在網際網路語音服務上設計支援視訊會議之網頁式電話及利用自由軟體開發互動式語音應答系統

Designing a Web-based Phone to Support Video Conferencing and the IVR System with Freeware over VoIP Networks

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摘要

對於網際網路語音服務的使用者和電信業者而言，網頁式電話是一個具有吸引力的解決方案。網際網路語音服務的使用者不論是在何時何地，只要透過網頁瀏覽器的使用並連接上網路，就可方便地使用網際網路語音服務。本論文是透過自由軟體來建立一個網際網路語音服務平台，並結合網頁式電話，提供語音、視訊會議、即時訊息及後付式/預付式的低成本之網際網路語音服務。互動式語音應答系統也能安裝在此平台上，有助於預付用戶用來查詢餘額及再計費。最後結論，我們的系統可以很容易地在花費最小成本的考量上來安裝及建立在一般大學，學院及小型公司上。

關鍵詞：互動式語音應答、視訊會議、網際網路語音服務、網頁式電話。

Abstract

The web-based phone is an attractive solution for VoIP users and the operators. A VoIP user only requires a web browser and an internet connection to use the services at anytime, anywhere. In this paper, freeware is used to build up a low-cost VoIP platform providing the web phone, voice communication, video conferencing, instant messaging and postpaid/prepaid services. An interactive voice response (IVR) is also installed to interact with the prepaid users for querying their balance and recharging. As a final remark, our system can be easily installed and established in universities, colleges and small companies with minimum cost.

Keywords – IVR, Video Conferencing, VoIP, Web-based Phone.

1.Introduction

Due to the low-cost and the convenience of Internet, the number of people who use VoIP (Voice over IP) services is growing rapidly. Many user agents (UAs) [6, 7, 13, 15, 16, 27] are freely provided to encourage customers to use the fascinating services. For example, Skype is one of the most used VoIP UA. The number of Skype users is over 100 millions in the world [10] and the user base is growing at a rate of about 6 Million users per month [1].

Table 1 compares the functions of these UAs. As shown by the table, instant messaging and video communication have become indispensable functions of a UA. Although these UAs are free, each user must join the corresponding member groups before using the VoIP services. For the security reason, lots of companies forbid their employees using these UAs while they are working.

Table 1. Comparisons of the functions of the UAs

Inc. / Services	Microsoft MSN		YAHOO	Google	Skype	Gogonet	Seednet
Product	MSN Messenger		Messenger	Google Talk	Skype	GogoTalk	Wagaly TelTel
Interface	Soft.	Web	Soft.	Soft.	Soft.	Soft.	Soft.
IM	O	O	O	O	O	O	O
Audio	O	X	O	O	O	O	O
Video	O	X	O	X	O	O	O
Dialout	X	X	X	X	O	O	O

Some advanced services (e.g., video conferencing) are not provided in these UAs. It is also difficult for the enterprise to modify these UAs to develop services which can meet with their requirements.

The web browser is a free and common tool for internet access. The portal service and e-commerce service are accessed via the browser using internet protocols. Hence, the web-based phone is an attractive solution for VoIP users and the operators. Some companies developed the web-based video conferencing [11, 25] with VoIP services. Any user with a broadband internet connection and a browser can easily log in the system and can take part in or host a web/video conferencing. However, the cost is still too expensive for the campus and small companies.

In this paper, freeware is used to build up the VoIP platform and services with minimum cost. Our system can provide voice communication, video conferencing, instant messaging, web phone and postpaid/prepaid services to the user. An interactive voice response (IVR) is also installed to interact with the prepaid users for querying balance and recharging. With the support of the NTP (National Science and Technology Program for Telecommunication) project, we had built up the platform with basic services [22, 26]. In this paper, we focus on designing the web-based phone supporting the video conferencing services and the IVR function with freeware.

2. Related works

In early days, many research topics of video conference were based on H.323 protocol. In [2], Houatra took experiments with a video-conferencing platform with the distributed object services. Chen [23] proposed a media synchronization mechanism, a CPU job scheduler and an adaptive flow control model to make H.323 multimedia conferencing systems more comfort to non-guaranteed QoS networking environment. In addition, video conferencing had been tested in the IPv6 network [28]. As the SIP protocol becomes a popular VoIP standard, many researches have been developed based on this new protocol. In [18], the authors proposed an experimental system internetworking between SIP and DMIF for video-conferencing applications over heterogeneous networks. Yang [29] provided a service model to support dynamic scalability. The multicast service for video-conferencing application was investigated in [4, 19]. Han [8] designed and implemented a dedicated module in the home server supporting four-way video conferencing.

In telecommunication networks, an IVR interacts with the subscribers and accepts user inputs to provide appropriate responses in the form of voice, fax and other media. It helps the operator to improve the level of customer's satisfaction. However, the operation of commercial IVR

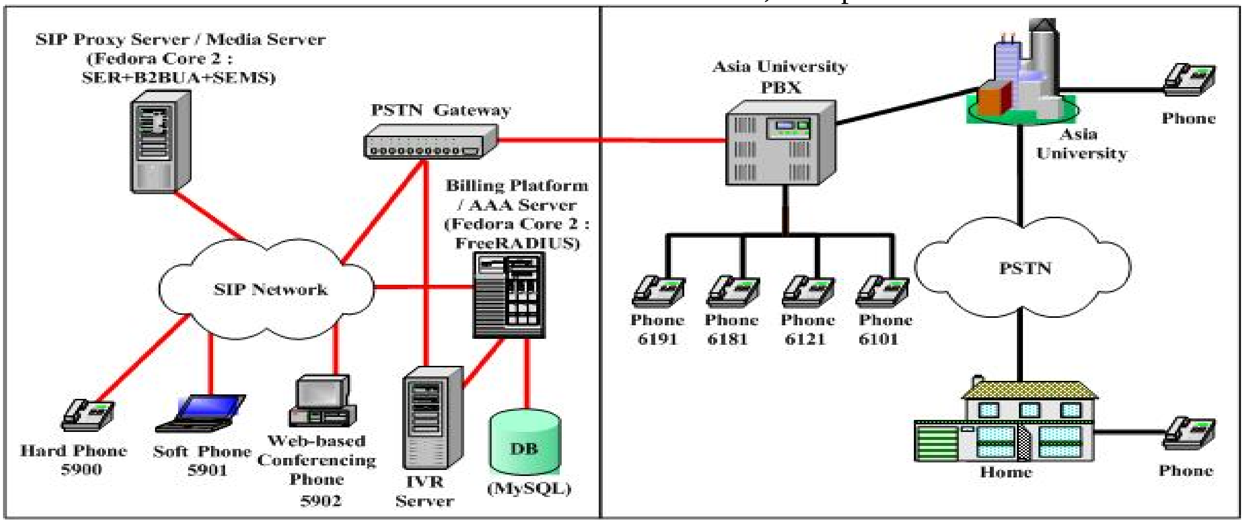

Figure 1. The architecture of the prepaid VoIP platform

software is complex and the cost of an IVR is too expensive for the campus and small companies. In this paper, the freeware－IVR with Magic Flow [17] is used to build our IVR system. This software is developed by PhoneBazooka [14]. It is simple to use and has abundance of features such as executing inbound/outbound IVR, VoiceXML text to speech, playing voice files and recording the user's voice to the hard disk. It can support 48 lines and can integrate with the voice modems and dialogic cards. The IVR with Magic Flow uses visualized tree view to help the developer to build the voice streams. Besides, it can access databases such as Microsoft Access, SQL server, MySQL and Oracle. It can also provide customized services by using the VBScript or the JavaScript.

The paper is organized as follows. Section 3 describes the architecture of our prepaid VoIP platform. Section 4 shows the architecture of the web-based phone and a demo of the video conferencing service. Section 5 describes the tree flow in IVR with Magic Flow for the IVR system. The conclusions are given in Section 6.

3. Prepaid VoIP system platform

Fig. 1 shows the architecture of our VoIP platform. The platform consists of hard/soft/web-based phones, a SIP proxy server/media server, a PSTN gateway, an IVR server, a database and the PSTN/mobile networks. The SIP proxy server/media server contains the SER (SIP Express Router) software [21], the B2BUA (Back-To-Back User Agent) software [24] and the SEMS (SIP Express Media Server) software [20]. The proxy server offers call control and call authorization functions. The media server provides audio conferencing services.

The SER has all the functions as a proxy, a registrar and a redirect server. It has very stable performance and many features (e.g., SMS gateway, SIMPLE2 Jabber gateway, RADIUS/syslog accounting and authorization, status monitoring).

The B2BUA can act as both a SIP user agent server (UAS) and a SIP user agent client (UAC). In our VoIP platform, the B2BUA is used as a prepaid call controller. It forwards a prepaid call to the RADIUS server for authorizing. The RADIUS server checks if the subscriber's credit is sufficient to make a prepaid call.

The SEMS provides the audio mixing function. When users start the video conferencing service, their voices are mixed by the SEMS and are transmitted to their UAs.

In our VoIP system, FreeRADIUS software [5] is used as our RADIUS (Remote Authentication Dial In User Service) server to provide authentication, authorization and billing functions [3, 9]. It is selected not only because it is free but also it has been proved to have a very high compatibility with the MySQL database and the PSTN gateway. The

MySQL database stores all prepaid/postpaid call detail records (CDRs) and the users' profiles.

Step 1 : A subscriber uses the browser to connect to the web server.

Step 2 : The subscriber downloads the WebPhone.CAB file from the web server.

Step 3 : After the download completes, the subscriber can enable the WebPhone.CAB to use the web phone service. The WebPhone.CAB executes the RTC client API component to send an "INVITE" message including the user's ID and password to SER. The SER will pass the subscriber information to FreeRADIUS to check if the subscriber is a legal user. If the verification is successful, the subscriber can start to use the web-based phone.

Step 4 : When the subscriber has logged in, the WinSock of WebPhone.CAB will be enabled to connect to the SQL Query Service.

Step 5 : The SQL Query Service can query the MySQL database about user's balance.

Step 6 : Finally, the user's balance information is sent back to the client.

4. Video conferencing with the web-based phone

In this section, we first introduce the architecture of the web-based phone. Next, a flowchart is used to describe the service flow of video conferencing. Finally, a video conferencing demo is shown in Section 4.3.

4.1 Architecture of a web-based phone

Fig. 2 shows the architecture of the web-based phone. The architecture includes the client, a web server and a SIP proxy server.

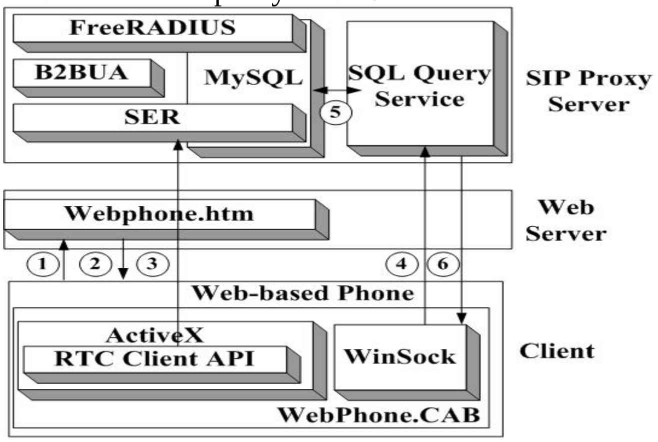

Figure 2: Architecture of the web-based phone

- Client :

The client includes a PC and an application program –WebPhone.CAB. The WebPhone.CAB is an ActiveX component that encapsulates the RTC client API [12]. When a user logs in, his/her PC will download this file from the web server. The user can use a web-based phone by activating the WebPhone.CAB.

- Web server :

The home page (i.e., Webphone.htm) file and the WebPhone.CAB file are stored in the working directory of a web server.

- SIP proxy server :

The SIP proxy server includes the MySQL database, SER, FreeRADIUS and B2BUA software.

A subscriber can use the browser to log in our system to use the web-based phone. The procedures are shown in Fig. 2 and are illustrated as follows.

4.2 The flow of video conferencing

Fig.3 shows the flow of a video conferencing using the web-based phone. The procedures are described as follows.

Step 1 : A subscriber inputs his/her account and password to log in the system.

Step 2 : The SER queries the MySQL database to verify the subscriber. If the verification is successful, the subscriber can start to use the web-based phone. Otherwise, the verification process is restarted or the service is stopped.

Step 3 : The subscriber chooses the IDs of the meeting members on the friend list to start the audio and video conferencing service.

Step 4 ~ Step 5 : The voices of all participants are transmitted from the SEMS in the media server to the mixer module. The mixer module mixes the voices and sends the voices back to all participants.

Step 6 ~ Step 9 : Every participant transmits his/her video frames to other participants. In this process, every participant must wait to receive video frames transmitted from other participants.

The video conferencing service continues as the loop going from Step 4 to Step 9.

Step 10：The user stops the video conferencing service.

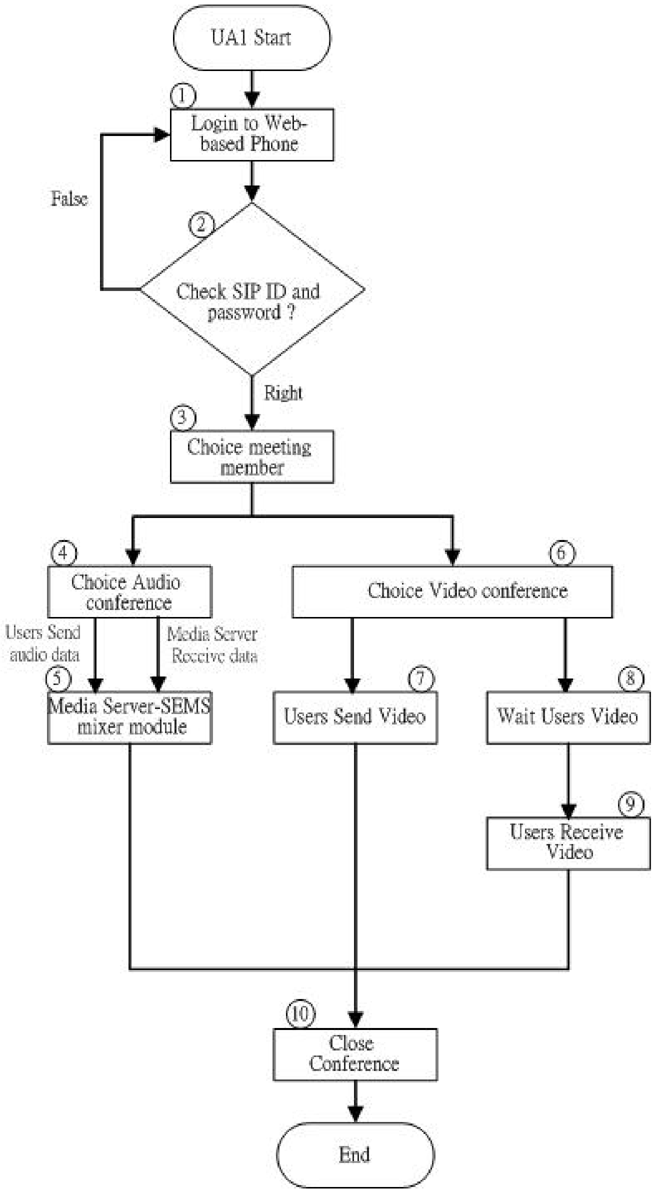

Figure 3. The flow of the video conferencing

4.3 A demonstration of video conferencing

Fig. 4 shows the web interface when a subscriber logs in to use the web-based conferencing services. The subscriber can use the VoIP call (PC-to-PC or PC-to-phone), video call (PC-to-PC) and instant messaging (PC-to-PC) services. The "DialOut" button on the top of the left enables the subscriber to make a PC-to-phone call. The left block is the friend list. A subscriber can insert or remove his/her friend's ID from the list. The middle block shows the contents of instant messaging when a subscriber uses the IM service. By double-clicking on the ID of his/her friend, the subscriber can easily use the audio, video and IM services.

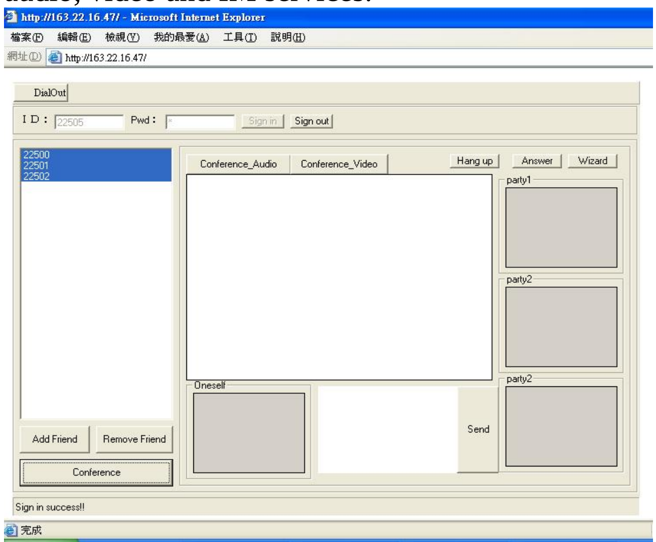

Figure 4. The web interface of web-based conferencing services

Fig. 5 shows the web page when a subscriber is using the web-based video conferencing service. The subscriber selects three IDs and clicks on the "Conference" button under the friend list. The initiator of the video conferencing can start the conferencing service by clicking two middle buttons of the page — "Conference_Audio" and "Conference_Video".

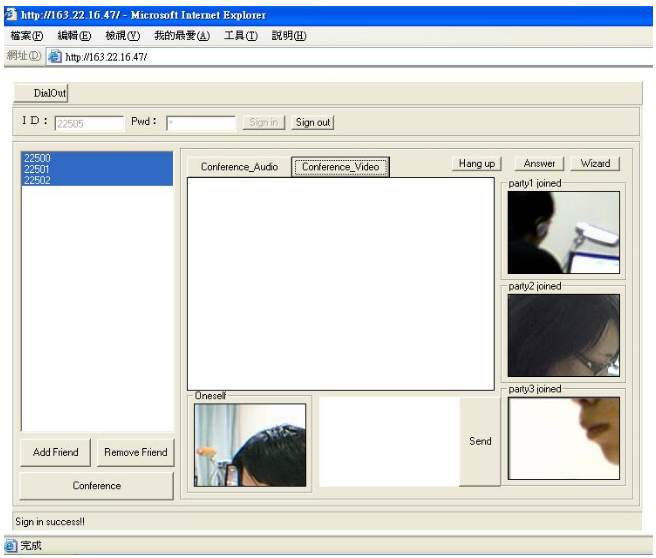

Figure 5. The web page when a subscriber is using the web-based video conferencing service

5. IVR system

The IVR system provides services and information to the subscriber without a human operator. A subscriber dials the service number and enters the IVR system through the PSTN gateway. Two services are provided in our IVR server: balance querying and recharging. The flow of IVR services is illustrated as below.

5.1 The flow of IVR services

Fig. 6 shows the flow of IVR services and Fig. 7 represents the services by a tree structure in Magic Flow. The tree contains nodes representing the triggered events that need to be processed. The whole flow is divided into three sub-flows as shown in Figs.8 ~ 10.

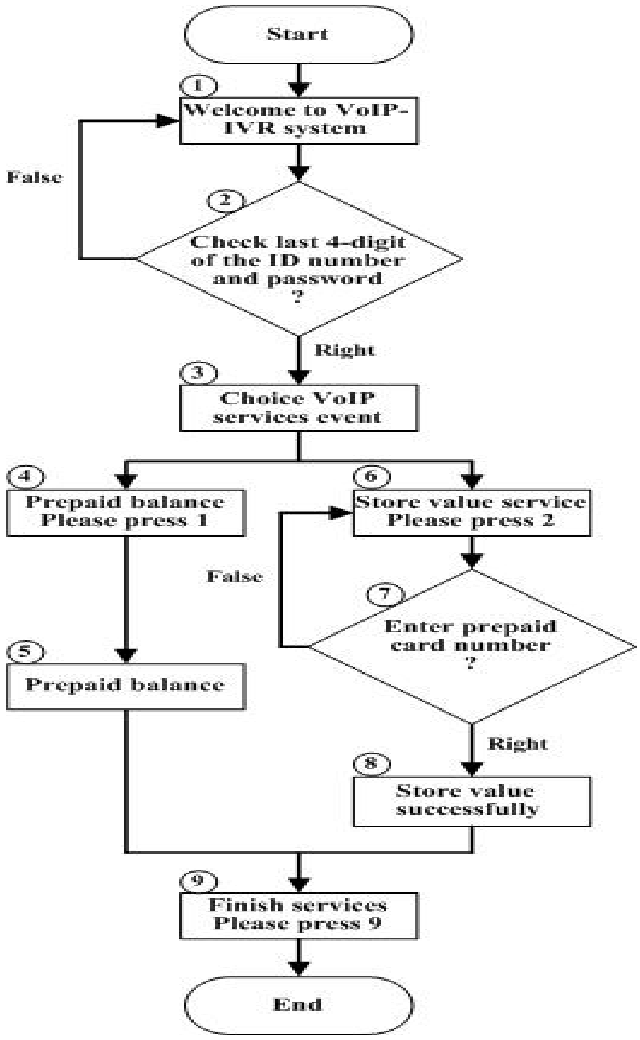

Figure 6. The flow of IVR services

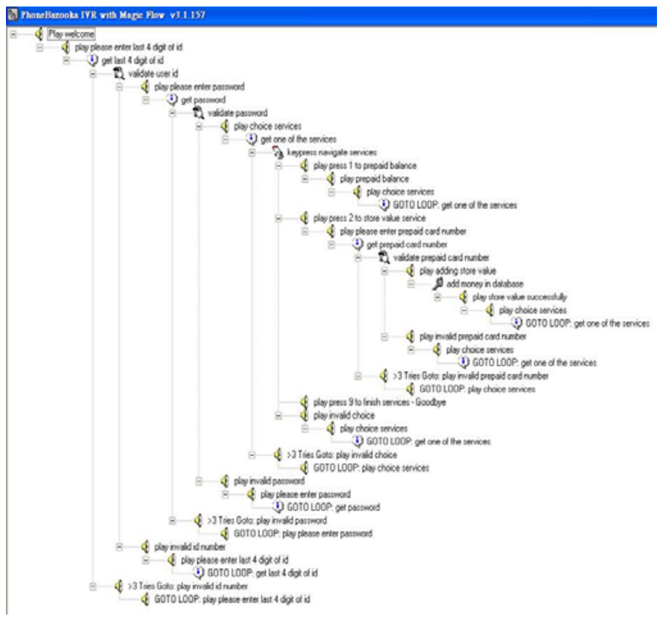

Figure 7. Tree structure in Magic Flow

- Sub-flow 1：

Fig. 8 shows the trees structure of sub-flow 1. Sub-flow 1 corresponds to the Step 1 and Step 2 in Fig. 6. The IVR plays a welcome announcement and prompts the subscriber to input the ID and password. The IVR interacts with the database through the "validate user id" and "validate password" nodes to verify the user's account.

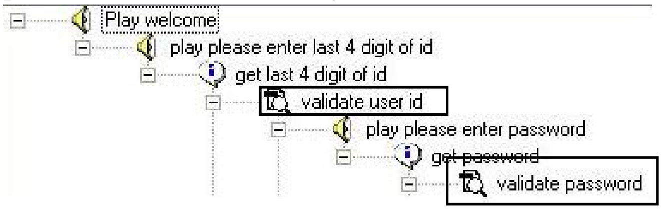

Figure 8. Sub-flow 1

- Sub-flow 2：

As shown by Fig. 9, sub-flow 2 corresponds to the Step 3 to Step 9 in Fig. 6. The IVR plays an announcement to prompt the subscriber to select the service. With "keypress navigate service" node, the IVR provides the service based on the input key. The IVR enables a subscriber to query his/her balance if he/she presses "1" key. If a subscriber presses "2" key, then the IVR provides the recharging function. The subscriber is prompted to input the number of a prepaid card. After receiving the card number, the IVR interacts with the database through the "validate prepaid card number" node to verify the card number. If the card number exists, then the credit can be recharged through the "add money in database" node. At last, the subscriber exits the service by pressing "9" key.

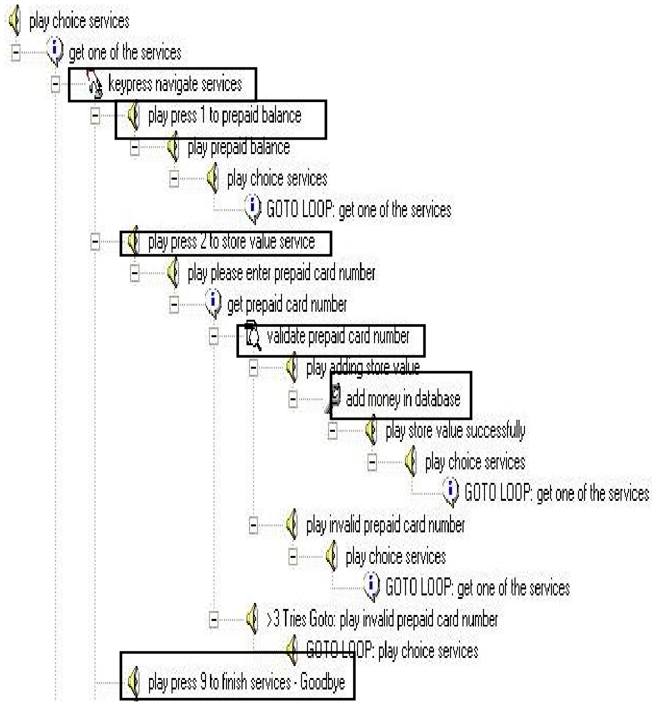

Figure 9. Sub-flow 2

- Sub-flow 3：

The Magic Flow inserts an error handling process as long as the IVR asks the subscriber to press a key. As Fig. 10 shows, sub-flow 3 corresponds to the Step 2 to Step 7 in Fig. 6. If the subscriber's input is invalid, then the "GOTO LOOP" event will be triggered to ask the subscriber to input the required information again.

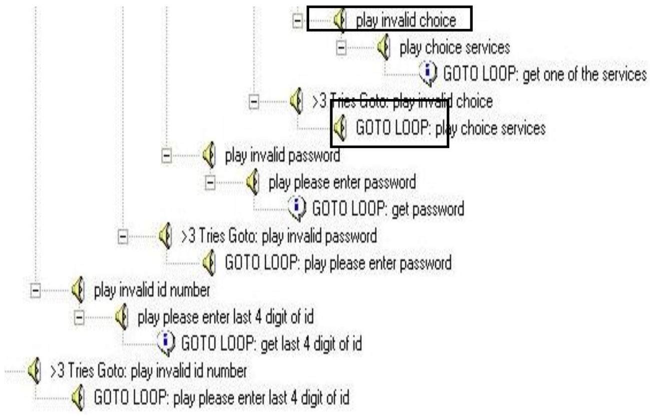

Figure 10. Sub-flow 3

Fig. 11 shows the line states in the IVR. With this table, the system administrator can know the line usages at any time.

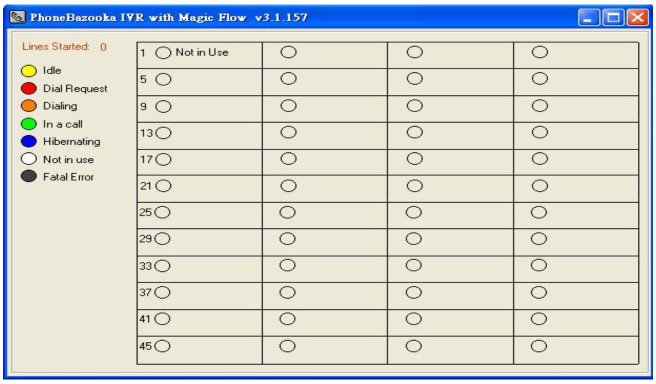

Figure 11. Line states in the IVR system

6. Conclusions

In this paper, freeware is used to build a low-cost VoIP platform providing the web phone, voice communication, video conferencing, instant messaging and postpaid/prepaid services. An IVR is also installed to interact with the prepaid users for querying their balance and recharging. The visualized tree view in IVR with Magic Flow is used to build up voice streams and work flows for the IVR system. Finally, our system can both be easily installed and be a good example of providing enhanced VoIP services in universities and small companies.

7. Acknowledgement

This paper was supported by the National Science Council under contract number NSC 96-2221-E-468-002.

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