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The Intelligent Agent of the Billing Service Between the PSTN and VOIP

[1]Jen-Chih Lin, [1]Steven K.C. Lo, [2]Huan-Chao Keh, [2]Shih-Jung Wu and [2]Yi-Hung Lin
[1]Department of Information Management, Jin-Wen Institute of Technology, Republic of China
[2]Department of CSIE, Tamking University Republic of China

Abstract: The Internet is most popular and is used in the widespread areas such as habitation, school, enterprises, the government and airports. Furthermore, the communication on the Internet is without occupied one line until terminating talking. The VOIP technology of H.323 protocol has transformed into significant technology since 1996. According to the American IDC corporation research report, the VOIP has been used at American and the total time is 1 billion minutes in 1999. The trend of VOIP is to integrate the Voice/Video/Data /Fax /LAN /WAN /Answer Machine with the billing system and becomes very convenient and easy to use at any time/location. Also, the new VOIP technology with intelligent billing system is instead of the traditional technology with the fix and expensive communication fees. The billing system receives all of CDR by PSTN or EDR by IP through the agent mechanism when the events interface(agent) is triggered. The Billing Production is according to factors such as the Rating, Billing Data, customer DB and Tariff and Sharing Schemes. After the Billing Production occurs, the Payment Checking, Bill Printing and Bill Delivery are launched at the same time.

Key words: Agent, VOIP, CDR

INTRODUCTION

According to the American IDC corporation research report, the VOIP has been used at American and the total time is 1 billion minutes in 1999. The trend of VOIP is to integrate the Voice/Video/Data /Fax /LAN /WAN /Answer Machine with the billing system. In the next generation, the billing system will refer the Call Data Record (CDR) or Event Data Record (EDR) to calculate the final bill list and send it to the subscriber by different cycles.

The technology of Voice Over IP (VOIP) has been become more useful than traditional VOIP technology. In the past, human used the Public Switched Telephone Network (PSTN) to dial up from the source site to the destination place, but this method had occupied one line until terminating talking. Nowadays, the Internet is most popular and uses in the widespread areas such as habitation, school, enterprises, the government and airports. Furthermore, the communication on the Internet is without occupied one line until terminating talking. The fix communication fees per each month are paid and there is no time and location limitation for users to search or query information on the Internet.

The Voice and Video data are captured from the hardware device (microphone/camera) and then transferred into the digital type. The data sizes whether the data is compressed are much bigger than the size of text messages. An important characteristic of this type of data is continuous and time sequence. If the quality of network is unsatisfactory, the voices or videos will intermittently show and be unacceptable for the user(s).

ARCHITECTURE

H.323: This study is based on the H.323 protocol to create the internet communication of voices and videos platform[1,2]. The H.323 protocol is most popular voice over IP (VOIP) communication protocol and uses around the world.

As above Fig. 1, the H.323 is a well collection of many protocols to identify the device of terminal communication rules. H.323 is a common standard and is used for the internet communication fields. This protocol is very useful for the video or audio packets to transmit on the internet. It includes the Control part (H.225.0, H.245), Data (T.120), RTP, Audio codec (G.7xx),Video codec (H.26x), A/V Cntl (RTCP),Gatekeeper and Register Admission Status (RAS).

The H.323 protocol is contained 4 aspects (Fig. 2): Terminal, Gateway, Gatekeeper[3] and Multipoint Control Unit (MCU). The H.323 protocol can make one to one or one to multiple communications.

System architecture: The diagram of integrated PSTN and VOIP is as Fig. 3. In traditional communication technology of telcom, it occupies one line on the PSTN when the sender dial up a phone call to receiver. In this

Corresponding Author: Yi-Hung Lin, Department of CSIE, Tamking University, Republic of China

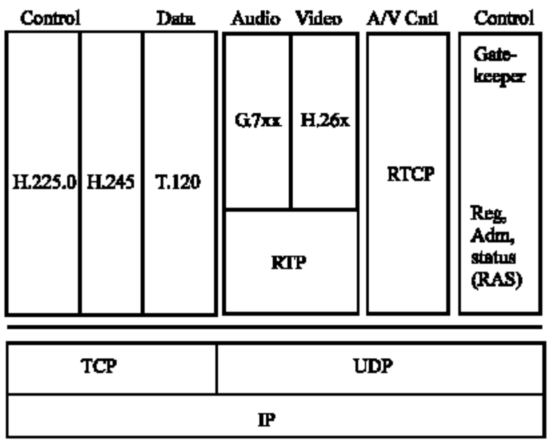

Fig. 1: H.323 Protocol Diagram

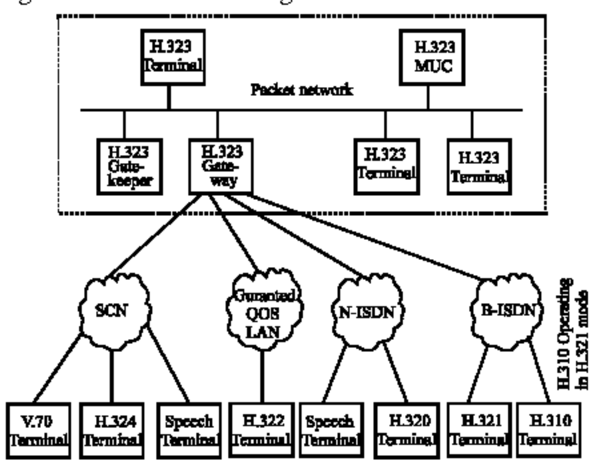

Fig. 2: H.323 architecture

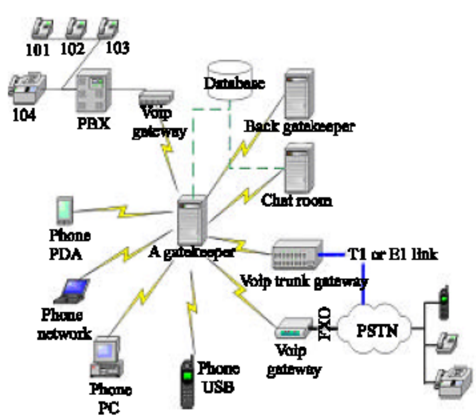

Fig. 3: The diagram of integrated PSTN and VOIP

diagram, it uses the internet technology to slice the data into the smaller packet size and it does not need to occupy one line.

The diagram of implementation is to demonstrate Register, Text, Voice, Video and on-net/off-net Call communication methods which are base on the H.323 protocol (Fig. 4). Audio and video uses the Codec to compress the data which is captured from the microphone or web camera device into the smaller packet sizes to reduce the traffic volumes on the internet.

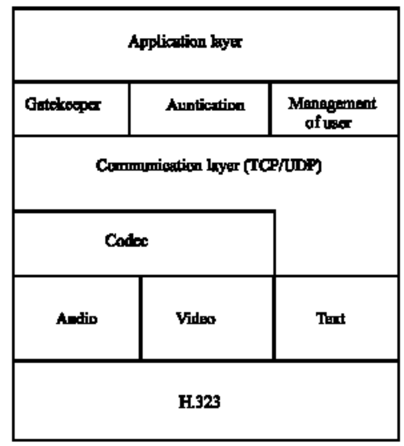

Fig. 4: The diagram of implementation

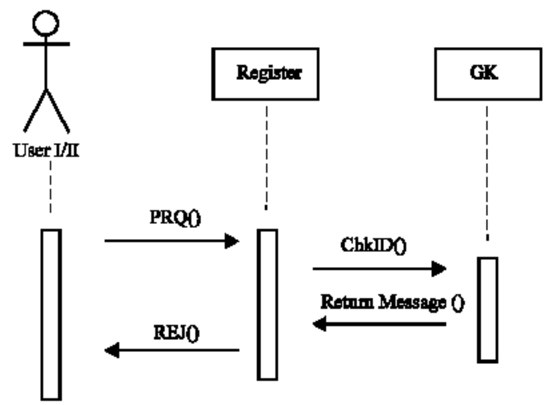

Fig. 5: Register Mode
RRQ: Register Request, REJ: Reject ChkID: Check ID. GK: GateKeeper

Register mode: Using the TCP/IP protocol[4] to provide the register function to Users. Its operation procedure is as below (Fig. 5)

Description:

- When the User I/II wants to use this communication tool at first time, User I/II will send the "RRQ" at first time to GK.
- GK will check the permit of RRQ, then give the ACK or REJ messages to User I/II.

Text mode: Using the traditional communication technology Client/Server structure to Send/Receive

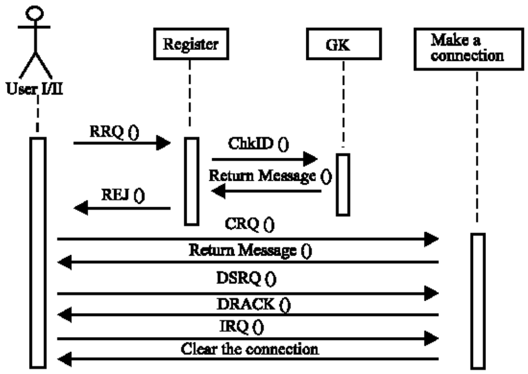

Fig. 6: Text mode
RRQ: Register Request ChkID: Check ID: REJ: Reject CRQ: Connection Request. DSRQ: Data Send Request, DRACK: Data Receive Acknowledge IRQ: Interrupt Request GK: GateKeeper

messages from Users. Its operation procedure is as below (Fig. 6).

Description:

- When the User I/II wants to create a chat mode of Text mode, User I/II will send the “RRQ” at first time to GK.
- GK will check the permit of RRQ,then give the ACK or REJ messages to User I/II.
- Once the User I/II get the ACK message, User I/II will send the CPQ to GK. If the User I/II get the REJ message, the system will be stopped.
- GK will check the connection pool or status, if connection pool is available, GK will send ACK message to User I/II, otherwise send the REJ message.
- If User I/II get the ACK message, User I/II can start using text mode to send whatever text to each of Client and send the DSRQ message to GK.
- If Client would like to give a response to sender, it will send the DRACK message to sender, sender will receive the text message from receiver. This connection has been created to each other.
- If User I/II want to stop the connection, it will send the IRQ message to GK. GK will interrupt the connection and the connection will be cancel.

Audio Mode: Using the H.323 protocol to make the voice communication with User I and User II. Its operation procedure is as below (Fig. 7).

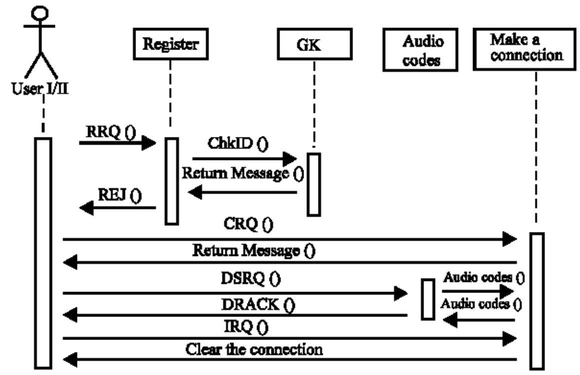

Fig. 7: Audio mode
RRQ : Register Request ChkID: Check ID: REJ: Reject RQ: Connection Request. DSRQ: Data Send Request; DRACK: Data Receive Acknowledge; IRQ: Interrupt Request GK: GateKeeper

Description:

- When the User I/II wants to create a chat mode of Audio mode, User I/II will send the “RRQ” at first time to GK
- GK will check the permit of RRQ and then give the ACK or REJ messages to User I/II.
- Once the User I/II get the ACK message. User I/II will send the CPQ to GK. If the User I/II get the REJ message, the system will be stopped.
- GK will check the connection pool or status, if connection pool is available, GK will send ACK message to User I/II. Otherwise send the REJ message.
- If User I/II get the ACK message, User I/II can start using audio mode to talk each other. The audio stream use the G.721 Codec to compress the stream to reduce the packet size and enhance the performance of communication.
- If User I/II want to stop the connection, it will send the IRQ message to GK. GK will interrupt the connection and the connection will be cancel.

Video mode: Using the TCP/IP protocol[4] to make the video communication with User I and User II. Its operation procedure is as following (Fig. 8).

Description:

- When the User I/II wants to create a chat mode of Video mode, User I/II will send the “RRQ” at first time to GK.

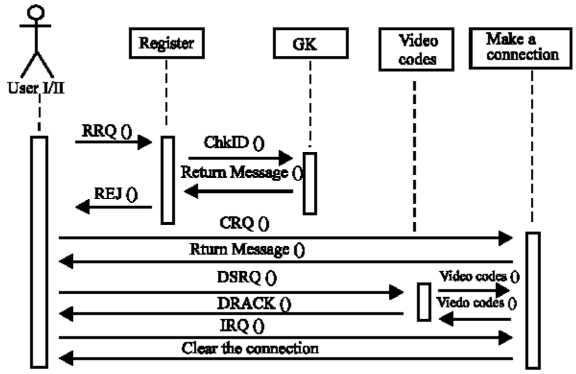

Fig. 8: Video mode
RRQ : Register Request ChkID: Check ID: REJ: Reject CRQ: Connection Request. DSRQ: Data Send Request; DRACK: Data Receive Acknowledge IRQ: Interrupt Request GK: GateKeeper

- GK will check the permit of RRQ and then give the ACK or REJ messages to User I/II.
- Once the User I/II get the ACK message. User I/II will send the CPQ to GK. If the User I/II get the REJ message, the system will be stopped.
- GK will check the connection pool or status, if connection pool is available, GK will send ACK message to User I/II, otherwise send the REJ message.
- If User I/II get the ACK message, User I/II can start using Video mode to see each other.
- If User I/II want to stop the connection, it will send the IRQ message to GK. GK will interrupt the connection and the connection will be cancel.

Call mode: Using the PSTN function to provide the dial out to phone or mobile phone. Its operation procedure is as below (Fig. 9).

Description:
- When the User wants to make a call to destination, User will dial the number of destination and wait for destination pick the phone up and response.
- Once the destination picks the phone up and answers for it, the connection has been already created.
- If destination or user wants to cancel the connection, it just hangs up the phone and system will clear the connection immediately.

BILLING SYSTEM

In traditional billing system is based on the Mediation system record of Call Detail Record (CDR). All

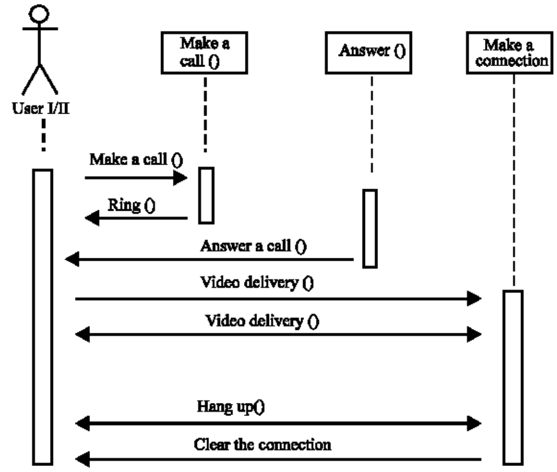

Fig. 9: Call mode

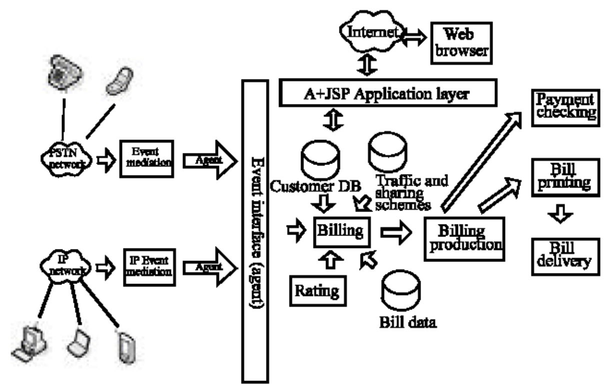

Fig. 10: Billing architecture

most of this traditional type of mediation, it includes the user profile, call starting time, call ending time, call from, call to, etc.

The Billing System Architecture is as Fig. 10. This billing system can obtain events from the PSTN or IP mode events through the agent mechanism. All of event will be processed from the PSTN event mediation agent or the IP event mediation agent to Event interface agent. This agent communicate state procedure as Fig. 11.

Bill production is based on Customer DB, Tariff and Sharing Schemes, Rating and Bill Data to produce the payment checking and Bill Printing. This is because the events will trigger the billing system, it also is called Event Detail Record (EDR), EDR should be included the follow parameters/attributes[5,6]:

- Source IP address and user profile (For example User ID, etc..).
- Destination IP address and user profile(For example User ID, Web site, etc)

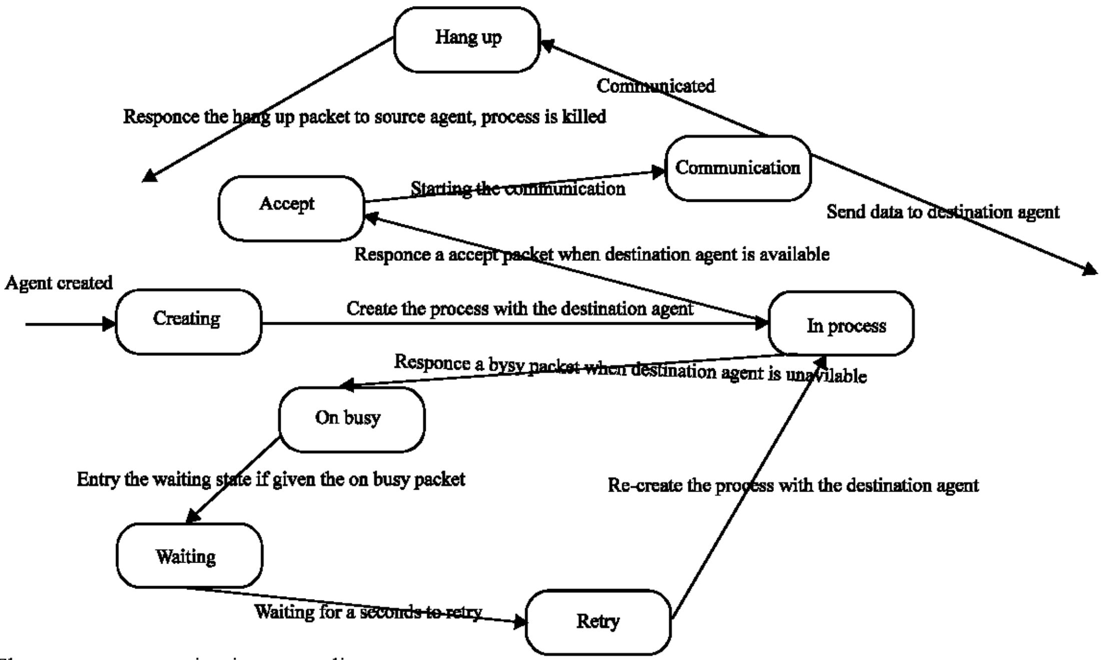

Fig. 11: The agent communication state diagram

- The Type of Net application program(For example EMAIL, etc.)
- Quality of Service (QoS)[7].

On-net: The IP Event Mediation will be triggered when user use this communication platform to make the interactive on the desktop PC or laptop PC or mobile device (PDA, cell phone).The user can get a high quality of service and do not need to pay extra communication fees.

Off-net: The PSTN Event Mediation will be triggered when user use this communication platform make the interactive on the telephone at home or use the cell phone at outside. Although the user dials up the long distance or international distance phone call, the user only needs to pay the local communication fees.

Billing: Billing will be according to the user profile (User ID, telephone, address, etc...), Rating Plan, Tariff and Sharing Schemes and Bill data to calculate the communication fees. The communication bill list will be produced after bill production finishes whole process. The Bill Printing will print the bill list by the bill cycle and send the bill list to user via mail or E-mail. The user can access the web site to know the information of bill and more.

CONCLUSIONS

In the early, internet communication technology has a lot of unknown problems such as the quality of voice, the voice delay and not easy to use. Users or enterprises do not like the voices or videos interactive on the internet. Only the Text communication is used on the internet. But the new technology of VOIP has become mature in the world. It uses the internet communication protocol and PSTN Gateway Server to solve many communication problems, like the delay of voice. The new VOIP technology has Network Communication Feature, more Convenience, Lower cost, Good quality and Multiple Function.

Network communication feature: Internet is a common standard protocol and most use in the world. Users can use any kind of terminal devices to connect the internet at any location to send the messages or dial up on-Net or off-Net to destination and leave some messages into destination voice mail box if receiver does not pick the phone up.

Convenience: This communication platform can be stand alone into any kind of device. The telephone machine can be embedded this communication platform to let user easy use this communication platform. User only needs to connect the Network line (RJ-45) with the telephone machine.

Lower cost: In past, our communication fee is very expensive at long distance or international distance. Enterprises or user can not afford it each month. Right now you can choice this communication platform to make the on-net free charge or off-net local distance charge. Only pay the local distance fee to dial up to the cell phone or international distance phone call or long distance phone call.

Good quality: New internet communication technology is coming; it will let the voice quality more clear. The voice codec G.721/G.723/G.726 is be used in this communication platform. Every voice capture from the microphone will be encode into the compression format and send it on internet. At receive site, the voice encode packets will be decode into the voice format and play to speaker device.

Multiple functions: Internet communication can combine the voice, video and text to transmit at one time. If users use the same protocol as this communication platform, users can communicate with this communication platform. This communication platform also support answer machine, conference, web camera functions.

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