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Telemedicine Transmission System Design Research Based on Session Initiation Protocol

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ABSTRACT

With the rapid development of the next generation Internet, the use of audio and video transmission applications are increasing, one of which is the typical applications of telemedicine systems. In this study, the telemedicine system overall structure and client software implementation process are presented based on Session Initiation Protocol (SIP) and the application of the system in Central Europe high-speed Internet IPV6 and its actual operation are described. Telemedicine models are researched and the system is designed. The advantages of SIP are finally analyzed and its prospects are described in the applications of telemedicine system.

Key words: Telemedicine, session initiation protocol, multi-threaded, audio and video communication, real-time transport protocol, signaling system

INTRODUCTION

Telemedicine may be as simple as two health professionals discussing a case over the telephone, or as complex as using satellite technology and video-conferencing equipment to conduct a real-time consultation between medical specialists in two different countries. Telemedicine is a new medical model which is combined with the modern medical technology, computer network technology and modern communication technology. The so-called telemedicine generally refers to two-way information transmission of network technology between patient and doctor far apart, the information search and diagnosis are completed by patient and the medical program is implemented (Gomez *et al*., 1996; Mitchell *et al*., 2004).

Telemedicine biggest advantage is the efficient sharing of resources, timeliness of emergency treatment, economy of saving hospitals and patients spending. If telemedicine can be provided through broadband network or telephone lines, the time protection is not only provided for the patient's treatment, patient expenses are reduced but also the awareness of doctors and hospitals are raised.

At present, although there are some telemedicine system at home and abroad (Li, 2006) but the relevant software (including server transfers, image viewing, etc.) is not only expensive but the software is not compatible between different manufacturers. For these systems, our hardware and software is relatively backward at the actual situation of the majority regions, there are apparently limitations. In addition, more worthy of note is that, due to various reasons, the data exchange is the more difficult between the many existing medical applications such as various Hospital Information System (HIS), Picture Archiving and Communication System (PACS), this situation affects the sharing of medical information, the overall progress of the medical information construction is seriously hindered (Xu *et al*., 2004; Wang *et al*., 2010).

Because the SIP protocol has featured with simple, good compatibility and strong scalability, so the development of telemedicine system has great advantages by using SIP protocol which can

be compatible with other software systems which is developed with SIP and the system can be ensured, the upgrade and expansion latter is convenience.

Session Initiation Protocol (SIP) is one of the core protocol in Next Generation Network (NGN). It was originally developed by the IETF Multiparty Multimedia Session Control (MMUSIC) working group and a standard was proposed in 1996, the signaling control is solved for IP network communications, SIP is a signaling work at the application layer, it is used to create, modify and terminate multimedia sessions process (Handley *et al*., 1999). Here the meaning of the session is that data is exchanged between the participants.

SIP protocol uses a client/server model work, SIP network includes two types of components: User Agent (UA) and Network Server (NS). Wherein the user agent is divided into a User Agent Client (UAC) which is responsible for initiating SIP call request and the User Agent Server (UAS) which is responsible for the call request, respond and the network server comprises a registration server, the proxy server, a location server and re-directional server. This series of servers primarily provide registration, authentication, authorization and routing services for users. SIP support the creation and end multimedia communications in five areas which are customer orientation, functionality, availability, call setup and call processing (Camarillo *et al*., 2003).

SIP is different with H.323 protocol, it is not an integrated communication system, it is just a signaling protocol, it is responsible for signaling transmission and negotiating sessions. SIP might be called a more appropriate component, it can be used as a part of other IETF protocol, it is used to construct the complete multimedia architecture. For example, these architectures will include Real-time data Transfer Protocol (RTP) which is used to transmit real-time data and to provide QoS feedback, Real Time Streaming Protocol (RSTP) for controlling the transmission of streaming media, media gateway control protocol which is used to control the Public Switched Telephone Network (PSTN) gateway and the Session Description Protocol (SDP) for describing multimedia sessions (Zhao and Cui, 2006; Murakami *et al*., 1994). Therefore, there are only SIP works with other protocols (Akselsoons and Folkow, 1993), a complete end-user services can be provided.

Thus, SIP itself does not provide specific services, it is a lightweight, versatile tool, whih can be used to create, modify and terminate a session which operate independently in the communication protocols and does not depend on the establishment session class. SIP can be used for different services, in current applications, SIP is mainly used to provide VOIP services. The SIP protocol is used in this study, the telemedicine software is designed and implemented. Audio and video data transmission is not only included in the telemedicine software and the instant messaging, file sending, whiteboard and real-time transmission of medical curve are also included.

MATERIALS AND METHODS

SIP Telemedicine methods

Overall framework design: Telemedicine systems uses the traditional C/S architecture design (Hu *et al*., 1999; Perednia and Allen, 1995). Design of the system include server configuration and software development of client terminal . Server is mainly responsible for SIP signaling interaction, when the session is established which is that the SIP signaling call is completed to start medical data transmission systems, interactive data is no longer needed to be supported by the server but the peer-to-peer, end-to-end transfer is used.

In systems, the function server are Proxy (Proxy server), it is the bridge unit of SIP network system, the role of routing, forwarding SIP messages are played. Registration Server (RS) provides

registration services for users, SIP messages are forwarded to local users, authorization services are provided for user, the proxy server location services are provided for users. Address of the server (Location server) storage SIP user registration information and IP address mapping table and address lookup service is provided for users together with the registration server. The system does not use the Redirect Server (RS). When functional testing, there will place the three functional entities on a single physical server which is built by using SER server, so that the resources can be saved and the basic requirements are meet. The entire software system was developed based on windows system.

Design of software terminals include collection and transmission of medical data, the software includes the following modules: user interface, SIP messaging, audio and video communication, instant messaging, file sending, whiteboard and medical curve. Terminal specific development uses hierarchical thinking, the development of each module is independent. Communication between the system is shown in Fig. 1.

Terminal software design: In telemedicine terminal software function design, there are mainly audio and video communication, instant messaging, file sending, whiteboard and medical curve. Telemedicine is to give "sight" feeling, where audio and video communication is essential. When the network environment is not good, the hope is to provide a minimum of communications security, because less data volume occupied in instant messaging but personal meaning can clearly conveyed and therefore these need to include instant messaging features. When there are interrogation between patients and doctors or exchange between the doctor and the doctor, they always want to exchange important documents, such as the patient's electronic medical records, it is a very important function that documents can be sent at this time. If the two sides need to see a picture at work together, such as the patient's X-ray images, then use the whiteboard is necessary to display and exchange between the two sides. Medical curve can display real-time patient's physiological parameters such as EEG curves, ECG curve.

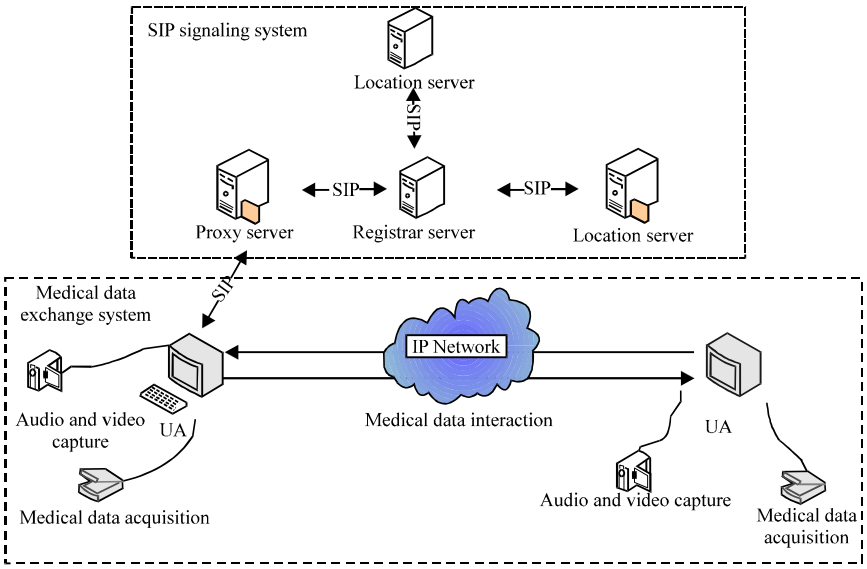

Fig. 1: Overall framework design

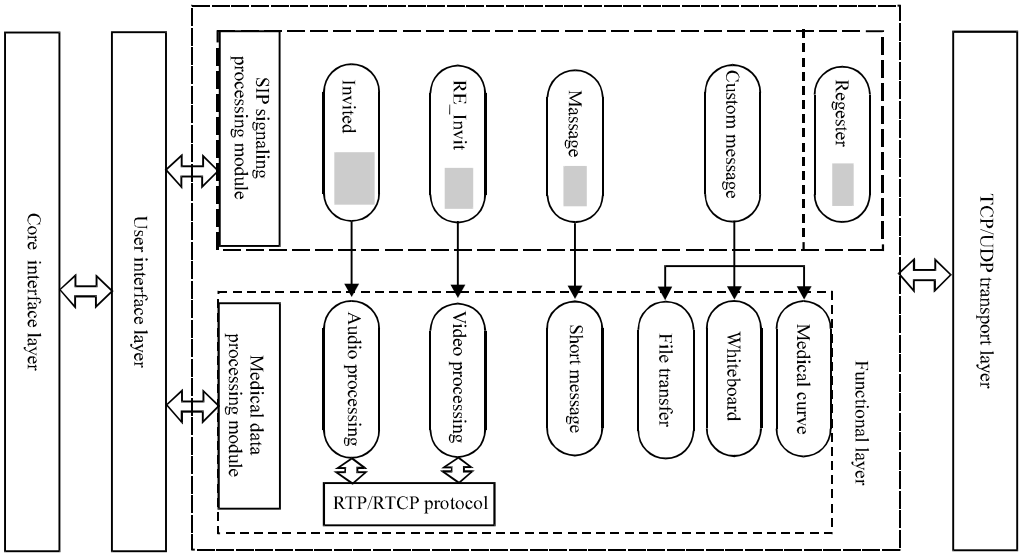

Fig. 2: Client terminal design

To achieve the above functions, telemedicine software terminal design uses a hierarchical design, the software is divided into the user interface layer, the core interface layer, the functional layer and the transport layer. Wherein the software function is realized in the functional layer, it mainly consists of two modules: SIP signaling processing module and a medical data processing module.

As shown in Fig. 2, the user interface layer is based on the visual studio, it is located on the top level of the entire application, it provides a graphical interface and defines the framework of the software. Core interface layer calls some of the interface functions, shields the details of the underlying modules and provide some simple interface, it is easy to invoke the callback function controls in the user interface layer and it achieves docking user interface with the functional layer. Functional layer achieves SIP signaling analysis and generation, it is responsible for completing the processing of medical data and it is the main part of the software, it is the key to achieve the various functional modules. TCP/UDP transport layer is responsible for implementing the underlying socket transmission of medical data.

Detailed design of telemedicine terminal modules: This design of each function is relatively independent, these functions are respectively realized by using the sub-thread implementation. When the program starts, the SIP signaling thread control is firstly run to monitor SIP request message. When the corresponding thread module only build services in other modules, child thread will be created. In addition to the real-time short messaging, each module is the same sub-thread with SIP signaling control module, the other modules are required to establish their own separate thread. The communication between the various sub-threads is shown in Fig. 3.

SIP message processing module: SIP message module is mainly responsible for the session establishment, maintenance and release in throughout the course of the telemedicine sessions. Before telemedicine session is established, the client initiates firstly registration (Register) to the registration server, sending REGESTER message, the server checks the user database and the user's IP address and user name are saved to the location server.

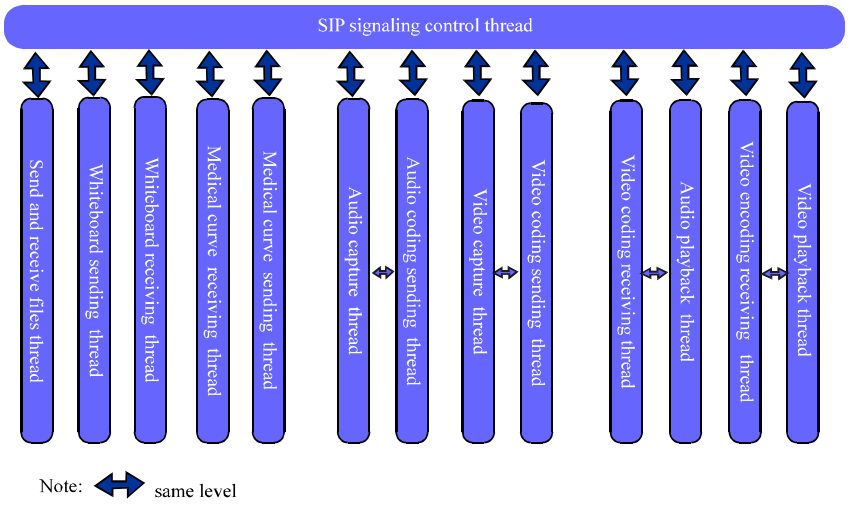

Fig. 3: Each child thread relations

Table 1: SIP messages of the corresponding services

Services division	Caller message	Callee message
Audio session	INVITE´ ACK	180´ 200OK
Video session	RE-INVITE´ ACK	180´ 200OK
Instant messaging	MESSAGE	MESSAGE
Send a file	FILESEND	FLSD-BACK
Whiteboard	WHITEBOARD	WHBD-BACK
Medical curve	MEDICALCURVE	MEDCUR-BACK

After registration is complete, calls and conversations can be made. SIP message processing module has two important functions, SIP signaling interaction initiates a call. Second, if SIP request message arrives are listened at time, the listens of SIP request message is achieved through the establishment of a separate SIP signaling control thread.

For audio and video calls and instant messaging, SIP protocol has provided well how to initiate a SIP call and the corresponding response. However, in the telemedicine terminal, the file sending, SIP Message of whiteboard and medical curve are designed, its expansion is needed own. The expansion of the SIP protocol new functions is also an important advantage of the rapid development of the SIP protocol. After the expansion of telemedicine terminals, the basic SIP signaling is in Table 1.

Among them, the audio session INVITE is used to call each other, after the called INVITE is received, the local ring and 180 provisional response message are replied, after the called is hooked, 200 OK message is sent, after the caller has received 200 OK replies ACK for confirmation message, the audio session is established.

A video session is similar with the audio session, except that this time , the call is 2 calls, the call messages use RE-INVITE.

Instant short message data is usually relatively small, so the message content is placed directly into the message body of the SIP MESSAGE message to send, as the caller need to respond to the short message, the message is replied same by using the MESSAGE.

For file sending, whiteboard, SIP signaling process and medical curve, “Three-way handshake” signaling mode is not used as audio and video sessions, the purpose is to facilitate the design and

implementation. When a caller initiates the call, in order to ensure that the called party can receive the call messages, a call message is sent at specified intervals (e.g., 1 sec) (e.g., documents sent for FILESEND), after the called party receives a call message, message is sent and replied once every once until no caller's message. This design can achieve the same audio and video session before "handshake" effect.

After sending the document, whiteboard and medical curve SIP signaling interaction, the document session, whiteboard sessions and medical curve session are established.

Audio and video processing module: After SIP signaling audio and video call is completed, i.e., the session establishment, SIP signaling control thread will notify the appropriate audio and video processing threads, real-time audio and video is captured, transmitted and playbacked. Real-time audio and video capture, playback and transmission use a plurality of threads, the specific embodiment are the audio data acquisition, encoding and transmitting, receiving and decoding audio data, the audio data playback of the audio data, each of which is achieve by using one thread. Video data acquisition, the video data encoding and transmitting, receiving and decoding video data, the video data playback use respectively a thread. Thus, audio and video processing module establish a total of eight threads.

For the collection of audio and video data, you can use wave Inxxx and wave Outxxx series functions and vfw32.lib library function which the Microsoft windows operating system comes. Transmit audio and video data uses RTP and RTCP protocol package.

Instant messaging module: SIP protocol instant messaging text transmission is not SIP core content but the transmission of instant messaging is achieved by extending the SIP protocol. SIP message of instant messaging extensions is MESSAGE, message format of this message is implemented according to the standard SIP protocol, instant text messages are transmitted by using XML language format which is sent to the called party via MESSAGE, after the called party receives a message of MESSAGE, XML content formatted text of message body is parsed, an instant message is gotten which the caller sends.

File transmission module: After the file is sent to establish a SIP session, the file is officially sent. Send the file sending process is divided into two parts which are the file header information sending and the file contents to send, the direct transmission mode uses a TCP socket transmission. We create a file header structure to store the file some basic information which includes the properties of files and so on.

The structure was named as SOCKET_STREAM_FILE_INFO, it is as follows:

```
typedef struct _SOCKET_STREAM_FILE_INFO
{TCHAR szFileTitle [128]; //File name title
DWORD dwFileAttributes; //Properties file
FILETIME ftCreationTime; //File creation time
FILETIME ftLastAccessTime;//File last access time
FILETIME ftLastWriteTime;//File last modified
DWORD nFileSizeHigh;//File size high double word
DWORD nFileSizeLow; //File size low double word
DWORD dwReserved0; //Reserved 0
DWORD dwReserved1; //Reserved 0}
SOCKET_STREAM_FILE_INFO, *PSOCKET_STREAM_FILE_INFO;
```

Whiteboard module: Whiteboard function is mainly to draw curves and curves plotted requires real-time display in the remote and local clients. In whiteboard data processing, in addition to the need to transfer the whiteboard background image, there are basically curve class information, the crossed coordinate information and the painted brush attribute selection information. Therefore, in the whiteboard function, the data to be transferred is relatively low and real-time transmission is required, data loss is minimized.

According to the characteristics of data transmission in whiteboard, socket data is directly transmitted by using TCP, structure of the data package is as follows:

```
typedef struct MSG_BUFFER_WHITE
{LPSTR operations;
Int Pentype; //Select the Brush Type
Int xPoint; //x coordinate draw point
Int yPoint; //y coordinates draw point}
MSG_BUFFER_WHITE,
*PMSG_BUFFER_WHITE;
```

Medical curve module: In medical curve module, the data to be processed is very similar with data whiteboard module, the coordinate points are depicted in the basic needs of data. Because it is need to select a variety of display curves in medical curve, so the curve type information must be marked in the transmitted data, curve attribute information must also be marked, so the data packaging structure body is used in medical curve is given as:

```
typedef struct MSG_BUFFER_CURVE
{LPSTR operations;
Int Curvetype; // curve type
Int xPoint; //x coordinate draw point
Int yPoint; //y coordinates draw point}
MSG_BUFFER_CURVE, * PMSG_BUFFER_CURVE;
```

Test analysis

Test environment: In the telemedicine system, different tests were performed in both IPv4 and IPv6 network environment. Telemedicine system test server uses SER server. The server is SIP open source server which is integration of registered servers, proxy servers and location server. When tested, the client runs under windows system platform, SER server runs on Linux environment, network environment under IPv4, is public network without regard to NAT transversal problem.

Testing process: First, client software registers to SER server in different locations. Secondly, after the registration is successful, clients to sign up for the legitimate ID number or user name landing. Then, another client is called from one client, when the call, the called user name can be entered directly. Finally, after a call is established, a variety of service dialogue can be made.

Because the designed software is relatively independent between each module, therefore, when a call, a variety of services are choiced based on the needs of users, this design shows that the software design is more humane philosophy.

RESULTS AND DISCUSSION

IP communications and SIP: SIP converged IP network technology has become a reality. SIP is a signaling protocol for multimedia, it can integrate internet service with e-mail, web, voice mail, instant messaging, multi-conferencing and multimedia collaboration. When SIP is used in conjunction with the IP infrastructure, SIP can be provided between any number of SIP enabled terminal alone voice or medical call, video call consultation and point to point video, web Collaboration and chat, instant messaging, wherein the terminal comprises IP telephony, PC, Laptop, Personal Digital Assistant (PDA) and mobile phones. In an open environment, participants can use any number of different vendors from terminal equipment, if the device supports the necessary SIP application and enough attention is given in the implementation process, a multimedia consultation call work very well.

SIP is an IETF standard, it is committed to open IP communications network to a new hardware and software operator, SIP can be used to build converged networks for telemedicine to offer more options and greater flexibility. Once upon a time, companies of using the TDM-based PBX have to rely on PBX vendors to supply the required features and functions; now, the integration of IP networks and SIP open the application development process, SIP provides stand for independent software vendors who have professional experience in the longitudinal medical applications. This process can be supported by the SIP IETF developed programs, the program for the basic functions required for interoperability are defined in level but also it retains the space for application level personalization features.

The base member of SIP system is a user agent and the agent server. SIP user agent is a deployment software in the end user device and server components, it is used to manage a SIP connection. User agents include IP telephony, SIP media gateways, conference servers and information processing systems and other terminals. SIP proxy server is responsible for routing the SIP request to the appropriate destination. Agents usually is located in a SIP registrar which is responsible for maintaining the contact details for a specific user or account IP domain. SIP uses a Real-Time Protocol (RTP) in the real-time transmission between the user agent packaged voice, video and data.

SIP Telemedicine is a interactive services, interactive telemedicine services provide real-time interactions between patient and provider, to include phone conversations, online communication and home visits, it help to make doctor's offices and medical facilities as close to one another as the nearest computer screen, it can be used in the remotest parts of the world or in places as close as a correctional facility, helping to eliminate the dangers and costsc associated with the transportation of prisoners to a medical center, it allows a surgeon to be in 2 places at once. SIP telemedicine is better than Store and forward telemedicine (Gao *et al*., 2014; Ma *et al*., 2014; Zhai *et al*., 2014). Store and forward telemedicine involves acquiring medical data (like medical images) and then transmitting this data to a doctor or medical specialist at a convenient time for assessment offline.

SIP is not a panacea: Unlike many emerging technologies, SIP some facts have been neglected and some of the features were exaggerated. For its part, SIP is not a panacea to communicate, it and many other standard support commonly open, reliable and rich multimedia communications. For important information about the SIP:

- SIP is a signaling protocol, it is independent of transport protocol; it can run on the basis of a number of transport protocols, such as User Datagram Protocol (UDP), TCP and the Stream Control Transmission Protocol (SCTP)

- SIP does not host, nor it does include the specific quality of service (QoS) capabilities; it may cooperate with other protocols to perform this function
- SIP is independent of all security protocols, it is shared with Transport Layer Security (TLS), IP Security (IP Sec) and other security protocols
- SIP is a peer protocol, rather than the IP to PSTN gateway control protocols, such as MGCP or H.248
- SIP provides a control method of a session but the session is not explicitly to use these applications and services; therefore, SIP can not ensure the application behavior
- SIP is independent of the used media, thus it provides flexibility for different types of media session initialization

CONCLUSION

Here, we propose a telemedicine architecture design and gives the detailed design of telemedicine terminal modules, such as SIP message processing module, audio and video processing module, instant messaging module, file transmission modul, whiteboard module and medical curve module. Through experimental tests, the program is feasible, there are good results.

With the rapid development of the Internet, SIP protocol can grow in a very short period of time, because it has a special advantage, first, SIP is considered as an IP network protocol from the beginning design. SIP is a text-based protocol, the distributed applications is based markup language in the field of compatible network. According to standard computing platforms and operating systems, it is easier to integrate SIP protocol into the application program, the integration of the network is promoted. Secondly, SIP is an extensible protocol, it is able to run a wide range of applications with a variety of media formats and load. By SIP protocol, a number of criteria is developed for instant messaging, resource management, media streaming (video) and online friend management applications. New applications can be created and innovatived by this extensibility. Third, SIP protocol is now part of most end point infrastructure, especially the desktop operating system. By this method, it enables developers to design easy application program based on SIP protocol.

Telemedicine system is developed by SIP protocol which is not only convenient, there but also is good compatibility, it can be interoperability with any other SIP protocol developed client. And it has great maneuverability for our future system upgrades and extensions.

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