

SIP-based VoIP Prepaid System on NTP VoIP Platform

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摘要

Session Initiation Protocol (SIP) 協定讓使用者間能夠簡單、快捷地建立網路語音連線。在電信國家型科技計畫下，我們建置了一個以 SIP 協定為基礎的網路語音通訊平台。透過這個平台，許多所大學的網路語音用戶可以使用 IP-IP 或 IP-PSTN 的電話服務。在這篇論文中，我們提出了一套網路語音的預付機制，使目前的網路語音通訊平台可以支援預付式之語音服務。而這個預付式系統只需修改一些 Call Server 的設定，便能夠整合到目前的網路語音通訊平台上運作。

關鍵詞： B2BUA、電信國家型科技計畫、預付式系統、RADIUS、SIP、網路語音通訊

Abstract

Session Initiation Protocol (SIP) provides simple and efficient handling of multimedia sessions among multiple users. Under the National Science & Technology Program for Telecommunications (NTP) in Taiwan, we have developed a SIP-based VoIP Platform for several university campuses in Taiwan. Subscribers can make IP-to-IP or IP-to-PSTN phone calls. In this paper, we propose a prepaid mechanism in the NTP VoIP Platform to handle the prepaid calls. This system can be easily integrated to the existing NTP VoIP Platform by re-configuring the Call Server.

Keywords: B2BUA, NTP, Prepaid System, RADIUS, SIP, VoIP

1. Introduction

Voice over IP (VoIP) is considered as one of the killer applications in the next generation telecommunications. By deploying a VoIP core network, service provider can reduce overall costs for

voice service and offer more advanced IP-based services. IETF has specified Session Initiation Protocol (SIP) [8] to provide simple and efficient handling of multimedia sessions among multiple users. 3GPP (Third Generation Partnership Project) adopts SIP for call control and signaling on its IP Multimedia Subsystem (IMS) [1] to provide a basis for rapid new service deployment and integration with IP services in the fixed network. Under the *National Science & Technology Program for Telecommunications (NTP)* in Taiwan, we have developed the *NTP VoIP Platform* [12] that provides SIP-based VoIP services for the university campuses in Taiwan. Subscribers in five universities can make IP-to-IP or IP-to-PSTN phone calls through this platform. Fig. 1 illustrates the NTP VoIP Platform architecture, which consists of the following components:

Call Server (Fig. 1(a)) provides primary capabilities for call-session control in the NTP VoIP Platform. It processes SIP requests and responses as a SIP proxy server. It also functions as a registrar that stores the contact information of each SIP user. At the current stage, the NTP Call Server is implemented on Window 2000 server running on an industrial PC.

PSTN Gateway (Fig. 1(b)) interworks between the NTP VoIP Platform and the Public Switched Telephone Network (PSTN), which allows IP phone users to reach PSTN users directly or indirectly through Private Branch Exchanges (PBXs; see Fig. 1(c)). Two models of PSTN gateways have been deployed in the NTP VoIP platform: the gateway developed by Industrial Technology Research Institute (ITRI) and Cisco 2600.

SIP User Agent (UA) is a hardware-based or a software-based SIP phone client (Fig. 1(d)) that provides basic call functions such as dial, answer, reject, hold/unhold, and call transfer. In the NTP VoIP

Platform, we have installed the SIP UA in software-based terminals including desktop computers, notebooks (with or without WLAN access), and PDAs (with WLAN access only). We also include hardware-based SIP phones manufactured by Cisco, Leadtek, Pingtel, and Snom.

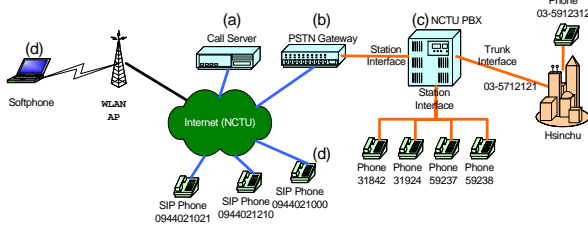


Figure 1: The NTP VoIP Platform

At the time of writing, the NTP VoIP Platform has been deployed in five universities: National Taiwan University (NTU; in Taipei City), National Tsing Hua University (NTHU; in Hsinchu City), National Chiao Tung University (NCTU; in Hsinchu City), National Cheng Kung University (NCKU; in Tainan City) and National Dong Hwa University (NDHU; in Hualien County). A Call Server and a PSTN gateway are deployed in each university, in accompany with several SIP hardphones and software-based SIP phone clients.

To control the domestic and long-distance call tolls, it is essential to deploy the appropriate components that collect and process charging information. In this paper, we describe the design and implementation of a prepaid mechanism for the NTP VoIP system.

2. NTP Prepaid System Architecture

Remote Authentication Dial-In User Service (RADIUS) [6] enables centralized authentication, authorization, and accounting functions (AAA) for network access. It is commonly used in network devices such as routers, switches and gateways. A RADIUS client sends user connection information in a RADIUS message to a RADIUS server. A RADIUS server authenticates and authorizes the RADIUS client request, and sends back a RADIUS response message. RADIUS clients can also send RADIUS accounting messages to the RADIUS server to log the session

accounting information [5].

A RADIUS message consists of a RADIUS header and RADIUS attributes. The RADIUS attributes are used to convey information between RADIUS clients and RADIUS servers. For example, the list of the attributes in the Access-Request message includes the user information of the connection attempt. On the other hand, the list of attributes in the Access-Accept message includes information about the connection constraints.

Fig. 2 illustrates the architecture for the NTP Prepaid System. The RADIUS protocol is utilized to provide AAA functions in the NTP Prepaid System. It includes four components. The RADIUS client resides in a SIP-based Back-to-Back UA (B2BUA; see Fig. 2(a)) can receive, process and send SIP messages. It exchanges SIP messages between the call parties and terminates the call when the authorized session time for the prepaid call is reached. Before a prepaid call starts, it sends an Access-Request message to the RADIUS server (Fig. 2(b)) for authorization. When the prepaid call terminates, it sends an Accounting-Request message to the RADIUS server to log the session accounting information (such as session time, etc.). The RADIUS server (Fig. 2(b)) processes user authorization and user accounting requests. It responds the RADIUS client with authorized information (such as maximum call time). It also processes the RADIUS accounting messages and stores the accounting record in non-versatile storage, e.g., the AAA database.

The AAA database (Fig. 2 (c)) stores user related AAA information. The RADIUS server queries this SQL-based database to retrieve user information. The AAA database also acts as the accounting database storing the user Call Details Records (CDRs). The OA&M system (Fig. 2(d)) can be accessed from web browsers over Secure Sockets Layer (SSL) protocol. Through this system, the administrators can view and/or modify user information and browse the prepaid CDRs generated by the NTP Prepaid System. The NTP Prepaid System is developed based on open-source packages including Vovida VOCAL B2BUA [11], FreeRADIUS [9] and PostgreSQL [10].

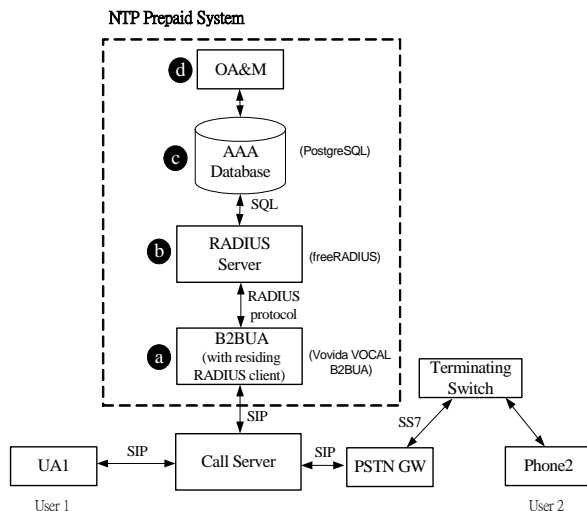


Figure 2: The NTP Prepaid System Architecture

By utilizing the back-to-back (B2B) UA technique, the NTP prepaid mechanism is inserted in a VoIP call by breaking the signaling session into two sub-sessions. Consider the scenario where the User 1 at UA1 attempts to establish an IP-PSTN call to User 2 at Phone2. The B2BUA of the NTP Prepaid System breaks the SIP session between UA1 and PSTN Gateway (PSTN GW) into two sessions, i.e., a session between UA1 and the B2BUA (Session 1) and a session between the B2BUA and the PSTN GW (Session 2). By inserting the B2BUA in the middle of a prepaid call session, the B2BUA can trigger the RADIUS Accounting-Request message with status “start” or “stop” when it receives SIP messages such as INVITE/200 OK and BYE; and terminates the prepaid call session when the prepaid credit of a user depletes.

Following Taiwan’s numbering plan (based on E.164 recommendation [2]), the telephone number for a SIP phone in the NTP VoIP Platform is of the format *0944-nnn-xxx*. The first four digits *0944* are the service code for NTP VoIP Platform approved by Directorate General of Telecommunications (DGT) of Taiwan for experimental usage. The next three digits *nnn* refers to the university number (e.g., *003* represents the NCTU). The remaining digits *xxx* represent a customer line number automatically generated by the NTP VoIP registration system. Some numbers are reserved for prepaid call services. When the Call Server receives SIP requests (such as INVITE, ACK, CANCEL or BYE) from a prepaid user, it

forwards the requests to the B2BUA for authorization. After the authorization, the B2BUA will send the authorized requests to the Call Server, and the Call Server then forwards them to the related SIP devices (such as the PSTN GW). On the other hand, if the Call Server receives SIP requests from non-prepaid users, it forwards the requests directly to the related SIP devices without involving the Prepaid System. The above mechanism is achieved by re-configuring the call routing rules on Call Server according to the callers’ phone numbers.

The NTP Prepaid System also collects data on the network resource usage for the prepaid services. For a prepaid call, the statistics are collected in a Call Detail Record (CDR) for billing purposes (i.e., for generating an itemized bill for a subscriber) and the statistical analysis of service usage. The CDR format used in the NTP VoIP Platform is shown in Table 1, where the CDR fields are the same as the attributes defined in the RADIUS protocol, as shown in Table 1.

<u>Field</u>	<u>Example Value</u>
radacctid	6201
username	Alice
nasipaddress	140.113.250.25
acctstarttime	2004-06-17 10:29:21
acctstoptime	2004-06-17 10:29:52
acctsesssiontime	00:00:31
calledstationid	035795128
callingstationid	0944003921
acctterminatecause	User-Request

Table 1: The CDR Format.

In this table, the *radacctid* is a unique CDR identity. The *username* is the user identity used in the NTP VoIP Platform. The *nasipaddress* indicates the IP address of the CDR generating node. The *acctstarttime* and the *acctstoptime* are the timestamps when the CDR was started and was ended, respectively. The $acctsesssiontime = acctstoptime - acctstarttime$ is the elapsed time of the call. The *calledstationid* and the *callingstationid* are the identities (the telephone numbers) of the called party and the calling party of the call, respectively. The *acctterminatecause* indicates how the session was terminated (ie., by User-Request or Session-Timeout).

3. Prepaid Call Setup and Termination

This section describes the message flows for the prepaid call setup and termination. For simplicity, the non-final SIP messages (such as 100 Trying or 180 Ringing) are not shown. In these message flows, SIP messages are exchanged through the Call Server, the B2BUA and the PSTN GW, and the SS7 messages are exchanged between the Terminating Switch of Phone2 and the PSTN gateway [4].

3.1 Message Flow for Call Setup

Fig. 3 shows the scenario where a prepaid User 1 at UA1 makes a call to User 2 at Phone2 of the PSTN. The call setup procedure is described as follows:

Step 1. UA1 sends the Call Server an INVITE message with Request-URI designating to User2. This INVITE message has a unique Call-ID with suffix “@UA1_IP”. According to the configuration in the Call Server, the Call Server forwards this message to the B2BUA for authorization.

Step 2. When the B2BUA receives the INVITE message, its residing RADIUS client sends an Access-Request message to the RADIUS server. The message includes the prepaid username (i.e., the prepaid phone number).

Step 3. When the RADIUS server receives the message, it queries the AAA database to retrieve the prepaid user’s record. If no record is found, the RADIUS server will reply an Access-Reject message indicating that the user is not an authorized prepaid user. Otherwise, the RADIUS server replies with an Access-Accept message. This message contains the prepaid credit in the Session-Timeout attribute. The Session-Timeout attribute defined in RFC 2865 [6] indicates the maximum session time for the authorized user. In our design, the definition of this attribute is generalized as the maximum quota representing either the amount of money or the amount of time.

Step 4a. When the B2BUA receives the Access-Accept message, it generates another INVITE message to User 2 (i.e., it starts Session 2). The Call-ID header field includes a unique call

identity ended with “@B2BUA_IP”. The contact address of the B2BUA is filled in the Contact header field. This new INVITE message is then sent to the Call Server.

Step 4b. After the Call Server receives the INVITE message (with Call-ID suffix “@B2BUA_IP”) from the B2BUA, it forwards the message to the appropriate SIP device. Since Phone2 is a PSTN phone, this message is sent to the PSTN GW.

Steps 5 and 6. The PSTN GW generates the Initial Address message (IAM) to the Terminating Switch of Phone2. Phone2 then starts ringing. When User 2 picks up the phone, the Answer message (ANM) is sent to the PSTN GW by the Terminating Switch. The message indicates that this call has been accepted by the called party.

Step 7. The PSTN GW generates a final response 200 OK. The contact address of the PSTN GW is filled in the Contact header field. The 200 OK message is routed to the Call Server and then forwarded to the B2BUA.

Steps 8 and 9. Upon receipt of the final response, the B2BUA starts a duration timer with the value based on the Session-Timeout attribute (obtains in Step 3). The RADIUS client sends an Accounting-Request message with status “start” indicating the beginning of a prepaid call session. The RADIUS server creates an accounting CDR in the AAA database (via the SQL command INSERT) and acknowledges the RADIUS client by an Accounting-Response message.

Step 10. When the RADIUS server creates the corresponding CDR, the B2BUA generates a 200 OK to UA1. The contact address of the B2BUA is set in the Contact header field. This response is routed to the Call Server and then forwarded to UA1.

Steps 11 and 12. Upon receipt of the 200 OK message, UA1 sends the ACK message to the B2BUA. The B2BUA sends the ACK to the PSTN GW. At this point, the PSTN GW opens a UDP port for the real-time transport protocol (RTP) [3, 7] so the voice packets sent by UA1 can pass through the PSTN GW. The prepaid call session is established.

Steps 13. The B2BUA sends an INFO message to inform UA1 about the authorized call time.

Steps 14. When UA1 receives the INFO message, it replies with a final response 200 OK message.

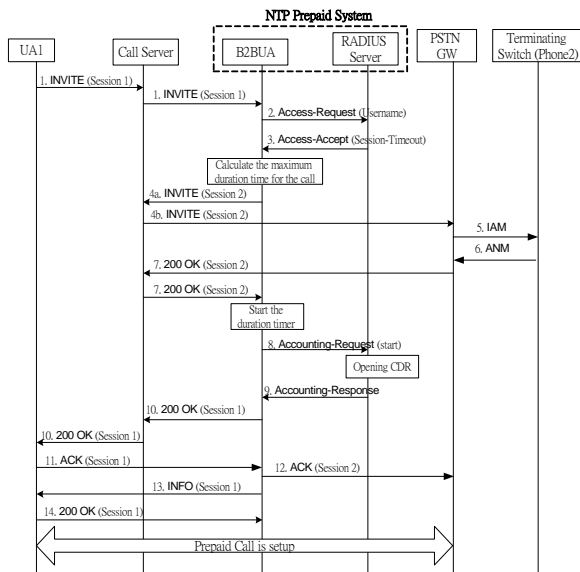


Figure 3: Message Flow for Call Setup

3.2 Message Flow for Call Termination

When the prepaid call is terminated by either call party, the BYE message is sent to the B2BUA directly. Then the B2BUA triggers the Accounting-Request message with status “stop” to close the CDR. Fig. 4 illustrates the message flows for the call session termination in different scenarios. In Fig. 4(a), the call session is terminated by UA1 with the following steps:

Step 1. UA1 sends a BYE message to the B2BUA.
Steps 2 and 3. When the B2BUA receives the BYE message, its residing RADIUS client invokes an Accounting-Request message with status “stop” to the RADIUS server. This message includes a Terminate-Cause attribute with value “User-Request”. That is, the call is terminated by the call party, not by the Prepaid System. The RADIUS Server responds with an Accounting-Response message when the accounting information is stored. In our deployment, the accounting information is stored in the AAA database via the SQL command UPDATE issued by the RADIUS server.

Step 4. The B2BUA sends a BYE message to the PSTN GW.
Steps 5 and 6. The PSTN GW sends the Release message (REL) to the Terminating Switch of Phone2. The Terminating Switch replies with the Release Complete (RCL) message. At this point, the PSTN GW closes the UDP port for the RTP connection so

the voice packets sent by UA1 cannot pass through the PSTN GW.
Steps 7 and 8. The PSTN GW replies a 200 OK message to the B2BUA, and the B2BUA generates a 200 OK message to UA1. The call session is successfully terminated.

In Fig. 4(b), the call session is terminated by the PSTN Phone2. The message flow is similar to the one shown in Fig. 4(a), so the description is omitted. In Fig. 4(c), the call session is force-terminated by the NTP Prepaid System with the following steps:
Step 1. When the authorized session timer in the B2BUA (started in Fig. 3 (Step 8)) is timed out, it implies that the prepaid credit is depleted. The B2BUA sends BYE messages to both UA1 and the PSTN GW.
Steps 2 and 3. The PSTN GW sends the REL message to the Terminating Switch of Phone2. The Terminating Switch replies with the RLC message. At this point, the PSTN GW closes the UDP port for the RTP connection so the voice packets sent by UA1 cannot pass through the PSTN GW.
Steps 4 and 5. The B2BUA then invokes an Accounting-Request message with status “stop” and Terminate-Cause “Session-Timeout” to the RADIUS server indicating force termination of the prepaid call session. The RADIUS Server responds with an Accounting-Response message when the accounting information is stored.
Steps 6 and 7. When the B2BUA receives the 200 OK messages from UA1 and the PSTN GW, the call is successfully terminated.

the voice packets sent by UA1 cannot pass through the PSTN GW.

Steps 7 and 8. The PSTN GW replies a 200 OK message to the B2BUA, and the B2BUA generates a 200 OK message to UA1. The call session is successfully terminated.

In Fig. 4(b), the call session is terminated by the PSTN Phone2. The message flow is similar to the one shown in Fig. 4(a), so the description is omitted. In Fig. 4(c), the call session is force-terminated by the NTP Prepaid System with the following steps:

Step 1. When the authorized session timer in the B2BUA (started in Fig. 3 (Step 8)) is timed out, it implies that the prepaid credit is depleted. The B2BUA sends BYE messages to both UA1 and the PSTN GW.

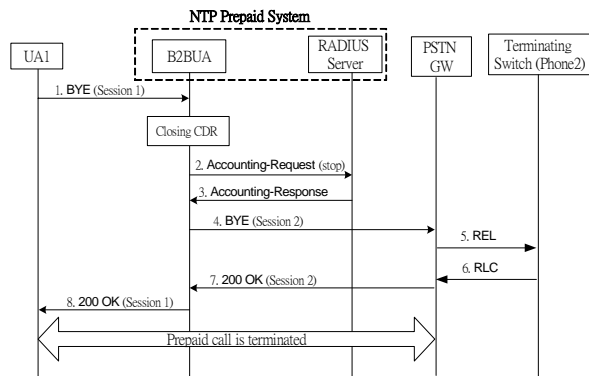
Steps 2 and 3. The PSTN GW sends the REL message to the Terminating Switch of Phone2. The Terminating Switch replies with the RLC message. At this point, the PSTN GW closes the UDP port for the RTP connection so the voice packets sent by UA1 cannot pass through the PSTN GW.

Steps 4 and 5. The B2BUA then invokes an Accounting-Request message with status “stop” and Terminate-Cause “Session-Timeout” to the RADIUS server indicating force termination of the prepaid call session. The RADIUS Server responds with an Accounting-Response message when the accounting information is stored.

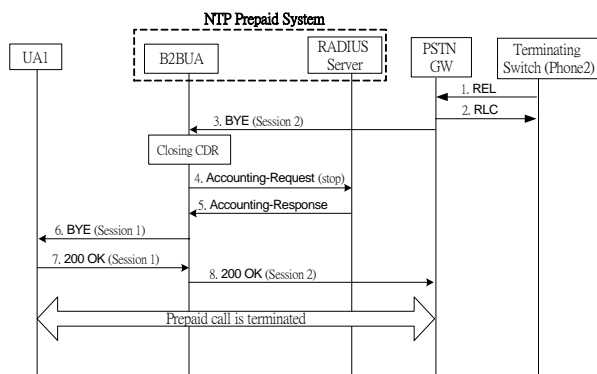
Steps 6 and 7. When the B2BUA receives the 200 OK messages from UA1 and the PSTN GW, the call is successfully terminated.

4. Conclusion

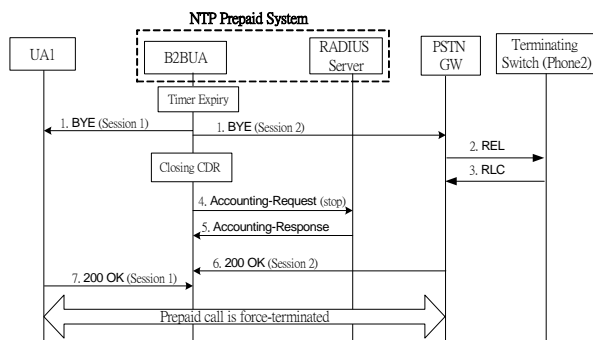
The NTP VoIP Platform provides SIP-based VoIP services for universities in Taiwan. To control the domestic and long-distance call tolls, this paper designed a SIP-based Prepaid System to handle the prepaid calls. This system can be easily integrated to the existing VoIP network by re-configuring the Call Server.



(a) Call session terminated by the SIP UA



(b) Call session terminated by the PSTN phone



(c) Call session terminated by the NTP Prepaid System

Figure 4: Message Flows for Call Termination

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