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ITJ

ISSN 1812-5638

INFORMATION TECHNOLOGY JOURNAL

ANSInet
Asian Network for Scientific Information
308 Lasani Town, Sargodha Road, Faisalabad - Pakistan

Information Technology Journal 12 (21): 6068-6072, 2013
ISSN 1812-5638 / DOI: 10.3923/itj.2013.6068.6072

SIP Extension and Implementation of Multimedia Communication System Based on SIP

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Abstract: This system adopts SIP as control signaling protocol of multimedia communication, combining the demand of multimedia communication and realizing the SIP-based multimedia communication system. This system is constructed on the basis of PC platform; the server uses Open Source Project, developed for core; Windows terminal signaling modules use GNU oSIP stacks; RTP transmission media stream uses oRTP stacks. Aiming at the support limitation of present multimedia communication, this study puts forward SIP extension methods supporting multipoint conference and proofs that the system having good expansibility and flexibility though analysis of system performance.

Key words: SIP, SIP extension, SIP server, SIP terminal, Multimedia communication

INTRODUCTION

SIP agreement is the agreement standard of multimedia communication in IP website which appears in the development of IETF. It is used to establish, amend or end one or more meetings, including Internet multimedia meeting, Internet phone call and multimedia distribution (Tang and Wang, 2009). It is a distributed agreement, pushing the complexity of network equipment to the network edge, so that the core network server in SIP system cannot keep calling statement (the SIP message itself contains all the information of one calling). Because the core network server needs to deal with a lot of calls, if it can keep un-calling statement, its call-dealing ability can be greatly improved (Rosenberg *et al*., 2002; Ha *et al*., 2010). Therefore, it laid a solid foundation for organizing large-scale multimedia communication system.

SIP system mainly consists of two parts: User agent and the network server. The User Agent (UA) can be divided into User Agent Client (UAC) and User Agent Server (UAS) (Rosenberg *et al*., 2002). While the UAC is used to start calls, the UAS is used to response calls. In this way, they constitute the necessary application of the client to accomplish starting and receiving calls. The network server has three categories: proxy, network registrar and redirect. In these three servers, the network registrar is used to register the user address; the proxy is used to route and transmit the SIP message; the redirect is responsible for returning target SIP user agent address information.

SIP agreement uses messages to let network elements to communicate with each other. It can be extended to meet the needs of multimedia system but need to follow a certain principle. There are 6 basic SIP messages: INVITE BYE, ACK, CANCEL, OPTION and REGISTER. The INVITE message is used to start calls (Liu *et al*., 2011; Wang and Lei, 2009). It includes two parts: Message Header and Data Area. The message header contains the address information of calling and called party and the call theme information, while the data area is about session media information which can be realized by SDP; BYE is used to end a conversation when the user want to; OPTIONS are used to enquire the called information, but the option itself can't start a call; ACK will confirm and answer the received information (Cheng, 2010; Wei *et al*., 2004), REGISTER is used for the user to transmit location and address information to SIP server; CANCEL is used to cancel the present request, but it can't end the established connection.

According to the analysis of the SIP system structure and the messages, it can be seen the whole SIP calling process is as follows: the user server starts a call, then the message passes the proxy and inquire the called party address from network registrar, then the proxy start a call to the called party address and at the same time transmit the present dealing message to user agent. The called party accepts the call and at last the call established.

THE FRAMEWORK STRUCTURE

Internet is formed by a great deal of heterogeneous communication subnet's interconnection. When the heterogeneous multimedia conferences clients who have different processing ability attend the same conference at

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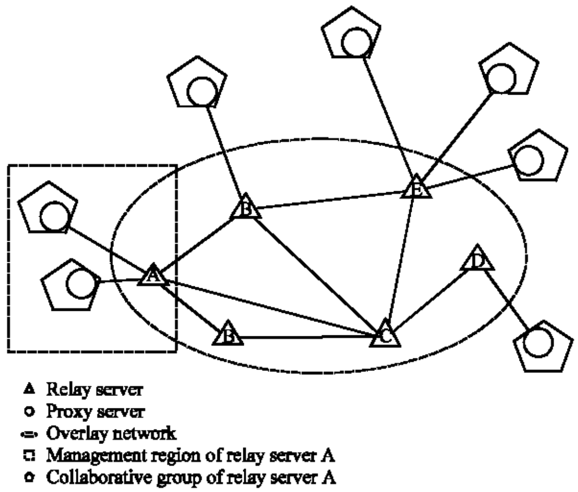

Fig. 1: The Network topology of framework

the same time, it is necessary that the unified coordination framework can identify each heterogeneous client's process ability and can intelligently adjust multimedia's encoding format and wide band transmitting. The framework will use agent mechanisms packing the multimedia conference to cooperative communication group that can intercommunicate which is called collaborative group. Each collaborative group need to set a agent node, being responsible for perceiving the messages from the other group, converting to the control command within the group and transmit the data to the other collaborative group (Li and Li, 2006). Then the communication and negotiation problem of collaborative group will change to agent nodes.

Each cooperative group is autonomous internally and they are connected by the overlay network formed by relay server. The network topology of framework is star shape structure (Fig. 1). Take the overlay network formed by relay server as center; each agent server corresponding collaborative group is redial distributed in the Internet. The organization and management of the overlay network is important to the efficiency and reliability of the message.

REALIZATION OF FRAMEWORKF

The realization of framework mainly depend on the cooperation of control server, relay server, SIP server and SIP terminal. The control server will provide directory services for overlay's network's establishment, admitting relay server register its location and maintaining the network topological structure. The principle for relay server is: different heterogeneous network has at least one relay node. Considering the scope and scale of each multi-medium communication system may contain and data processing ability of relay server, it will control the node quantity flexibly. Each relay server is responsible for more than zero collaborative groups transmitting data. This relay node and the collaborative it's responsible for formed an area which is called management area of relay node. SIP server mainly provide with registration and session management of SIP terminal, providing with the communication mechanism based on SIP for the whole system.

SIP TERMINAL

Because most users use the Windows system at present, the development of this Windows system's terminal decides its promotion. The terminal software based on the Windows platform consists of five parts, such as show in Fig. 2.

Operation system layer is between the hardware and upper application software, helping the application layer to realize the hardware management, memory management, process management and document management. This layer mainly includes the initialization of TCP Socket function, receiving and sending messages and release of the Socket.

Core component layer includes all kinds of protocol stacks supported by terminal, such as SIP stack, SDP stack, RTP stack, XML stack, A/V codec stack and so on (Wang and Bo, 2010).

Application Service Component Layer (ASCL) including calling API function encapsulated by underlying protocol stack for upper layer.

Application service layer provides the realization of specific application business, including sound, video communication, timely news and online service and so on.

The user interface layer provides operation interface for external layer, realizing the parameter configuration of the terminal system, communication control and the management of user friends list and groups.

The realization of SIP and RTP use oSIP stack and oRTP stack. The oSIP stack and oRTP stack are open source projects, providing SIP protocol stack analysis and a state machine mechanism, providing all calls for upper user agent. The application service layer is mainly responsible for handling signaling message (Teng *et al*., 2011). It uses oSIP stack to realize such as sending calls and responding calls, the media flow parameter consultation, ending message and so on. In addition to application service layer and interface layer, the other layer modules are formed by dynamic link library, using the user interface to call this dynamic library.

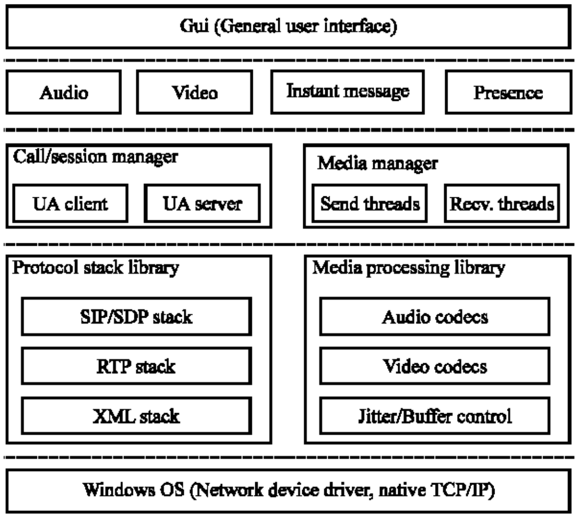

Fig. 2: SIP terminal software

Terminal Solutions and Development Environment.

Signaling protocol uses the SIP protocol in RFC3261, RFC3265 proposal.

Audio and Video collection and playing adopt the DirectShow technique

Media codec format: audio coding using G.711, g.723, video using H.261, H.263 compression.

Streaming media transmission using RTP protocol.

Interface realization using MFC.

Terminal parameter configuration using XML data file.

Development platform: PC, Windows operating system, VC++2005, DirectShow, Windows Media SDK.

SIP EXTENSION

SIP is essentially a point-to-point agreement. Although it is put forward by multimedia application, SIP itself doesn't support the application in multimedia communication (Liao *et al.*, 2003). Therefore, the SIP agreement must be extended when used in multimedia communication at the premise the SIP system structure not changed. Since the SIP itself can be extended easily, the SIP message extension is necessary in multimedia communication.

According to the analysis SIP agreement, SIP doesn't response to the member change. The study put forward a method to notice the user state according to CONF way. There is no definition for multiple Proxy to communicate each other's state, so it is not enough to use the original SIP method.. We must extend SIP. We define a PROX method for Proxy to communicate each other's state. If there is a member join or leave, the domain will use the CONF method and the domain state will use the PROX method. PROS extension method is not emerged by agent server itself, but emerges when there is user change.

As for all the proxy participate in the conference, it is necessary to first understand the other Proxy in this conference. This can come into realization according to LS. For some conference, if some domain request to join UA first time, UA will call to control server and request the address of the other Proxy in this conference at the same time. Every Proxy will keep the other Proxy list so that when there is other UA want to join, they will need to request. Then Proxy will send Proxy message to the other Proxy, including its own address and member state attribute. This can improve its existence to the other Proxy. The other Proxy will update the list according to this information and at the same time sending ACK message to original Proxy and informing the other state member according to CONF method. Only when there is member change, Proxy will sending Prox message to the other Proxy actively. If the last member in the domain leave, its sending message will contain one state indicating its existing the conference and cancelling from the positioning server (Cheng, 2011). After that, the other Proxy will not sending Prox message and only keep Proxy data in positioning server.

Proxy will handle the Prox message according to the Prox message send by positioning server, but not only simply relay the message. According to the above analysis, it builds up Prox structure, not describing the other field of Prox, but mainly describing the following key field in multi-party conference:

```
status=(“statue” ”=” ”alive”|” leave”)
proxy= (”proxy”|”P”) :
1#((name-addr | addr-spec)
proxy=(”proxy” | ”P”):
1#((name-addr|addr-spec)
{*(“;”proxy-params)})
proxy-params=(“status””=”activc”|” leave”)
extension-attribute=extension-name[“=”extension-value]
```

Here, this study only take the Proxy message send from Proxy 3 to Proxy 1 as example, but not to list the detailed message streaming between Proxies. This message is Prox message sends from Proxy to Proxy 1 in agent server list after Proxy processing the UAE's joint in this domain.

```
proxy3??>proxy 1
PROX sip:proxy1@proxyl_kmu.edu.cn SIP/2.0
Via:SIP/2.0/UDP@proxy3_kmu.edu.cn
From:sip:proxy3@proxy3_kmu.edu.cn; tag=53453
To: sip: proxy1@proxyl_kmu.edu.cn; tag=542389
Call-ID:54623187@proxy3_kmu.edu.cn
CSeq: 3 PROX
Status: alive
Proxy: < proxy1@a_kmu.edu.cn >; status=active, < proxy2@b_kmu.edu.cn
>, status=active
Participant: < e@kmu,edu,cn >; stauts=active
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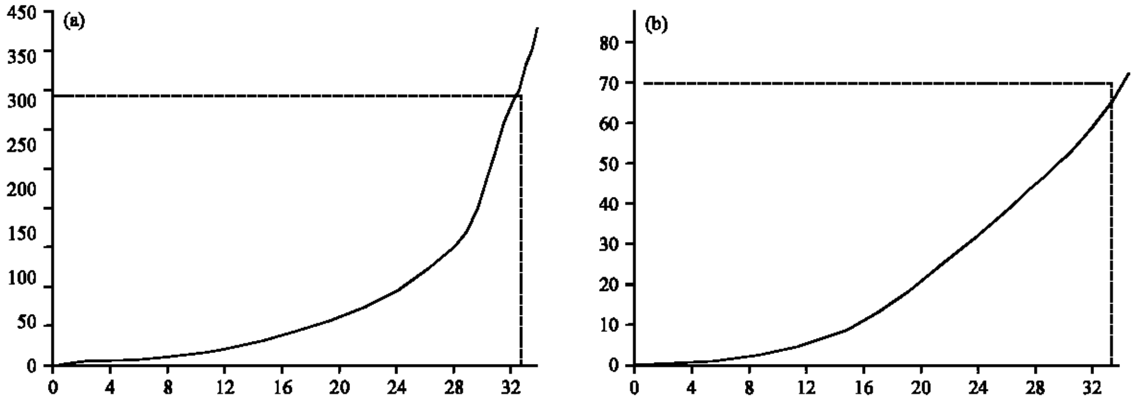

Fig. 3: Relationship between access number and forwarding delay or CPU use ratio

The status shows three present conditions of proxy 3. Alive indicate that it is still alive in the current conference. If there is no conference member in this domain, the status will be leave. This also includes the proxy list and state proxy 3 get from positioning server and the newly joint UAE. It is similar to process the leaving member.

Because the proxy only keeps its domain state member, the member keeps the member list. The Proxy quantity will be far less than the conference member number, so there will not cause conference bottleneck. This improves the system extension, especially suitable for the multimedia application in present Internet network.

ANALYSIS

In the test, we mainly quantizing and analyzing the Proxy system access number of core component, packet relaying delay, resources occupancy and their relationship. The tested resource occupancy mainly referred to CPU service condition when terminal access number rises. The access capacity of Proxy is the access number when the delay reaches the maximum value (According to ITU provision, the communication delay must be less than 350ms)

Test method: test uses scheme based on control point scheme that is to add test code in some terminal code, making the test terminal sending a detection bag in every period of time when sending a data packet. At the same time, Proxy data packet receives detection bag in the entrance and transmitting to the original end. According to Dotal=DTX+DRX+Dfoucs, the relay delay can be calculated (He and Zhao, 2012).

Three PCs are respectively responsible for management server, control server and Proxy as well as registrar server. The logic function of Proxy and registrar is afforded by one same PC. The speech coding algorithm is G.7231, video coding algorithm is H.263, the image format is CIF, conference wide band set is 640kbps. The test result is shown in the Fig. 3.

From the Fig. 3, it can be clearly that when the terminal access 33 road, the packet relay time delay slightly more than 350ms. At that time, the CPU utilization ratio is around 70%. Therefore, when the Focus capacity is more than 32 roads, it can meet the needs of medium-sized conference.

CONCLUSION

SIP protocol is a multimedia signal control protocol of great potential. It plays an important role in the next network generation. Because the SIP multimedia communication system model is in accordance with the layered network idea, it can packaging the heterogeneous multimedia communication system to collaborative group, realizing the large-scale video collaborative communication among different communication groups in Internet. This framework uses group management, autonomous management within the group. It has good expansibility and manipulate. The SIP extension method which supports multipoint conference can easily upgraded though software to support new standard, new function; it has a broad application prospect.

ACKNOWLEDGMENTS

The study was supported by Applied Basic Research Project of Yunnan Provincial Science and Technology Department (2010ZC168).

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