QoS-Guaranteed Realtime Multimedia Service Provisioning on Broadband Convergence Network (BcN) with IEEE 802.11e Wireless LAN and Fast/Gigabit Ethernet

Young-Tak Kim, Young-Chul Jung, and Seong-Woo Kim

Abstract: **In broadband convergence network (BcN), heterogeneous broadband wired & wireless subnetworks and various terminal equipments will be interconnected. In order to provide end-to-end realtime multimedia services on such heterogeneous networking environment, as a result, two major problems should be resolved: i) Multimedia session establishment & negotiation that adjusts the differences in the capability of multimedia data processing at the end terminal nodes, ii) quality of service (QoS)-guaranteed connection establishment or resource reservation with connection admission control (CAC) in each heterogeneous subnetworks along the path. The session layer signaling (e.g., SIP/SDP) should be extended for QoS negotiation, and must be tightly cooperating with network layer signaling or resource reservation with CAC function. In this paper we propose a session and connection management architecture for the QoS-guaranteed realtime multimedia service provisioning on BcN, with Q-SIP/SDP, resource reservation protocol with traffic engineering extension, and CAC functions. The detailed interaction scenario and related algorithms for QoS-guaranteed realtime multimedia session, resource reservation and connection establishment are explained and analyzed. From the experimental implementation of the proposed scheme on a small scale BcN testbed, we verified that the proposed architecture is applicable for the realtime multimedia service provisioning. We analyze the network performance and QoS parameters in detail.**

Index Terms: **Broadband convergence network (BcN), IEEE802.11e, multi-protocol label switching (MPLS), quality of service (QoS), session initiation protocol (SIP)/session description protocol (SDP).**

I. INTRODUCTION

In broadband convergence network (BcN), heterogeneous broadband wired & wireless subnetworks and various terminal equipments will be interconnected. As an example, the backbone transit network may be implemented by internet protocol (IP)/multi-protocol label switching (MPLS) or IP/wavelength division multiplexing (WDM) technology with xDSL or passive optical network (PON) access loop, while the home/office intranet may be implemented based on IEEE 802.3 Fast/Gigabit Ethernet or IEEE 802.11 wireless local area network (WLAN) (i.e., WiFi) technology. Each subnetwork may have different signaling mechanism for connection establishment or resource reservation with connection admission control (CAC). Since the practical home/office intranets are mostly implemented with connectionless and contention-based Ethernet or WLAN, the quality of service (QoS)-guaranteed service provisioning is strictly limited unless well-designed CAC is supported.

Also, the end terminal may have different capability in multimedia data processing. For example the caller may be a PC-based multimedia device with high capability in video data encoding/decoding and large display size, while the called party may be a PDA-based multimedia device with limited capability and small display size. In this case, the end-to-end realtime multimedia session should be established that can be supported by the both end terminal nodes. As a result, we must resolve two major problems in order to provide QoS-guaranteed end-to-end realtime multimedia service on such heterogeneous networking environments: i) Multimedia session establishment & negotiation that adjusts the difference in the capability of multimedia data processing at the end terminal nodes, ii) QoS-guaranteed connection establishment or resource reservation with CAC in each heterogeneous subnetworks along the path.

For the multimedia session establishment & negotiation among heterogeneous multimedia terminals with different capabilities in audio/visual data processing, extended session initiation protocol (SIP)/session description protocol (SDP) should be used. SIP is a signaling protocol on application-layer for managing sessions with one or more participants [1]. The session may be voice over IP (VoIP) telephone call or real-time multimedia conferences. SDP is used for describing multimedia sessions for the purpose of session announcement, session invitation, and other forms of multimedia session initiation [2]. In order to establish a multimedia communication session, SIP/SDP messages must be exchanged among participants to check the availability of participant, capability of terminal, determination of media type, transport protocol, format of media, related network and port addresses [3]. After determination of parameters for a multimedia session, QoS-guaranteed connection for the session must be established between the participant's terminals.

When the QoS parameters and traffic parameters for the multimedia session are determined via the exchanges of SIP/SDP messages between the end terminals, a network layer connection establishment in connection-oriented subnetwork and a resource reservation with CAC should be provided. For IP/MPLS

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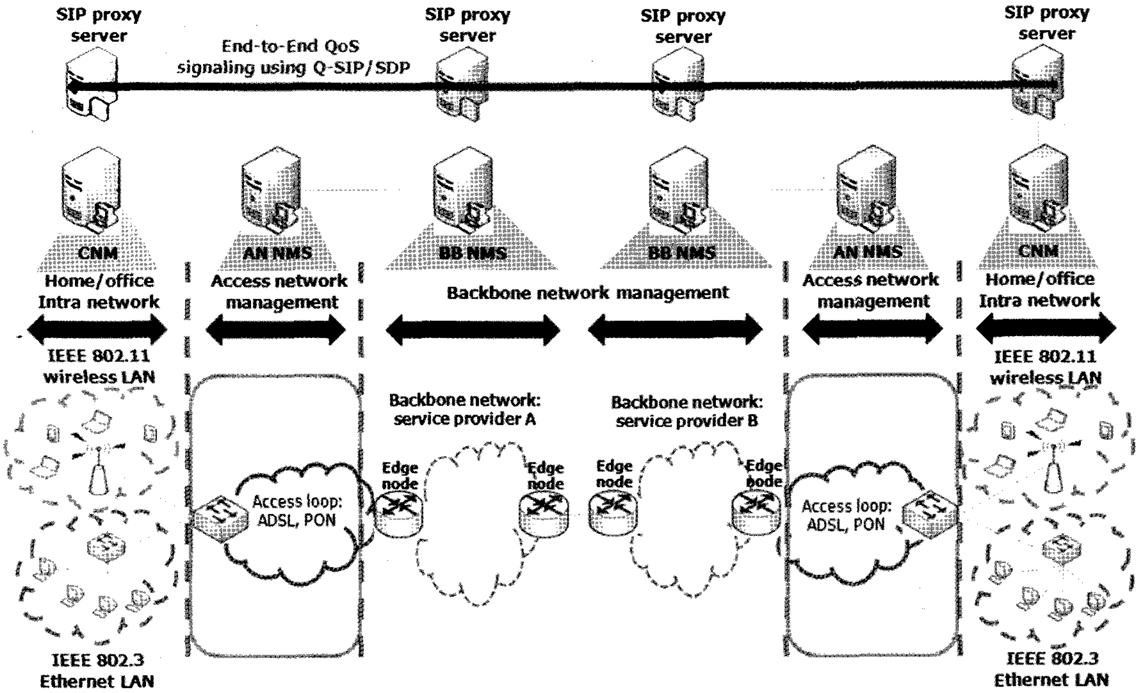

Fig. 1. End-to-end session layer signaling and per-domain subnetwork management for QoS-guaranteed multimedia service provisioning.

transit network, the resource reservation protocol with traffic engineering extension (RSVP-TE) [4] can be used to establish MPLS label switched path (LSP) for the multimedia session. In the non-connection-oriented subnetworks, such as IEEE 802.3 Fast/Gigabit Ethernet or IEEE 802.11 WLAN, however, there is no explicit connection establishment. Instead, we must provide the bandwidth allocation by configuration of Fast/Gigabit Ethernet switch ports with proper CAC function for the guaranteed QoS provisioning. The end-to-end connection establishment or resource reservation in network on such heterogeneous networking environment, should be tightly cooperated with the session layer signaling (e.g., SIP/SDP).

In this paper, we propose a session and connection management architecture for the QoS-guaranteed realtime multimedia service provisioning on BcN, with Q-SIP/SDP, RSVP-TE, and CAC functions. The detailed interaction scenario and related algorithms for QoS-guaranteed realtime multimedia session, resource reservation and connection establishment are proposed. From the experimental implementation of the proposed scheme on a small scale BcN testbed, we verified that the proposed architecture is feasible for the QoS-guaranteed realtime multimedia service provisioning. We analyze the network performance and QoS parameters.

The rest of this paper is organized as follows. Section II briefly explains the related work, including heterogeneous networking technologies in BcN, SIP/SDP, and RSVP-TE. Section III explains the functional model of the proposed session and network layer signaling with related resource reservation and CAC. Section IV explains the performance analysis of QoS provisioning with the proposed scheme based on experiments on a test-bed. Finally, Section V concludes this paper.

II. BACKGROUND AND RELATED WORK

A. Heterogeneous Subnetwork Technologies in BcN

In BcN, various broadband networking technologies, such as IP/MPLS or IP/WDM transit network, very high-data rate digital subscriber line (VSDL) or PON access loop, and IEEE 802.3 Gigabit/Fast Ethernet or IEEE 802.11 WLAN home/office intranet, will be integrated. Each subnetwork may have different signaling mechanism for connection establishment or resource reservation with CAC. Fig. 1 shows the end-to-end session layer signaling and per-domain subnetwork management for QoS-guaranteed multimedia service provisioning.

The IP/MPLS or IP/WDM backbone transit network will be equipped with (G)-MPLS signaling, such as RSVP-TE, that provides constraint-based LSP establishment in the MPLS network domain. The transit network may be implemented by DiffServ-over-MPLS virtual overlay networks, in order to efficiently provide QoS-guaranteed differentiated multimedia services.

The access network interconnects commercial backbone transit network and home/office intranet. Nowadays, most access loops are implemented by VDSL or PON technologies that provide fixed bandwidth channel. Since most access loop does not include packet switching functions, bandwidth-guaranteed channel allocation with simple CAC will be good enough for the QoS-guaranteed multimedia service provisioning in the access loop.

The IEEE 802.3 Fast/Gigabit Ethernet and IEEE 802.11 WLAN are the most popular home/office intranet technologies because of the simplicity and low installation cost. The carrier sensing multiple access with collision detection (CSMA/CD)-based Ethernet and carrier sensing multiple access with collision avoidance (CSMA/CA)-based WiFi, however, are contention based networking technology where bandwidth & QoS cannot be guaranteed by themselfs. Well designed CAC should be supported in the Ethernet switch or WiFi access point (AP) that may be included in the home/office gateway.

B. SIP/SDP

SIP is a session-layer signaling protocol that is used for establishment, modification, and release of sessions among participants [1]. The session may be VoIP, video conference, multimedia distribution, and IP multimedia subsystem (IMS) applications. SIP supports five facets of establishing and terminating multimedia communications: user location, user availability, user capabilities, session setup, and session management. SIP may run on top of several different transport protocols, such as UDP and TCP.

SIP is composed of three major functional elements: User agent (UA) and servers.

1. *UAs* originate SIP requests to establish media sessions and to exchange media. A user agent can be a SIP client software running on a desktop PC, laptop, PDA, or other available devices.
2. *SIP proxy server* is used to help the routing request to the user's current location, to authenticate and authorize users for services, to implement provider's call-routing policies, and to provide features to users. *Redirect server* receives a request from a user agent or proxy, and returns a redirection response, indicating where the request should be redirected. *Registrar server* provides a registration function that receives SIP registration request and updates the users agent's information into location server or other database.
3. *Location servers* support the database of location information of users, such as URLs or IP addresses, scripts features and other preferences.

SIP invitations are used to create sessions and carry session descriptions that allow participants to agree on a set of compatible media types. SDP [2] has been designed to convey session description information, such as session name and purpose, the type of media (e.g., video, audio, etc.), the transport protocol (e.g., RTP/UDP/IP or H.320), and the media encoding scheme (e.g., H.261 video, MPEG video, etc.). SDP also includes information associated to QoS (e.g., bandwidth, delay, and jitter) and security (e.g., encryption key).

When a multimedia communication session is accepted by the SIP/SDP signaling, the related traffic and QoS parameters should be determined. The traffic parameters for realtime multimedia services are committed information rate (CIR), committed burst size (CBS), excess burst size (EBS), and peak information rate (PIR). The QoS parameters include end-to-end transfer delay, delay variation tolerance, packet error rate (PER), and packet loss rate (PLR). Based on the determined QoS & connection parameters, a network-layer connection establishment is requested using user-network interface (UNI) and network-node interface (NNI) signaling [5].

When the subnetwork access mechanism is based on connectionless or contention mode (e.g., IEEE 802.3 Fast/Gigabit Ethernet and IEEE 802.11 WLAN), however, the network layer should allocate the required resources with appropriate CAC mechanism. In following session, we propose a detailed procedure of network resource allocation scheme for QoS-guaranteed service provisioning.

C. RSVP-TE for IP/MPLS Subnetwork

After determination of the QoS and traffic parameters for a multimedia session using the SIP/SDP session layer signaling, QoS-guaranteed per-class-type connection must be established among the participant end terminal nodes [5]. In network layer, a connection establishment is accomplished by UNI and NNI signaling. For UNI signaling between the user terminal and the ingress edge router, RSVP-TE can be used to carry the connection request [4]. MPLS [6] NNI signaling standard for inter-AS (autonomous system) domain network, however, is not yet well standardized to establish per-class-type QoS-guaranteed connections across multiple AS domain networks of different network operators. As a consequence, an efficient alternative transit networking scheme must be provided so as to configure a per-class-type QoS-guaranteed differentiated service (DiffServ) provisioning and to support the scalable CAC [5]. In order to support the per-class-type DiffServ provisioning, RSVP-TE must provide traffic engineering extensions so as to deliver the traffic and QoS parameters.

The user agent in multimedia terminal must provide RSVP-TE client function, while the ingress router must support the RSVP-TE server function. Since RSVP-TE establishes only unidirectional connection, two PATH-RESV message exchanges should be implemented to establish a bidirectional path between the user terminal and the ingress router [5]. The RSVP-TE UNI signaling message carries the QoS and connection parameters for the specified application data flows. RSVP-TE NNI signaling message is used by routers to deliver QoS requests to all nodes along the path in IP/MPLS subnetwork, in order to establish and maintain the state to provide the requested QoS-guaranteed service. The routers should be able to reserve network resources according to the requested QoS & traffic parameters [7].

III. SESSION ESTABLISHMENT FOR QOS-GUARANTEED REALTIME MULTIMEDIA SERVICE PROVISIONING ON BCN

A. Q-SIP/SDP Interaction for End-to-End QoS Session Negotiation

Fig. 2 depicts the overall functional block diagram and interaction model of session and connection management in QoS-guaranteed multimedia service provisioning [5], [8]. The user terminal will firstly discover available service from the service directory of network operator that provides service profiles. The service subscription will define the service level agreement (SLA) of the service, and service level specification (SLS)/traffic level specification (TLS) will further specify the detailed traffic parameters and QoS parameters. For QoS-guaranteed service provisioning, the network operator must configure DiffServ-over-MPLS virtual overlay networks for differentiated services. The scalability can be achieved by using vir-

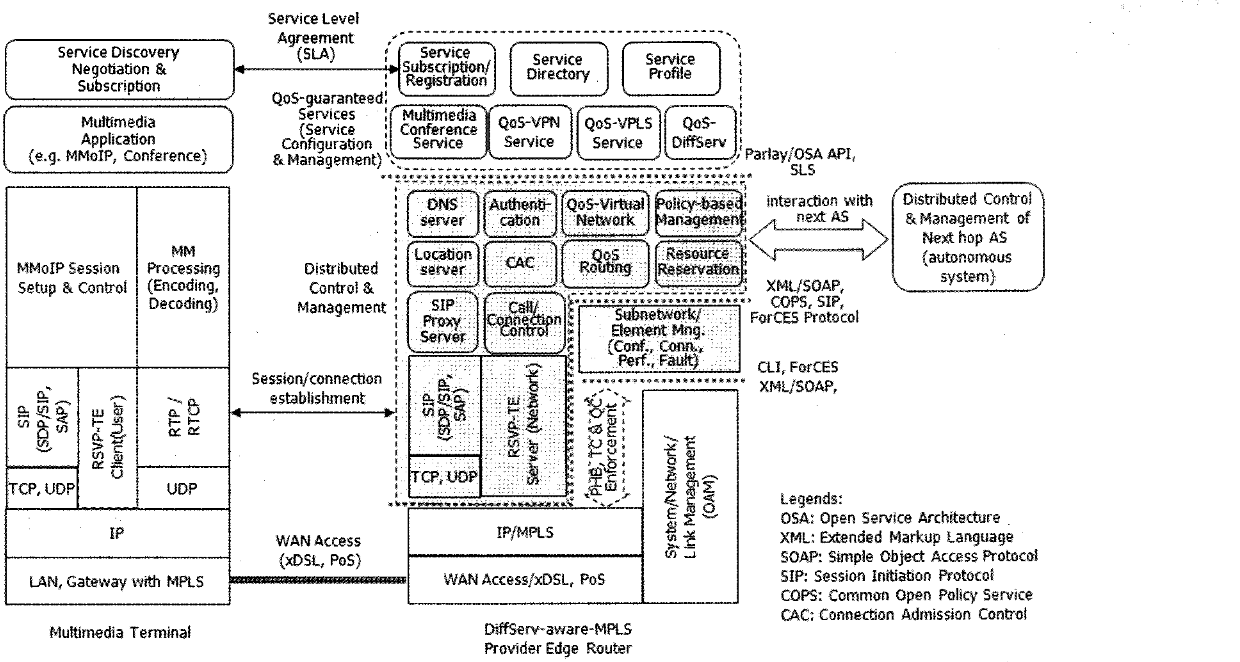

Fig. 2. Functional block diagram and interaction model of session & connection management in QoS-guaranteed multimedia service provisioning.

tual overlay network to configure the predefined paths for different traffic classes.

If a specific application service is agreed via SLA, a session setup and a connection establishment for the media application should be initiated using SIP/SDP and RSVP-TE signaling. SIP/SDP will be used to find the current location of the destination, to determine the availability and capability of the terminal for the requested multimedia application. The SIP proxy server may utilize location server to provide some value-added service based on presence and availability management functions. The sending SIP/SDP user agent will offer the list of possible media types and encoding standards, and the receiving SIP/SDP user agent, considering its processing capacity, will select some acceptable media type and its encoding format, and reply the acceptable list to the sender. Finally, the sending SIP/SDP user agent will determine the selected media type and the encoding format to be used in the multimedia session.

Once a multimedia session description is agreed by the participants, a QoS-guaranteed per-class-type end-to-end connection establishment will be requested through UNI signaling. If the multimedia communication service is using separated connection for each media type, multiple connections should be established for each media-type packet flow. As an end-to-end connection establishment signaling, RSVP-TE can be used in the backbone transit IP/MPLS network. The edge node (router) of ingress provider network should take the role of connection control and management functions for on-demand connection establishments.

When the user's terminal is attached to a contention-based intra network, such as IEEE 802.3 Fast/Gigabit Ethernet or IEEE 802.11 WLAN, or participant's terminal does not support RSVP-TE signaling, the end-to-end connection cannot be established; instead, per-class-type packet flow should be registered and controlled by the connection management function of the ingress provider edge (PE) node with CAC. The customer network management (CNM) system may support the procedure of per-class-type packet flow registration [5]. In the following Sections III-C and III-D, we explain the packet flow management in contention-based intranet for QoS-guaranteed multimedia service provisioning.

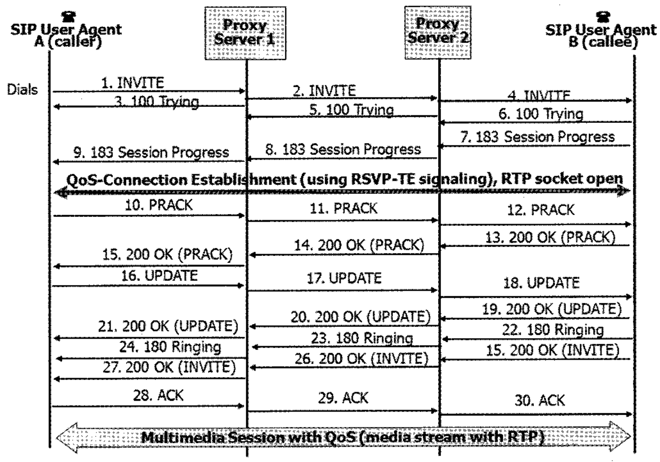

Fig. 3. SIP/SDP messages for QoS-guaranteed multimedia session establishment.

Fig. 3 depicts the SIP/SDP message exchanges for QoS-guaranteed session and connection establishment in realtime multimedia service provisioning [3], [8]. User agent A (caller) invites user agent B (callee) through INVITE request message that is relayed by the SIP proxy server to the destination (callee). When the INIVITE message is created by the caller, the message body includes the information of services that the caller can provide. This information is contained in SDP message that describes the details of media, QoS, bandwidth, and security related attributes. When the callee receives the INVITE request from the caller, the callee checks the requested QoS parameters in SDP message, checks the availability of resource and ca-

Table 1. SIP messages using multimedia session establishment.

SIP Message	Direction	Major Contents
INVITE	A(caller) → B(callee)	Media information of the caller, QoS or security information
100 Trying	A(caller) ← B(callee)	Typically does not contain a *To* tag
183 Session Progress	A(caller) ← B(callee)	Information about the progress of the session (call state)
PRACK	A(caller) → B(callee)	Acknowledges receipt of reliably transported provisional response (1xx)
UPDATE	A(caller) → B(callee)	Used to change media session parameters in early dialogs (before the final response to the initial *INVITE*)
180 Ringing	A(caller) ← B(callee)	QoS or security information or ring tone or animations from user agent server (UAS) to the user agent client (UAC)
200 OK	A(caller) ← B(callee)	Request has completed successfully
ACK	A(caller) → B(callee)	Acknowledgment of final response to *INVITE*

Table 2. Functionality of RSVP-TE messages.

RSVP-TE Message	Description	Major Contents
PATH	Used to set up and maintain reservations	SENDER_TEMPLATE, SENDER_TSPEC, ADSPEC
RESV	Sent in response to PATH messages to set up and maintain reservations	FLOWSPEC, FILTER_SPEC
RESV-Conf	Optionally sent back to the sender of a RESV message to confirm that a given reservation actually got installed	FLOWSPEC, FILTER_SPEC, ERROR_SPEC
PATH-Error	Sent by a recipient of a PATH message who detects an error in that message	ERROR_SPEC, POLICY_DATA
RESV-Error	Sent by a recipient of a RESV message who detects an error in that message	ERROR_SPEC, POLICY_DATA
PATH-Teardown	Analogous to PATH messages, but used to remove reservations from the network	SENDER_TSPEC, ADSPEC
RESV-Teardown	Analogous to RESV messages, but used to remove reservations from the network	FILTER_SPEC

pacity, and sends response message (183 Session Progress response) to the caller. In this response message, the callee specifies the acceptable media types, QoS parameters, and bandwidth & traffic parameters. When the caller receives the 183 message, it checks the parameters that have been selected by callee, and sends a provisional response acknowledgement (PRACK) message to the callee as the answer to the 183 message. When the callee receives the PRACK message, it sends a 200 OK message as the acknowledgement to the the PRACK. Table 1 presents the SIP messages used in multimedia session establishment.

After receiving the 200 OK message, the caller initiates the connection establishment procedure using the network layer signaling, such as RSVP-TE UNI signaling in IP/MPLS subnetwork. Details of the connection establishment procedure in IP/MPLS subnetwork using RSVP-TE UNI/NNI signaling are explained in next section.

When the two connections are successfully established for each direction, then the caller sends an UPDATE message to inform that the multimedia stream may be turned on to deliver the multimedia audio/video data. By receiving the UPDATE message, the called party finally accepts the multimedia call/session by replying a 200 OK (INVITE) message to the caller. The caller sends an ACK message that informs the successfully established call/session. Bidirectional QoS-guaranteed multimedia communication can be provided at this moment. When either the caller or the called party sends a BYE message, then the bidirectional connection will be released, and the call/session will be closed. VOVIDA project provides a good example of SIP/SDP implementation, and VoIP applications [9].

B. Connection Establishment in IP/MPLS Transit Network

RSVP-TE is required as the UNI signaling to request QoS-guaranteed connection establishment [8]. In our implementation of RSVP-TE, we expanded the objects for PATH message and RESV message to contain the traffic parameters (such as peak data rate, peak burst size, committed data rate, committed burst size, and excess burst size) and service class-type that implicitly specifies the QoS parameters (such as end-to-end delay, jitter boundary, packet error ratio, packet loss ratio, service availability, protection mode, and service reliability). Table 2 describes the RSVP-TE messages with their major contents.

To establish a bidirectional QoS-guaranteed connection, a coordinator module of the caller provides the related traffic/QoS parameters to the RSVP-client. These parameters are based on the session parameters which have been negotiated at the previous SIP/SDP INVITE procedure between the caller and the callee via SIP proxy servers. In current implementation of multimedia terminal, as shown in Fig. 4, the MM service coordinator is managing the overall session status and connection establish-

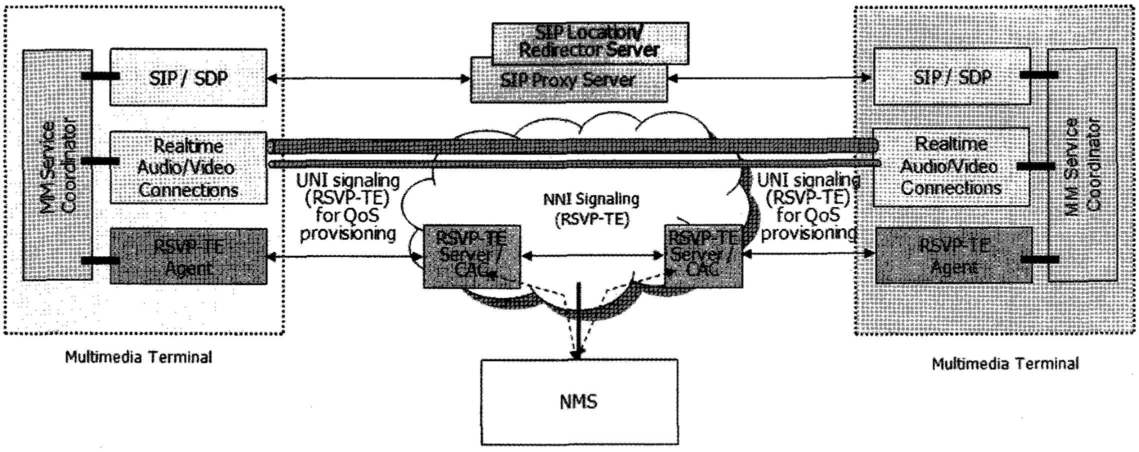

Fig. 4. Proposed QoS-guaranteed multimedia service system.

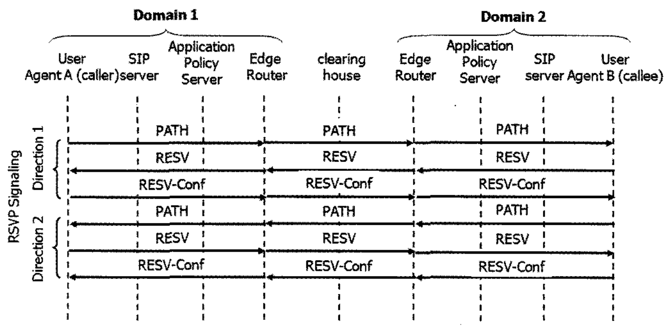

Fig. 5. RSVP-TE signaling procedure to establish TE-LSP in IP/MPLS subnetwork.

ments.

A PATH message is delivered to the destination, and the destination terminal sends a RESV message to request the resource reservation and the connection establishment. Since SIP/SDP has been used to configure the bidirectional session, the PATH/RESV message exchange is executed for each direction. We assume that the QoS-parameters are defined in the SLA/SLS, and TE-LSPs are established among PE-pairs for simplified traffic grooming. A multimedia application may require the multiple QoS-guaranteed connections for multiple media streams.

Since the RSVP-TE is using "soft state" connection management [7], the end user's terminal should periodically exchange the PATH and RESV message to keep the connection in normal operational mode. When the required network resource is changed for the multimedia session, the traffic parameters of the connection must also be updated by the PATH/RESV message exchange. Fig. 5 shows the procedure of MPLS TE-LSP establishment using RSVP-TE signaling.

Another consideration in the UNI & NNI signaling is the scalability of connection status management. Since each QoS-guaranteed connection must be maintained by the provider network, an efficient traffic grooming scheme must be used. In the current implementation, we are using edge-to-edge TE-LSP and node-to-node TE-Link for each QoS class-type based on the virtual overlay networks [8]. In this hierarchical traffic grooming, the ingress edge router is managing the detailed connection status of each QoS-guaranteed connection; the transit routers are not keeping the detailed information for each connection, but maintains the aggregated traffic parameters of each QoS class-type.

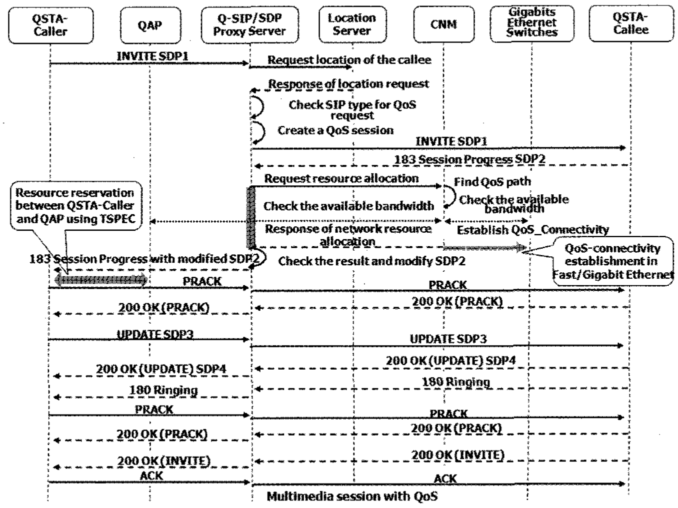

Fig. 6. Interactions between Q-SIP/SDP proxy server and CNM for QoS-connectivity establishment.

C. Resource Reservation and CAC on IEEE 802.3 Fast/Gigabit Ethernet Switch

Nowadays, most broadband end user terminals are attached to the home network or office network via IEEE 802.3 Fast/Gigabit Ethernet or IEEE 802.11 WLAN. In order to provide an end-to-end QoS-guaranteed multimedia services across heterogeneous network environment, as shown in Fig. 1, each subnetwork should be able to provide QoS-guaranteed packet delivery with appropriate resource allocation and CAC.

In IEEE 802.3 Fast/Gigabit Ethernet, the legacy dummy hubs cannot provide guaranteed QoS; while some recent Fast/Gigabit Ethernet switches, such as Cisco Catalyst 3550 and 2950, can provide QoS scheduling and queuing capability with strict priority queue and weighted random early detection (WRED) functions [10], [11]. In Cisco Catalyst 3550, each port has four different output queues, and one of the queues can be configured as a strict priority queue, while the remaining queues are configured as non-strict priority queues and are serviced with the use of weighted round-robin (WRR). The strict priority queue is specially designed for delay/jitter-sensitive traffic, such as voice and realtime video. Since strict priority queue is always emptied first regardless of the status of other queues, it may eventually cause starvation of the other WRR queues.

Since the currently available Gigabit Ethernet switch support neither SIP/SDP session layer signaling nor RSVP-TE network layer signaling, the CNM for home/office intra network must provide the QoS-scheduling and the queue configuration functions with CAC. Fig. 6 depicts the procedure of QoS-connectivity establishment in an intranet with IEEE 802.11e WLAN and Fast/Gigabit Ethernet switch.

When a caller QSTA initiates a session establishment with INVITE message, the Q-SIP/SDP proxy server firstly checks the location of the called party (callee), determines the next hop, and relays the SIP/SDP message. When the callee accepts the request of session establishment by replying 183 Session Progress message, the Q-SIP/SDP proxy server receives it, and sends a resource allocation request message to the CNM that manages the intranet resources. Based on the CAC function, the CNM firstly checks the availability of the requested bandwidth in the Fast/Gigabit Ethernet switch, confirms that the addition of the new QoS-connectivity does not deteriorate the QoS provisioning of the existing flows. When the new QoS flow is admitted, the CNM configures the QoS-scheduler and the strict priority queue.

D. Resource Reservation and CAC in IEEE 802.11e WLAN

Guaranteed QoS provisioning on currently available IEEE 802.11 WLAN products is very limited, since the data delivery on most WLANs is based contention-based medium access control, i.e., CSMA/CA. IEEE 802.11e [12] has been standardized to support QoS provisioning with enhanced channel access mechanism: Enhance DCF channel access (EDCA) and HCF controlled channel access (HCCA). The QoS provisioning on IEEE 802.11e is based on the enhanced distributed channel access by EDCA and the centralized channel access by HCCA. These two channel access functions are managed by a centralized controller called hybrid coordinator (HC) which is a module in the QoS access point (QAP). The IEEE 802.11e EDCA is a contention-based medium access method, and is realized with the introduction of traffic category (TC). The EDCA provides differentiated distributed accesses to the wireless medium with 8 access priorities from stations. The EDCA defines the access category (AC) mechanism that provides support for the priorities at the stations. Each station may have up to 4 ACs (AC_VO, AC_VI, AC_BE, and AC_BK) to support 8 user priorities. One or more user priorities are assigned to one AC.

Even though EDCA provides differentiated access categories, it does not guarantee the QoS parameters of hard realtime applications, i.e., jitter and delay [13]. In order to provide realtime services with guaranteed QoS-parameters, HCCA that has been designed for parameterized QoS support with contention-free polling-based channel access mechanism must be used. The QAP scheduler computes the duration of the polled-transmission opportunity (TXOP) for each QSTA based upon the traffic specification (TSPEC) parameters of an application flow. The scheduler in each QSTA then allocates the TXOP for different traffic stream (TS) queues according to the priority order. In IEEE 802.11e, TSPEC is used to describe the traffic characteristics and the QoS requirements of a TS to and from a station.

In practical QoS-guaranteed multimedia service provisioning on IEEE 802.11e WLAN, there are two problems yet to be solved: i) IEEE 802.11e WLAN AP node and network interface card that supports both EDCA and HCCA are not commercially available, ii) the IEEE 802.11e WLAN AP is limited and does not provide the SIP/SDP session layer signaling and RSVP-TE network layer signaling. So, we have to use the CNM that can manage the resource utilization in WLAN and interacts with SIP proxy server. Similar to the networking scenario in the QoS provisioning in Fast/Gigabit Ethernet, the CNM nodes in the source and destination intranets receive the QoS-connectivity requests from the SIP proxy server, and check the available network resources as CAC operation. If the new QoS flow is accepted, the CNM controls the configuration of the QoS scheduling on appropriate AC in the IEEE 802.11e WLAN AP. The actual QoS-connectivity establishment is requested by the QSTA via IEEE 802.11e MAC protocol with TSPEC parameter. The CNM can manage the overall aggregated throughput of each AC in the intranet, by adjusting the beacon interval, and the proportion of contention-free period (CFP) and contention period (CP).

E. QoS-aware CNM

In order to support QoS-aware service provisioning in Fast/Gigabit Ethernet and IEEE 802.11 WLAN environment, the QoS-aware customer network management (Q-CNM) system should provide QoS-aware resource allocation with CAC function. In this study, we designed and implemented web-based enterprise management (WBEM)-based Q-CNM. The Q-CNM provides QoS-aware resource allocation and CAC in the intranet, configures the details of the per-port queue management and scheduling of Fast/Gigabit Ethernet switch. It also configures the policy-based operational parameters in IEEE 802.11e WLAN. The Q-CNM maintains the detailed operation status of each QoS-aware basic service set (QBSS) in IEEE 802.11e WLAN and Fast/Gigabit Ethernet subnetwork. The details of the functional blocks and implementations of Q-CNM are out of the scope of this paper.

IV. EXPERIMENTAL IMPLEMENTATION AND PERFORMANCE EVALUATION OF QOS PROVISIONING

A. Testbed Network of Converged Heterogeneous Wired & Wireless Network

Fig. 7 depicts the testbed network of converged heterogeneous wired & wireless network for the performance evaluation of the overall QoS provisioning of multimedia realtime services. In order to analyze the performance of QoS provisioning in practical intranet environment, we configured two kinds

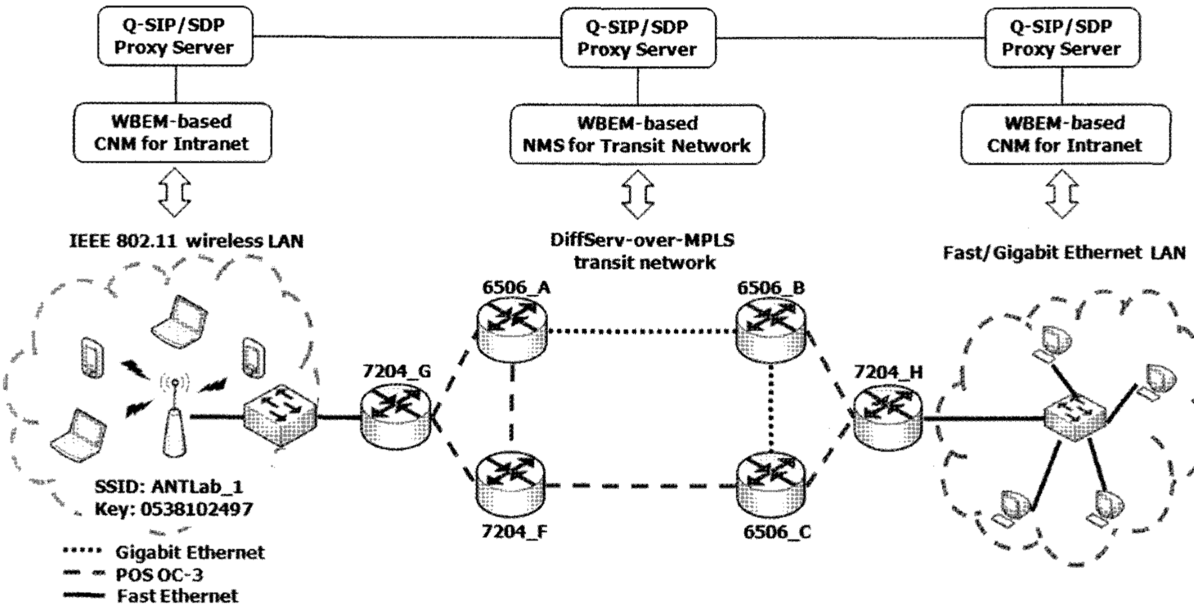

Fig. 7. Testbed network of converged heterogeneous wired & wireless network.

of intranets with commercially available products: i) Wired intranet with Cisco 2950 IEEE 802.3 Fast/Gigabit Ethernet switch, ii) wireless intranet with Cisco Aironet 1242 AG IEEE 802.11e WLAN AP and Cisco Aironet 802.11a/b/g PI21AG network interface card (NIC). The backbone core network is configured by IP/MPLS network with Cisco 7204 and 6505 IP/MPLS routers.

According to the request of a TE-LSP establishment from the Q-SIP/SDP proxy server and the WBEM-based NMS for the transit network, a bidirectional TE-LSP between the boundary edge label edge routers (LERs) in the IP/MPLS backbone network is established using the RSVP-TE signaling of Cisco IP/MPLS routers. The scheduling of TE-LSP (e.g., bandwidth and class-type) is controlled by the NMS with traffic engineering function. As we explained in Section III, the QoS-connectivity establishment in intranets is controlled and managed by the CNM of each intranet.

For performance analysis of the end-to-end QoS provisioning for realtime multimedia services, two kinds of end-to-end video telephony service sessions are configured: i) QoS-guaranteed video telephony session, and ii) best-effort video telephony session. For the QoS-guaranteed video telephony session, an EDCA AC_VI is used in the IEEE 802.11e WLAN, a strict-priority queue is used in the Fast/Gigabit Ethernet switch port, and a TE-LSP is configured in the IP/MPLS transit network. For the best-effort video telephony session, an EDCA_BE is used in IEEE 802.11e WLAN, a WRED queue is used in the Fast/Gigabit Ethernet switch port, and a best effort LSP is configured in the IP/MPLS transit network.

B. Performance Analysis of QoS-provisioning at Each Subnetwork

B.1 QoS Provisioning at IEEE 802.3 Gigabit Ethernet

This section explains the performance evaluation results of the access network. In this scenario, we use two kinds of service categories, best effort and video, to show the QoS-guaranteed service provisioning in the access network. Totally, eight stations are used for both service categories, and each service category uses four stations to generate traffic flow. To provide a QoS-guaranteed service for AC_VI service, we use classification, metering, marking, WRED, priority queuing and priority scheduler in the access network switches. Fig. 8 shows the detailed system configuration architecture. Aggregated 60 Mbps bandwidth at IP-layer is allocated for the AC_VI service in the access network switch. Consequently, AC_VI service can receive the QoS-guaranteed service unless its aggregated data rate does not exceed 60 Mbps.

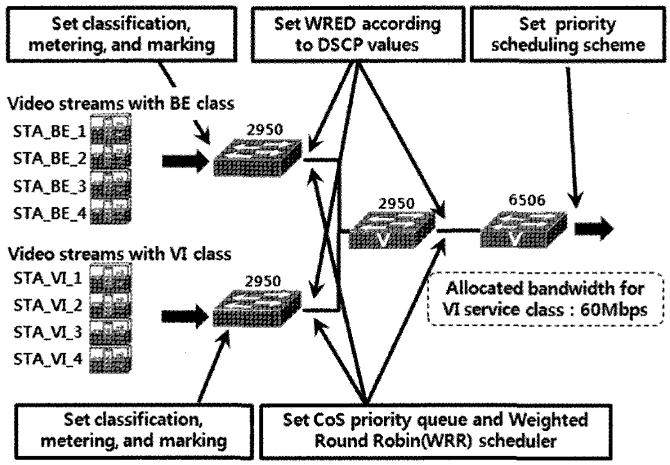

Fig. 8. The detailed system configuration architecture for provisioning QoS-guaranteed service.

Fig. 9(a) depicts the per-flow average delay while increasing per-station traffic load (at IP-layer) continuously. This result shows that the delay of the traffic belonging to AC_VI service is maintained in the range of 3 ms. Fig. 9(b) depicts the aggregated throughput at IP-layer for the same scenario. When the aggregated network load exceeds the link capacity of 100 Mbps, the AC_VI class traffic has higher priority than BE traffic.

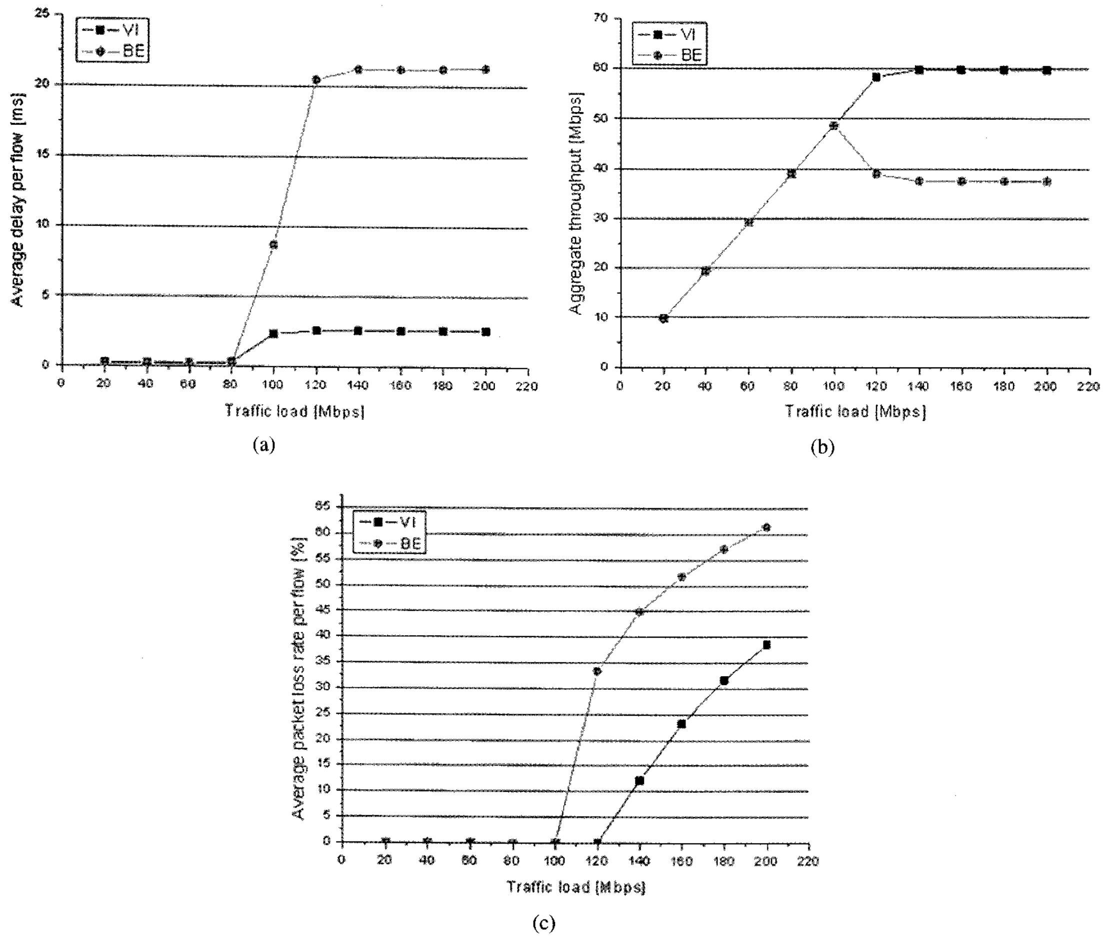

Fig. 9. Traffic load vs. (a) average delay, (b) aggregate throughput, and (c) average packet loss ratio in Gigabits Ethernet network.

However, when the aggregate network rate exceeds 120 Mbps, AC_VI traffic rate exceeds its bandwidth allocation limit of 60 Mbps and does not occupy the transmission portion of BE service anymore. Fig. 9(c) shows the packet loss rate over the aggregate network load. The packet loss rate is also shown that the AC_VI service can receive the QoS-guaranteed service unless the aggregated data rate is within the limitation of allocated network resource which is set to 60 Mbps.

The experiments results shown in Fig. 9 show that the network resource in access network also has to be treated carefully in the QoS provisioning, and the classification, active queue management, scheduling and others, must be carefully used in the access network in order to provide end-to-end QoS-guaranteed service.

B.2 QoS Provisioning at IEEE 802.11e WLAN

The test-bed is configured with IEEE 802.11g WLAN. The request to send (RTS)/clear to send (CTS) and fragmentation mechanisms are not used as in most practical WLAN environment with default settings. In the case of IEEE 802.11e, we classify the QoS-guaranteed service and map the QoS required multimedia services into the AC_VI in IEEE 802.11e EDCA mechanism by using type of service (ToS) field in IP header. The other best effort traffic is processed as AC_BE in IEEE 802.11e EDCA. The QoS-aware traffic and background traffic are generated in the same traffic pattern. The packet size is 1400 bytes, and the packet inter-arrival time is adjusted according to the target throughput. The data rate shown in this paper is measured at IP-layer, not at physical layer.

Fig. 10 depicts the packet loss ratio and the delay in IEEE 802.11e/g WLAN, according to per-station increasing traffic load. Because of the overhead in the IEEE 802.11 MAC and physical layer protocols, the packet loss and delay are increased rapidly when the offered traffic load at IP-layer is increased beyond 23 Mbps. Even though the maximum transmission rate in IEEE 802.11a/g physical layer is 54 Mbps, the typical throughput has been reported as around 25 Mbps [14]. In Fig. 10(a), we can find that the practical aggregated throughput at IP-layer in IEEE 802.11g WLAN is around 30 Mbps with reasonable packet loss ratio and packet delay. And, when the traffic load is increased beyond 23 Mbps, the packet loss and packet delay increase rapidly; so we need careful design of CAC and EDCA channel allocation scheme.

Fig. 11 depicts the performance of QoS provisioning with differentiated IEEE 802.11e EDCA access categories. In this experiment, two groups of 4 STAs are generating traffic in different ACs: 4 STAs are generating AC_VI traffic, while the other 4 STAs are generating AC_BE traffic. The amounts of the generated traffic load from each group are configured to be same.

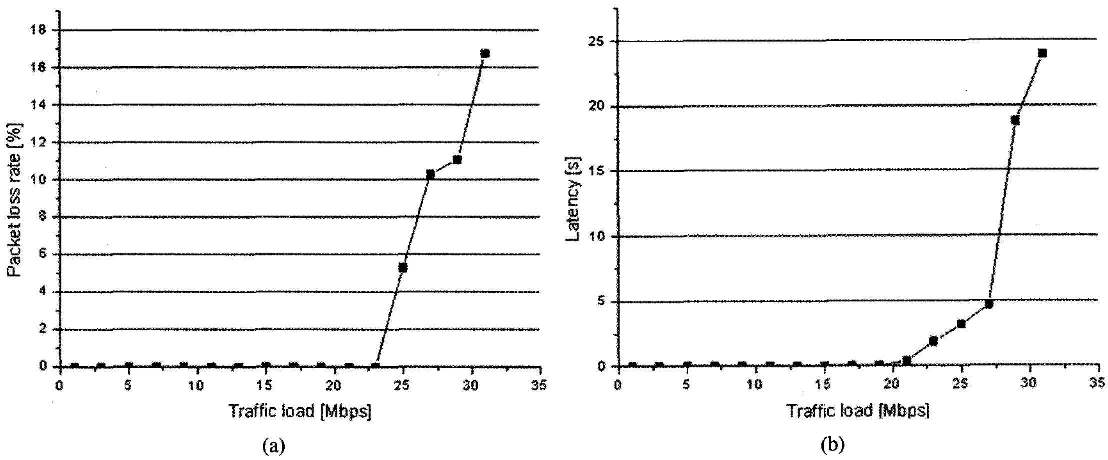

Fig. 10. Traffic load vs. (a) aggregated packet loss ratio, and (b) average delay in IEEE 802.11e/g WLAN.

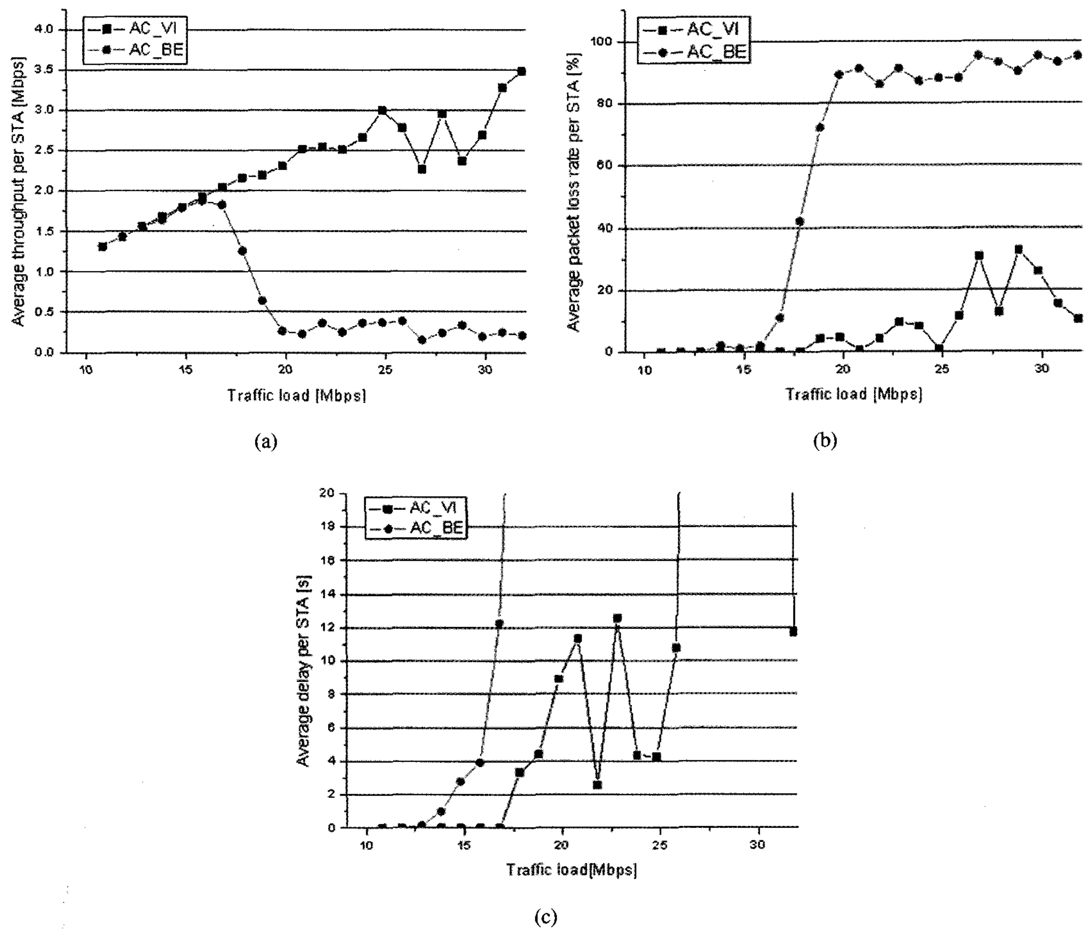

Fig. 11. Traffic load vs. (a) per-station throughput, (b) packet loss ratio, and (c) average delay in IEEE 802.11e/g WLAN.

From Figs. 11(a) and (b), we can see that the AC_VI is relatively protected from AC_BE when the aggregated traffic load is increased beyond 16 Mbps. The throughput of AC_VI is continuously increasing beyond 20 Mbps according to the increased offered traffic load. From Fig. 11(c), however, we can find that average delay is increased rapidly when the offered traffic load is more than 15 Mbps. The average delay of AC_VI is also increased beyond 2 seconds, when the aggregated offered load is more than 17 Mbps.

From the series of experiments shown in Figs. 10 and 11, we can see that the QoS-provisioning performance with currently available commercial IEEE 802.11e WLAN is possible with differentiated ACs and CAC functions. In order to provide QoS-guaranteed realtime services, the amount of traffic load should be carefully controlled, and EDCA should be employed that can provide the differentiated access category. The

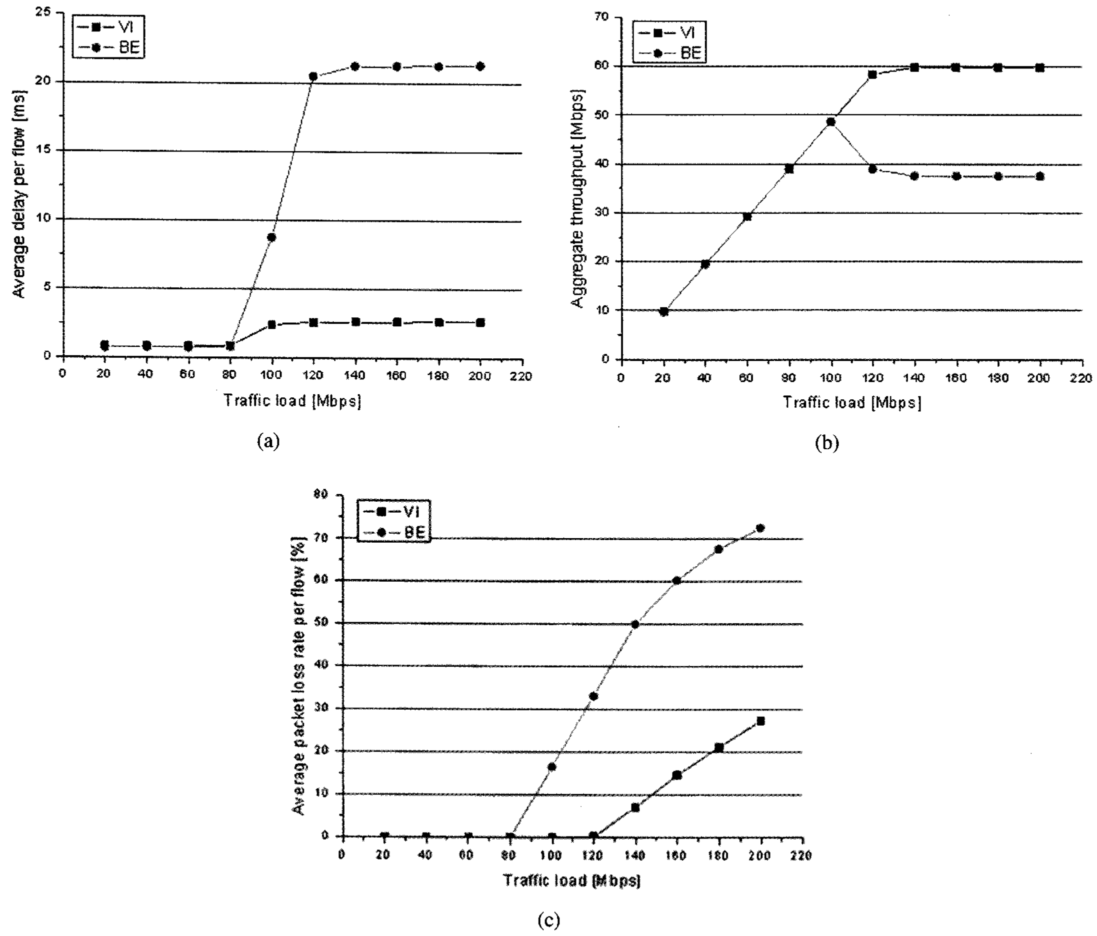

Fig. 12. Traffic load vs. (a) per-flow delay, (b) aggregate throughput, and (c) per-flow packet loss ratio in IP/MPLS transit network.

EDCA, however, cannot guarantee QoS if the offered traffic load is more than a certain threshold, as we can see in Fig. 11(c). Especially, the average delay is increased rapidly beyond 2 seconds, while the end-to-end delay of realtime multimedia services is required to be less than 100 ms for realtime conversational applications [15]. In order to control the overall aggregated traffic load in a QBSS of IEEE 802.11e WLAN, the CAC mechanism must be provided in the MAC layer or in the intranet Q-CNM that controls the operational configuration of IEEE 802.11e QAP. For QoS-guaranteed realtime multimedia service provisioning, the HCCA with centralized polling mechanism should be provided.

B.3 QoS Provisioning at IP/MPLS Backbone Network

This section analyzes the performance of QoS provisioning in the IP/MPLS backbone network. The aggregate traffic load increases continuously and is evenly distributed among 8 stations. In the same way as in previous scenarios, 4 stations are assigned for each category. We configured two MPLS TE-LSPs for AC_VI and AC_BE with allocated bandwidth limit of 60 Mbps each. The TE-LSP for AC_VI is configured with higher priority than the TE-LSP for AC_BE.

Fig. 12 shows how the QoS provisioning is supported for AC_VI and AC_BE traffic classes in IP/MPLS backbone network. When the aggregated traffic load in IP-layer is less than 80 Mbps, both AC_VI and AC_BE are receiving good QoS level. When the aggregate traffic load is increased beyond 120 Mbps, however, the delay and packet loss of AC_BE traffic is increased gradually, while the QoS level of the AC_VI is continuously maintained unless the traffic load is not increased beyond 60 Mbps which is the allocated bandwidth limit of AC_VI for guaranteed-QoS provisioning.

C. *Performance Analysis of End-to-End QoS Provisioning*

Fig. 13 depicts the end-to-end performance analysis of QoS-provisioning on the testbed network with IEEE 802.11e WLAN, IP/MPLS transit network, and Fast/Gigabit Ethernet. The end-to-end realtime application is video conference with H.323 video coding at 10 frames per second, 320×240 pixels display size, and the target average data rate of 384 Kbps at IP-layer. In order to compare the end-to-end quality of video conference according to the service provisioning scenario, we measured the end-to-end QoS performances of two networking scenarios: QoS-aware transport networking, and best-effort transport networking. In the QoS-aware transport networking, the IEEE 802.11e EDCA AC_VI is used in the WLAN subnetwork, and the strict-priority queue is used in Fast/Gigabit Ethernet switch. In the best-effort transport networking, the IEEE 802.11e EDCA AC_BE is used in the WLAN subnetwork, while the non-strict priority WRED queue is used in Fast/Gigabit Ethernet switch.

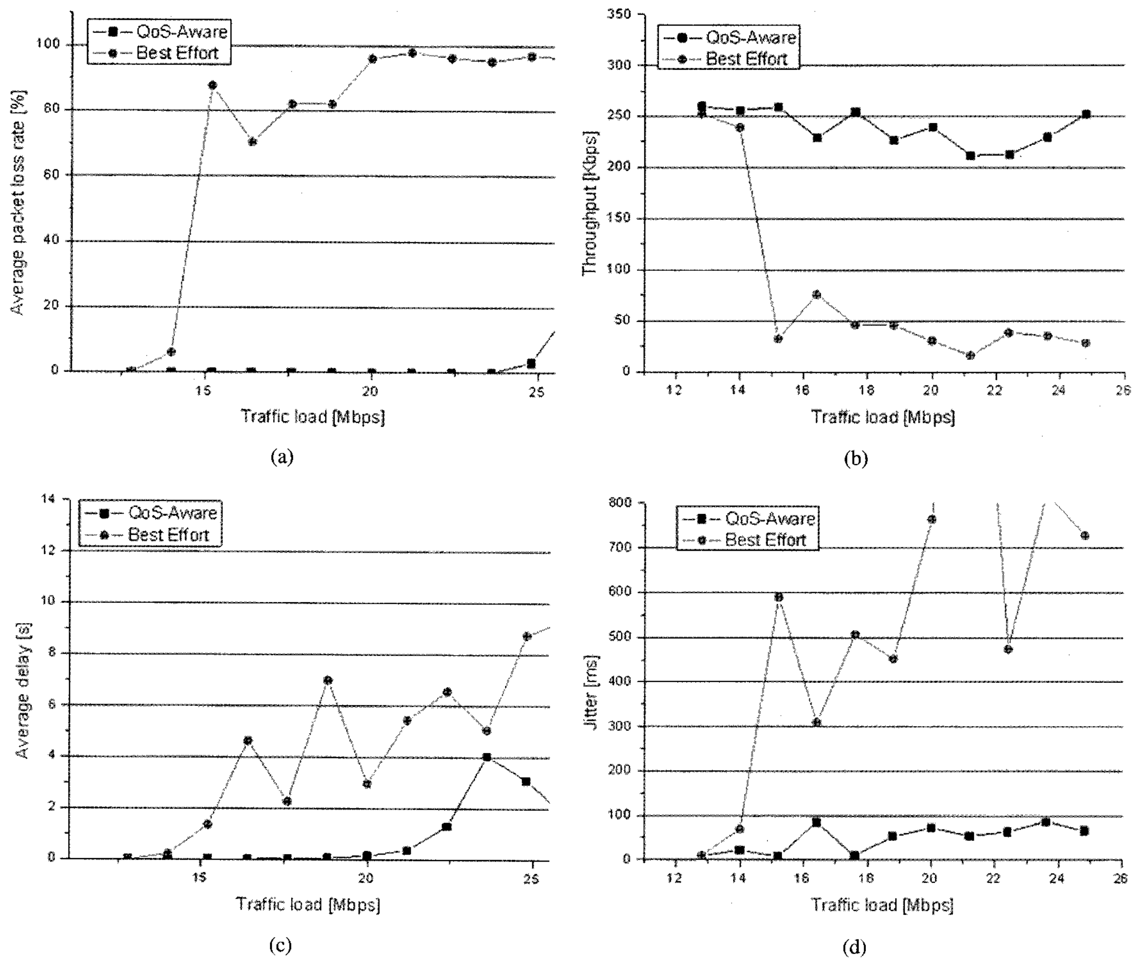

Fig. 13. Traffic load vs. (a) packet loss, (b) throughput, (c) delay, and (d) jitter.

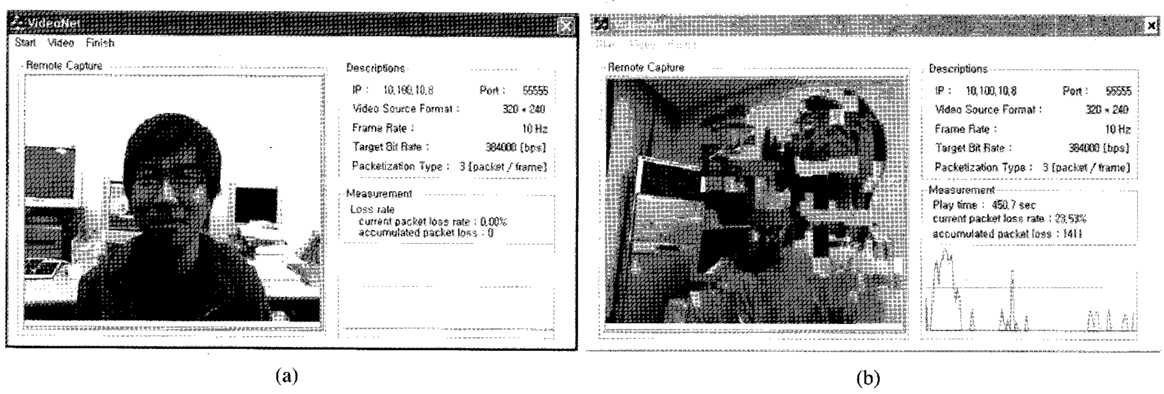

Fig. 14. Capture images of video conference: (a) QoS-guaranteed and (b) best effort.

The backbone transit network is configured to provide TE-LSPs for both cases.

From Fig. 13, we can see that the end-to-end packet delivery of QoS-aware video traffic is guaranteed until 23 Mbps, while the best-effort video traffic is suffering from massive packet loss. The average delay of QoS-aware video traffic is guaranteed to be less than 100 ms until the aggregated traffic at IP-layer is less than 20 Mbps, while the end-to-end delay of best-effort video traffic is suffering longer delay which is more than 2 seconds. Since the currently available IEEE 802.11e WLAN AP and NIC do not support HCCA mechanism, QoS-guaranteed realtime multimedia service provisioning is not possible if the aggregated traffic load is beyond 23 Mbps which is around 60% of the practically available throughput of IEEE 802.11g WLAN. From these experiments, we found that the QoS-aware connectivity provisioning in WLAN is the most critical factor in the end-to-end QoS-guaranteed realtime service provisioning in broadband convergence network with heterogeneous networking technologies. Fig. 14 shows two examples of captured images in the experiment: (a) QoS-guaranteed and (b) best-effort video conferences.

V. CONCLUSION

In this paper, we proposed a session and connection management architecture for the QoS-guaranteed realtime multimedia service provisioning on BcN where various contention-based intranets, e.g., IEEE 802.11e WLAN and Fast/Gigabit Ethernet, are included. Detailed interaction scenarios among Q-SIP/SDP, intranet CNM, IEEE 802.11e WLAN AP, and Fast/Gigabit Ethernet switch are proposed. Based on a small scale BcN test-bed network, the performances of the QoS provisioning for realtime video conference have been analyzed at the IEEE 802.11e-based WLAN intranet, Fast/Gigabit Ethernet-based intranet, and IP/MPLS transit network. Also, the end-to-end QoS-performance has been analyzed.

We performed a series of experiments to analyze the performance of QoS provisioning in wired & wireless intranet subnetwork, access loop, and IP/MPLS-based transit network. The QoS-guaranteed service provisioning in IP/MPLS transit network is relatively easy to configure, since the transit networks are mostly well-equipped with network-layer signaling, such as MPLS RSVP-TE, and traffic engineering with TE-LSPs. The access loop, such as VDSL is also not a critical bottleneck, since most wired access loop technologies are expected to provide more than 25 Mbps without big problem. The QoS-aware realtime service provisioning in Fast/Gigabit Ethernet can be easily configured by priority-based scheduling and differentiated queuing at each port.

From these experiments, we found that the QoS-aware connectivity provisioning in WLAN is the most critical factor in the near future end-to-end QoS-guaranteed realtime service provisioning in broadband convergence network with heterogeneous networking technologies. Even in the IEE 802.11g WLAN where the maximum physical layer transmission rate is 54 Mbps, the practical limitation of QoS-aware service provisioning with IEEE 802.11e EDCA mechanism is around 23 Mbps at IP packet layer. In order to provide QoS-guaranteed multimedia service provisioning in IEEE 802.11e WLAN, the HCCA/DLS scheme must be implemented.

Another key factor in QoS-guaranteed realtime multimedia service provisioning is the end-to-end Q-SIP/SDP signaling with tight cooperation with the network-layer signaling and the resource management function with CAC, in each subnetwork. In this paper we proposed the functional architecture of this tight interaction between the session-layer and the network-layer. As future work, we are developing the IEEE 802.11e distributed CAC scheme and channel allocation algorithm to optimize the resource utilization, while providing the end-to-end QoS provisioning according to the user's request.

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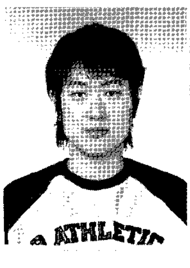

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