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FRAMEWORK OF SIP-BASED CENTRALIZED MULTIMEDIA CONFERENCING SYSTEM

K.NIVETHITHA[1], JACKULIN.C[2], C.VIJAYALAKSHMI[3], M.GEETHA[4].

[1]K.Nivethitha, AP.Department of CSE, Panimalar Engineering College, Chennai, Tamilnadu, India, nivgiri@gmail.com
[2]Jackulin.C, AP.Department of CSE, Panimalar Engineering College, Chennai, Tamilnadu, India, chin.jackulin@gmail.com
[3]C.Vijayalakshmi, AP.Department of CSE, Panimalar Engineering College, Chennai, Tamilnadu, India, vijidurai88@gmail.com
[4]M.Geetha, AP.Department of CSE, Panimalar Engineering College, Chennai, Tamilnadu, India, geetha114@gmail.com

Abstract: - Multimedia conferencing is becoming a hot topic of communication in recent years. However, the model and architecture of multimedia conferencing are several. This paper first introduces some SIP multimedia conferencing architectures, and their advantages, disadvantages and performance. According to a centralized multimedia conferencing model, feasible system architecture is constructed, which can support middle scale multimedia conference. Most of the researches on SIP-based multimedia conferencing, however, have still remained on theories or experiments. This paper introduces the whole architecture of the practical system, and expatiates on the flows of conference process. The SIP-based control mechanism for achieving the scalability has been designed in details. Finally, the system's performance is evaluated and discussed

Keywords: - SIP, International Telecommunication Union (ITU), IETF & IP multimedia subsystem (IMS)

I. INTRODUCTION

Multimedia conferencing has been on the research agenda for many years. It has desirable advantage for people who don't want to spend the time and money flying all over the world for meetings which require face-to-face contact.

International standardization bodies have defined protocols for urgent requirement of multimedia conferencing. The International Telecommunication Union (ITU) has made H.323 standard [1], and the Internet Engineering Task Force (IETF) carries out SIP [2]. Compared with these two protocols [3], H.323 is widely deployed and more mature due to its early adoption by the market, but has several problems, including scalability issues. At the same time, SIP is more lightweight, flexible and extensible. It is a text-based protocol which can easily interact with other internet protocols. SIP is gradually becoming popular because of its excellent characteristics. The IP multimedia subsystem (IMS) which is a standardized next generation networking (NGN) architecture exploits SIP as its core protocol. It is reasonable to enrich the SIP conferencing architecture with the features that H.323 is not equipped with.

SIP Conferencing

The standards of conferencing requirements [4] and framework [5] by SIPPING working group are earlier than XCON working groups. We can call the system published by SIPPING working group SIP conferencing, which uses SIP for session management and conference control.

As shown in Fig. 1, the SIP conferencing architecture consists of a centralized conference server and some participants. A focus is a SIP user agent which is responsible for the management of the conference using SIP signaling protocol. The focus which is addressed by a conference URI maintains a SIP signaling relationship with each participant, and handles the requests from participants by referring to the conferencing polices. The conference policy stored in policy server contains the rules that guide the decision-making process of the focus for the management of various conference requests from the participants. A conference notification service is a logical function provided by the focus. The focus can act as a notifier [10], accepting subscriptions to the conference state, and notifying subscribers about changes to that state. A mixer is responsible for handling the multimedia streams, and generating output streams which can be distributed to participants. It is also controlled by the focus.

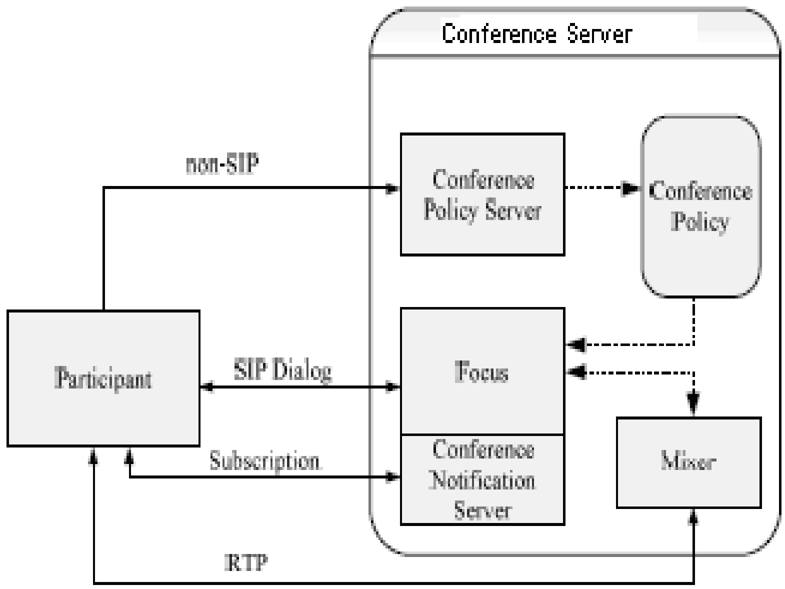

Figure 1. The architecture of SIP conferencing.

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II. DESIGN OF CONFERENCING ARCHITECTURE

The SIP-based centralized multimedia conferencing system architecture we designed is a development of the previously mentioned XCON model. Because of the strongpoint described in Section 1, we surely select SIP as call signaling protocol. In this section, we first list the characteristics of this conferencing system. Then we illuminate our design of the conferencing framework.

A. Characteristics

To build an available centralized multimedia conferencing, we introduce four major functions of this system:

- Besides the scenario that participants can initiatively dial-in to a conference, which is already implemented in some articles, this conference can also dial-out to users who are already registered in the server. The dial-out list can be determined at the beginning of a meeting or added during the conference by someone (e.g., the moderator of a conference) who has right to invite participants.

- We can reserve a conference at our willing time in the future. When the conference begins, it invites users in the dial-out list.
- The conference server can distinctively send full or partial conference state information to the authorized participants who have subscribed the notifications with different demands.
- This multimedia conferencing system does not only support audio or video like traditional products. We can definitely call it a data conferencing, as it also enables document and image exchanges among multiple participants, desktop sharing, which lets users remotely view and control one another's desktops, whiteboard, text messaging and chat.

B. Framework

We illustrate the framework of SIP-based centralized multimedia conferencing system in Fig. 2. We can divide the whole system into two parts: server and participants. A conference participant can be a personal computer or a phone. A telephone including mobile phone can join in a conference as long as it is SIP-compliant to correctly support SIP dialog, that we call it SIP phone. People using SIP phone can enter a conference to talk with other participants, or sometimes can display his performance or receive videos of speakers if this SIP phone has camera and scream.

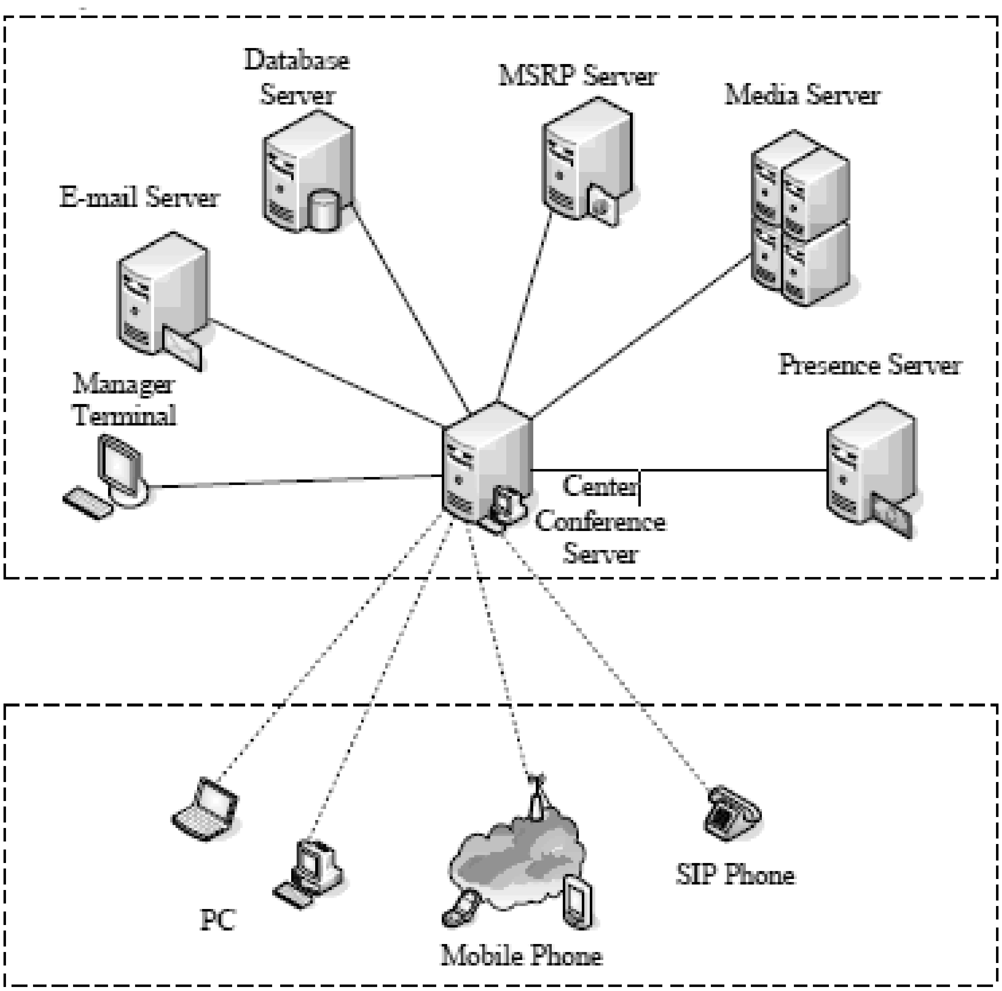

Fig 2. Framework of SIP-based centralized multimedia conferencing system

III. SYSTEM DESIGN

As in centralized conferencing the heavy load of media process will become the bottle-neck of the whole system, we take media process function out of original server. We use a set of media servers to share the system load. When a participant enters a conference, a RTP stream is established between the participant and a media server in the set. The focus control the media server relying on a protocol called MCP defined by ourselves. A MCP client is settled on the center conference server, and a MCP server is on the media server.

Focus

Focus is a SIP user agent which is responsible for the management of the conference using SIP signaling protocols. The focus handles the requests from participants by referring to the conferencing polices which are stored in the membership and media policy databases.

Conference policy server

The conference policy server manages the membership and media policy databases.

Conference policy

The conference policy contains the rules that guide the decision-making process of the focus for the management of various conference requests from the participants.

Mixer

Mixer is responsible for handling the multimedia streams, and generating output streams, which can be distributed to participants. Mixer can be located either in the focus or in the participant's user agent. In both cases, mixer is controlled by the focus.

Conference notification server

A conference notification server is responsible for notifying according to message and status of conference for the participants. The participants can pre concert conference servers of messages and status by SUBSCRIBE.

IV. MODELS OF MULTIMEDIA CONFERENCING

In Table 1, the characteristics of the various models are compared. In a centralized server model, both the focus and the mixer are located together in a centralized conferencing server. In an endpoint server model, both the focus and the mixer are located together in one of the participants. A participant plays the roles of both the server and the participant. In a media server component model, the focus and the mixer are separately located into two centralized conferencing servers. In a distributed mixer model, the focus is located in a centralized conferencing server and the mixer is located in the participants. In a cascade mixer model, the focus is located in a centralized conferencing server, and the mixer is located in several distributed conferencing servers [9-10].

In Table 1, for the case of the cascade mixers model, although it is known to be designed for a large scale conference, the specific details on the control and signaling mechanisms have not been proposed. The distributed conference model can eliminate some problems by making the focus server configured in a distributed way, thereby achieving the greater scalability than that of the cascade model. Furthermore, since the model depends on the centralized conference server, the communication bottleneck may occur due to the concentrated traffic to the single sever. Centralized server model only has a server, and achieves a medium scalability.

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Table 1.The comparison of Multimedia conferencing Models

	Location of Focus	Location of Mixer	Number of Servers	Relationship between Focus and Mixer	Scalability
Centralized Server Model	Central conferencing server	Central conferencing server	1	Co-located	Medium
Endpoint Server Model	One of participants	One of participants	0	Co-located	Small
Media Server Component Model	Central conferencing server	Central conferencing server	2	Separated	Medium
Distributed Mixing Model	Central conference server	Every participant	4	Separated	Medium
Cascade Mixers Model	Central conferencing server	Distributed conferencing server	Many	Separated	Large

V. FLOWS OF THE MODEL

In this section, we depict the flows of conference running mechanism. It is mostly composed of conference creation, running and destroying, participants joining and leaving, and diverse manipulations of participants. Participants have different authorities depending on their roles in the meeting. A special role called moderator has the highest priority.

A. Conference initiation

Based on whether conference identity is created in advance, conference initiation is classed into beforehand initiation and immediate initiation.

(1) Beforehand initiation

Beforehand initiation means conference identity has already been created, and terminal has acquired
Conference identity in advance and asks conference
Server for creating conference.

(2) Immediate initiation

As shown in Fig.3, immediate initiation means that the terminal asks conference server for creating conference and server immediately creates conference identity and returns it to the terminal. Firstly, the terminal asks the URI of the conference server for a quest of INVITE and creates a

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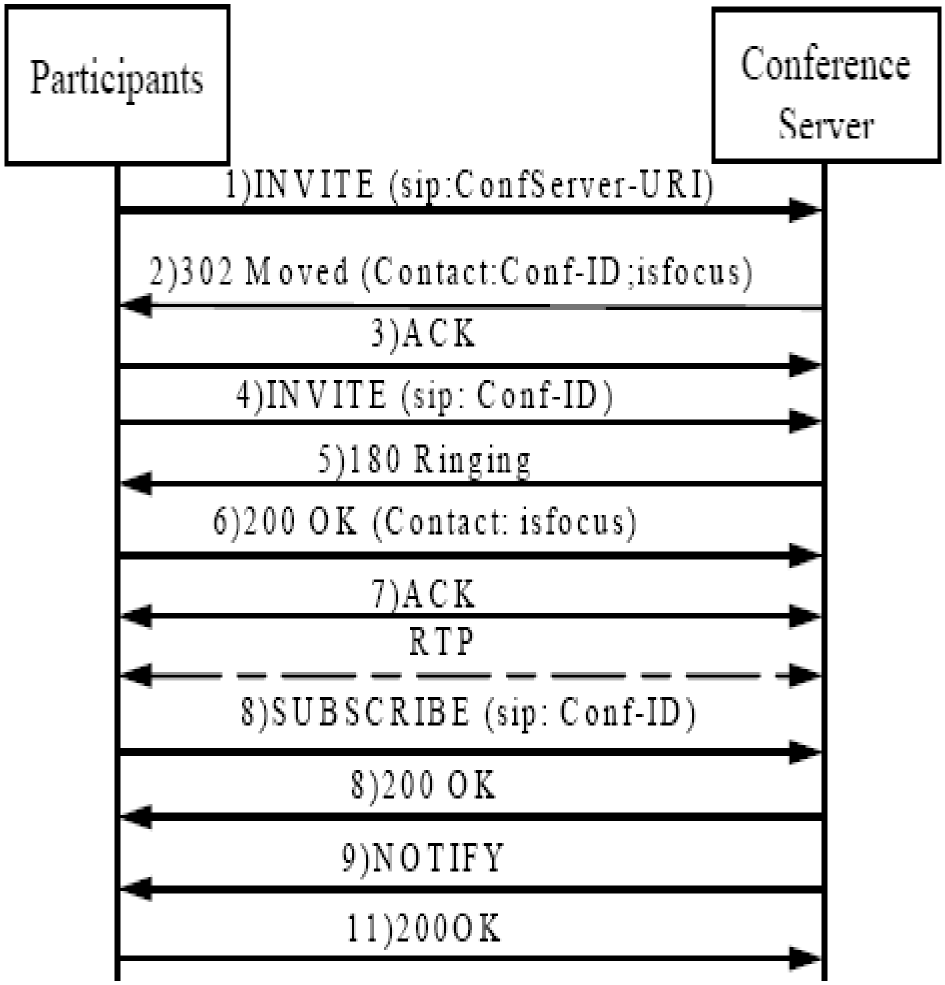

Fig.3 signaling procedures for immediate initiation

Conference identity. Then the terminal asks for INVITE again according to the conference identity, thus entering into the session creation.

B. Invitation and joining

According to different initiations, there are three ways to add a joining to a conference: terminal initiation, conference server initiation and third party initiation.

(1) Terminal initiation

Terminal initiation is the terminal which expects to join the conference positively asks for the joining request. The flow of the terminal initiation is the flow of session creation of two common parties. This way is the dial-in of the traditional telecommunication conference.

(2) Conference server initiation

Conference server initiation is that conference server actively invites a certain terminal to join the conference. The flow of the conference server initiation is similar to that of session creation of two common parties. This way is the dial-out of the traditional telecommunication conference.

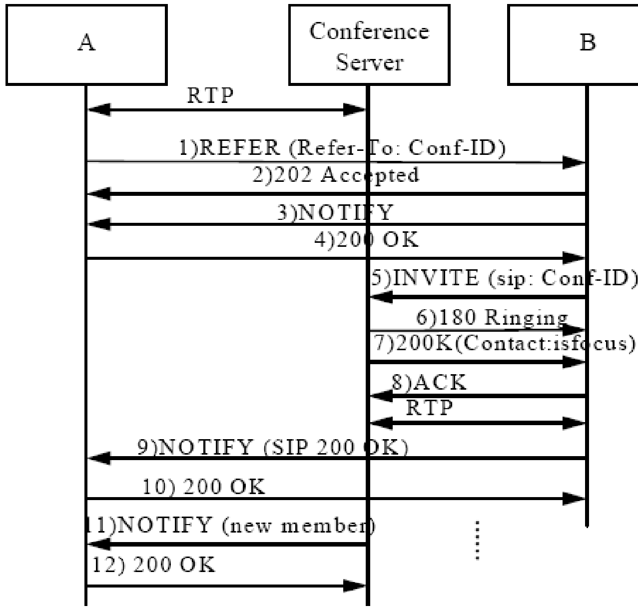

Fig.4 signaling procedure for intimation

(3) Third party initiation

Third party initiation means the third party asks for the joining request rather than the terminal or the conference server. This way is usually used as shown in Fig. 4. It is assumed that terminal H is the participant who has joined the conference, H invites I to join the conference. H sends the message of REFER to I, whose contents of conference identity is denoted by refer-to. If agree to join the conference; it answers the response of 202 to accept the message. Based on RFC315 SIPREFER [8], the accepter needs to report the message to the sender by SIP. Thus, B sends the message of NOTIFY to A, which represents B is relating with the conference identity. Then, B initiates the session. After joining the conference, B needs to send the message of NOTIFY to A to inform a successful relation. As a new membership B joins the conference, the state of the conference has changed. Conference server informs A. As shown in Fig. 4, A sends INVITE to B.

C. Leaving

Similar to the joining, there are three ways to remove a participant from a conference: terminal initiation, conference server initiation and third party initiation

(1) Terminal initiation

Terminal voluntarily leaves a conference as shown in Fig. 5. Similar to the basic SIP, the ending course of the session is simpler. Terminal asks conference server for the message of BYE, which represents leaving the conference. The server accepts the message in the response of 2000k. It is noted that each participant subscribes the case information serve when he joins the conference. So, it needs to cancel the subscription when participants leave the conference. That is to say, terminal sends the message of SUBSCRIBE to the server, in which it represents subscription termination when Expires is 0, the server sends the message of NOTIFY to the terminal and denotes that Subscription-State is terminated.

(2) Conference server initiation

Conference server voluntarily removes a participant from the conference, the flow of which is similar to that of the terminal initiation. Only the message of BYE is sent by the server rather than the terminal. Besides, whether the terminal sends the message of SUBSCRIBE to cancel the subscription, the server needs to send the message of NOTIFY to the terminal, in which Subscription-State is terminated to terminate the subscription

(3) Third party initiation

Third party initiation is a participant who asks another to leave the conference or asks conference server to remove another participant. Similar to the joining, the participant initiated by the third party also uses the way of REFER

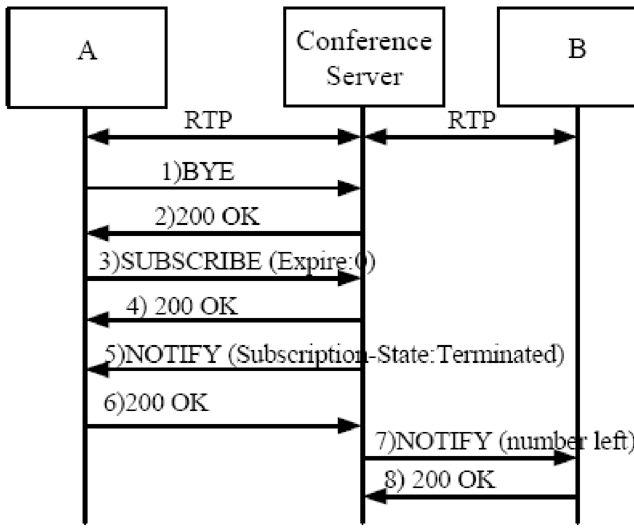

Fig 5. Leaving from participants

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VI. CONCLUSIONS

Based on the implementation, SIP provides a suitable multimedia conferencing platform that allows advanced scenarios and services without requiring that end systems are conferencing-aware. It is possible to build medium scale conferencing servers in software form. In addition to audio and video conferences, various services can be provided at the conference server, such as whiteboard applications and multi-user games. This may lead to centralized conferencing server architecture with different services. Participants can also join the conference from the PSTN if they use SIP to PSTN gateways.

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