile terminal to communicate with others at any time and at any place. The voice communications of the GSM network is based on circuit switched (CS) domain. To support packet data transport in GSM network, the General Packet Radio Service (GPRS) [2] technology is developed and we call it 2.5G mobile system. The GPRS system is evolved from the GSM network and provided packet switched (PS) data service. In the GPRS core network, there are two types of nodes: Serving GPRS Support Node (SGSN) and Gateway GRPS Support Node (GGSN). A new mobile technology which we call third generation (3G) mobile system supports higher transport speed and larger system capacity than 2G. The Universal Mobile Telecommunications System (UMTS) [3] proposed by the Third Generation Partnership Project (3GPP) is a 3G mobile system evolved from GPRS network.

3G mobile system will be an all IP (Internet Protocol) network. That means the voice and data will be transported over IP. Voice over IP (VoIP) may be the trend in the future. The reasons to use VoIP include lower bandwidth requirement, lower equipment cost, and the widespread availability of IP. Protocols have been developed for providing internet telephony, such as Session initiation Protocol (SIP) [4] for signaling transport and Real-Time Transport Protocol (RTP) for voice and data transport.

3G mobile system is a mobile access to the Internet. 3G will give us more and more features, such as multiple sessions, always on line, world-wide services and so on. In order to

support these multimedia services, there is one IP Multimedia Subsystem (IMS) [5] in the UMTS network. 3G multimedia entertainment may become the most popular things for the mobile users. There may be applications that make the mobile service more popular, such as Push-to-talk service. In the following section, I will describe protocols and systems used in this thesis.

1.1 SIP

The Session Initiation Protocol (SIP) proposed by the Internet Engineering Task Force (IETF) is a text based and application layer control protocol, and it can handle the setup, modification, and termination of multimedia sessions. SIP allows two or more participants to establish a session that may consist of multiple media streams. SIP can be used in conjunction with several other IETF protocols, such as the Real-Time Transport Protocol (RTP) for transporting real-time data and providing Quality and Service (QoS) feedback, and the Session Description Protocol (SDP) [6] for describing multimedia sessions. SIP defines a number of network entities, including user agent client (UAC), user agent server (UAS), registrar, redirect server, and proxy server. A UAC can generate SIP requests and a UAS can respond to them. A registrar is responsible for keeping the registration information about the user, such as the user's location. A redirect server receives SIP requests and returns a response that indicates where the user agent should send the request next. A proxy server accepts the SIP requests from the client and then relays the SIP request to route the calls.

SIP defines a number of request methods as Table 1-1 illustrate. A REGISTER is used to register a user's current location. An INVITE request is used to initiate a session, and an ACK is used to confirm a session establishment. A BYE request is used to terminate a session, and a CANCEL request is used to stop the call setup process. A SUBSCRIBE request is used to get some information and a NOTIFY request is used to confirm the subscriber.

■ Table 1-1: SIP methods

Method	Function
REGISTER	Register with a SIP server
INVITE	Set up session
ACK	Acknowledgment of an INVITE request
BYE	Terminate a session
CANCEL	Stop the call setup
SUBSCRIBE	Subscribe information such as user status
NOTIFY	Notify information such as user status

SIP defines six classes of responses. These responses respond the SIP requests to indicate success of failure, distinguished by status codes, 1xx for progress updates, such as 100 Trying and 180 Ringing, 2xx for success, such as 200 OK, 3xx for redirection, such as 302 Move temporarily, and higher number for failure.

1.2 IMS

The Third Generation Partnership Project (3GPP) defines the IP Multimedia Core Network Subsystem (IMS) which comprises all Core Network (CN) elements to support multimedia services such as voice telephony, video, real-time interactive games, instant messages, and multimedia conferencing. IMS is a subsystem of 3GPP ALL IP Network, and it uses the components of the PS domain to supply multimedia services. Figure 1-1 illustrates the 3GPP ALL IP Network architecture and it is also viewed the Universal Mobile telecommunications System (UMTS) architecture. In this figure, we can see that the packet data services of a User Equipment (UE) are provided by the SGSN via UMTS Terrestrial Radio Access Network (UTRAN). The SGSN connects to the external data network through the GGSN. In IMS, the multimedia services are provided by Call Session Control function (CSCF) using SIP. There are three kinds of CSCF: Proxy-CSCF (P-CSCF), Interrogating-CSCF (I-CSCF), and Serving-CSCF (S-CSCF). The Home Subscriber Server (HSS) is a database to store user information. The interface between HSS and CSCFs is Cx [7, 8] interface. S-CSCF connects to the Application Server (AS) by using ISC (IMS service control) interface. The node called Multimedia Resource Function (MRF) provides performing multiparty call and multimedia conferencing functions. The Media Gateway (MGW) is a transport termination of PSTN and media stream termination of PS domain and it provides media change and codec processing. The Media Gateway Control Function (MGCF) is a signaling termination of the PSTN, and it provides protocol conversion between all IP network and legacy network and controls MGW resources. The Breakout Gateway Control Function (BGCF) is the gateway between all-IP network and the PSTN.

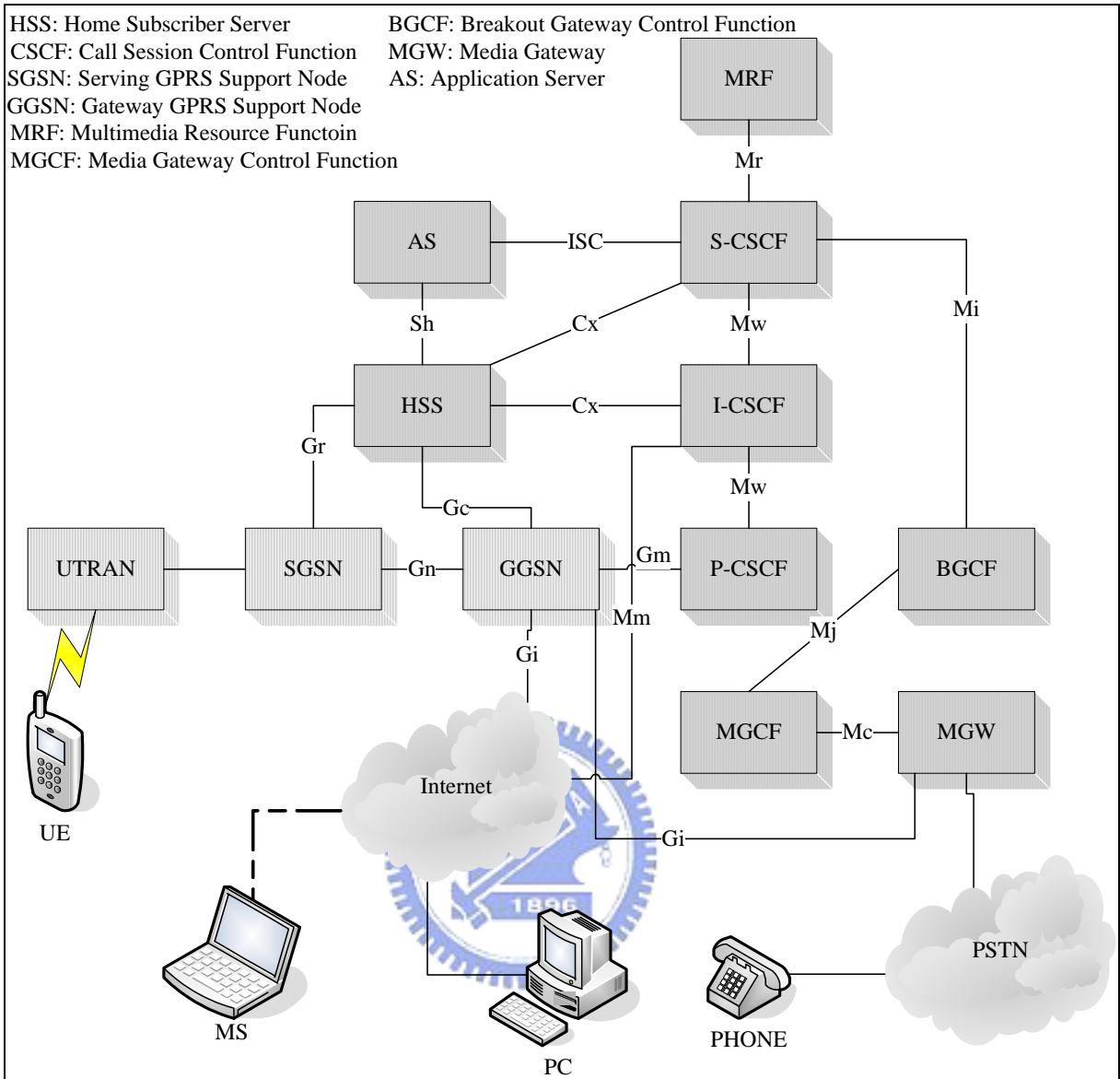


Figure 1-1: 3GPP ALL IP Network architecture

1.2.1 Proxy-CSCF

The P-CSCF is the first contact point for UE within the IMS. Its address is discovered after UEs use PDP (Packet Data Protocol) context activation procedure. The P-CSCF behaves like a proxy server, because it accepts SIP requests and forwards them to next node. The P-CSCF may forward the SIP REGISTER received from the UE to I-CSCF determined using the home domain name and may forward the SIP messages between UE and I-CSCF/S-CSCF.

And P-CSCF may provide authorization of bearer resources and QoS management. Figure 1-2 illustrates the P-CSCF discovery using (Dynamic Host Configuration Protocol) DHCP and Domain Name Server (DNS). At the first step, the UE process the PDP Context Activation procedure to create PDP Context bearer. At the second step, the UE requests a DHCP server and requests the domain name of the P-CSCF and IP address of DNS server. At the third step, the UE performs a DNS query to get the P-CSCF IP address.

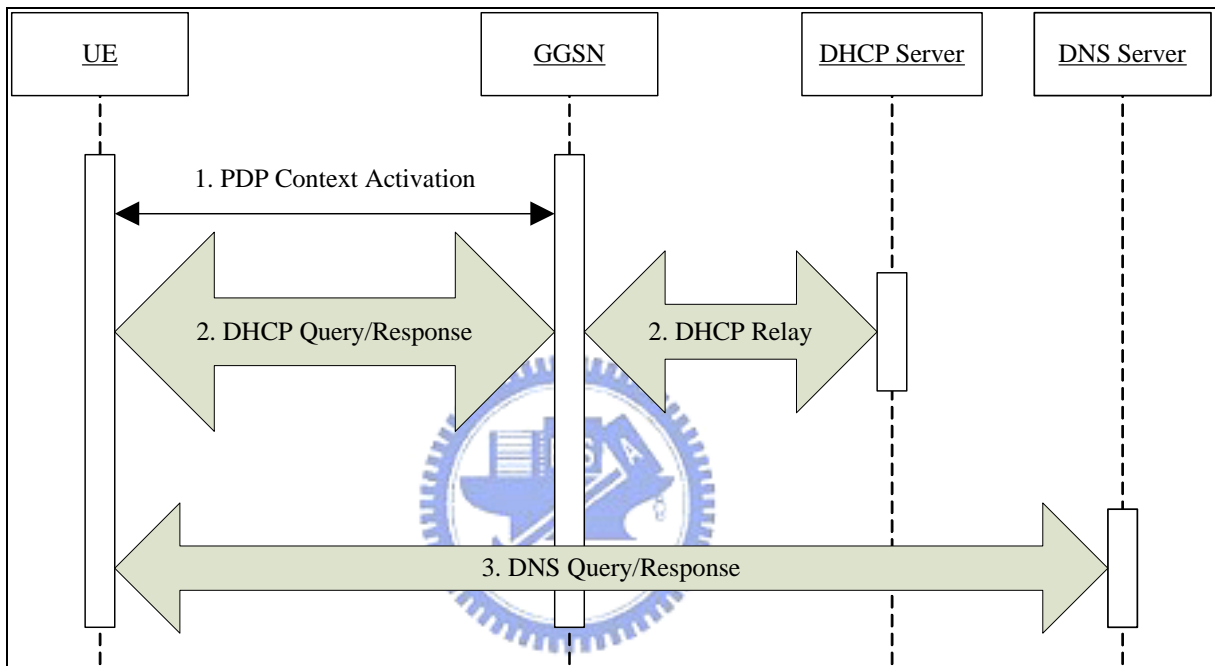


Figure 1-2: P-CSCF discovery using DHCP and DNS

1.2.2 Interrogating-CSCF

The I-CSCF is the contact point within an operator's network for all connections destined to a UE of that network operator. The I-CSCF can be viewed as a firewall between the operator network and the external network. There may be many I-CSCFs in one operator network. The I-CSCF may perform some functions, such as registration and some session flows. When the I-CSCF processes the SIP REGISTER, it will assign one S-CSCF to the subscriber. When performing session flows, it routes the SIP requests towards the S-CSCF.

The I-CSCF will connect to HSS using Cx interface and connect to S-CSCF/P-CSCF using SIP.

1.2.3 Serving-CSCF

The S-CSCF [9] may perform the session control services for the mobile terminal. The session case consists of registration, mobile originating request, mobile terminating request, IP multimedia session release request, subscription and notification. There are many interfaces defined for the S-CSCF including S-CSCF – CSCF (Mw and Mm) interface, S-CSCF – Application Server (ISC) interface, S-CSCF – HSS (Cx) interface. Mw, Mm and ISC are based on SIP. When receiving the SIP REGISTER from the mobile user, the S-CSCF acts as a Registrar, it will register the user through HSS and get user information from HSS. When receiving the SIP message, the S-CSCF may act as an outbound proxy or an inbound proxy. When handling mobile originating request, it is outbound proxy. When handling mobile terminating request, it is inbound proxy.

1.2.4 SIP Application Server

SIP Application Server connects to the S-CSCF in IMS using ISC interface and connects to the HSS via the Sh interface to access subscriber related information. An Application Server can utilize some basic modes of operation for processing SIP request, such as terminating user agent (UA), originating UA, redirect server, proxy server, B2B UA, or no longer involved. When acting as a terminating UA or a redirect server, the incoming SIP request is proxied from the S-CSCF. When acting as an originating UA, Application Server will create one SIP request to the S-CSCF which will route this message. When acting as a sip

proxy, Application Server proxies the SIP request to the S-CSCF. When acting as a B2B UA, Application Server receives one SIP request and then creates a new SIP request for a different SIP dialog, besides, sends the new request to the S-CSCF.

1.2.5 IMS service control

The S-CSCF is the mainly point to control multimedia service in IMS. Each IMS user has a user profile which is talking about the services of the user and stored in the HSS. Each user profile has a formal format and it consists of one or more service profile. The service profile consists of one or more public identity and initial Filter Criteria (iFC) which comprises one Application Server location and one or more Service Point Trigger (SPT). SPTs are those points in the SIP signaling on which Filter Criteria can be set in the user profile. When S-CSCF receives one SIP request, it will get some SPTs from SIP request and match these SPTs to the Filter Criteria. If the result is match, and S-CSCF will get one Application Server, it will pass this SIP request to the Application Server to process the service for the subscriber.

1.2.6 Cx

The interactions between HSS and I-CSCF/S-CSCF are referred to as the Cx interface, which is a reference point in IMS and supports user information transfer between HSS and CSCF. The operations on the Cx reference point can be classified into three functional purposes: location management, user data handling, and user authentication procedures. According to the specification, 3GPP TS 29.228, those functional purposes are achieved by using the Cx operation. Each Cx operation acts as a text-based message exchanged between CSCF and HSS, thus, the Cx operation are also called the Cx messages. We will describe

some Cx message below including Cx-Location-Query message, Cx-Put message, and Cx-Pull message. The Cx messages transfer based on the Diameter protocol. Here we will not describe the Diameter but we will give some Cx-Diameter mapping: Cx-Location-Query/Resp is mapped to LIR/LIA, Cx-Put + Cx-Pull is mapped to SAR, and Cx-Put-Resp + Cx-Pull-Resp is mapped to SAA.

■ Cx-Put / Cx-Put-Resp

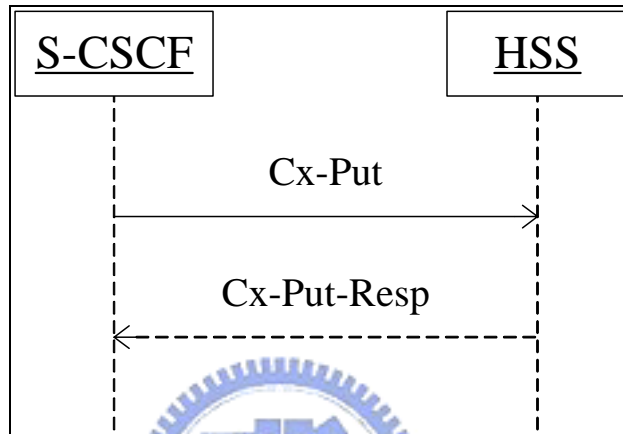


Figure 1-3: Cx-Put/Cx-Put-Resp

The S-CSCF uses the Cx-Put message to inform the HSS that the S-CSCF is assigned to serve the UE in the case of registration/re-registration, or that, in the situation of de-registration, the name of the S-CSCF assigned to the UE is cleared from the HSS. The Cx-Put-Resp is a response for the Cx-Put and also an acknowledgement returned from the HSS to the S-CSCF. It specifies the result of this registration or de-registration.

■ Cx-Pull / Cx-Pull-Resp

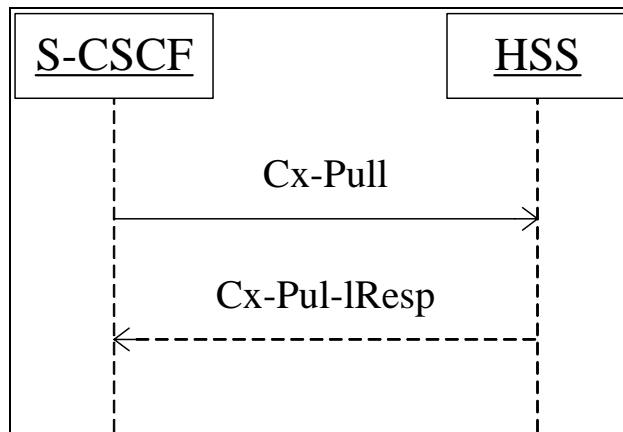


Figure 1-4: Cx-Pull/Cx-Pull-Resp

The Cx-Pull and Cx-Pull-Resp message is only used in the case of registration or re-registration, immediately following the Cx-Put and Cx-Put-Resp message. The Cx-Pull is invoked by the S-CSCF to download, from the HSS, the relevant user profile information that the S-CSCF needs to serve the user. In response, the HSS sends the user profile via the Cx-Pull-Resp message to the S-CSCF.

■ Cx-Location-Query / Cx-Location-Query-Resp

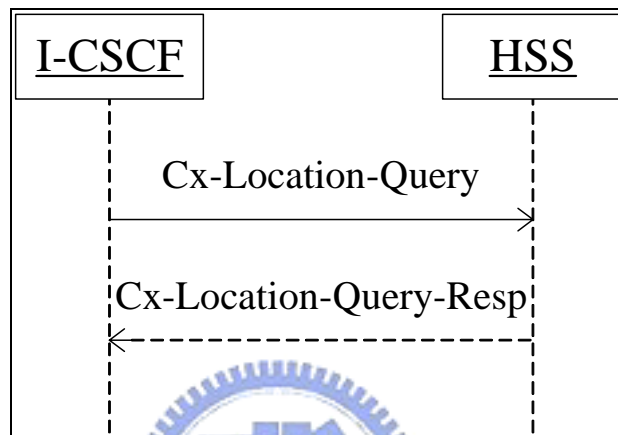


Figure 1-5: Cx-Location-Query/Cx-Location-Query-Resp

When a request for session set-up is received by the I-CSCF, the I-CSCF shall send the Cx-Location-Query message to the HSS to query the name of the S-CSCF that serves the terminating UE. If the terminating UE has registered to an S-CSCF, the HSS shall return the Cx-Location-Query-Resp message which contains the name of the S-CSCF. Otherwise, the Cx-Location-Query-Resp shall contain the server capability set to help the I-CSCF in the selection of the S-CSCF.

1.3 Push-to-Talk service

Always on could become a killer service in the next generation internet, such as Push-to-talk (PTT) service. PTT allows a man to connect to another man instantly or group which is predefined as part of the group list. There is a new standard for PTT, and it is called

Push-to-talk over Cellular (PoC) [10] and is proposed by Ericsson, Motorola, Nokia, and Siemens. Figure 1-6 illustrates the PoC architecture. There are two new nodes including the PoC Server and Group and List Management Server (GLMS), and we will describe them below.

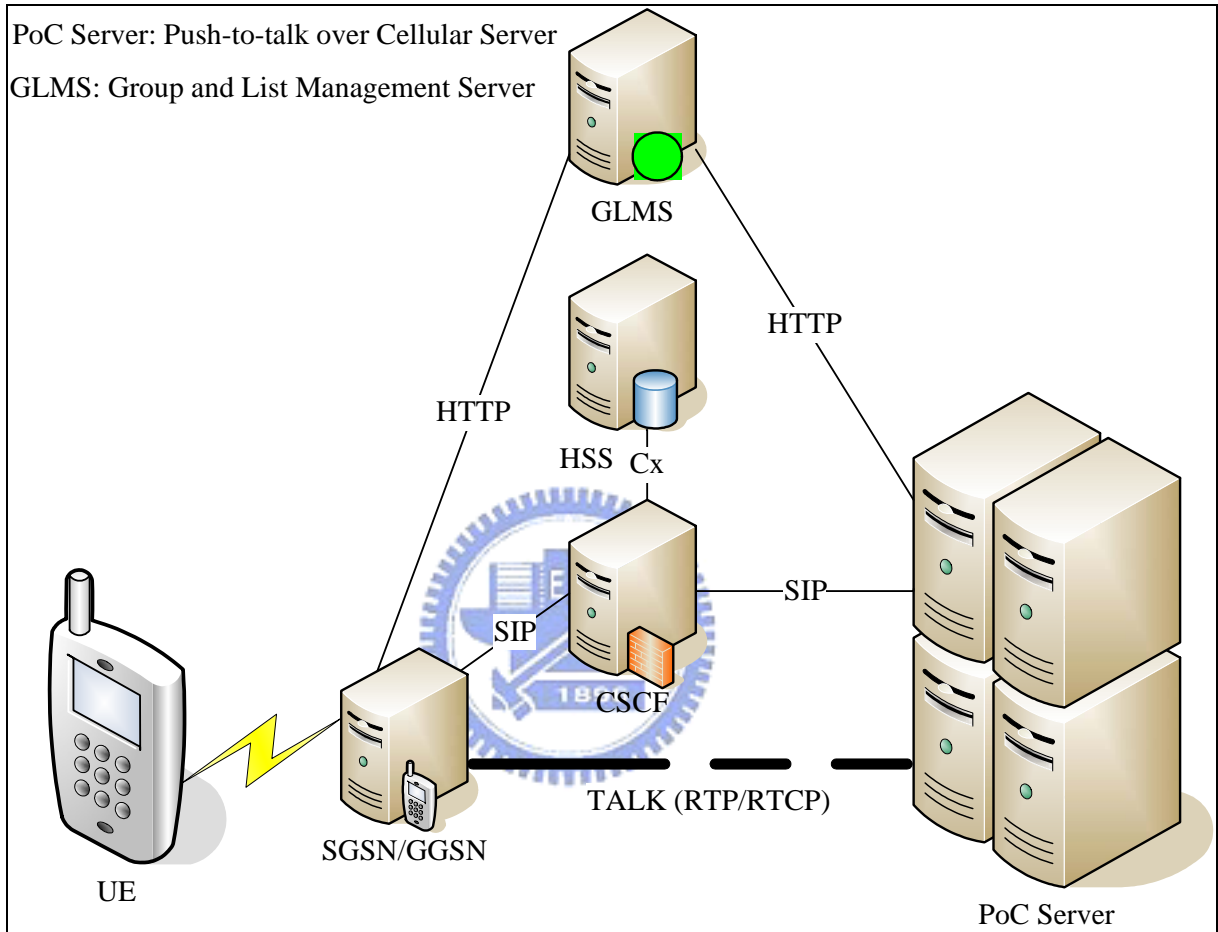


Figure 1-6: the PoC architecture

1.3.1 PoC

By using the PoC service, a mobile user will communication with other users by pressing a button on a PoC enabled UE. The PoC Server is the principal point to supply the PoC services. The PoC Server performs several functions, such as end-point for SIP signaling and RTP/RTCP signaling, provides SIP session handling, policy control for access to groups, group session handling, access control, do not disturb functionality, floor control functionality,

talker identification, media distribution.

1.3.2 GLMS

The mobile users can use the Group and List Management Server (GLMS) to manage his group lists, contact lists and access lists. A PoC group can be defined by a mobile user. A mobile user can select a group form GLMS to initiate an instant group talk or a chat group talk. A contact list is just an address book, and it can be used by user to initiate an instant talk with other users or groups. An access list is used by the user to control who is allowed to make call with the user.

1.3.3 Session establishment

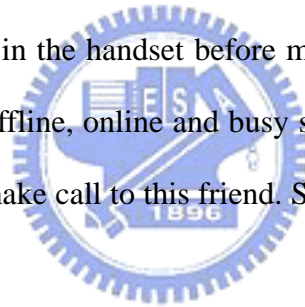


The mobile user must register to the IMS before using the PoC services. The SIP REGISTER is used to do this registration, and this SIP request shall consist of a Contact header including a media tag to indicate the PoC service. When UE wants to establish a session, the SIP INVITE shall contain the Accept-Contact header with a media tag to indicate the PoC service. There are four high level procedures:

- Instant Personal Talk: Two PoC users establish a PoC session. The connection setup may be done either with SIP INVITE or SIP REFER.
- Ad-hoc Instant Group Talk: A user shall be able to invite selected users.
- Instant Group Talk: A member of the instant talk group shall be able to invite the members of the group.
- Chat Group Talk: One user creates open chat group or restricted chat group, and the other users may join the group.

The data network which consists of 3G network and Wireless 802.11 network is popularization of the day. The mobile user in 3G cellular network can use multimedia services through the IMS, but the user in Wireless 802.1 network could not use the IMS services. This shows the integration of 3G network and WLAN network is an important issue. In order to achieve this goal, we purposed a component called WLAN-based Call Session Control Function (W-CSCF) which integrate the I-CSCF and the S-CSCF in IMS network to support the WGSN network. The mobile user in Wireless 802.11 network could use the IMS service through WGSN and W-CSCF.

We use the Push-to-talk (PTT) application to demonstrate the design of W-CSCF. Moreover, we will like to talk to someone by wasting time never. In order to achieve this goal, we proposed a method to show the user status in advance. By using this method, the user will watch the status of his friends in the handset before making a call. The status in our method will be three kinds including offline, online and busy status. When the busy status showed in the handset, the user will not make call to this friend. So this will decrease the time of making a call.



1.6 Overview of this thesis

The thesis presents the design and the applications of W-CSCF. The Chapter 2 describes the concept and architecture of W-CSCF. The Chapter 3 describes the applications of W-CSCF including PTT, SUBSCRIBE/NOTIFY system and a distributed system for W-CSCF applications. The Chapter 4 shows the implementation of system. The Chapter 5 finally presents a conclusion about this thesis.

Chapter 2 The Design and Architecture of the W-CSCF

In this chapter, we will propose our concept about one system including IMS network and wireless network. In this system, the major node is the W-CSCF which will connect with the WGSN in the wireless network and the HSS in the IMS network. We design the W-CSCF which integrates the I-CSCF and S-CSCF in the IMS to support the integration of 3G network and WLAN network. In order to demonstrate the design of W-CSCF, we will propose some application which is used with the W-CSCF in the Chapter 3.

Before we start to describe the concept and architecture of our approach, we depict the W-CSCF landscape and some of the basic assumptions in our approach.

2.1 W-CSCF Landscape and Basic Assumptions

The W-CSCF landscape is an end-to-end All-IP network comprising wireless access network and a 3G IP network. An All-IP network is a network where IP is used end-to-end from the user equipment to the network entities in the network. Figure 2-1 depicts the end-to-end All-IP network of the W-CSCF landscape. We described the W-CSCF landscape in the following. This landscape is composed of 3G IP network and wireless network connected together with the Internet. In this landscape, we define domain concept including IMS home domain and W-CSCF domain. The IMS home domain is composed of an I-CSCF, S-CSCF and one HSS. A W-CSCF domain is composed of a W-CSCF and some WGSN and many user equipments. A brief description of these elements and their characteristics is given below.

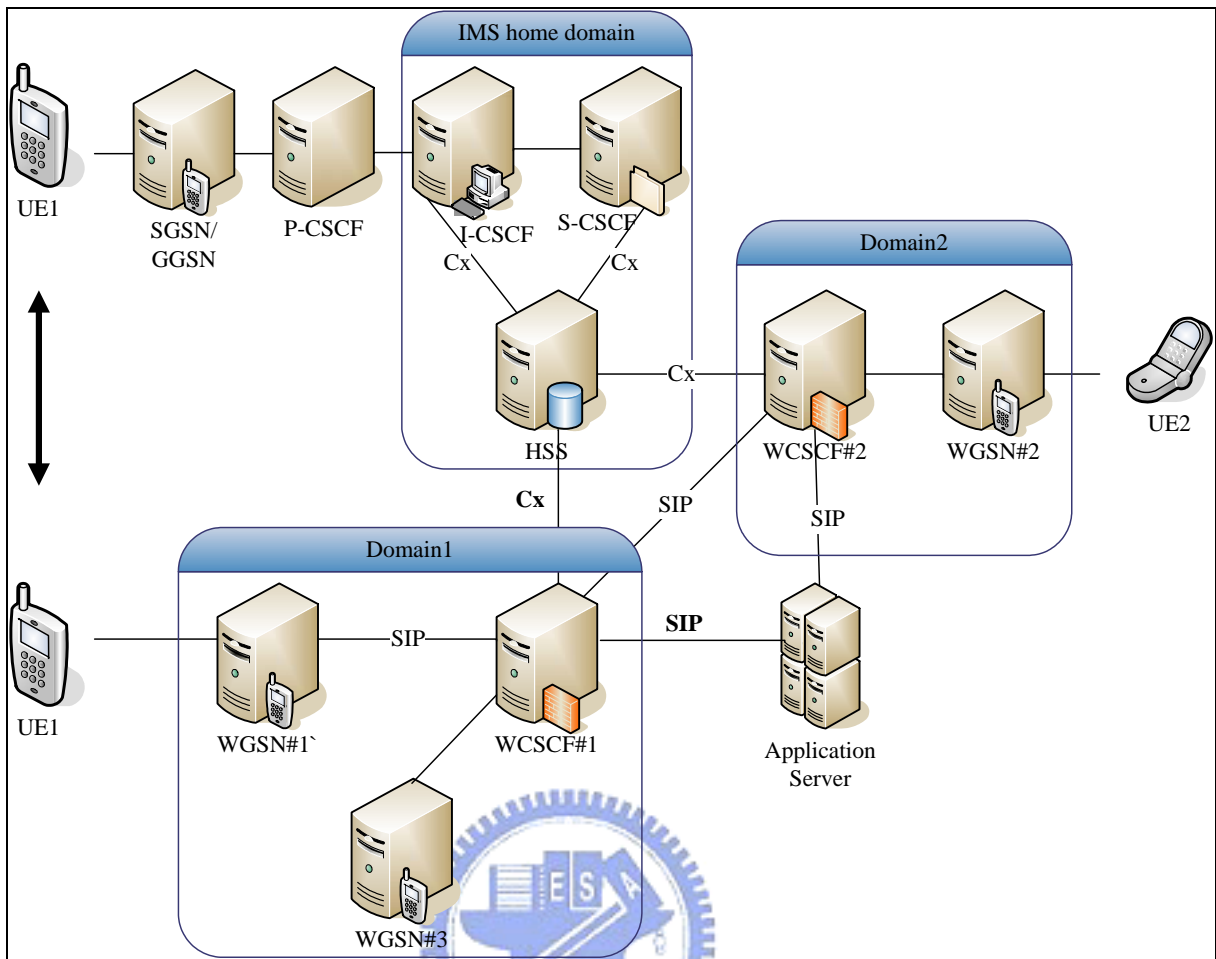


Figure 2-1: the scenario of the W-CSCF landscape

2.1.1 User Equipment (UE)

A User Equipment is the mobile terminal that allows users to connect the wireless access network and 3G IP network. We assume that a user has his subscribed network operator and has a unique network access identifier which we call it private identifier. If the user equipment is attached to a wireless access network or 3G IP network, a user should send a registration with the private identifier to the network of his subscribed operator. If the user equipment is roaming, it can move to wireless access network or 3G IP network.

2.1.2 Domain

Each domain consists of one W-CSCF and several WGSN. In the wireless access network, UE will connect to the W-CSCF through the WGSN. We assume that each domain has a W-CSCF to provide mobility management for the UEs. When a UE moves within the domain, from WGSN#1 to WGSN#2, the signal of movement is handled within the domain to decrease the global location updates and global signals of movement since the W-CSCF will control the movement.

2.1.3 Basic Assumptions

Besides the wireless environment and some definition that we describe above, some of the assumptions in the thesis that we explicitly make are listed below:

- When the UE roams to a WGSN network, it performs GPRS attach procedure to trigger UMTS authentication through WGSN to the HSS.
- The UE gets the address of the W-CSCF through DHCP server. W-CSCF is the first contact point of the UE toward IMS.
- The W-CSCF connects the MS through WGSN.

2.2 Proposed Architecture of the W-CSCF

2.2.1 The architecture of W-CSCF

A W-CSCF is a proxy that integrates the I-CSCF and the S-CSCF in the IMS network

and it let WLAN user access the IMS services through the WGSN network. The main module of W-CSCF is the S-CSCF module which contains a call control function to route the SIP signaling to the destination and a check SPT function to trigger some applications according to the SIP register message. Figure 2-2 shows the architecture of W-CSCF and we can see that W-CSCF connects to other SIP node and HSS in the IMS home network. We will describe all functions of W-CSCF in the next section.

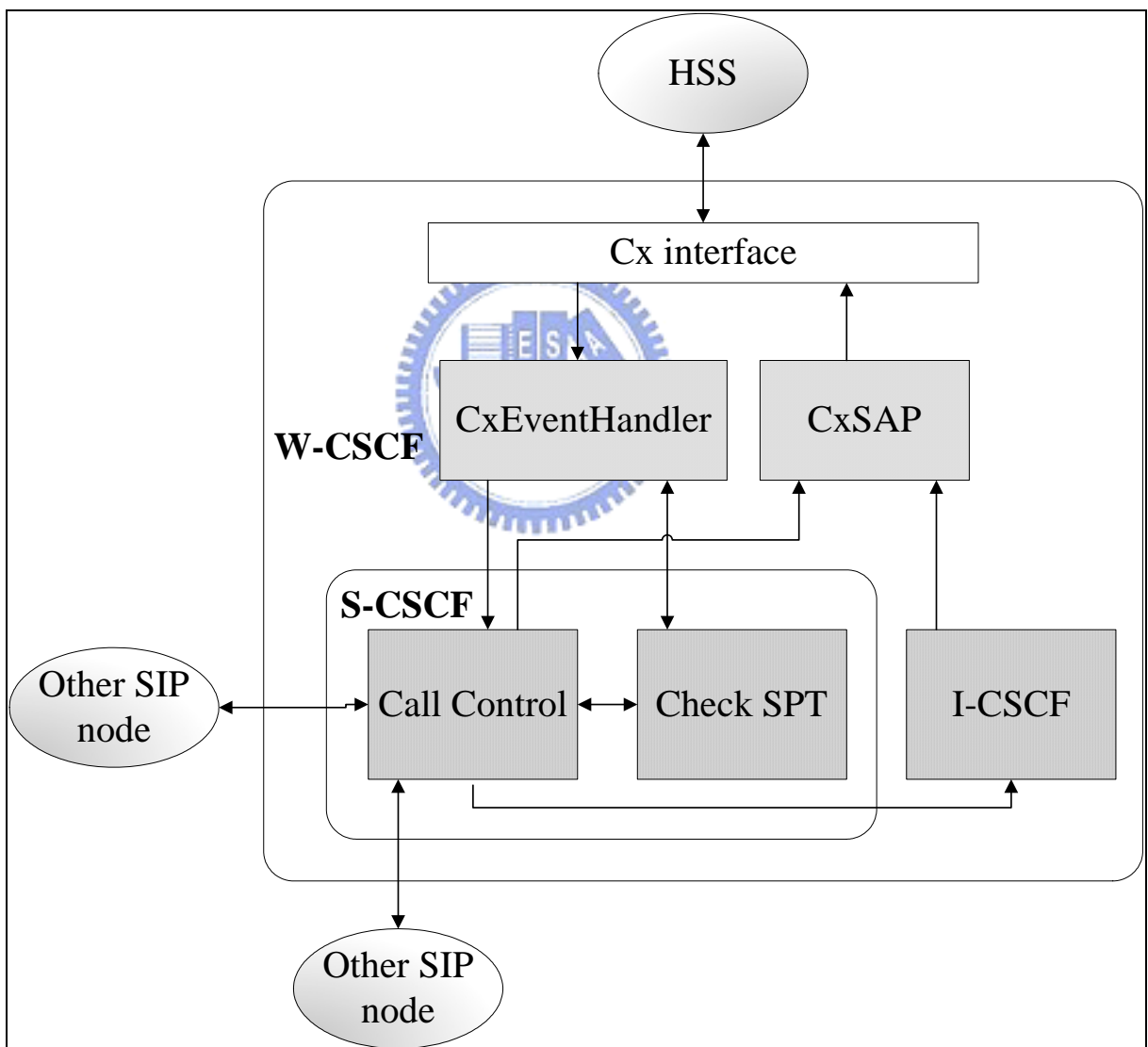


Figure 2-2: the architecture of W-CSCF

2.2.2 The functions of W-CSCF

A W-CSCF consists of S-CSCF module containing a call control function, check SPT function, and I-CSCF function and CxSAP function and CxEventHandler function. Here we will describe each function below.

- S-CSCF: The S-CSCF module consists of call control function, check SPT function. Any message that the W-CSCF received will be passed to the S-CSCF module. The S-CSCF module will accept SIP request and make its information available through the HSS to do the registration operation.
- Call Control function: A W-CSCF is a SIP proxy basically. It receives the SIP message from the UE and routes the SIP message to another W-CSCF, Application Server and another UE. W-CSCF will connect to the HSS in the IMS home network using Cx interface to do the registration and query user's location. Because W-CSCF supports SIP, the call model is based on SIP transaction. There are two transactions in one call. One is responsible for communication with the calling party and maps the messages to the call states of the Server Transaction. The other is responsible for communication with the called party of the Client Transaction. Figure 2-3 shows the call control function. Server Transaction and Client Transaction will both be split to two parts containing INVITE part and Non-INVITE part. The INVITE part is different from those of other methods because of its extended duration. When the W-CSCF receives SIP INVITGE from other SIP node, it will be passed to the INVITE Server. After the INVITE Server processed this message, it will pass this message to the INVITE Client. The INVITE Client will send this message to next SIP node. Other SIP messages containing REGISTER, SUBSCRIBE, NOTIFY, BYE, MESSAGE will be received by Non-INVITE Server and be sent by Non-INVITE Client. The call control

function will call the check SPT function to trigger some points to route the message to the application server.

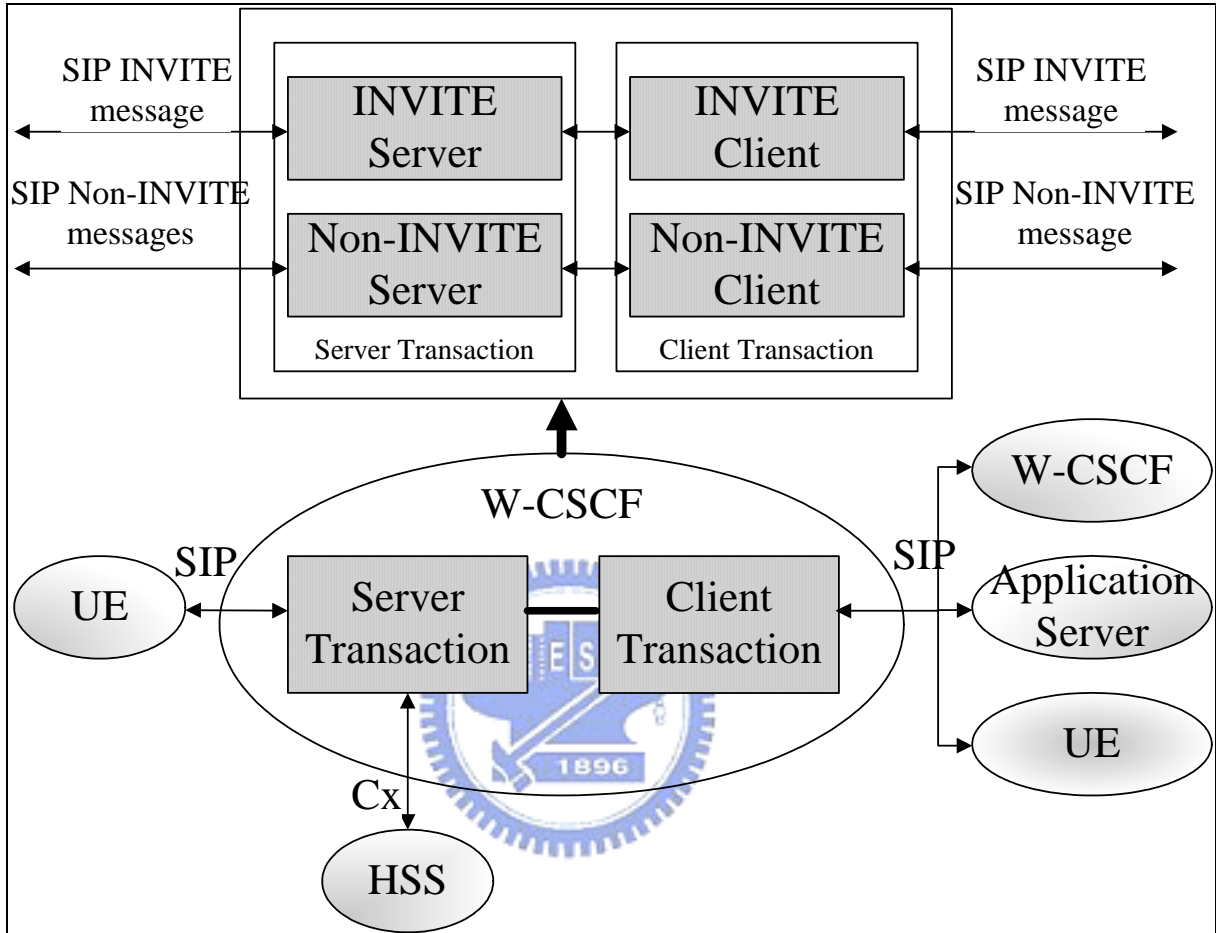


Figure 2-3: call control function

- I-CSCF function: When this function is called, it will send Cx message, Cx-Location-Query, to query HSS where is another W-CSCF, and will receive Cx message, Cx-Location-Query-Resp, to get W-CSCF name.
- CxSAP function: When this function is called, it will call Cx library to send Cx message including Cx-Location-Query, Cx-Put, and Cx-Pull. The CxSAP function will be put several information from SIP request to full the data structure of Cx message. The Cx-Location-Query function will query HSS to get another W-CSCF

location. The Cx-Put and Cx-Pull function will assign an W-CSCF to a public identify, or to clear the name of the W-CSCF assigned to one or more public identities and download from HSS the relevant user profile information that the W-CSCF needs to serve the user.

- CxEventHandler function: When the Cx message is back from the HSS, the Cx library will call this CxEventHandler function in the W-CSCF. This CxEventHandler function consists of CxLocationQueryRespHandler function, CxPutRespHandler function, and CxPullRespHandler function. The CxLocationQueryRespHandler function will get a W-CSCF location and return to call control function. The CxPutRespHandler and CxPullRespHandler function will get user profile information and save it in the database of the W-CSCF.
- Check SPT function: Before using this function, the relevant user profile must be downloaded and stored in the database of the W-CSCF. The user profile of each user will consist of service profiles. And each service profile consist one or more Initial Filter Criteria which consists of many Service Point Trigger and one Application Server location. When the W-CSCF received one SIP message, this function will be called. The input of this function is any SPT like Request-URI, SIP Method, SIP Header, Session Case and Session Description. The output of this function is one Application Server location. If the W-CSCF gets one Application Server location from the check SPT function, the W-CSCF will send the SIP message to this Application Server.

2.2.3 Operations of the W-CSCF system

This section will illustrate the operations of each network entity in the W-CSCF system.

Here we will describe some operations such as registration operation, session establishment operation, and roaming operation. We use the message flow to describe the inter-working of each network entity.

- Registration: When a UE is powered on in a network, it should register with the network. Figure 2-4 shows the message flow when the UE1 powers on in the W-CSCF#1 domain and the UE2 powers on in the W-CSCF#2 domain.

1. When UE1 powers on in the W-CSCF#1 domain, it performs GPRS Authentication with the WGSN#1 to authenticate itself to the IMS service.
 2. After GPRS authentication, the UE1 performs the PDP context activation to obtain GRPS access.
 3. The UE1 sends the SIP REGISTER to the WGSN#1.
 4. When the W-CSCF#1 receives the SIP REGISTER, it will call the CxSAP function to send Cx-Put/Cx-Pull to the HSS to register the UE1.
 5. When the HSS receives the Cx-Put/Cx-Pull, it will return Cx-Put/Pull-Resp which consists of one the user profile of the UE1.
 6. When the W-CSCF#1 receives the Cx-Put/Pull-Resp from the CxEventHandler function, it will store the user profile of the UE in its local database and send SIP 200 ok to the WGSN#1. At the moment, the W-CSCF#1 will call the Check SPT function to check the SPTs in the SIP REGISTER with the Filter Criteria in the user profile. If match, the function will return one Application Server location and the W-CSCF#1 will send a SIP REGISTER to this Application Server.
- 7-12. The registration of the UE2 is the same as the registration of the UE1.

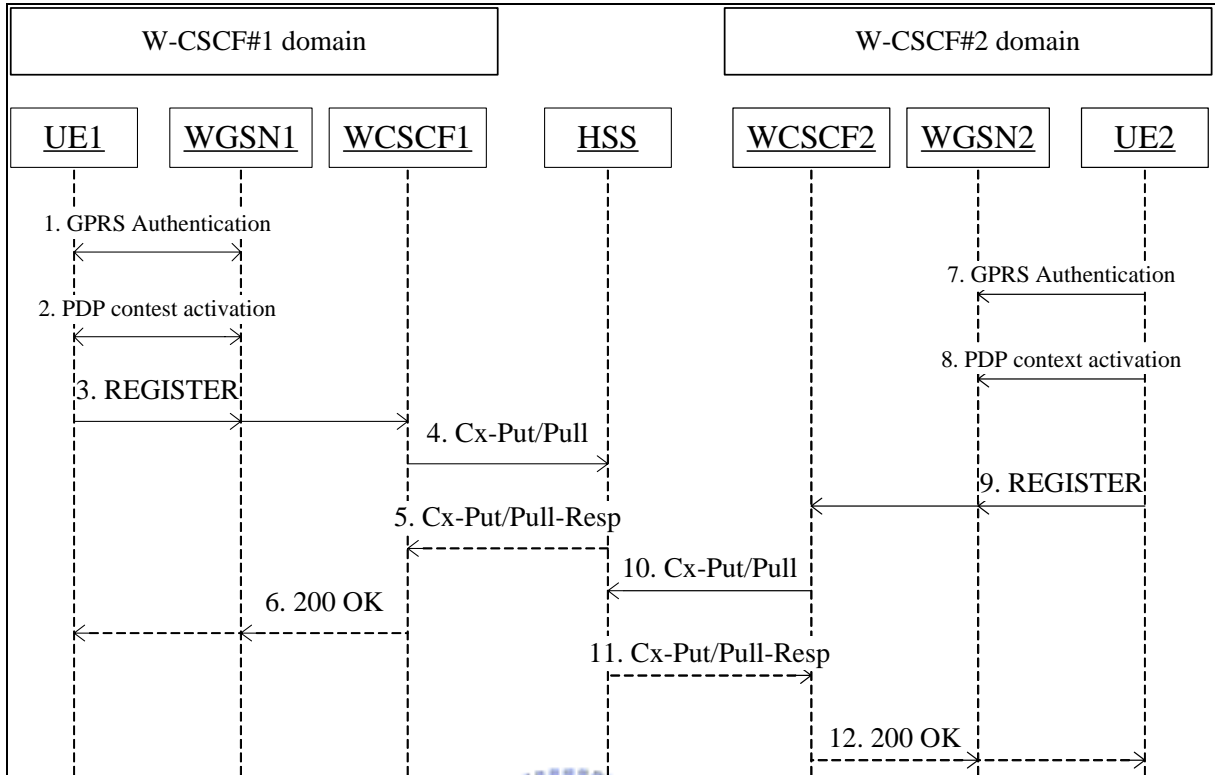


Figure 2-4: the operation of registration

- Session establishment: Figure 2-5 shows the message flow that the UE1 in the W-CSCF#1 domain makes call to the UE2 in the W-CSCF#2 domain.

1. The UE1 wants to make call to the UE2 and it sends a SIP INVITE message to the WGSN#1.
2. The W-CSCF#1 receives the SIP INVITE and it will call the Call Control function to check the original of the SIP INVITE. It will become aware of that it acts as the mobile originating CSCF. Then, it will call the I-CSCF function to send the Cx-Location-Query by calling the CxSAP function to get the CSCF location of the UE2.
3. When the HSS receives the Cx-Location-Query, it will query the database where is the CSCF of the UE2 and it will send the Cx-Location-Query-Resp which consists of the CSCF location to the W-CSCF#1.

4. When the W-CSCF#1 receives the Cx-Location-Query-Resp from the CxEventHandler function, it will get the CSCF location of the UE2 and it will send the SIP INVITE to the CSCF of the UE2. Here the CSCF is the W-CSCF#2.
5. The W-CSCF#2 receives the SIP INVITE and it will call the Call Control function to check the original of the SIP INVITE. It will become aware of that it acts as the mobile terminating CSCF. Then, it will call the I-CSCF function to send the Cx-Location-Query by calling the CxSAP function to get the CSCF location of the UE2.
6. When the HSS receives the Cx-Location-Query, it will query the database where is the CSCF of the UE2 and it will send the Cx-Location-Query-Resp which consists of the CSCF location to the W-CSCF#2.
7. When the W-CSCF#2 receives the Cx-Location-Query-Resp from the CxEventHandler function, it will get the CSCF location of the UE2. Here the W-CSCF#2 will become aware of that it is the serving CSCF of the UE2 and then it will send the SIP INVITE to the WGSN#2.
- 8-13. The UE2 sends the SIP 200 OK to the UE1, and then the session will be established.

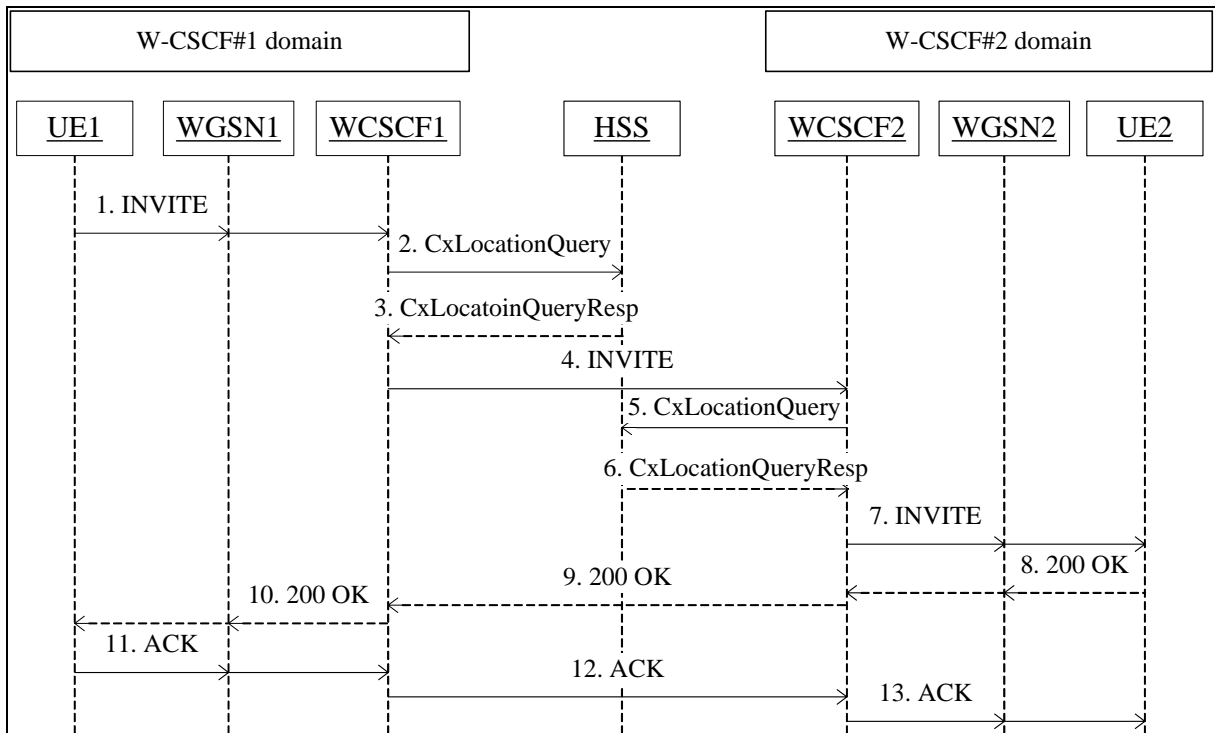


Figure 2-5: the operation of session establishment

- Roaming: Figure 2-6 shows the message flow that the UE1 roams from the W-CSCF#1 domain to the W-CSCF#2 domain, and we will omit the message flow through the WGSN.

- The UE1 will send the SIP REGISTER to de-register itself before it roams from the W-CSCF#1 domain to the W-CSCF#2 domain. The SIP REGISTER contains an expiration value of zero seconds.
- When the W-CSCF#1 receives the REGISTER, it will send the Cx-Put/Cx-Pull to the HSS.
- When the HSS receives the Cx-Put/Cx-Pull, it will know the intention of the message is to de-register the UE1 and the HSS will clear the CSCF name of the UE1. Then, the HSS sends the Cx-Put/Pull-Resp to the W-CSCF#1.
- When the W-CSCF#1 receives the Cx-Put/Pull-Resp, it will clear the user profile of the UE1 in local database and return the SIP 200 OK to the UE1.

5-8. When the UE1 roams to the W-CSCF#2 domain, it will do the registration the same as the action we talked before.

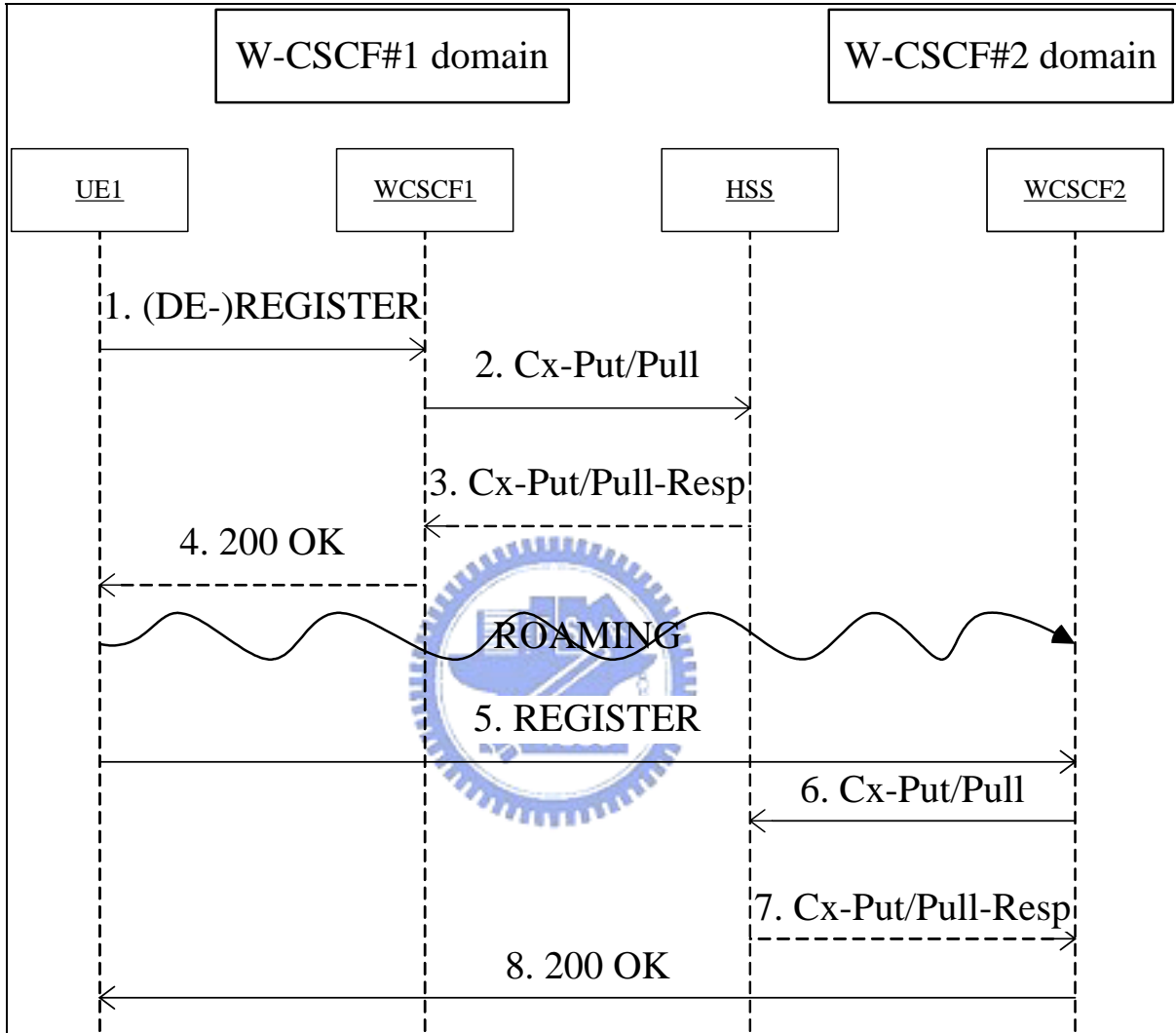


Figure 2-6: the operation of roaming

Chapter 3 The applications of W-CSCF

In this chapter, we will describe some applications which are used with the W-CSCF, such as call forwarding service, call filtering service, and Subscribe/Notify for user status service. We purpose the Subscribe/Notify application to support that the caller can see the status of the PTT users in advance and we support the status monitoring when PTT user roams from one W-CSCF domain to another W-CSCF domain. In order to demonstrate the design of the status monitoring, we describe three scenarios containing the registration, the user status change and the subscription below.

3.1 Basic Applications



In the telephony network, the call forwarding and call filtering services are considered as the basic applications. Here we will purpose a simple scenario as figure 3-1 to show how the W-CSCF to make it. The scenario expresses that the UE-A wants to make call to the UE-B, but the UE-B has registered a call forwarding service in the AS1. The service points out that the man who wants to make call to the UE-B and the UE-B marked the man with a notation will be forwarded to make call to the UE-C. Moreover, the UE-C has also registered a call filtering service in the AS2. The service points out that the man who wants to make call to the UE-C will be accepted if the UE-C marked the man with acceptability or be rejected if the UE-C marked the man with disagreement.

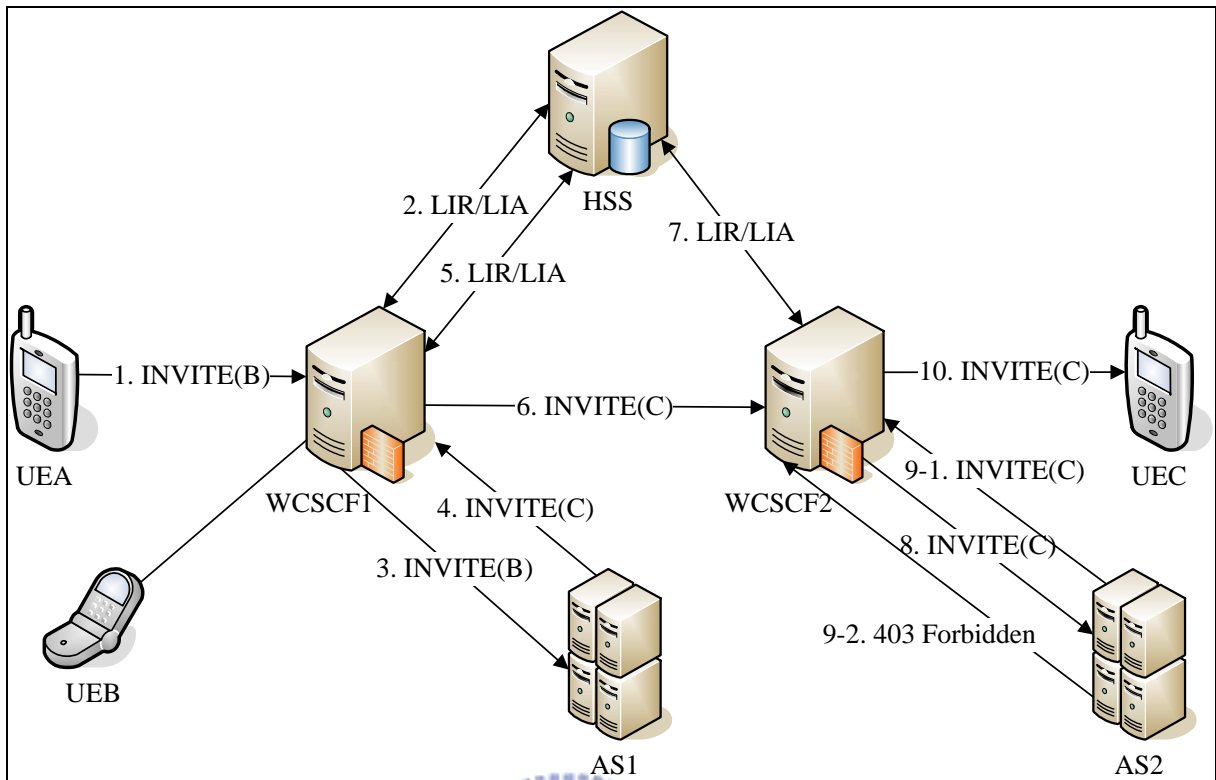


Figure 3-1: the scenario of forwarding and filtering services

Figure 3-2 shows the message flow of the scenario of forwarding and filtering services and we omit the message flow through the WGSN.

1. The UE-A wants to make call to the UE-B and it sends a SIP INVITE message to the W-CSCF#1.
2. The W-CSCF#1 receives the SIP INVITE and it sends the Cx-Location-Query (LIR) to the HSS to get the CSCF location of the UE-B.
3. The HSS sends the Cx-Location-Query-Resp (LIA) which contains the CSCF location of the UE-B to the W-CSCF#1.
4. When the W-CSCF#1 receives the LIA, it will become aware of that it is the serving CSCF of the UE-B and it will check the SPTs from the SIP INVITE for the UE-B. By calling the Check SPT function, it returns an Application Server named AS1 and the W-CSCF#1 passed the SIP INVITE to the AS1 to trigger some service for the UE-B.

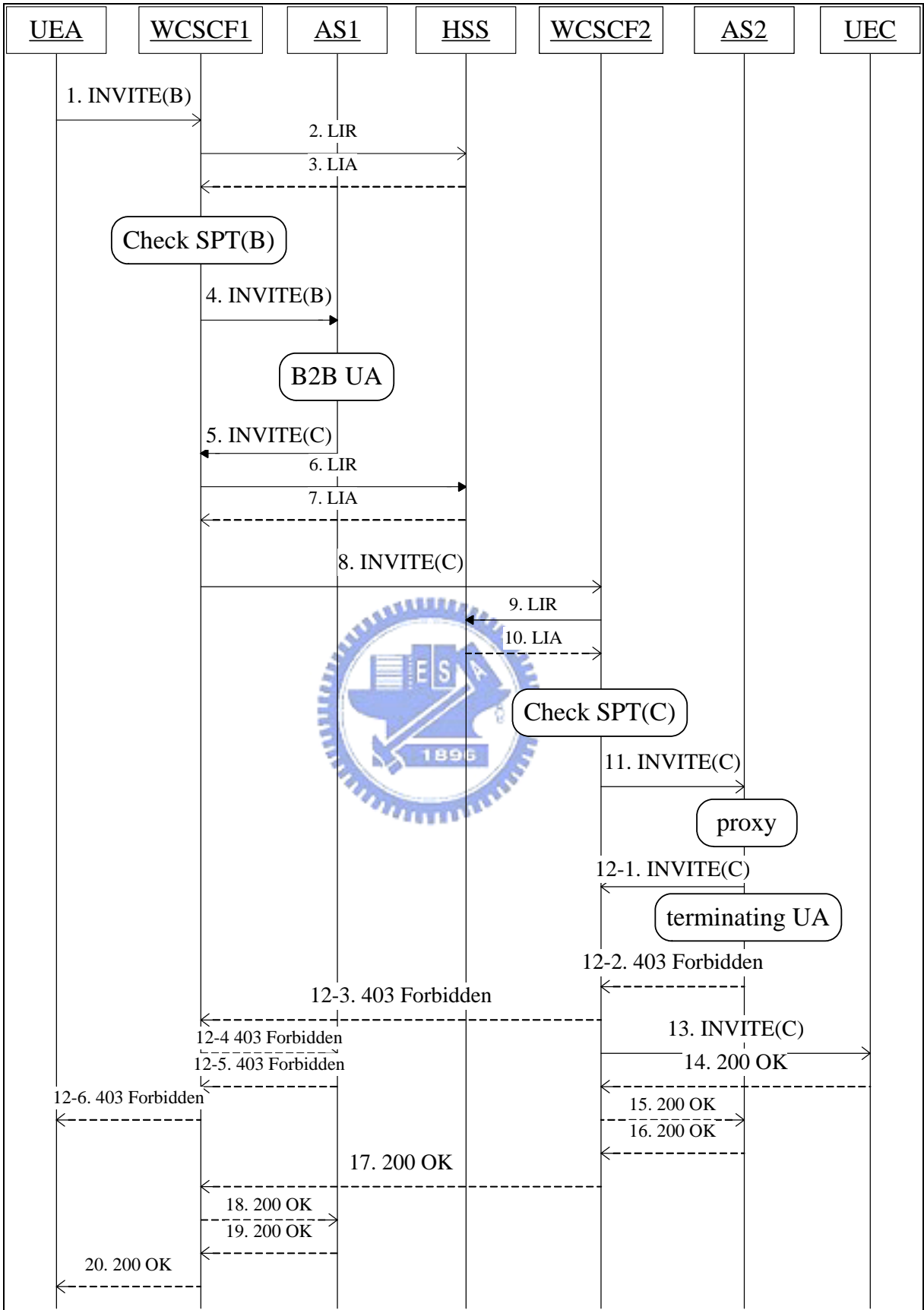


Figure 3-2: the message flow of the forwarding and the filtering service

5. When the AS1 receives the SIP INVITE, it will check the user profile of the UE-B in the local database and it will realize that the UE-B registered a call forwarding service for the UE-A. The services points out that this call should be forwarded to the UE-C. Then, the AS1 will act as a B2B UA to send a new SIP INVITE with a different SIP dialog to the W-CSCF#1.
6. When the W-CSCF#1 receives the SIP INVITE message, it will send a LIR message to the HSS to ask the CSCF location of the UE-C.
7. The HSS sends the LIA which contains the CSCF location of the UE-C to the W-CSCF#1.
8. The W-CSCF#1 sends the SIP INVITE to the W-CSCF#2 according to the LIA.
9. When the W-CSCF#2 receives the SIP INVITE, it sends the LIR to the HSS to ask the serving CSCF of the UE-C.
10. The HSS sends the LIA which contains the CSCF location of the UE-C to the W-CSCF#2.
11. When the W-CSCF#2 receives the LIA, it will become aware of that it is the serving CSCF of the UE-C and it will check the SPTs from the SIP INVITE for the UE-C. By calling the Check SPT function, it returns an Application Server named AS2 and the W-CSCF#2 passed the SIP INVITE to the AS2 to trigger some service for the UE-C.
12. When the AS2 receives the SIP INVITE, it will check the user profile of the UE-C in the local database and it will realize that the UE-C registered a call filtering service. Here there are two conditions including acceptability or disagreement.
 - 12-1 If the UE-C marked the UE-A with acceptability, the AS2 will act as a proxy to proxy the SIP INVITE to the W-CSCF#2.
 - 12-2 If the UE-C marked the UE-A with disagreement, the AS2 will act as a terminating UA to return a SIP 403 Forbidden message to abort the SIP session establishment.

12-2~12-6 The SIP 403 Forbidden message will return to the UE-A.

13. When the W-CSCF#2 receives the SIP INVITE, it will check SPT of the UE-C and the result is no match. Then, it passes the SIP INVITE to the UE-C.

14. The UE-C receives SIP INVITE, and it returns the SIP 200 OK to answer the session.

15-20 The SIP 200 OK message will return to the UE-A and then the UE-A can talk to the UE-C.

3.2 Subscribe/Notify Applications

Push-to-talk is a good application for each mobile user because it can reduce the ringing time that the people must wait. However, there is something to be left out. For instance, if Mr. Wang pushes to talk to Miss Wang, but the Miss Wang is not available, the result is that the Mr. Wang still wastes the time to push to talk. Therefore, we purposed a method to solve this problem by using Subscribe/Notify to show the user status in advance. Figure 3-3 shows the landscape of our method. We modify PoC Server to become a PoC/USV (User Status Server) which will conform our system by using the HTTP protocol to communicate with the GLMS server. Table 3-1 shows the SIP methods which we should modify to fit in with our method. The user status consists of the offline, online, busy status now.

■ Table 3-1: SIP methods

SIP method	usage
SUBSCRIBE	query user status for mobile user
NOTIFY	send user status for PoC/USV to mobile user
MESSAGE	Set new status for mobile user

We explain the message flow in Figure 3-3 step by step.

1. The UE-B changes his status and it sends the SIP MESSAGE to set his new status to the W-CSCF.
2. The W-CSCF passes the SIP MESSAGE to the PoC/USV.
3. The PoC/USV server sets the new status of UE-B into the GLMS using HTTP PUT message.
4. The UE-A wants to get the status of the UE-B and it sends the SIP SUBSCRIBE to the W-CSCF.
5. The W-CSCF passes the SIP SUBSCRIBE to the PoC/USV.
6. The PoC/USV server gets the status of UE-B using HTTP GET message.
7. The PoC/USV sends the SIP NOTIFY including the status of the UE-B to the W-CSCF.
8. The W-CSCF passes the SIP NOTIFY to the UE-A.

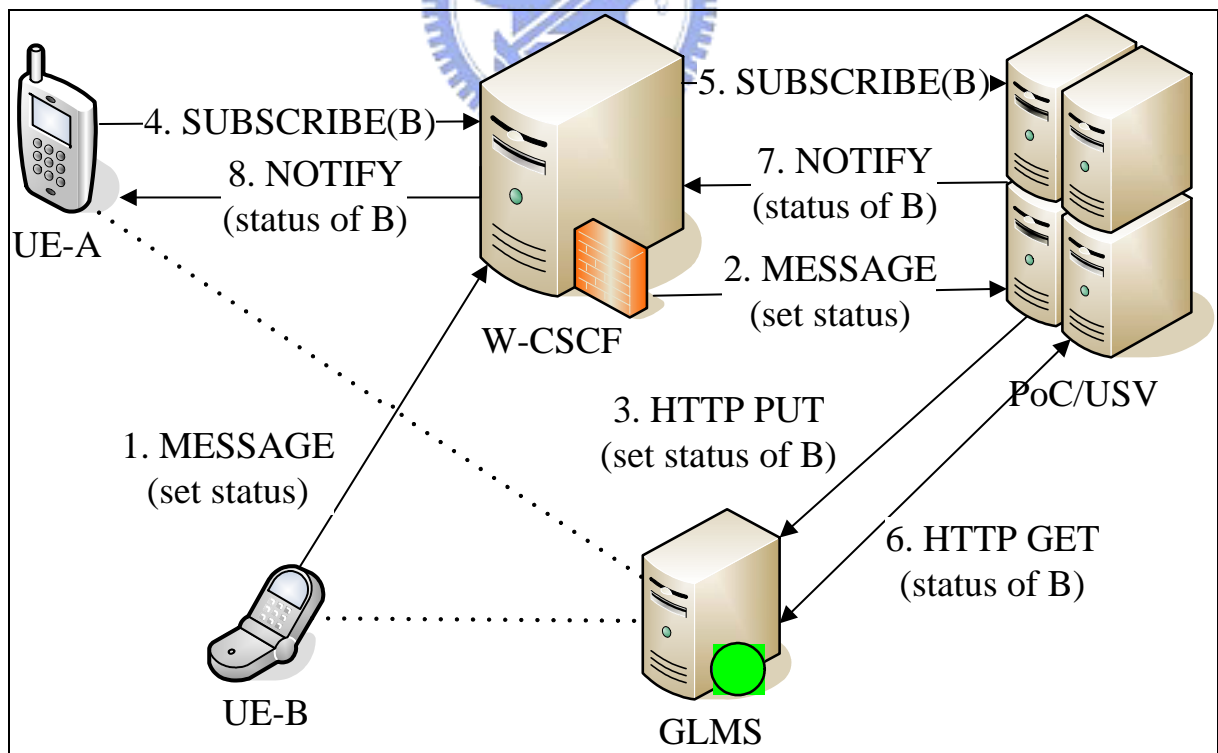


Figure 3-3: the Subscribe/Notify landscape

3.2.1 The functions of each component

■ The functions of the W-CSCF:

1. When receiving the SIP REGISTER message, the W-CSCF must check the SIP Contact header whether the request is PTT message or not. If the Contact header consists of one PTT media flag, the SIP REGISTER is a PTT message. Then, the W-CSCF will send 200 OK including the GLMS location of this domain to the UE. Moreover, the W-CSCF will pass this SIP REGISTER to the PoC/USV server to register the service and set the new user status of online.
2. When receiving the SIP INVITE message, the W-CSCF must check the SIP Accept-Contact header whether the request is PTT message or not. If the Accept-Contact header consists of one PTT media flag, the SIP INVITE is a PTT message. Then, the W-CSCF will pass this SIP INVITE to the PoC/USV server of this domain to trigger the Push-to-talk service.
3. When receiving the SIP MESSAGE, the W-CSCF must check the SIP Accept-Contact header whether the intention of this request is to set status or not. If it is for setting status, the W-CSCF will pass this SIP MESSAGE to the PoC/USV.
4. When receiving the SIP SUBSCRIBE message, the W-CSCF must check the callee is in this domain or not. If the callee is registered in this domain, the W-CSCF will pass the SIP SUBSCRIBE to the PoC/USV. If the callee is not registered in this domain, the W-CSCF will send LIR to the HSS to query where is the CSCF location of the callee. After getting the CSCF location of the callee, the W-CSCF will pass this SIP SUBSCRIBE to the CSCF.
5. When receiving the SIP NOTIFY message, the W-CSCF will check whether the man who subscribes the status is in this domain or not. If the man is registered in

other domain, the W-CSCF will query HSS to get the CSCF location of the man. Then, the W-CSCF will pass the SIP NOTIFY to the CSCF of this man.

■ The functions of the PoC/USV:

1. When receiving the SIP REGISTER message, the PoC/USV server sets online status for the UE to the GLMS and gets the friend list and status of the UE. Then, the PoC/USV server will send the SIP NOTIFY including the online status of the UE to the friends of the UE and send the SIP NOTIFY including the friend's status of the UE to the UE.
2. When receiving the SIP INVITE message, the PoC/USV server will pass this SIP INVITE to the W-CSCF of this domain.
3. When receiving the SIP MESSAGE, the PoC/USV server sets the status of the UE to the GLMS and gets the friend list of the UE. Then, the PoC/USV server will send the SIP NOTIFY including the status of the UE to the friends of the UE.
4. When receiving the SIP SUBSCRIBE message, the PoC/USV server will query the GLMS for the status of the user and set the subscriber to be a temporary friend of the user. Then, the PoC/USV server will send the SIP NOTIFY including the status of the user to the subscriber. There will be one timeout for the temporary case.

■ The functions of the GLMS:

1. When receiving the HTTP PUT message from the PoC/USV server, the GLMS will change the status of the UE.
2. When receiving the HTTP GET message from the UE, the GLMS will return the address list and friend list to the UE.
3. When receiving the HTTP GET message from the PoC/USV server, the GLMS will query the local database for the UE and return the status and friend's list of the UE to the PoC/USV server.

■ The functions of the UE:

1. The UE could send the SIP REGISTER to get GLMS location.
2. The UE could connect to the GLMS to get the address list and friend list.
3. The UE could send the SIP INVITE to initiate a PTT session.
4. The UE could send the SIP MESSAGE to change his current status.
5. The UE could send the SIP SUBSCRIBE to query other user status.
6. The UE could receive the SIP NOTIFY to update the status of his friend.

3.2.2 The scenarios of the status monitoring

There are several scenarios for the status monitoring at this distributed system. The first scenario is registration. The second scenario is change user status. The third scenario is subscription. We will describe these scenarios below.

- Registration: This scenario is to say that the UE2 is roaming from the W-CSCF#1 domain to the W-CSCF#2 domain and executes the registration to the W-CSCF#2 domain. Then, the friends including the UE1 and the UE3 of the UE2 will be notified that the UE2 is online. Here the UE1 and the UE3 is in the W-CSCF#1 domain. Figure 3-5 shows the message flow of the scenario of the registration.

1. The UE2 roams to the W-CSCF#2 domain. Then, The UE2 sends the SIP REGISTER message to the W-CSCF#2 to execute the registration.
2. When the W-CSCF#2 receives the SIP REGISTER, it sends the Cx-Put/Pull (SAR) message to the HSS to register the UE2. The HSS returns the Cx-Put/Pull-Resp (SAA) including the user profile of the UE2 to the W-CSCF#2.
3. The W-CSCF#2 returns the SIP 200 OK including the GLMS2 location to the UE2.
4. The UE2 gets the address list and friend list from the GLMS2 using the HTTP GET.

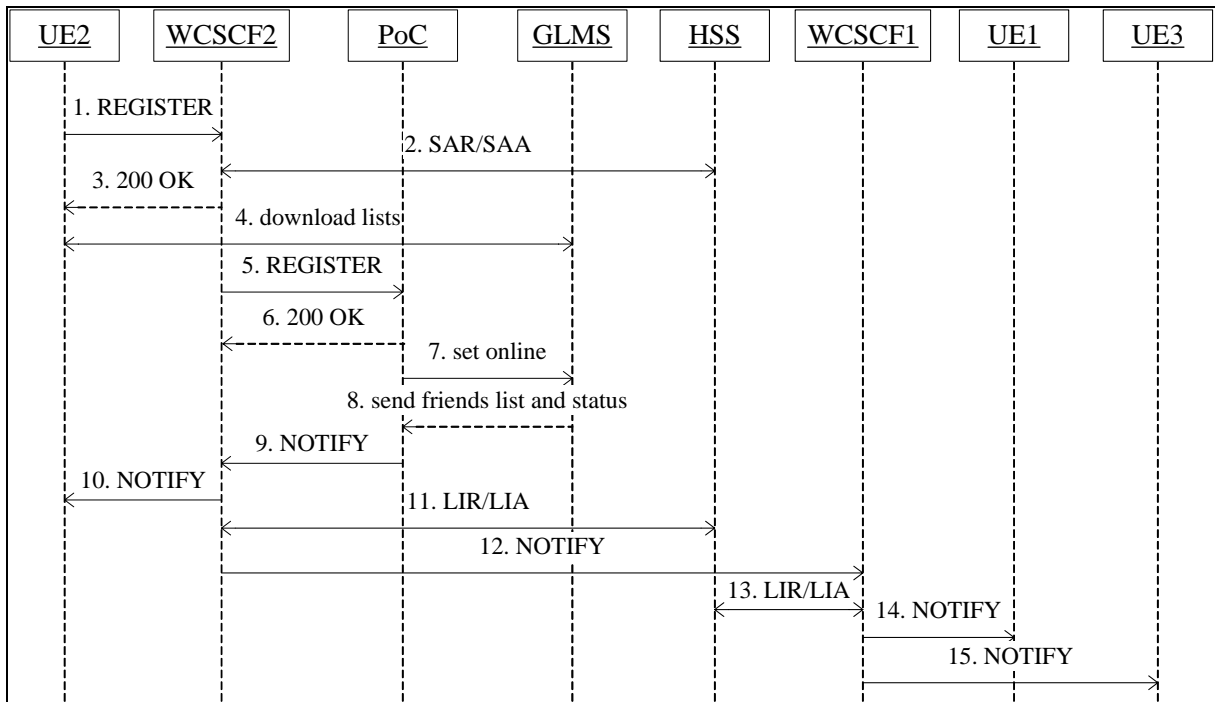


Figure 3-4: the scenario of the registration

5. The W-CSCF#2 checks the SIP REGISTER and discovers that the REGISTER is a PTT message. Then, the W-CSCF#2 sends the REGISTER to the PoC/USV2 server.
6. The PoC/USV2 server returns the SIP 200 OK to the W-CSCF#2.
7. The PoC/USV2 server sets the online status for the UE2 to the GLMS2.
8. The PoC/USV2 server gets the friends list and status from the GLMS2.
9. The PoC/USV2 will send three SIP NOTIFY message to the W-CSCF#2. One NOTIFY message including the statuses of the UE2's friends is sent to the UE1. The other Notify messages including the current status of the UE2 are sent to the UE1 and UE3.
10. The W-CSCF#2 passes the SIP NOTIFY to the UE2.
11. The W-CSCF#2 sends the Cx-Location-Query (LIR) to the HSS to query the CSCF location of the UE1 and the UE3. The HSS returns the Cx-Location-Query-Resp (LIA) containing the CSCF location of the UE1 and the UE3 to the W-CSCF#2.
12. The W-CSCF#2 sends two SIP NOTIFY messages to the W-CSCF#1.

13. The W-CSCF#1 sends the LIR to the HSS to query the serving CSCF of the UE1 and the UE3. The HSS returns the LIA to the W-CSCF#1 containing the serving CSCF location of the UE1 and the UE3.
 14. The W-CSCF#1 discovers that it is the serving CSCF of the UE1 and passes the SIP NOTIFY to the UE1.
 15. The W-CSCF#1 passes the SIP NOTIFY to the UE3.
- Change user status: This scenario is to say that the UE2 makes call with the UE4 using Push-to-talk and the UE2 sends the SIP MESSAGE to change his current status. Then, the friends of the UE2 will be notified that the UE2 is busy now. Figure 3-6 shows the message flow of this scenario.
1. The UE2 sends the SIP INVITE to initiate a Push-to-talk session with the UE4.
 2. The W-CSCF#2 receives the SIP INVITE and it discovers that the message is a PTT message. Then, it passes the SIP INVITE to the PoC/USV2 in this domain.
 3. The PoC/USV2 receives the SIP INVITE and passes the message back to the W-CSCF#2.
 4. The W-CSCF#2 sends the LIR to the HSS to query the CSCF location of the UE4.
 5. The HSS returns the LIA containing the CSCF location of the UE4.
 6. The W-CSCF#2 sends the SIP INVITE to the W-CSCF#1 which is contained in the LIA.
 7. The W-CSCF#1 sends the LIR to the HSS to query the serving CSCF of the UE4.
 8. The HSS returns the LIA containing the serving CSCF location of the UE4.
 9. The W-CSCF#1 discovers that it is the serving CSCF of the UE4 and it passes the SIP INVITE to the UE4.
 - 10-12. The UE4 returns the SIP 200 OK to the PoC/USV2 server and the PoC/USV2 server modifies the information in the 200 OK for RTP/RTCP connection.
 - 13-14. The PoC/USV2 server passes the SIP 200 OK to the UE1.

15. The UE1 connects direct to the PoC/USV2 server using RTP/RTCP and the PoC/USV2 server connects direct to the UE4 using RTP/RTCP. Then, the Push-to-talk session is established and the PoC/USV2 server can through the W-CSCF domain by using RTP/RTCP protocol.

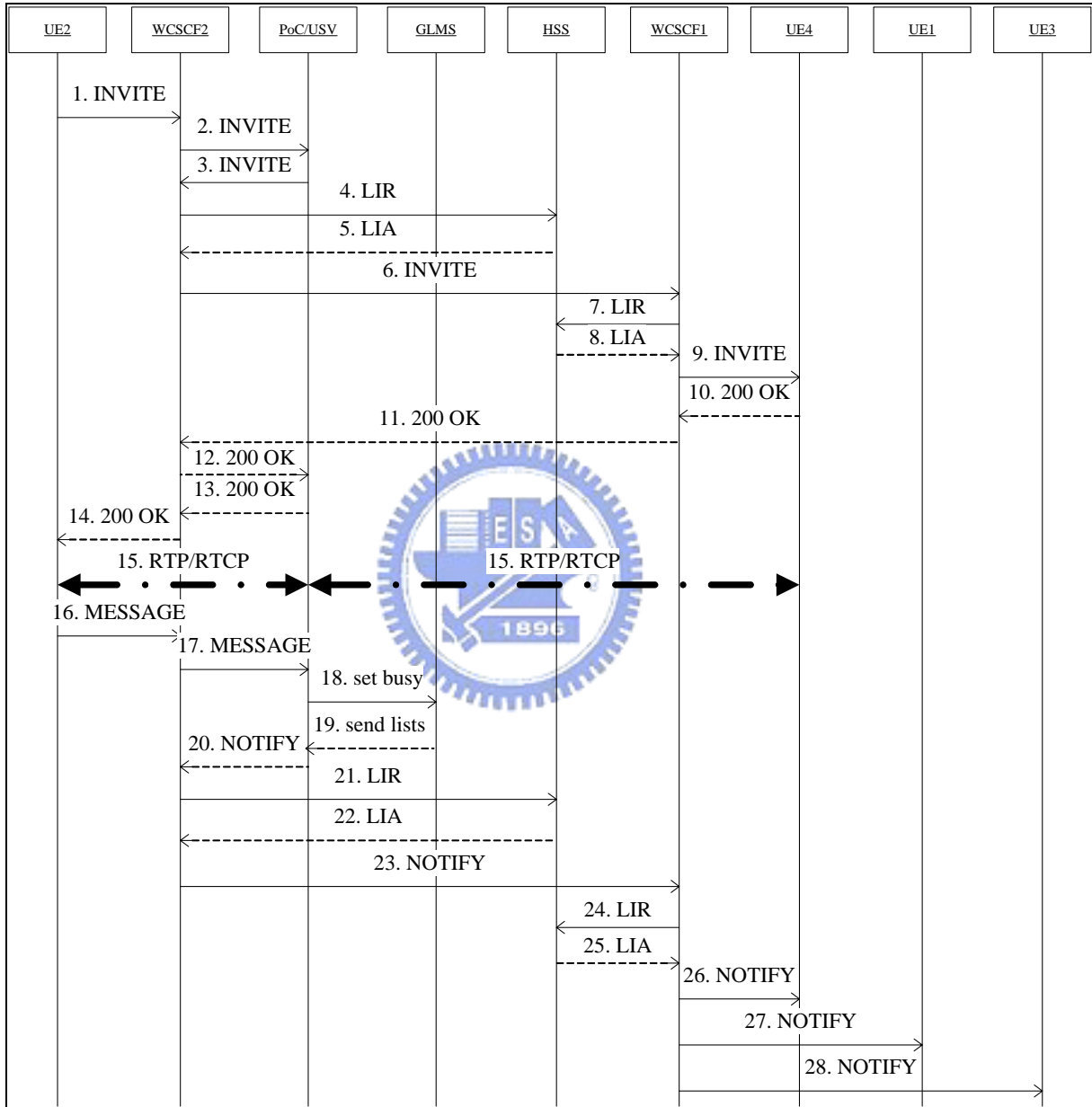


Figure 3-5: the scenario of the change user status

16. After the session is established, the UE2 sends the SIP MESSAGE to change his status and to notify his friends.

17. The W-CSCF#2 receives the SIP MESSAGE and passes the message to the PoC/USV2 server.
18. The PoC/USV2 server sets the busy status for the UE2 by using the HTTP PUT to the GLMS2.
19. The PoC/USV2 server gets the friend list of the UE2 from the GLMS2 by using the HTTP GET.
20. The PoC/USV2 server sends the SIP NOTIFY to the W-CSCF#2 according to the friend list.
- 21-28. The SIP NOTIFY message will be sent to the UE1 and the UE3 and the UE1 and the UE3 will know that the UE2 is at busy status now.

- Subscription: This scenario is to say that the UE5 wants to subscribe the status of the UE2, but the UE5 is not in the friend list of the UE2. When the UE5 sends the SIP SUBSCRIBE to ask the status of the UE2, the PoC/USV server will return the current status of the UE2 to the UE5 and initiate a subscription timeout for the UE5. The PoC/USV server will notify the UE5 of the UE2's new status every time until the subscription timeout terminates.

- 1-4. The UE2 sends the SIP MESSAGE to set his current status and the PoC/USV2 server gets the friend list of the UE2.
- 5-7. According to the friend list, the PoC/USV2 server sends the SIP NOTIFY containing the status of the UE2 to the UE6.
8. The UE5 sends the SIP SUBSCRIBE to the W-CSCF#1 to ask the current status of the UE2.
9. The W-CSCF#1 sends the LIR to the HSS to get CSCF location of the UE2 and the HSS returns the LIA to the W-CSCF#1.

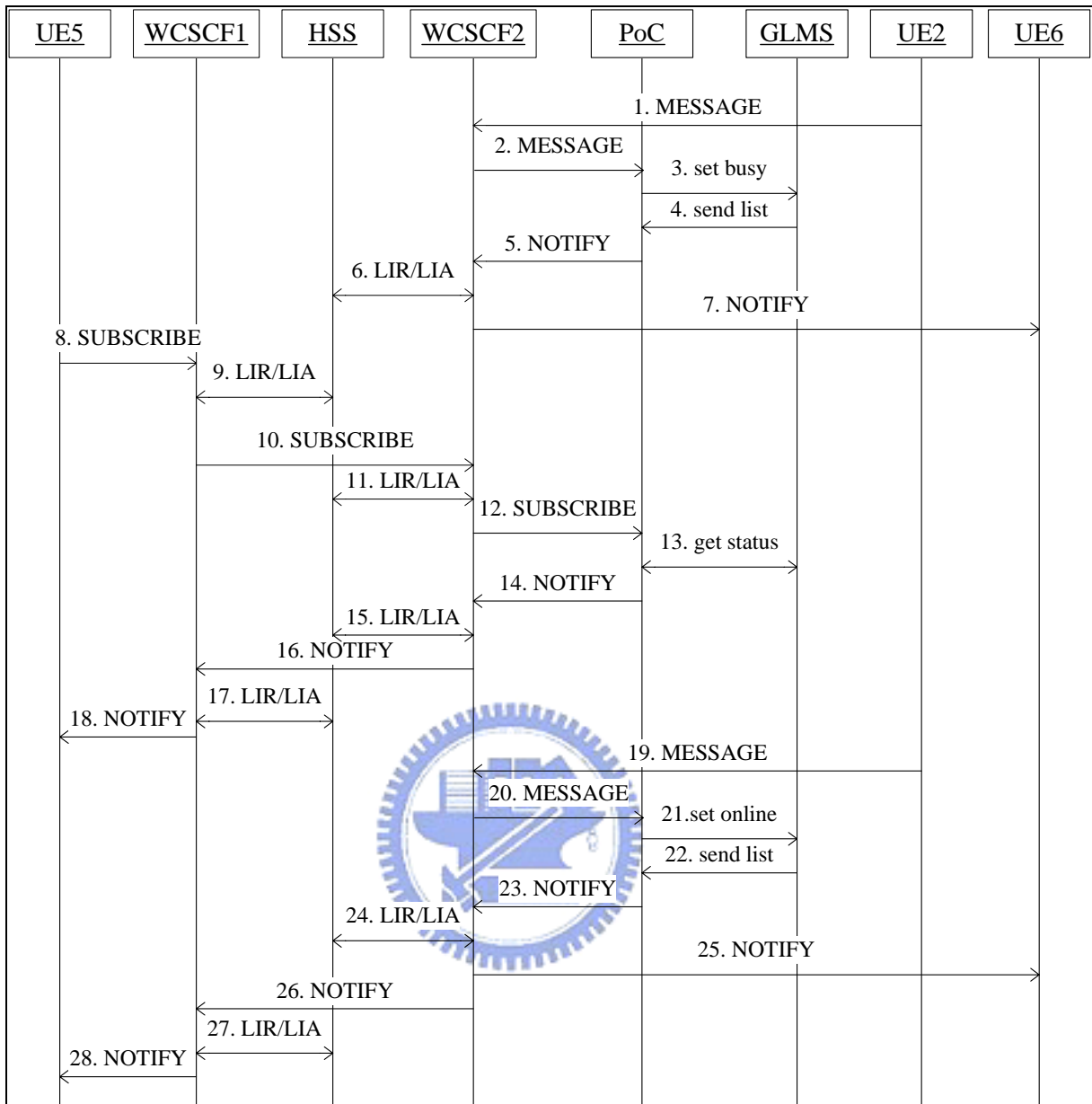


Figure 3-6: the scenario of the subscription

10. The W-CSCF#1 sends the SIP SUBSCRIBE to the W-CSCF#2.
11. The W-CSCF#2 uses the LIR/LIA to get the serving CSCF of the UE2.
12. The W-CSCF#2 passes the SIP SUBSCRIBE to the PoC/USV2 server and the PoC/USV2 server initiates a subscription timeout for the UE5.
13. The PoC/USV2 server gets current status of the UE2 from the GLMS2.
- 14-18. The PoC/USV2 server sends the SIP NOTIFY containing the UE'2 status to the

UE5.

19-22. The UE2 sends the SIP MESSAGE to set his current status and the PoC/USV2 server gets the friend list of the UE2.

23-28. The PoC/USV2 server sends the SIP NOTIFY to the UE6 according to the friend list of the UE2 and sends the SIP NOTIFY to the UE5 according to the subscription timeout.



Chapter 4 System Implementation

This chapter describes our system implementation. At first, we briefly describe the developing platform and programming tools. Then, we describe the implementation of each network entity containing the W-CSCF, the PoC/USV server and the SIP UA.

4.1 Platforms and Tools

We have implemented the W-CSCF on the Linux Platform and the PoC/USV server, SIP UA and GLMS on the Microsoft Windows Platform. We use the Microsoft Visual C++ 6.0 as our developing tool on the Microsoft Windows Platform and we also use the MySQL and MySQL++ for database management. In our implementation, we extend the SIP proxy to turn into the W-CSCF and modify the SIP UA to fit in with our demands. The following sections describe how each network entity is implemented.

4.2 W-CSCF

As the description of the Chapter 2, we can realize that the architecture of the W-CSCF consists of two blocks containing the Non-INVITE Server and INVITE Server. Here we will describe the two blocks by flow chart and we will also describe the CxEventHandler function.

4.2.1 The Non-INVITE Server

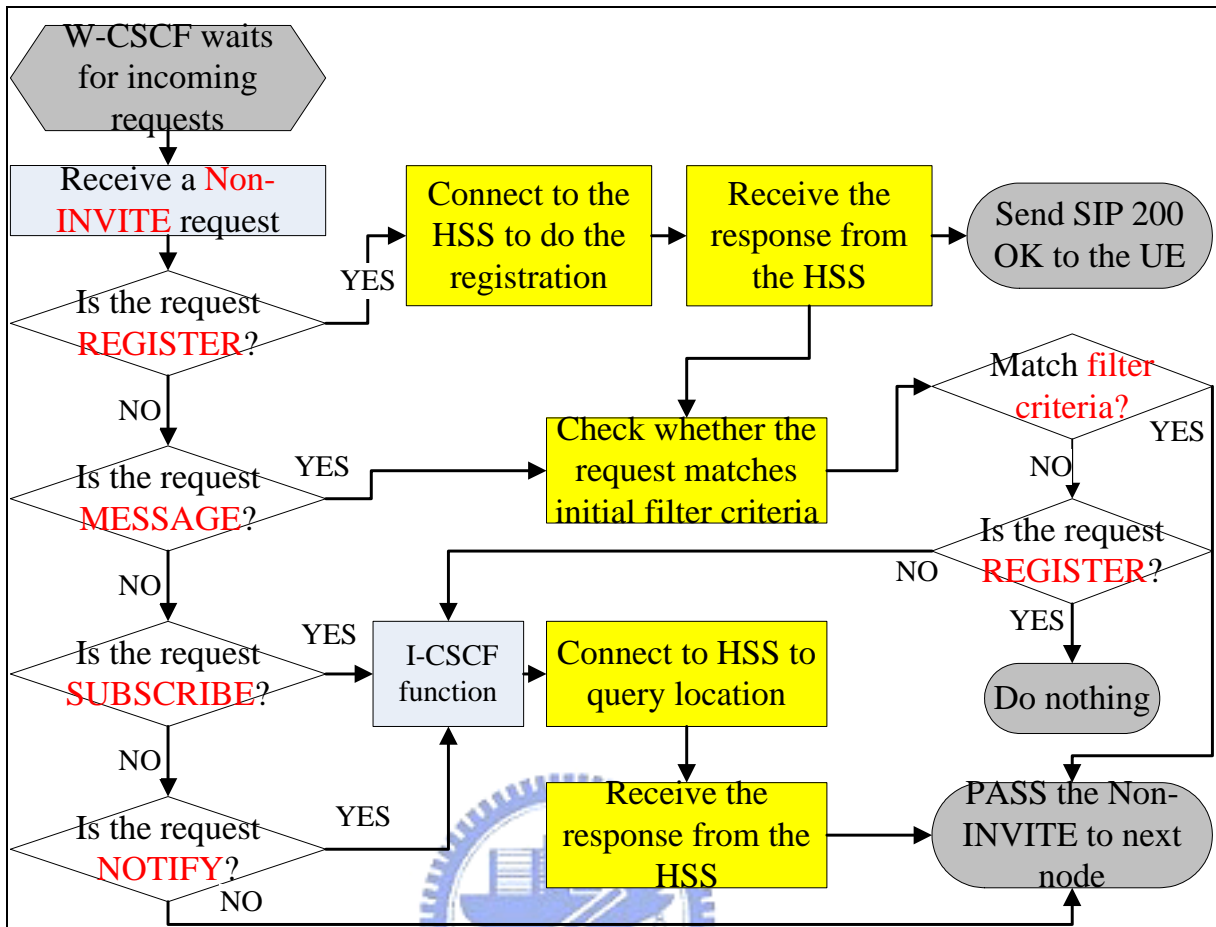


Figure 4-1: The flow chart of the Non-INVITE Server

The figure 4-1 shows the flow chart of the Non-INVITE Server that we have implemented in the W-CSCF. When the W-CSCF is ready to listen for incoming requests, the W-CSCF can accept a Non-INVITE request by the Non-INVITE Server. In our system, we will only deal with the SIP REGISTER, OPTIONS, SUBSCRIBE and NOTIFY messages and the other Non-INVITE messages will be direct passed to the next node. When receiving the REGISTER, we use SAR/SAA to register the UE and return SIP 200 OK to the UE. Moreover, we check the SPTs that we will send the REGISTER to the next node if the return value of the function is existence. By the way, we will do nothing if the return value is NULL. When receiving the OPTIONS, we check the SPTs to get one Application Server since we know that the message is used to set status for the UE. When receiving the SUBSCRIBE and NOTIFY, we use the LIR/LAA to get the CSCF location of the target and pass the message to next node.

4.2.2 The INVITE Server

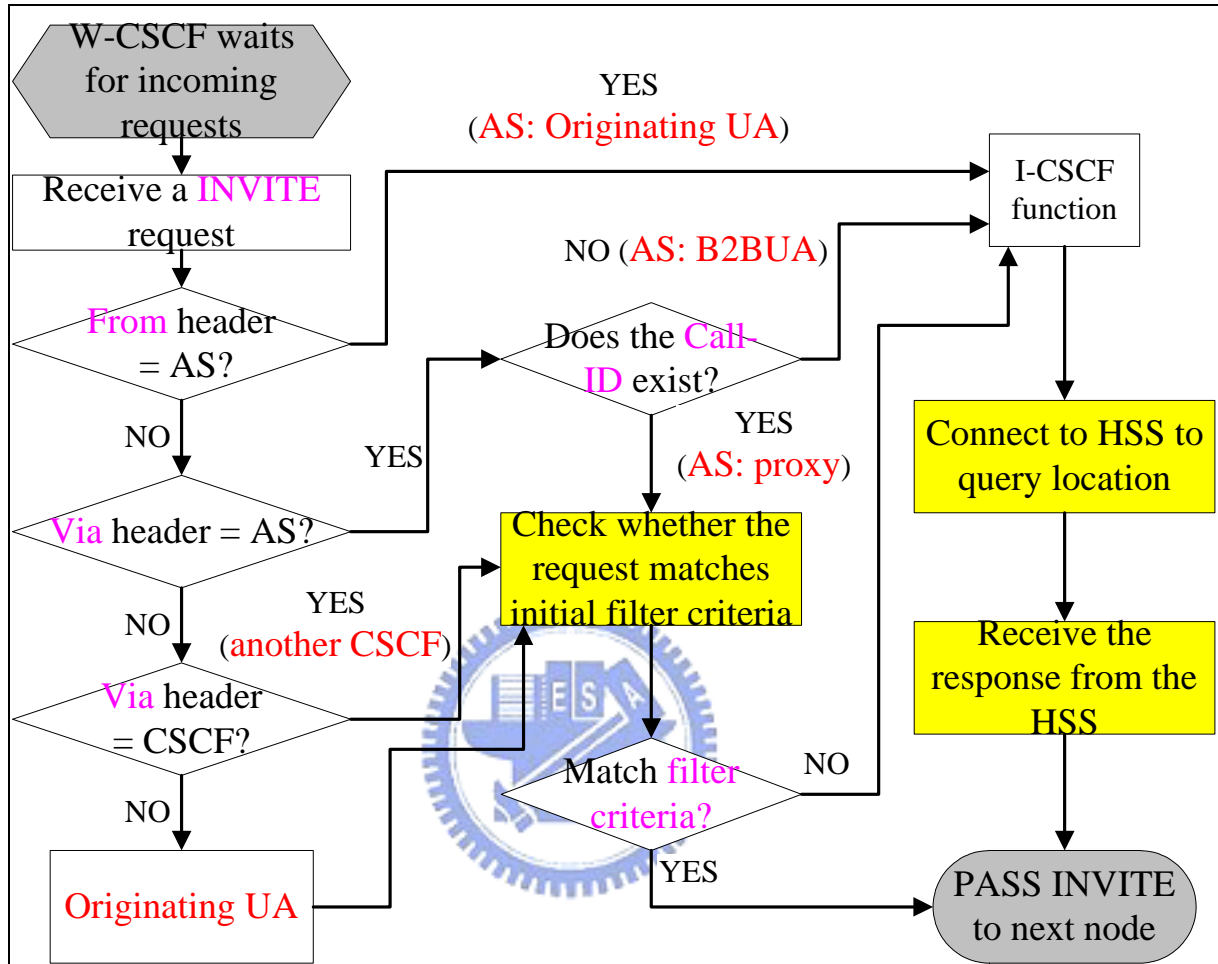
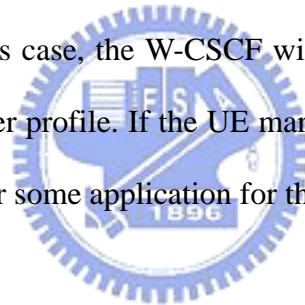


Figure 4-2: The flow chart of the INVITE Server

The figure 4-2 shows the flow chart of the INVITE Server that we have implemented in the W-CSCF. When the W-CSCF is ready to listen for incoming requests, the W-CSCF can accept an INVITE request by the INVITE Server. In our system, we will check where the INVITE message comes from by using the information which is included in the SIP headers. In order to achieve the goal, we create two table files containing the AS file and the CSCF file. We add all locations of the AS and the CSCF which are close to the W-CSCF in the files and the two files are used with the XML format. At first, we check that is the SIP From header including the AS location. If this INVITE is from the AS, the AS must act as originating UA

and the W-CSCF will use the LIR/LIA to find the serving CSCF of the target and pass the message out. If the SIP From header is not including the AS location, we will check that is the SIP Via header including the AS location. If the INVITE is via the AS, we will check that does the SIP Call-ID exist. If the Call-ID is not existence, the AS must act as B2B UA and the W-CSCF will use the LIR/LIA to find the serving CSCF of the target. But if the Call-ID is existence, the AS must act as proxy and the W-CSCF will check SPTs. If the return value is NULL, the W-CSCF will pass the message to next SIP node. If the SIP Via header is not including the AS location, we will check that is the SIP Via header including the CSCF location. If the INVITE is via the CSCF, we will check the SPTs as the same before and take down the Call-ID of this message. If the SIP Via header is not including the CSCF location, the originator of the INVITE must be the UE and the W-CSCF must act as a mobile originating (MO) CSCF. In this case, the W-CSCF will check the SPTs to find that does the UE mark any service in the user profile. If the UE marked the service, the W-CSCF will pass the INVITE to the AS to trigger some application for the UE.



4.2.3 The CxEventHandler function

The figure 4-3 shows the flow chart of the CxEventHandler function that we have implemented in the W-CSCF. When the Cx library triggers the CxEventHandler function, the function will check that the Cx Response is the Cx-Location-Query-Resp (LIA) or the Cx-Put/Pull-Resp (SAA). If the response is LIA, we will check that is the original SIP request INVITE message. If the original message is INVITE, we will check that does the W-CSCF act as a mobile terminating (MT) CSCF. If the W-CSCF acts as the MT CSCF, the W-CSCF will check the SPTs to match Filter Criteria. If one AS is matched, the W-CSCF will pass the INVITE to the AS. If no AS is matched, the W-CSCF will pass the INVITE to the WGSN

which the UE is located. If the W-CSCF does not act as the MT CSCF, the W-CSCF will pass the INVITE to the other CSCF. If the original request is not INVITE, the request will be SUBSCRIBE or NOTIFY message. Then, we will check that does the W-CSCF act as a MT CSCF and do the same as before. On the other hand, if the Cx response is SAA, it must do for registration. The W-CSCF will store the user profile into the local database and return the SIP 200 OK to the UE. Moreover, we will check the SPTs to match the Filter Criteria. If SPTs match, the W-CSCF will pass the REGISTER to the AS. If SPTs do not match, we will do nothing at all.

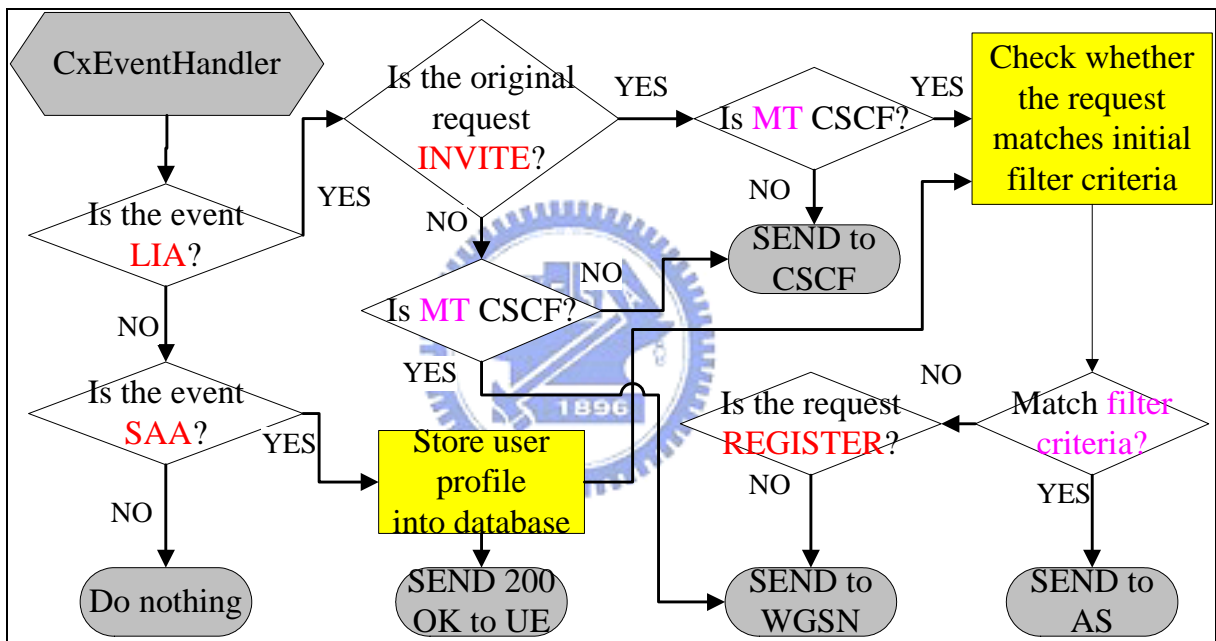


Figure 4-3: The flow chart of the CxEventHandler function

4.3 PoC/USV server

The PoC Server has implemented according to the PoC Specification by the member of our laboratory and we enhance the PoC Server to become the PoC/USV server. In each W-CSCF domain, there is only one PoC/USV server and it is close to the W-CSCF. On the other hand, the SIP UA cannot connect direct to the PoC/USV server. The figure 4-4 shows

the flow chart of the PoC/USV server. There are seven functions to support status monitoring containing the set online function, the set offline function, the set busy function, the get friend list function, the get friend status function, the set timer function and the check timer function. The USV server will check the incoming requests to trigger the registration function, the un-registration function, the change function and the subscription function. The termination of these functions is to send SIP NOTIFY message to the W-CSCF.

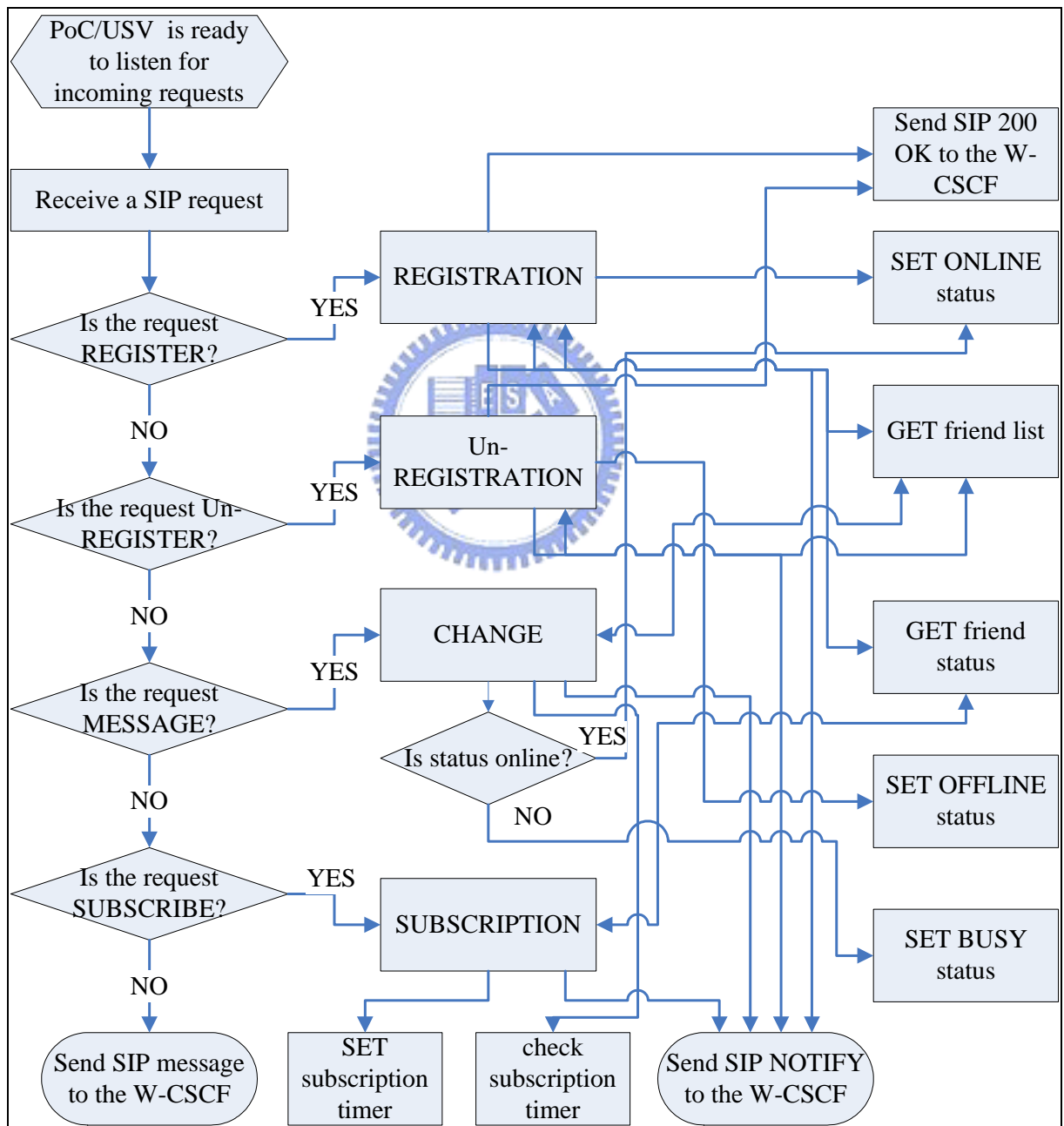


Figure 4-4: The flow chart of the USV server

4.4 SIP UA

The CCL SIP User Agent was implemented by the Computer & Communication Research Laboratories (CCL) of the Industrial Technology Research Institute. The UA does not support the Push-to-talk service and the function of the SIP OPTIONS, SUBSCRIBE and NOTIFY message and some SIP header. Therefore, we enhance it based on the CCL SIP protocol stack and Microsoft Windows API to fit in with our demands. By the way, we use the function which is supported by the MFC in the Microsoft Visual C++ 6.0 to connect to the GLMS to get the lists of the user.



Chapter 5 Conclusion

In this thesis, we present a W-CSCF that supports the WGSN users to access the IMS services. A UE, equipped with a 3G/WLAN dual module card, can roam between the WLANs and the GPRS network. Therefore, a mobile user can obtain the IMS services at any place, any time. We have implemented the W-CSCF which integrates the functions of I-CSCF and S-CSCF. A W-CSCF can process the SIP messages received from the UE and route the messages to the destination. A W-CSCF also handles the registration and location queries for a UE by sending Cx commands to the HSS and downloads the user profile of the UE. According to the SPTs in the SIP messages, the W-CSCF can trigger the local database to initial application for the UE. The advantage of W-CSCF is to reduce the number of messages exchanged and simplify the function model of S-CSCF. However, the W-CSCF may be overloaded because it serves as I-CSCF and S-CSCF.

In addition, we have implemented a method to enable a caller to see the current status of PTT users in advance. The method uses SIP messages for the status monitoring and to notify the mobile user before the user wants to make call to his friends. SIP MESSAGE is used to set new status for a UE and SIP NOTIFY is used to notify other user's current status.

In summary, we proposed a W-CSCF to route the SIP messages from the WGSN, a PoC server and application servers. By using the W-CSCF, we can make sure the authentication of SIP messages and maintain the registration status of the mobile user. Our future work is to reduce the frequency of SIP NOTIFY messages.

Chapter 6 Reference

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