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Design and Implementation of P2P instant messaging tools in LAN

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Abstract. A tool for instant communication of voice was designed and implemented in this study, in order to achieve instant messaging and the sharing of resources in LAN, The network communication of the software used TCP / IP protocol; and its procedural framework used the mode combing C/S and P2P, that is to say, P2P architecture was to build the communications among clients and C/S structure was to connect the clients and the server. Simulation results showed that the tool can be easily interconnected to LAN, having good performance of voice communication.

Introduction

With the development of computer and Internet technology, local area network (LAN) in many domains such as companies, government agencies and schools has been widely applied. In order to take full advantage of the Internet, making the Intranet more effective utilization of resources and its sharing, and making communications between the staffs in the origination more convenient, quicker, it is necessary to develop a LAN environment instant messaging tool .[1]

IM (Instant Messaging, IM) [2] is a kind of instant communication, The communication object can be a dialogue between two people or may also be more than two people, Communication platform is a personal computer or software to the client. At present, more popular instant messaging software, such as QQ, ICQ[3], MSN, and Skype [4], etc., these software tools have good function to chat, provide users with many choices [5]. However, at present most of the communication software is based on main C/S architecture, not fully realize the peer to peer communication, it still mainly rely on the server. [6]. Communication system based on C/S architecture of the server is used to transmit messages, also controls the communication between the clients, so it is required that every client to connect to the server. Moreover, once there is a problem in the server, the system will be paralyzed. This way of communication, therefore, cannot meet satisfaction of the users' needs, there are certain application limitations in it.

P2P (Peer to Peer) is a kind of distributed application program framework, the end-to-end communication mode; the part of every node has some functions of the server, provides resources and also takes advantage of resources [7]. P2P structure overcomes the bottleneck of the server restrictions, with superior data transmission property [8]. In recent years, the development of P2P technology in instant communication system has been widely used; many systems have turned to a P2P architecture model. P2P model in the aspect of management mechanism, however, is insufficient and its security is relative poor, the C/S model has the strength of the centralized management. [9].

Compared with the traditional text communication mode, voice communication is more flexible, convenient, quick, can adapt to almost all types of user demand. Moreover, the bandwidth of the interior LAN in an organization infrastructure provides a convenient platform for voice communication. Based on this background, in this paper we present the design and implementation of instant voice communication tool in the local area network (LAN) environment. The software of network communication apply TCP/IP protocol, procedural framework based on the combination of C/S and P2P model [10], namely, to apply the P2P architecture to perform the communication between the software client, to apply C/S architecture to realize communication between LAN users and their servers. Therefore, this software is able to realize public resources sharing, and be able to share private resources by individuals; both can regulate resource management, and to realize peer to peer communication between users; not only to ensure information safety, but also can solve the problem of network congestion.

The overall design

We design the instant voice communication software by the TCP/IP protocol to design the network communication in order to realize the audio data transmission; to design procedural framework by adopting the combination of C/S and P2P model, which is applying the P2P architecture to realize the communication between the client and local area network (LAN), by using C/S architecture to realized the connection between the user and the server. The software system architecture is shown in figure 1.

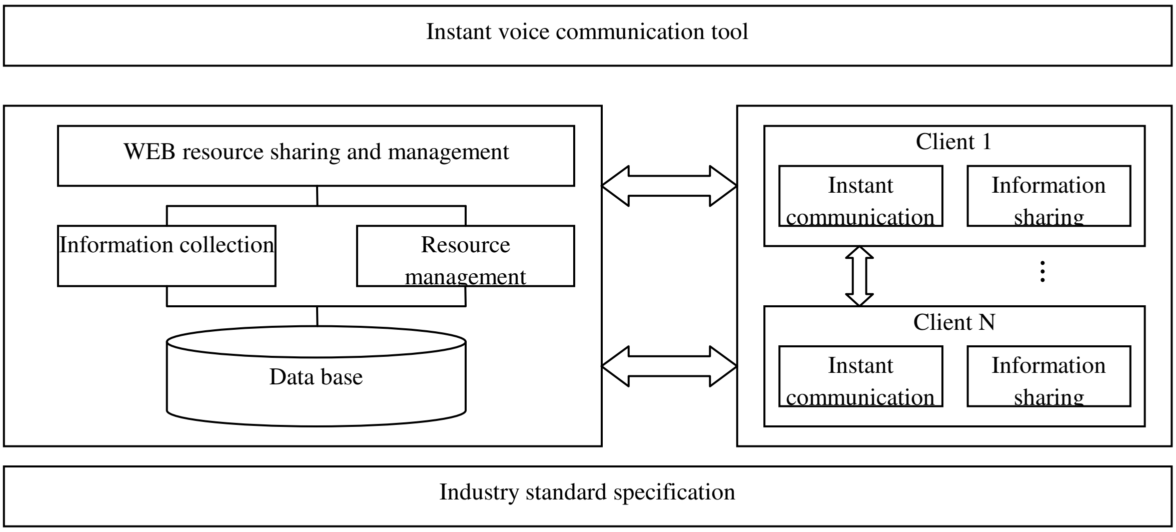


Fig.1 System architecture of the software

Peer to peer voice communication tool includes two main parts: one end of the instant acquisition, processing and display voice; on the other side receiving voice. Through the network protocol to reliable transmission of voice, and connect the two ends of the communication, can realize voice communication. WIN32 API provides a set of user multimedia services function; we adopt the API for the designing of related waveform audio. The overall work process is shown in figure 2.

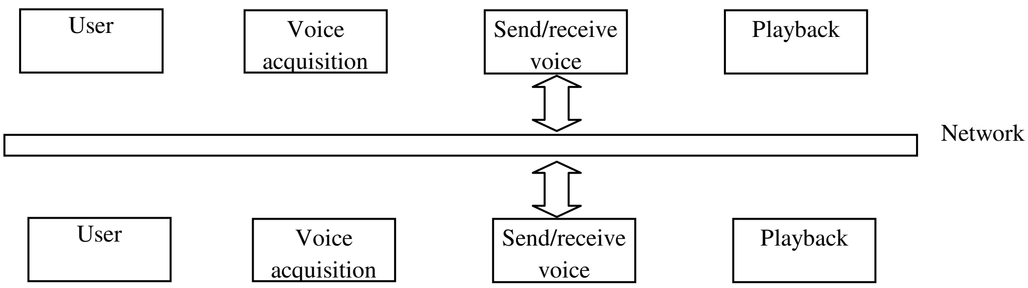


Fig.2 Overall workflow of the software

For voice acquisition, processing and broadcasting, we apply Windows MDK lower layers audio services; the callback mechanism can raise the continuity of sound. When the application routine continuously to provide audio data to the device drivers, device driver control audio equipment in the background to complete the recording and playback of specific operation. Through the callback mechanism, can detect the over time of data block be used, and transmit the next block of data, to ensure the continuous of the voice, so instant acquisition and playback can operate in an individual computer.

In a peer to peer network transmission, we should choose a connection-oriented TCP protocol. TCP transport protocol process automatically loss data and delivery disorder problem, so only need before collecting audio playback, on the one hand, to transmit voice to the network, receive the network from the voice on the other hand, thus realize the peer to peer voice communications.

Function Designing

2.1 Instant acquisition, processing and playback of voice
Phonetic acquisition and play through the Windows lower layers audio services waveform data block WAVEHDR to complete the operation, in the process of allocation buffer (memory) to correspond distribution WAVEHDR structure of data blocks, and then assign to the indicator of the buffer to the member variables lpData of corresponding data block structure, such as a buffer to be filled, is filled audio data block. Through the message mechanism can be processed in the message function and play, then can send buffer to audio equipment input driver, to continue their acquisition and playback. The method mainly includes two parts initialization and news operation. Each part of the design process is as follows:

2.1.1. **Initialization**

The operation is mainly on the system input and output devices, audio format, audio data block, and audio buffers for initialization settings. They are used for phonetic acquisition and processing of the voice. The flow chart is shown in figure 3.

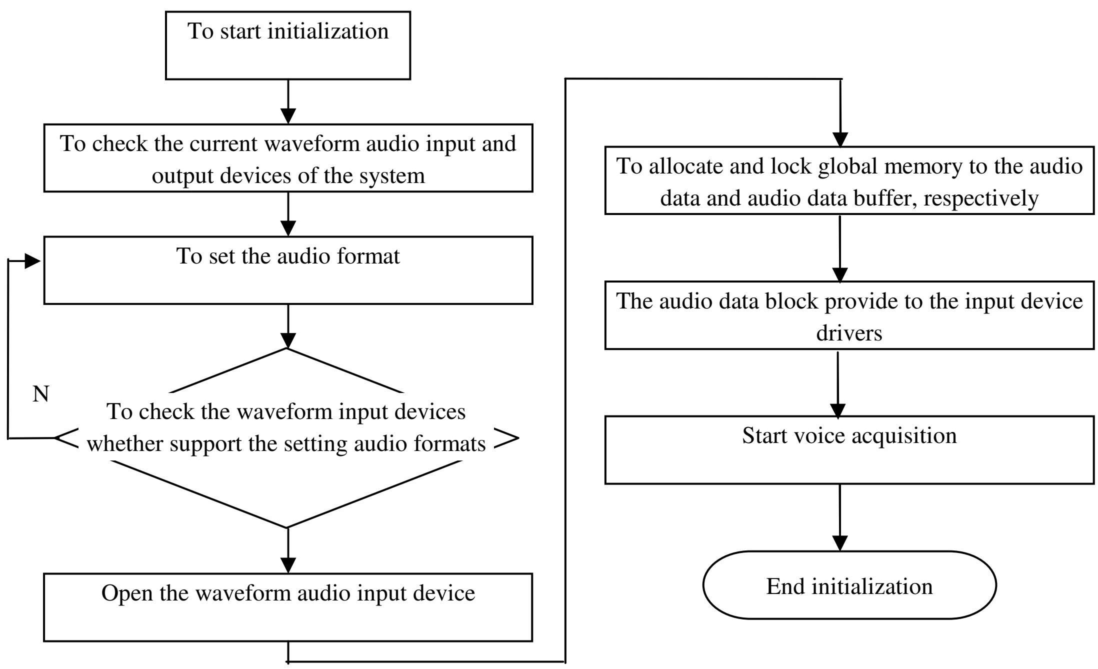


Fig.3 Workflow of the initialization

We can set the voice frequency in accordance with the 11025 Hz, 16 Bit, and mono, 22 k /S format, also can be carried out in accordance with the other settings format.

2.1.2. The message operation

The main function of this part is to controlled through the message completion of the instant voice acquisition, processing and playback.

(1) after the beginning of voice acquisition, whenever there is a sample data to fill after the data block, device driver will send the user window MM_WIM_DATA message, the appropriate message callback function On MM_WIM_DATA (...), the block of data sampling data processing, and then send it to the output device for playback.

(2) Whenever an audio data block after the broadcast, device driver and emit MM_WOM_DONE message, the message of the corresponding callback function MM_WOM_DONE (...), record audio data and sent to the input device, which is ready to receive the sample data that follow.

By the above two steps, Under message controlling, the original audio data blocks for input device recycle between the input and output devices, without the need of artificial control that can achieve instant acquisition, processing and playback of voice.

To shut down the audio input devices at the end of the communication, audio device driver will send MM_WIM_CLOSE message, we can clear the assigned to input and output devices of audio data block in the appropriate message function OnMmWimClose (..)

2.2 Peer to peer voice transmission based on TCP protocol

Using a connection-oriented TCP protocol to send and receive voice, and apply the Windows Socket network to realize programming. In order to ensure acquisition data block after fill to be sent, and after data be received completely to be played, function interface of sending and receiving data will be set in the function of OnMmWimData (...).

Voice communications between the client and software is done by P2P architecture, between users by peer to peer communication. So between the client and the server adds a monitor Socket, once the call connection is established, between two peers will create a data flow, even if both sides don't talk, each peer is instantly sending and receiving data, if one party has a voice, will be as the data stream is sent to the other party. Peer-to-peer communications between the clients, therefore, the key problem is how to read the voice and data flow.

Run to TCP streaming service does not guarantee boundary, performing the Send function does not ensure that the sender one-time send complete data, neither will the receiver receive complete data one time. And in order to achieve the broadcast, we hope to perform a time function can send out the voice data in the buffer zone, and perform one time receiving function can receive the voice sent by each other, so, when using the TCP protocol to send data, we add a logo on each packet header. This logo header contains long (4 bytes) speech data value and a symbol of string, in the program can be set up for the two sign corresponding offset, also set the offset for voice and data at the meantime, to receive by overloading OnReceive (...).The process of receiving data as shown in figure 4.

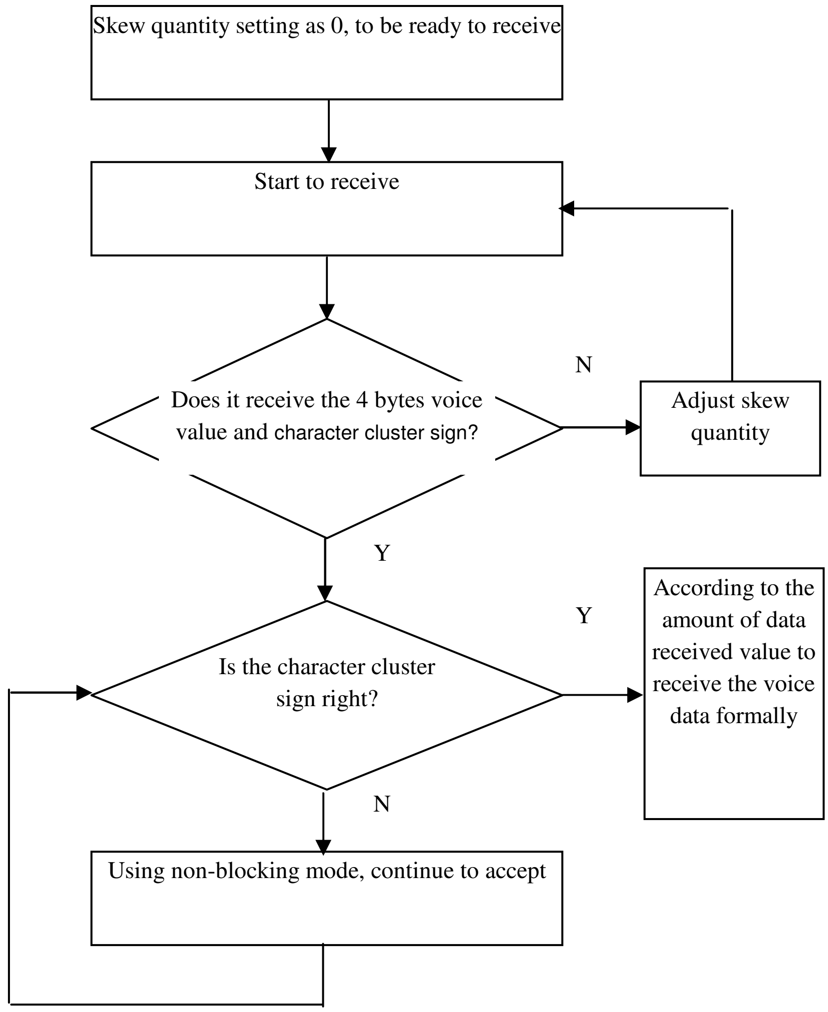


Fig.4 The receiving process of the speech data

In this way, after reloading OnReceive (...)receiving function can receive each other sending data at the time. In the same way, the reloading OnSend function of CAsyncSocket, can realize to finish data in a buffer in one sending. So as long as setting the interface of sending and receiving function in OnMmWimData (...), we can have a instant rate of receiving and sending data stream, so, realizing the network voice transmission and playback.

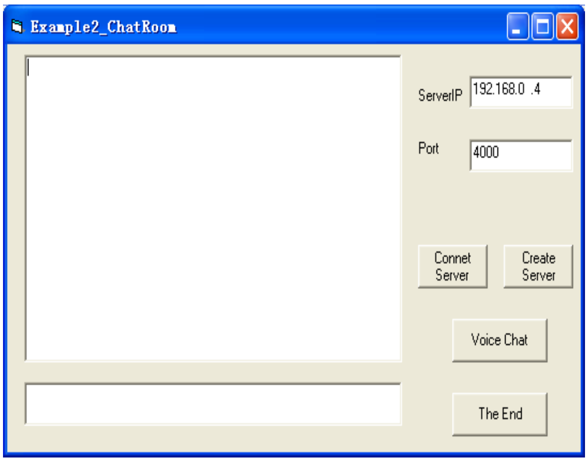
The simulation results and test

To test and verify the validity of this software, a large amount of data communication experiment was carried out in the local area network (LAN). Experimental platform can be PC computer of the Intel Celeron (R) (R) E3400 2.60 GHz 2.59 GHz, 988 MB of memory, programming environment as Microsoft Visual Studio 2008, c + + programming language.

3.1 The result of the experiment

3.1.1 The software operation interface

Program automatic access to the machine IP address, this software show the IP address of the demo for 192.168.0.4, set the port number to 4000. Software running interface as shown in figure 5.



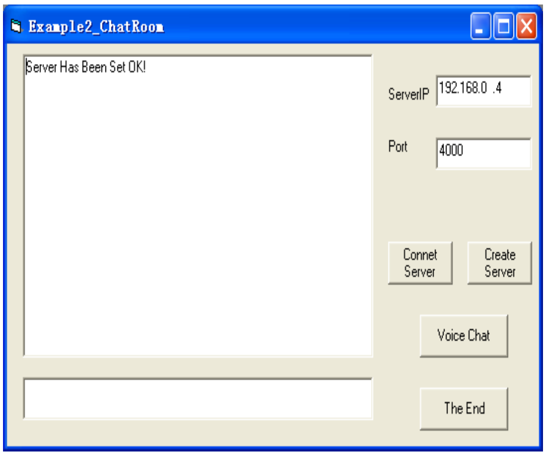


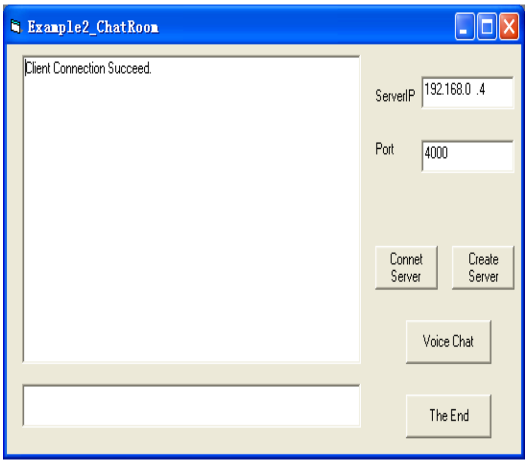
Fig.5 Running interface of the software Fig.6 Interface of creating server

3.1.2 To create a server

IP address should be the machine address, click on to create a server to create the network to listen to, and in the text box to display the connection is successful, as shown in figure 6.

3.1.3 The client connection process

Attention, we should guarantee the LAN is connected, set the port on the client process, the IP address for Server IP address. Click connect to the server, and will show the client connection success message, as shown in figure 7.



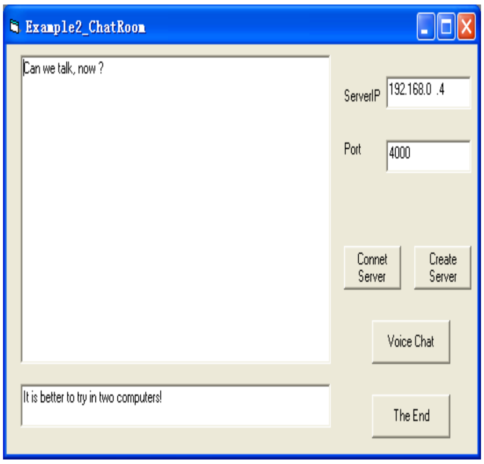


Fig.7 Interface of client connection Fig.8 Interface of text communication

3.1.4 Communication function

After the successful connection, it can undertake voice communication. To this software communication function, we use the text mode for testing. As shown in figure 8, below the interface for the text edit box, above the chat for text display area.

Communication both sides click the chat button, turn on the voice input device; you can proceed to voice chat.

3.2 Software function testing

In order to further explains the software performance in the local area network (LAN) environment, the response time of the landing and sending a message from our customer to test the software performance.

As shown in table 1, for five clients access software test results.

Table 1 Test results on software login

Client	Server	Responding time	Login result
lyd	192.168.0.163	280ms	success
Zhy	192.168.0.163	270ms	success
ykm	192.168.0.163	290ms	success
Wb	192.168.0.163	260ms	success
lyy	192.168.0.163	280ms	success

The data in table 1 shows that the client login time vary slightly, and can successfully login, landing module is stable.

As shown in table 2, 5 pair clients perform voice communications, the test results of sending messages response time.

Table 2 Test results of sending messages

Client1	Client 2	Sending time	Communication quality
192.168.0.4	192.168.0.1	1.8s	good
192.168.0.4	192.168.0.2	1.7s	good
192.168.0.4	192.168.0.3	1.9s	good
192.168.0.4	192.168.0.5	1.6s	good
192.168.0.4	192.168.0.6	1.8s	good

The transmission time in table 2 is from voice and data acquisition; send to the receiving and playing time. The data in table 2 shows that the response time of the voice communications between the clients can satisfy the general requirements of users, and the communication quality is high voice communication function of this software has reached the design requirements.

Conclusion

Based on the TCP/IP protocol for network communication design, by adopting the combination of C/S and P2P model for application architecture design, realized the instant voice communication software in a local area network. The simulation experiment results and the performance test show that the software can easily achieve LAN interconnection; it also has a good voice communication function. In order to make the software to learn more powerful function, on the basis of the voice communication, try to increase the function of video communication, making it a fully functional video communication tool.

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