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Design and Development of IP Network Phone Based on SIP Protocol

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Abstract: With the rapid development and popularity of the Internet, the Internet has spread all over the world, which has led to the rapid development of various network applications based on IP networks. The design is based on IP telephony research conducted by the Internet. The design is based on the SIP protocol, TCP/IP and other related technologies to design the client for IP telephony. This design is based on the Linux environment, using MiniSipserver as the SIP server, using C language as the development language, using SIP protocol, RTP protocol, TCP/IP protocol, UDP socket communication, MD5 encryption algorithm and other related technologies to achieve the basic Call function: call, answer incoming calls, reject incoming calls, call reminders, etc. And in the development process, we optimize the existing VoIP technology and use our own packaged OSIP function library to make the call more efficient and the response time faster. The development process was developed in accordance with the SIP protocol call initiation process. In order: registration, heartbeat keep-alive, call request, answer, hang up.
Keywords: Internet, IP phone, SIP protocol, RTP protocol, TCP/IP.

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1. INTRODUCTION

Since the birth of the Internet in the 20th century, the Internet has become the backbone of the economic development of all countries in the world, and has greatly promoted the development of the world economy. The development of the network has gone through the previous wired network to the current wireless network, from 2G to 3G to 4G. At this moment, we will step into the 5G network world, and the wireless network brings us more and more convenience. At present, Internet access to the Internet has spread all over the world[14]. Therefore, the high penetration rate of the Internet has made it possible to cover the world with telephone communication using the Internet as a communication medium.

In 2014, the Ministry of Industry and Information Technology proposed “three networks integration”. The so-called triple play refers to the integration of telecommunication networks, telephone networks and broadcast television networks into one network cable. The telephone network here refers to a fixed landline, which utilizes the principle of time division multiplexing TDM circuit switching of the PSTN network, the cost is relatively high, and the call quality is general. The use of IP networks for communication industry is in response to the call for national network convergence, saving resources and improving resource utilization[16]. Thereby achieving a simple cost and a reduced cost. The design is intended to use existing network resources to achieve calls, reduce call costs, improve resource utilization, improve call efficiency, and facilitate public life.

2. IP VOICE COMMUNICTION RELATED TECHNOLOGY

This design supports two transport protocols, TCP transport control protocol and UDP transport control protocol[17]. Both of these network protocols can implement this design function. TCP has three handshakes and four mobile phones. It is relatively reliable compared to UDP, but its development process is more complicated. UDP is a simple and unreliable message transmission for things, which is relatively simple to develop. Therefore, after comparing the characteristics of the two network protocols, the UDP transmission control protocol was chosen to develop

2.1 UDP Transmission Control Protocol

The communicationtransmission protocol used in this design is a connectionless UDP communication communication protocol, which uses sockets for data packet transmission.
The socket communication flow chart is shown in Fig.1.

2.2 SIP protocol

The SIP protocol is developed on the basis of the Internet, and it is at the application layer in the computer network OSI model. It mainly uses TCP/IP network protocol for transmission, and also uses RTP real-time transmission protocol. The SIP protocol is generated by exchanging multimedia data over an Internet network[1]. The SIP protocol is mainly used to manage sessions. The SIP protocol can achieve point-to-point, point-to-multiple, many-to-many sessions, and it is convenient to add or delete in the media session.

The full name of the SIP protocol is the Session Initiation Protocol, the initial protocol of the session. Its main function is the session. The session can be a two-session session or a conference session between multiple parties. This is also the advantage of the SIP protocol. The media for the data exchange of these participants can also be a variety of media, such as: text, multimedia, audio, video, pictures, etc., can be used as a medium for exchange between participants. The SIP protocol is developed on the basis of the Internet, and it is at the application layer in the computer network OSI model[2]. It mainly uses TCP/IP network protocol for transmission, and also uses RTP real-time transmission protocol.

2.3 RTP real-time transport protocol

In the design of the session, the media stream communication between the two clients uses the real-time transport protocol RTP. This protocol is built on top of UDP communication and transmitted over UDP.

This is the communication protocol structure diagram:

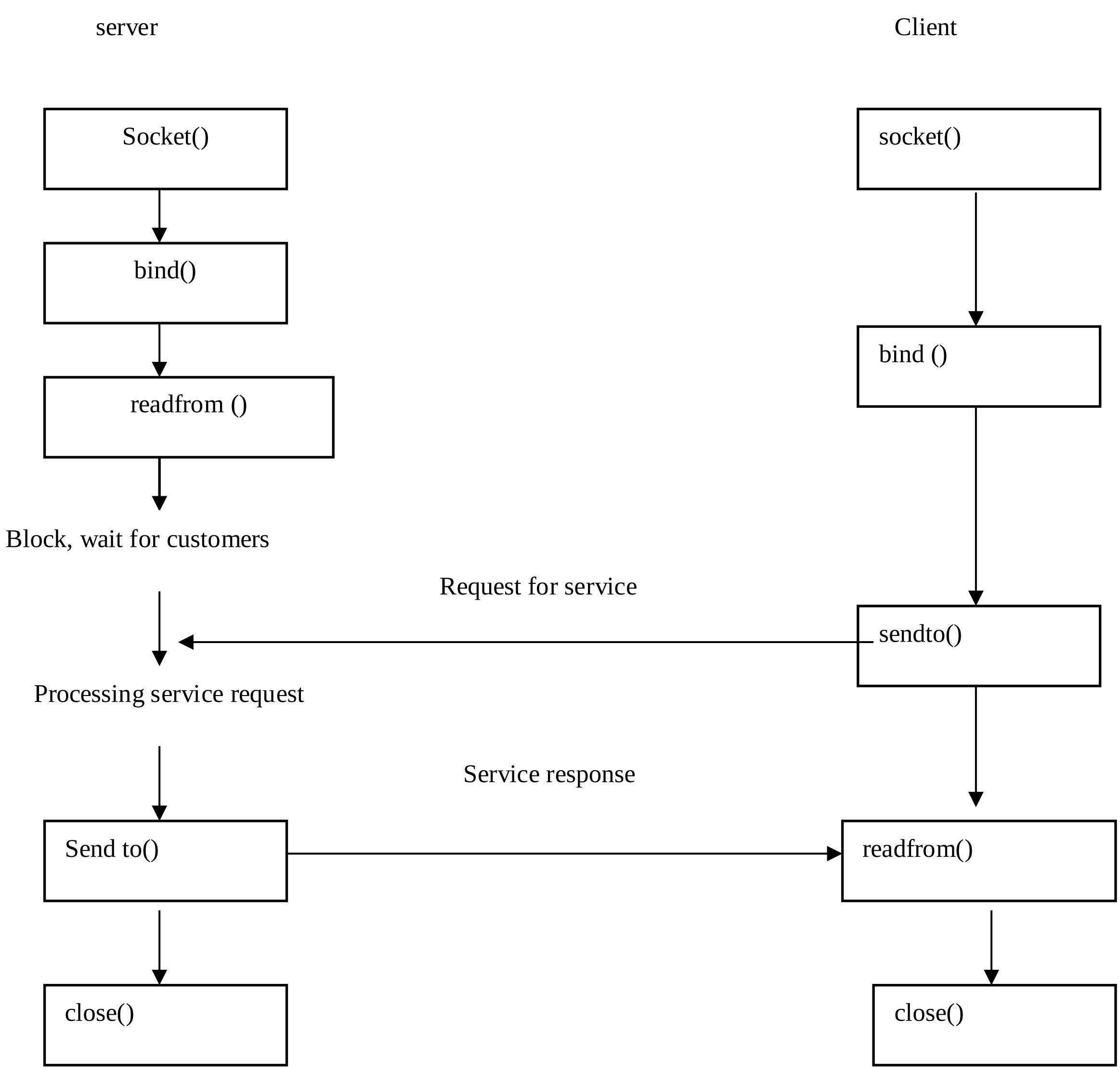

Fig. 1 SOCK_DGRAM socket communication flow chart

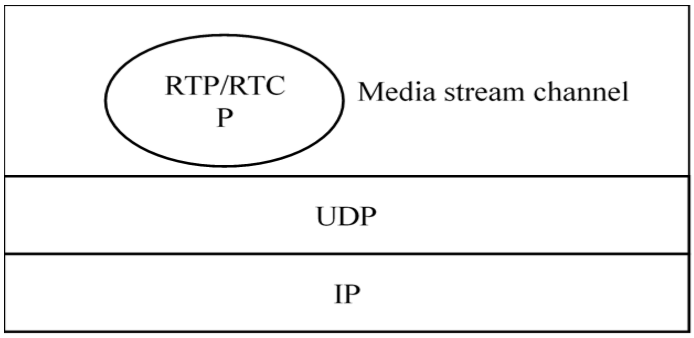

Fig. 2 Communication protocol structure diagram

3. DESIGN AND IMPLEMENTATION OF IP TELEPHONY

3.1 Design summary

this design is based on the SIP protocol to achieve IP phone design. The SIP protocol is mainly divided into two modules: SIP phone and server.

This is the data exchange diagram between SIP client and SIP server:

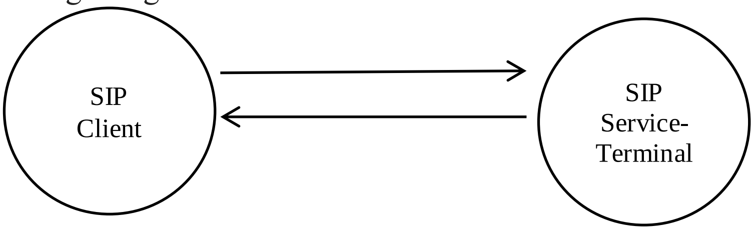

Fig. 3 SIP client and SIP server data exchange

3.1.1 SIP client

The SIP client, the SIP user agent UA, mainly implements the following functions:

(1) The SIP client sends registration information to the server for registration.

(2) Initiate a conversation

(3) Accept the conversation

(4) Terminate the session

(5) Heartbeat keeps alive

3.2 SIP message analysis

The SIP protocol uses the UTF-8 character set, which refers to the message body format in the RTSP protocol. All its functions are implemented by command messages. After the server receives the request from the client, the server will respond[2]. Used to indicate the state of the process. The response message body contains a status code indicating the result of the SIP server's execution of the message body command sent by the client.

The SIP specification defines a total of six normative method names:

REGISTER: The client requests registration from the server.

INVITE: When a call is made, the request sent by the originator can also be used for the invitation of the session during the session, and the session update can also be performed during the session.

ACK: After the INVITE acknowledges receipt, the media stream is requested by ACK to complete the final request response.

BYE: Terminates the session for which a connection has been established.

CANCEL: Cancel the body of the message that is not responding.

OPTIONS: Query the related functions owned by the server.

3.3 IP Phone Overall Framework

This section mainly introduces the IP network phone workflow.

The workflow is shown in the figure:

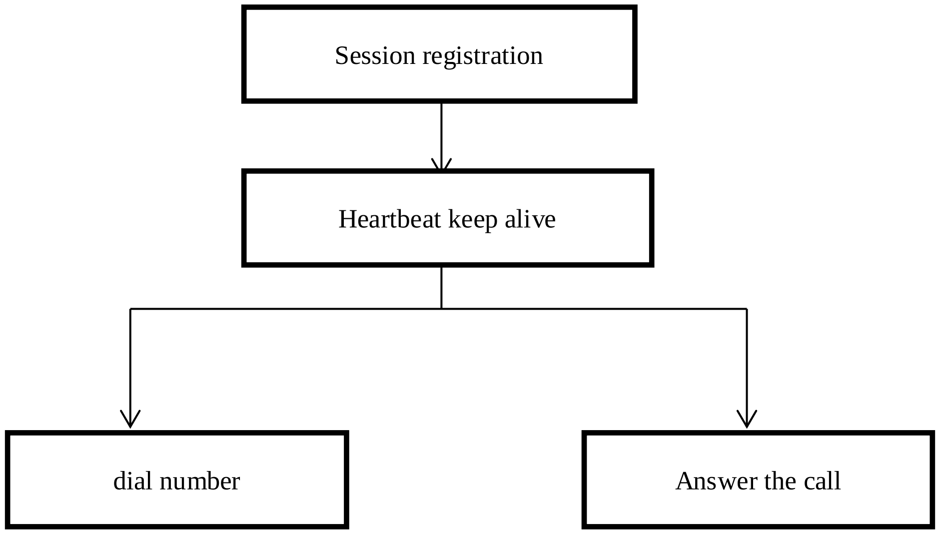

Fig.4 IP phone framework flow chart

3.3.1 Session registration

To use the SIP client in a SIP server, you must first register it.

Registration process description:

(1) Register signaling is a registration request sent by the SIP proxy to the SIP server.

(2) The SIP server wants the SIP proxy to send a response message 401 and carries the message body with the authentication code.

(3) The SIP proxy again sends a Register Request message with an authentication code and the included authentication information to the SIP server. The authentication code is the primary identifier of a device[12]. It is generated by the device IP, device name, device port, and device Call-ID.

(4) The SIP server is authenticated again, and the identity is legally sent to the proxy server with 200 OK, indicating that the registration is successful, and the identity is invalid, and the response is rejected.

3.3.2 Heartbeat keep alive

When the SIP proxy registers on the SIP server, there is a 3-minute time limit. If there is no signaling between the SIP proxy and the SIP server within three minutes, it will be disconnected, must be re-registered, or periodically send heartbeat keep-alive signaling. .

The method of adopting heartbeat keep-alive in this design is to send registration information every 2 minutes through sub-threads.

The way of working is as shown:

3.3.3 Telephone call

Client network configuration: In the LAN, the IP address in the IP segment of the IPV4 network 192.168.20.0-255 is used. Mainly through the data message socket for data transmission, to achieve interaction.

Telephone call flow: In the process of using, you do not need to use the IP address to make a call when making a call. Instead, when the SIP protocol is registered, there is a corresponding number. When we make a call, we can only dial the number and dial the corresponding number.

Server-side function introduction: The server uses a miniSIPserver server.Its role is mainly the identification and function completion of SIP communication message instructions. Such as registration, heartbeat keep-alive, call and other functions related to telephone registration, dialing, etc., it is mainly communication between SIP messages.Media server mainly realizes the data exchange between media streams[5]. For example, when the phone is dialed, the exchange of media streams between the calling end and the called end is realized by the media server.

Telephone call flow: The call is based on the SIP protocol point-to-point communication. The call and the call between them are realized through the third-party server to complete the point-to-point communication.

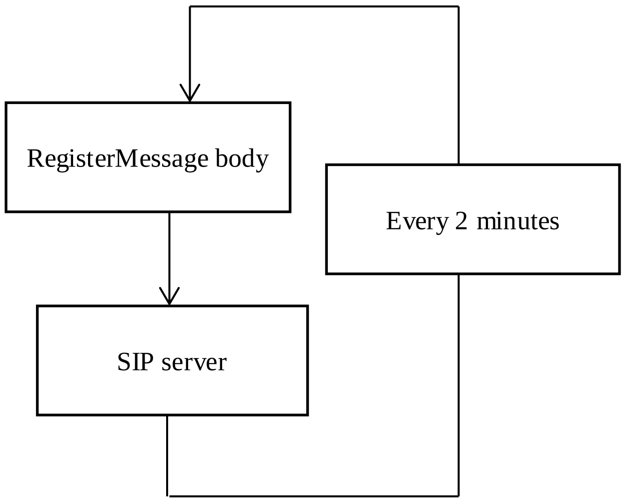

Fig.5 SIP heartbeat keep-alive mechanism schematic

The relationship between the calling client, the called client the server is as shown in the figure:

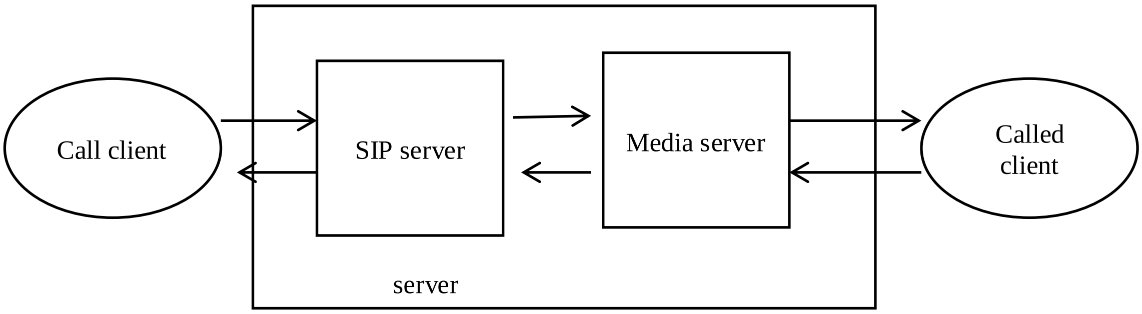

Fig.6 Call flow

3.3.4 Hang up

There are two situations in which the hang-up call is designed. One is that the calling client hangs up, and the other is that the called client hangs up. These two cases are the same in the basic process principle, and there are certain differences in the writing of the code. During the call, one party hangs up and informs the other party to hang up, otherwise the other party has been receiving data streams to send data streams.

3.3.5 Calling the entire process

A full phone call involves dialing a phone, talking a conversation, and hanging up a call. The Fig.7 is the complete flow chart for making a call.

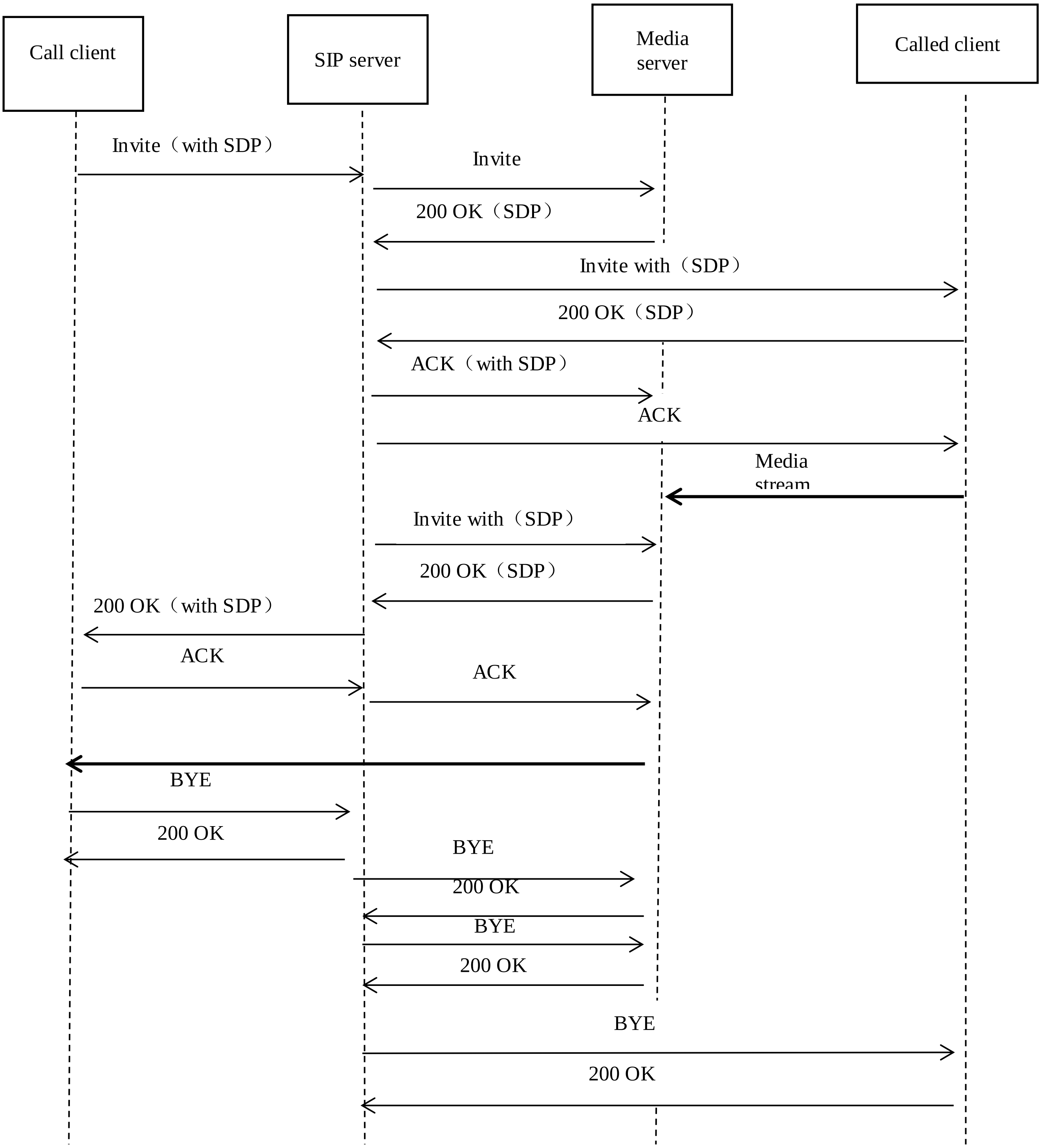

Fig.7 complete flow chart for making a call

3.3.6 IP phone other situations

In real life, the phone is no stranger to each of us. Calling is to achieve communication between two points. However, it has a variety of special circumstances, and special circumstances must be dealt with during design and development.

When the call is made, the other party refuses to answer the call: When the call is made, the other party cannot hang up the phone for other reasons, and needs to hang up the call[7]. If the call is not answered, the call is hung up. Different. Directly rejecting the answering call, the called client will send a message body BYE to the server. After receiving the BYE, the server will disconnect the Invite call request and send a 302 temporary migration response to the calling end to inform the calling party that the other party is busy. Refused to answer the call.

The caller is not answered: the caller initiates a call request to the SIP server, and the SIP server redirects the call to the called party, and forwards the INVITE call request to the called party[10]. The called party does not answer the call for a long time. Here, the server has timeliness. The time is 32 seconds. Considering that 32 seconds is short, the design takes 64 seconds of effective time. After the valid time expires, the server automatically disconnects the called party's call request, and the server sends a 408 response message to the caller to inform the caller that no call is answered.

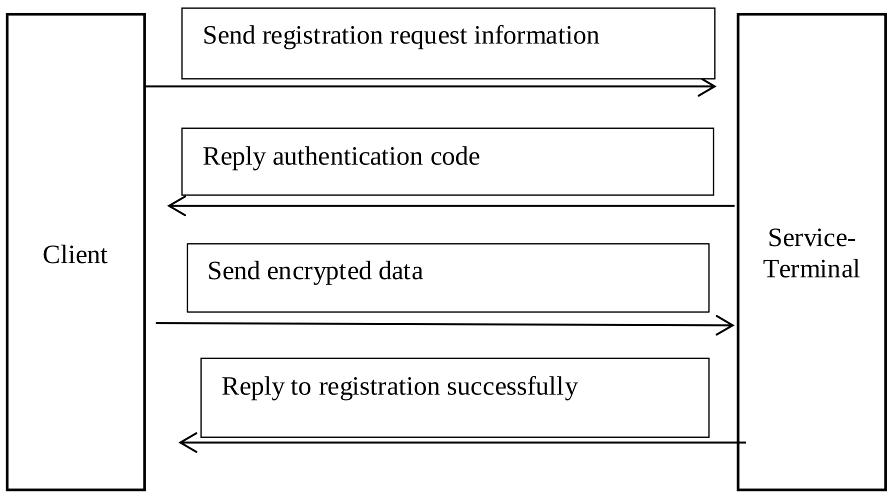

Fig.8 Application of MD5 algorithm in SIP protocol

4. MD5 ENCRYPTION ALGORITHM

The SIP protocol is an Internet-based communication protocol with high requirements for security. The security of the SIP protocol is mainly reflected in the user registration process. The SIP protocol uses IP addresses to communicate[4]. The IP address of each device is unique, and each IP address can only register one user. Therefore, each user can be found by IP address. When a user authenticates to a server, it needs to be encrypted by an encryption algorithm, and the service will be decrypted accordingly, thereby improving security. The encryption in the SIP protocol mainly encrypts information such as IP geology, user identification, and user number into a series of digital passwords. Therefore, this design uses the MD5 encryption algorithm[8]. Because the security of the SIP protocol is high, it needs to be encrypted when registering the identity. Therefore, the MD5 encryption algorithm is used to encrypt the data. The encrypted data mainly includes the IP address, the phone number, the authentication code of the server reply, and the REGISTER message body. After the encryption is performed by the MD5 encryption algorithm, the encryption is generated and sent to the server.

Application of MD5 algorithm in SIP protocol:

5. SYSTEM DEBUGGING

The system test mainly uses the Wireshark network analyzer to perform packet capture analysis, track data flow, and detect the data stream format and response. The eyeBeam phone client performs call test and tests the call function implementation.

5.1 Wireshark Network Analyzer

The Wireshark Network Analyzer is a must-have test software for the development of related projects involving the network. It can capture packets from the IP network and discover the details of the analyzed packets to find out the problem. In this design, Wireshark's main job is to capture packets and analyze the sending and response of SIP messages to better test the development situation, and timely detect errors and correct them in time. The Wireshark network analyzer can also view and track the data stream in the captured packet, and can clearly see the data stream transmission and reception, and can find the problem in time.

5.2 eyeBeam

eyeBeam is an existing network IP phone client, an instant messaging software. eyeBeam is mainly used in the windows environment, it can be registered in the MiniSIPserver server, and interaction, so this design uses eyeBeam as the test software. In this design research process, eyeBeam mainly acts as a client. In the functional test process, eyeBeam and the client developed by the design conducted call test to test the problems and function realization of the design client development process. And completed the test very well.

6. EXPERIMENTAL RESULTS AND CONCLUSIONS

The design is based on the research and development of the SIP protocol network telephone. It mainly uses the Internet network as the transmission medium to realize the IP call function, and mainly realizes the functions of telephone call, telephone answering, rejecting the call, and telephone reminding.

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