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A Framework for QoS-Oriented SIP Multimedia Conference over DiffServ/MPLS Networks

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***Abstract*—Multimedia conferencing is a real-time interactive multi-party communication service with multimedia applications, e.g., audio, video, whiteboard etc. due to the Internet characteristics of resource sharing and ubiquity, the multimedia conferencing would provide lower cost and higher flexibility than the traditional video conferencing in telecommunications. However, the QoS (Quality of Service) issues of packet-switching are also inherent in the multimedia conferencing. In this paper, we propose the QoS-oriented multimedia conferencing framework over DiffServ/MPLS networks. Since DiffServ (Differentiated Services) and MPLS (Multi-Protocol Label Switching) technologies enhance the capability of QoS guarantee on IP network, the quantity and quality of Internet resource would be guaranteed well. In the framework, a virtual multimedia conferencing overlay network, which is composed of focuses, mixers and media gateways, is constructed. Unlike best-effort Internet, the routing path, COS (Class of Service) and bandwidth reservation of data streaming in our framework can be controlled well by the overlay network. Therefore, the QoS-oriented multimedia conferencing service would be approached by our framework.**

***Index Terms*—multimedia conference, video conference, SIP, QoS, DiffServ, MPLS**

I. INTRODUCTION

Multimedia conferencing provides users a multi-party communication service, which connects two or more endpoints through IP networks. The users can communicate to all participants in the same conference with multimedia applications, such as audio/video, whiteboard, file transfer, etc. In order to support the service reliability, availability and security, the multimedia conferencing system should help users to handle the conference management and control. Since it realizes a real-time interactive conference as sharing a virtual working space through Internet, the users can join and leave a conference remotely, and they would feel like the same in the real conference. Additionally, due to the Internet characteristics of resource sharing and ubiquity, the multimedia conferencing would provide lower cost and higher flexibility than the traditional video conferencing in telecommunications. However, the Internet weaknesses are also inherent in the multimedia conferencing. In a commercial multimedia conferencing service, how to guarantee the QoS (Quality of Service) is very significant.

Several researches on multimedia conferencing have been developed including system architectures, signaling standards, and service profiles. However, the most of them assume that the quantity and quality of Internet resource are available and stable. They focus on the application developments, but almost omit the inherent transmission characteristics of Internet. Actually, the real-time interactive multimedia conferencing service is very sensitive to the transmission quality of Internet. Therefore, if the transmission quality is guaranteed well, the conferencing quality should be handled easily, e.g., the quality of audio/video, call-setup delay, service reliability, etc.

We propose the QoS-oriented multimedia conferencing framework over DiffServ/MPLS networks. Since DiffServ (Differentiated Services) [1] and MPLS (Multi-Protocol Label Switching) [2] technologies enhance the capability of QoS guarantee on IP network, i.e., differentiated services, bandwidth reservation, traffic engineering, fast forwarding, fast reroute and VPN (Virtual Private Network), the quantity and quality of Internet resource would be guaranteed. In the upper layer, a virtual multimedia conferencing overlay network, which is composed of focuses, mixers and media gateways, is constructed by the multimedia conferencing service provider. Particularly, the routing path, COS (Class of Service) and bandwidth reservation of data streaming in our framework can be controlled well by the overlay network, not by the best-effort Internet with shortest-path routing mechanism. Thus, the QoS-oriented multimedia conferencing service would be approached.

II. RELATED WORKS

Nowadays, SIP (Session Initial Protocol) [3] and H.323 [4] are the two major open communication standards for VoIP conferencing. H.323 standard based on tightly coupled model was proposed by ITU-T (International Telecommunications Union Telecommunications) in 1996. However, due to the high complexity and the low flexibility, the development of H.323 tends to be stable. Otherwise, IETF (Internet Engineering Task Force) has been discussed the issues to VoIP and IP network conferencing in recent years. SIP has been proposed to simplify the development of VoIP applications. Since SIP has the advantage of high scalability and flexibility, novel and various SIP applications are developed recently. In addition,

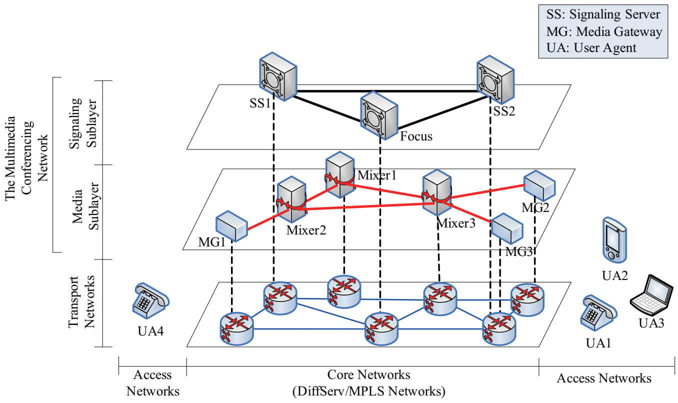

Fig. 1. The system architecture of QoS-oriented multimedia conferencing over DiffServ/MPLS networks.

IETF proposed several multimedia conferencing architectures according to SIP and their related standards. However, the architectures work over the best-effort Internet, and the characteristics of IP networks would not be considered realistically.

Therefore, DiffServ/MPLS networks are adopted in this paper. The QoS-oriented multimedia conferencing framework is proposed to improve the system reliability and availability. Our system enhances CPCP [5] to support the conference management that is short-lived and on-demand, i.e., Ad-hoc conferencing.

III. Multimedia Conferencing System Architecture

A. System Architecture

The QoS-oriented multimedia conferencing system over DiffServ/MPLS network is shown in Fig. 1. The system architecture can divide into two logical layers, the multimedia conferencing network and the transport networks, according to the roles. Additionally, the multimedia conferencing network is separated into the signaling sublayer and the media sublayer. The signaling sublayer is constructed from signaling servers and focuses, and the media sublayer is constructed from mixers and media gateways.

In the environment, the conference users access the conferencing service from their local Internet access networks. The sufficient and available bandwidth in the local access networks for the multimedia conferencing service is assumed, and the transportation quality could be upheld. Additionally, the core networks below the multimedia conferencing network, i.e., DiffServ/MPLS networks, provide differentiated services, bandwidth reservation, traffic engineering, fast forwarding and fast reroute. According to the features of MPLS and DiffServ, the performance and reliability of IP networks could be enhanced. The DiffServ-enabled LSPs are established by the network provider in advance after contracting network-level SLAs. In the media sublayer, the traffic of each conference streams would be aggregated into the proper LSPs on demand to deal with the impact of long distance and shared-media network transport after conferences are set up. According to

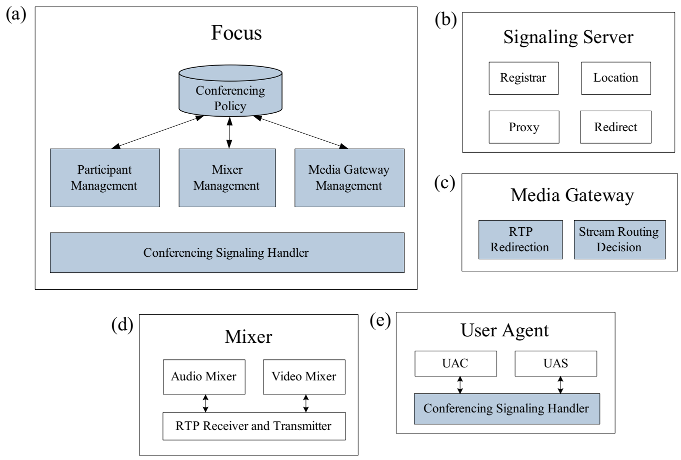

Fig. 2. The functional blocks of Focus, Signaling Server, Media Gateway, Mixer and User Agent.

the cost and network stability, the proper network service class could be purchased by the multimedia conferencing service provider, e.g., Expedited Forwarding class (EF), Assured Forwarding class (AF) and Best Effort class (BF). Each class represents different QoS treatment in the core networks. Besides, the entities in the signaling sublayer are also connected with the DiffServ-enabled LSPs. Finally, the network transportation QoS in the multimedia conferencing network could be assured.

In the situation, the virtual multimedia conferencing service overlay network composed of focus, signaling servers, mixers and media gateways is constructed. The research issues of multimedia conferencing control, resource planning and resource management should be discussed in the overlay network. However, this paper focus the design of system framework, and the resource planning and resource management falls out of the interest of this work.

B. System Elements

The five major elements in our multimedia conferencing system are defined, focus, signaling server, media gateway, mixer and user agent. The functionalities of each element are shown in Fig. 2 and explained below. Note that the gray function blocks are designed additionally by this paper.

1) **Focus** is the centralized conferencing signaling control. Its function blocks are shown in Fig. 2 (a). All the participants and service elements need to negotiate with the focus using conferencing signaling, i.e., SIP with CPCP. Focus deals with participant management, mixer management and media gateway management in the multimedia conferencing overlay network. The conferencing control and management are done according to the conference policy of the multimedia conferencing service provider.
2) The functional blocks of **Signaling Server (SS)** are illustrated in Fig. 2 (b). The signaling server provides the functionalities of proxy server, redirector server, location server and register server defined in SIP.

3) The functional blocks of **Media Gateway (MG)** are illustrated in Fig. 2 (c). Media gateway is a new element in our system, and never discussed in the related works. The media gateway deals with RTP (Real Time Protocol) redirection and streaming aggregation into the suitable DiffServ-enabled LSPs. Due to the long-distance and shared-media communications of Internet, the end-to-end transportation could be suffered from congestion and latency. In the conventional environments, the routing paths of multimedia streams are controlled by network service provider with shortest-path routing mechanism and network policy routing. However, actually, the states of conferencing resource, users' requirements and network resource are known by the multimedia conferencing service provider. Therefore, the routing paths of multimedia streams are dominated by the media gateways in our framework. Additionally, because of the low complexity and the low cost, the many media gateways could be deployed in every region near our customers.
4) The functional blocks of **Mixer** are shown in Fig. 2 (d). A Mixer dominates several regional MGs and combines the audio/video streams not only from local MGs but also from other mixers. Finally, the Mixer delivers the mixed media stream to each participant via the MGs and the other mixers. Except the MGs, the structure is similar to the cascaded mixers.
5) **User Agent** is illustrated in Fig. 2 (e). Except of the functionalities of SIP UAC (User Agent Client) and SIP UAS (User Agent Server), the User Agent deals with the conferencing signaling, i.e., SIP with CPCP.

IV. An Use Case of Multimedia Conferencing

For example, we assume that there are three participants in the same area. A MG and a mixer have been deployed in the area to provide the multimedia conferencing service. In this case, Tom who is the original conference initiator wants to invite the other two participants (Bill and Joe) to start a multimedia conference. Based on the participant model and the conference model, the sequence chart of constructing a conference is illustrated in Fig. 3, and the participant management can be demonstrated.

In addition, the sequence charts of MG control and media channel negotiation are illustrated in Fig. 4. The detail to negotiate the video channels is explained below.

1) Tom sends an INVITE message to focus. The INVITE message includes a SDP offer, which indicates media type, endpoint location and transport port for media [6] [7]. The enhanced CPCP is also embedded in the message body. In message F1, the CPCP represents the configuration for conference initiation. Some parameters in CPCP are defined to achieve conference settings. For example, $< Setting >$ describes the conference URI or ID, maximum and minimum participants, and security level. $< Time >$ describes the time of conference begin and terminate. $< DialOut - list >$ carries the participant list that the focus needs to invite. $< RuleSet >$ describes the rules related to conference control.
2) After receiving the message from Tom, the focus negotiates the resource allocation with the MG and the mixer shown in messages F2 and F4, respectively. During the negotiation, the focus also configures the media channel. The media channel is set up by using the third party call control (3pcc) mechanism defined in RFC 3725 [8]. The messages F1, F2, F3, F9 are negotiated for the media channel between Tom and MG; the messages F4 to F8 are negotiated for the media channel between MG and mixer. Consequently, Tom transmits the multimedia data to MG, and then MG aggregates the related streams into the appropriate LSPs and delivers the multimedia data to the mixer.
3) After the conference is initialized, the focus invites the participants according to the dial-out list according to the same way. The details are omitted.

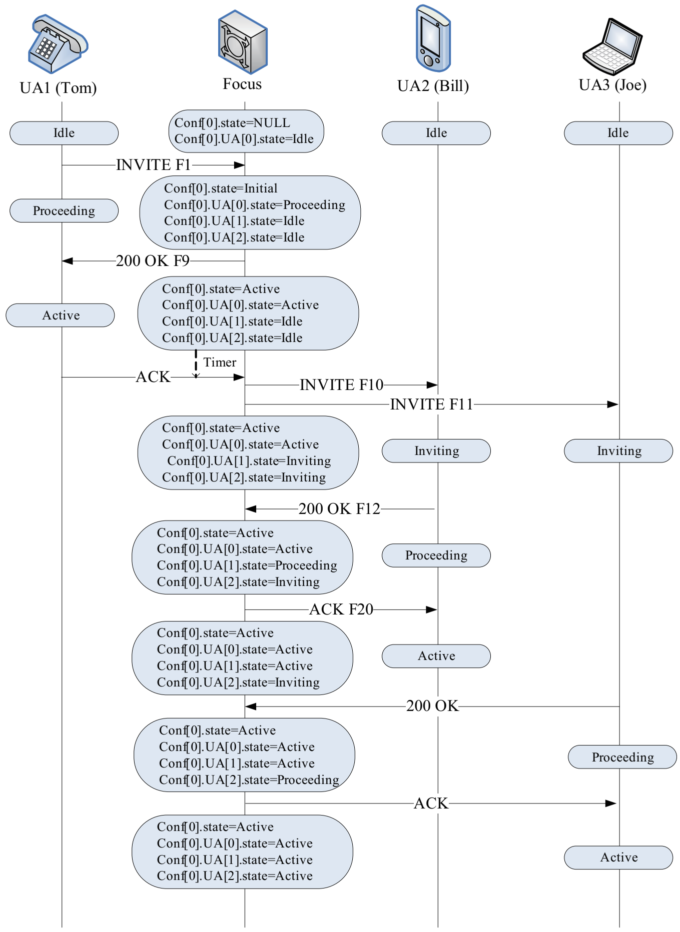

Fig. 3. The sequence chart of constructing a conference.

Therefore, after finishing the conference control procedure, all participants can communicate together. All multimedia streams are aggregated into DiffServ-enabled LSPs to solve the QoS issues of long-distance transportation between mixers and

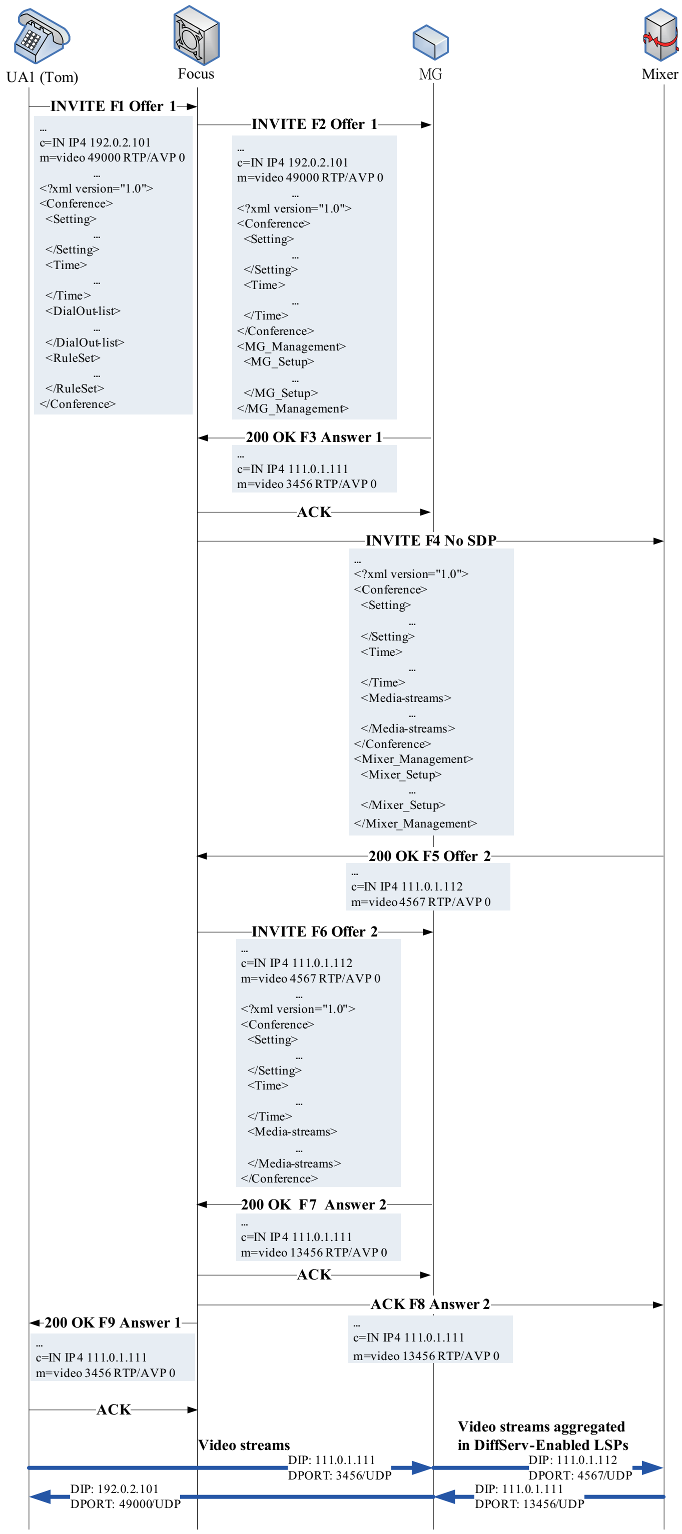

Fig. 4. The sequence chart of session negotiation while a conference is initialized.

MGs. Based on the framework, the QoS-oriented multimedia conferencing service would be approached.

V. Conclusion

We propose the QoS-oriented multimedia conferencing framework over DiffServ/MPLS networks. Because of DiffServ and MPLS technologies for IP networks, the QoS issues of long-distance transportation in the current Internet would be improved. Additionally, unlike the best-effort Internet with shortest-path first routing mechanism, the routing path of multimedia streams could be controlled well by the multimedia conferencing service provider. Finally, the conference management and related protocol enhancement are proposed. According to the demonstration of the use case and our system prototype, the framework is realizable and approachable. However, conference service is a point-to-multipoint communication model; it is more complex than the traditional client-server and point-to-point communications. Many research topics are still ongoing, such as resource allocation, call admission control, floor control, etc. In the future, we will focus on the resource planning and dynamical allocation based on our framework. We believe that the QoS-oriented multimedia conferencing framework over DiffServ/MPLS networks and the related works can give a good solution to multimedia conference development.

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References

[1] S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang, and W. Weiss, “An architecture for differentiated services,” RFC 2475, Dec. 1998.

[2] F. L. Faucheur, L. Wu, B. Davie, S. Davari, P. Vaananen, R. Krishnan, P. Cheval, and J. Heinanen, “Multi-protocol label switching (MPLS) support of differentiated services,” RFC 3270, May 2002.

[3] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, “SIP: Session initiation protocol,” RFC 3261, Jun. 2002.

[4] *H.323v4: Packet-based multimedia communications systems*, ITU-T Std., Nov. 2000.

[5] H. Khartabil, P. Koskelainen, and A. Niemi, “The conference policy control protocol (CPCP),” IETF Draft, Oct. 2004, draft-ietf-xcon-cpcp-01.txt.

[6] M. Handley, V.Jacobson, and C. Perkins, “SDP: Session description protocol,” RFC 4566, Jul. 2006.

[7] J. Rosenberg and H. Schulzrinne, “An offer/answer model with the session description protocol (SDP),” RFC 3264, Jun. 2002.

[8] J. Rosenberg, J.Peterson, H. Schulzrinne, and G. Camarillo, “Best current practices for third party call control (3pcc) in the session initiation protocol (SIP),” RFC 3725, Apr. 2004.