Session & Connection Management with SIP and RSVP-TE for QoS-guaranteed Multimedia Service Provisioning

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Abstract

In the realtime multimedia communication, various multimedia terminals with different processing capabilities (i.e. desktop PC vs. PDA) must be considered. In the transit networking, the on-demand multimedia connection requests must be carefully controlled with connection admission control (CAC) and hierarchical traffic grooming function for guaranteed QoS provisioning. In order to provide QoS-guaranteed realtime multimedia services on IP-based networks, tightly integrated session and connection management is essential. SIP and RSVP-TE have been proposed as practical implementation solutions for the session signaling and the network connection managements.

The current research works on the QoS-guaranteed realtime multimedia service provisioning have not studied the detailed integration of SIP, RSVP-TE, CAC and hierarchical traffic grooming of transport networks. Most of the approaches just use SIP/SDP to establish a session on the best-effort Internet without guaranteed QoS & bandwidth reservation. In this paper, we analyze the detailed operations of SIP and RSVP-TE for QoS-guaranteed multimedia service provisioning. The detailed specification of QoS & traffic parameters using SDP is explained, and the handshaking procedure is proposed for offer-and-selection model for the choice of proper multimedia service type for various terminal capabilities. We also propose a multimedia service terminal platform and hierarchical traffic grooming for QoS-guaranteed service provisioning. The overall performance of the QoS-guaranteed multimedia service is analyzed.

Introduction

- Signaling for QoS-guaranteed realtime multimedia service provisioning
  - End-to-end session initiation signaling to establish & configure a multimedia session
    - Different terminal capabilities: Desktop PC, Notebook, PDA, Cellular Phone
    - types of media, encoding / decoding format: video, audio, text, etc.
    - required bandwidth, QoS parameters
  - UNI/NNI signaling to establish QoS & bandwidth-guaranteed virtual connection for media packet flow
    - on-demand request for connection establishment, modification, release
    - two possible connection types :
      - a single connection with multiple packet flows of different media streams
      - multi-connections for each media data stream
  - Connection Admission Control (CAC)
    - for guaranteed QoS provisioning, every packet flow must be controlled

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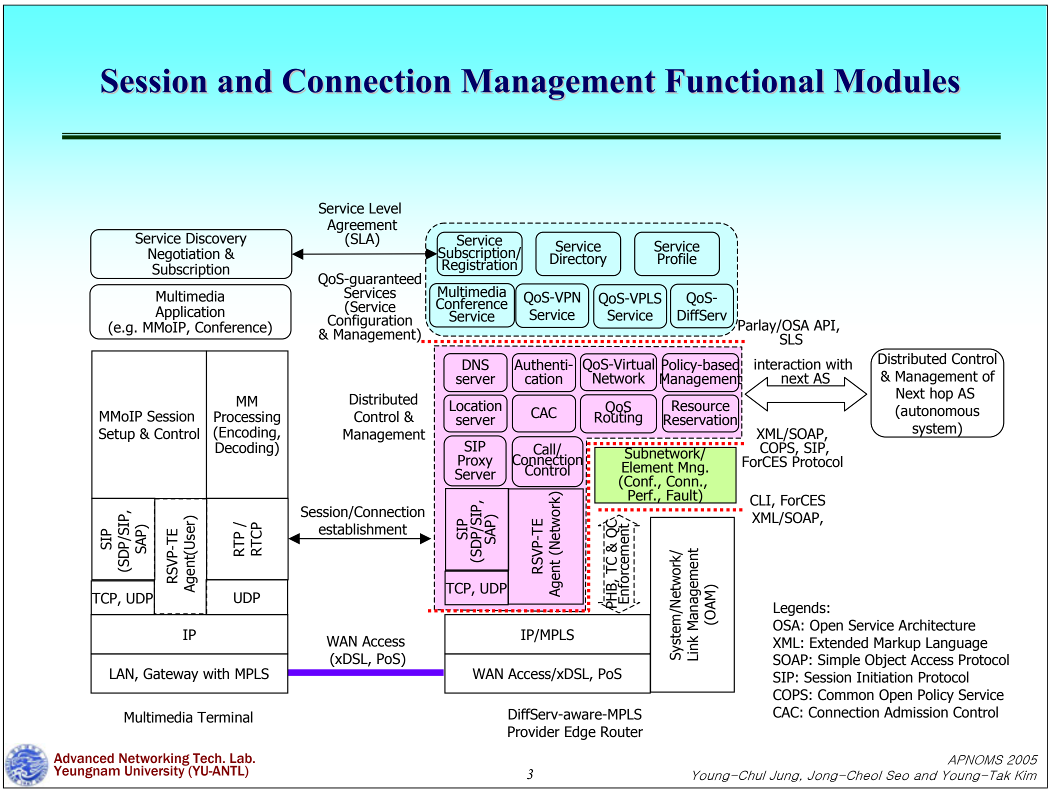
In order to provide QoS-guaranteed realtime multimedia services on IP-based network, two kinds of end-to-end signaling functions are required: (i) end-to-end signaling to establish a session, and (ii) UNI/NNI signaling to establish QoS and bandwidth-guaranteed connection for data flow. SIP/SDP (Session Description Protocol) [1,2,3] has been developed for the session initialization, while RSVP-TE [4,5] has been developed as the UNI and NNI signaling to establish QoS-guaranteed connection. In the realtime multimedia communication, various multimedia terminals with different processing capabilities (i.e. desktop PC, PDA and cellular phone) must be considered. In the transit networking, the on-demand multimedia connection requests must be carefully controlled with connection admission control (CAC) and hierarchical traffic grooming function for guaranteed QoS provisioning.

The current research works on the QoS-guaranteed realtime multimedia service provisioning have not studied the detailed integration of SIP, RSVP-TE, CAC and hierarchical traffic grooming. Most of the approaches just use SIP/SDP to establish a session on the best-effort Internet without guaranteed QoS & bandwidth reservation.

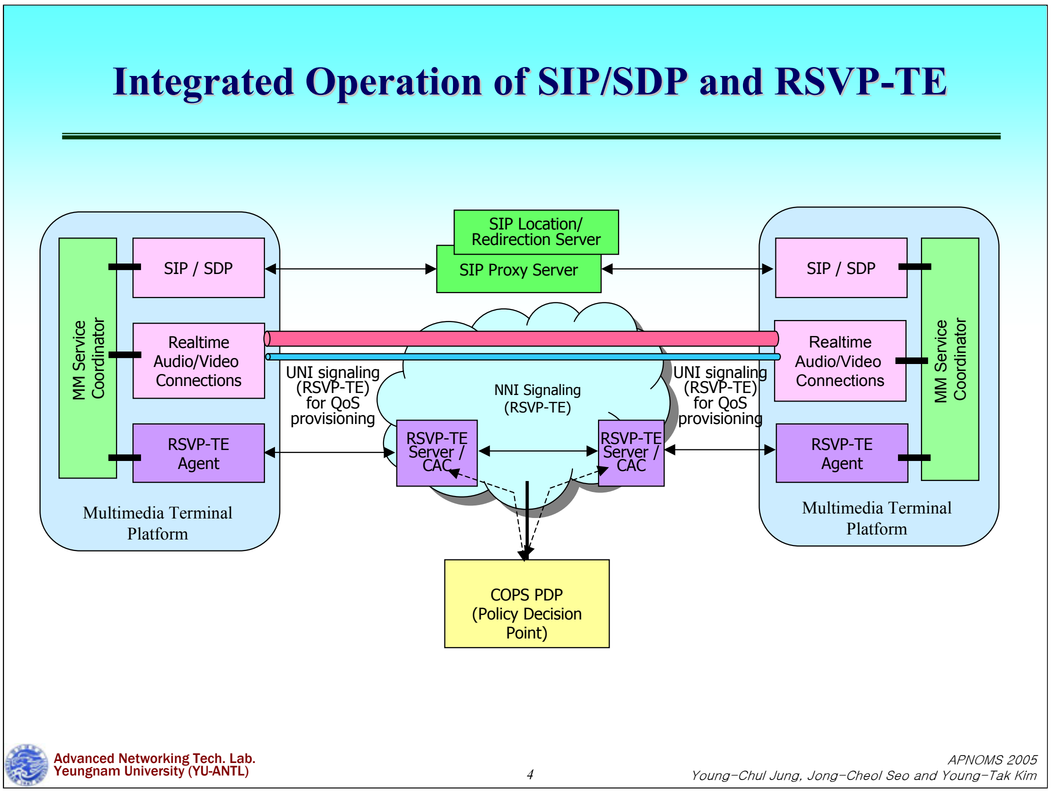
In this paper, we analyze the detailed operations of SIP/SDP, RSVP-TE, CAC and hierarchical traffic grooming for QoS-guaranteed multimedia service provisioning. The detailed specification of QoS & traffic parameters using SDP is explained, and the handshaking procedure is proposed for offer-and-selection model for the choice of proper multimedia service type for various terminal capabilities. We implement a multimedia service terminal platform, and a hierarchical traffic grooming in the IP/MPLS transit network for QoS-guaranteed service provisioning.



The end user will firstly discover and subscribe the required service from service directory of the network operator who provides multimedia service profiles. The subscription will define the SLA (service level agreement) of the service, and SLS (service level specification) / TLS (traffic level specification) 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 [6]. The scalability can be achieved by using virtual overlay network to configure the pre-defined paths for different traffic classes. For the service provider who is configuring service provisioning networks based on the transport network, open service architecture (OSA) based API (application programming interface) must be provided by the network operator.

For an end-to-end multimedia service invocation, the session setup & connection establishment for the multimedia application must be initiated using SIP/SDP and RSVP-TE signaling. SIP and SDP will be used to find the location of destination, to determine the availability and capability of the terminal. 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 the acceptable media type and its encoding format, and reply the acceptable list. Finally the sending SIP/SDP user agent will determine the selected media types and encoding format to be used for the multimedia service.

Once session configuration is agreed by both participants, QoS-guaranteed per-class-type end-to-end connection (or packet flow) establishment will be initiated using UNI signaling. If the multimedia communication service is using separated connection for each media type, multiple connections should be established for each media type. As end-to-end connection establishment signaling, RSVP-TE may be used in IP/MPLS network. The edge IP/MPLS router of ingress provider network will contain connection control and management functions for on-demand connection establishments.

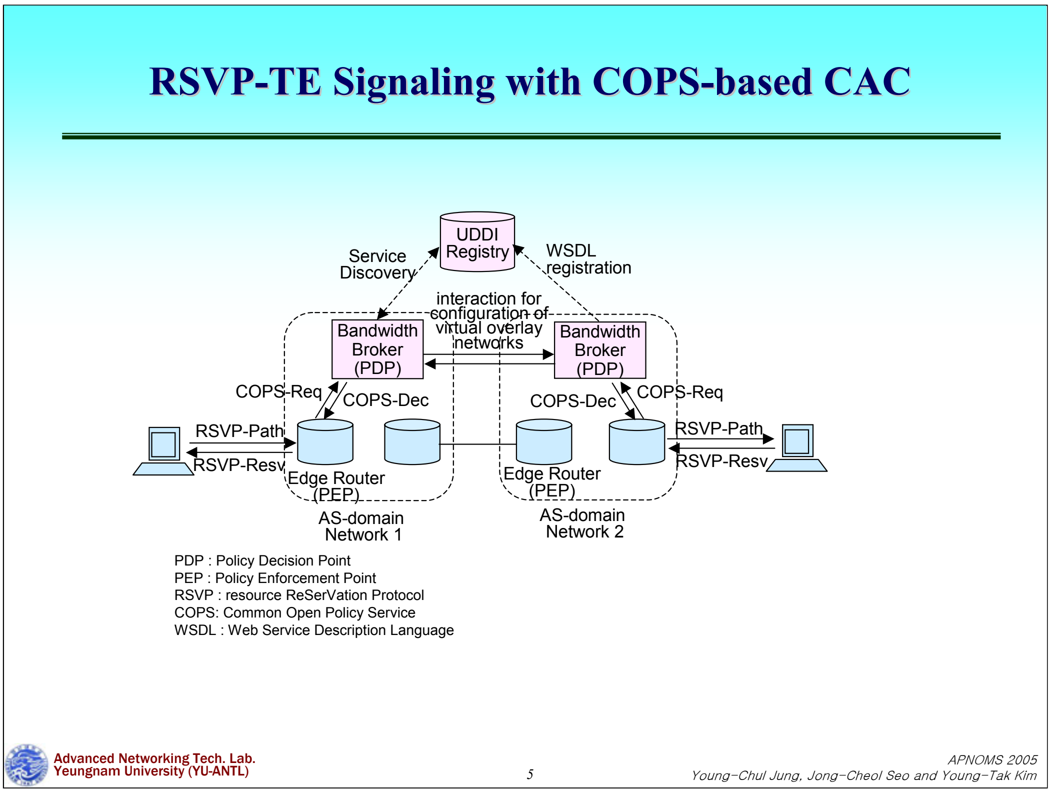


SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls. In order to establish multimedia communication session, SIP/SDP messages must be exchanged between session participants to check out the participant capability, media type negotiation and format type. SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility - users can maintain a single externally visible identifier regardless of their network location [2].

SIP is not a vertically integrated communications system. SIP is rather a component that can be used with other IETF protocols to build complete multimedia architecture. SIP does not provide services. Rather, SIP provides primitives that can be used to implement different services. For instance, SIP can locate a user and deliver an opaque object to his current location. If this primitive is used to deliver a session description written in SDP, for instance, the endpoints can agree on the parameters of a session.

SIP does not offer conference control services such as flow control or voting, and does not prescribe how a conference is to be managed. SIP can be used to initiate a session that uses some other conference control protocol. Since SIP messages and the sessions they establish can pass through entirely different networks, SIP cannot, and does not, provide any kind of network resource reservation capabilities. SDP was proposed to convey information about media streams in multimedia sessions to permit the recipients of a session description to take part in the session [3]. Session description information, such as information about the bandwidth to be used by the session, type of media, time(s) the session is active, transport protocol (RTP/UDP/IP, H.320, etc), the format of the media (H.261 video, MPEG video, etc) can be specified based on the SDP.

After the offering-and-selection of available multimedia session parameters, QoS-quaranteed connection for the present session should be established between session participants. For that purpose RSVP-TE (Resource Reservation Protocol with extension of Traffic Engineering) will be used. Since multimedia communication session will be fully determined by SIP/SDP, the connection establishment based on the required connection parameters will be claimed by UNI signaling function. The multi-media coordinator must provide RSVP-TE client function, while the ingress edge IP/MPLS 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 bi-directional path between user terminal and ingress router[6].

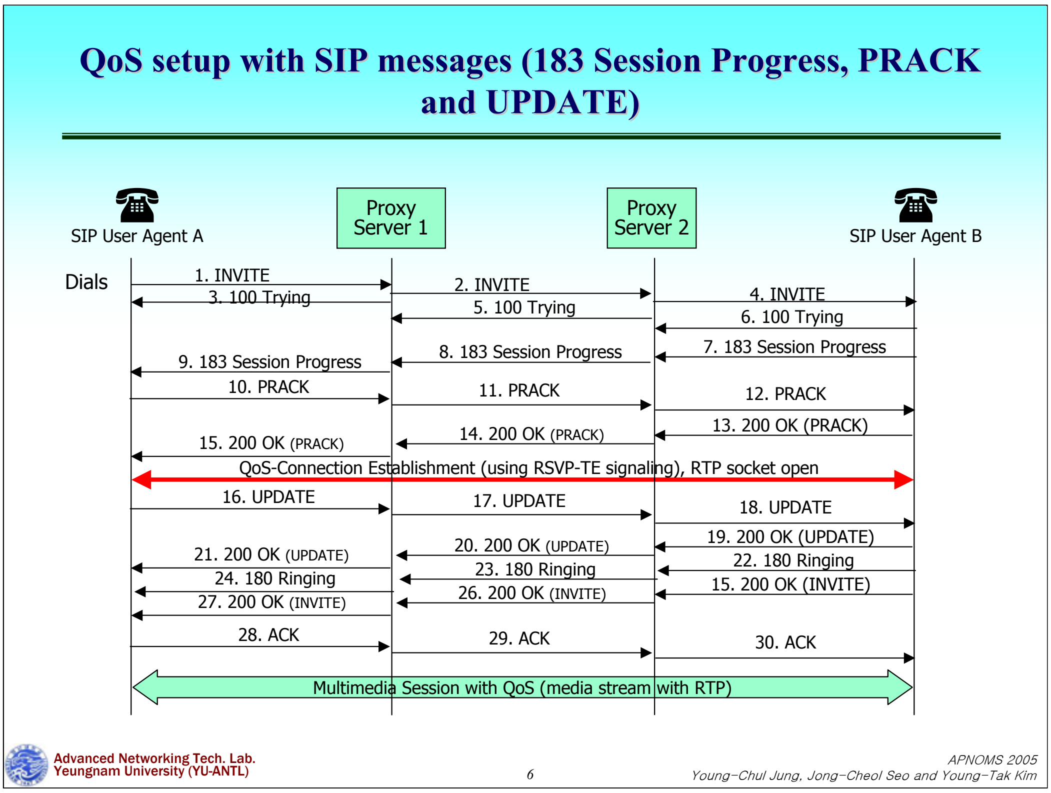


After determination of QoS & traffic parameters for a multimedia session, QoS guaranteed per-class-type connections for the session must be established among the participant terminals. In IP/MPLS network, connection establishment is accomplished by UNI (user-interface-interface) and NNI (network node interface) signaling. For UNI signaling between user terminal and ingress edge router, RSVP-TE can be used to carry the connection request. In order to support per-class-type DiffServ provisioning, RSVP-TE must provide traffic engineering extensions so as to deliver the traffic & QoS parameters

The RSVP protocol is used by a host to request specific qualities of service from the network for particular application data streams or flows. RSVP is also used by routers to deliver QoS requests to all nodes along the path(s) of the flows and to establish and maintain state to provide the requested service. RSVP requests will generally result in resources being reserved in each node along the data path[4]. In order to support traffic engineering, RSVP-TE expands the definition of traffic flows to include label switched paths, and makes the actual resource reservation optional. With this flexibility, routers can use RSVP-TE to distribute labels even when the paths do not need reserved network resources[5].

The ingress provider edge (PE) node is providing COPS PEP (policy enforcement point) role where CAC (connection admission control) and differentiated per-class-type packet processing are implemented. COPS-based CAC can provide various flexibilities in policy-dependent traffic engineering criterions, such as link utilization level, excess traffic handling, weight-based packet scheduling, and time-dependent admission policy. As PDP (policy decision point), a centralized server is implemented with bandwidth broker (BB) function. In each AS-domain network, a bandwidth broker centrally collects network resource, configures virtual overlay networks for QoS class-type, and balances network loads. The above figure shows the interactions of COPS, RSVP-TE, bandwidth broker across multiple AS domain networks.

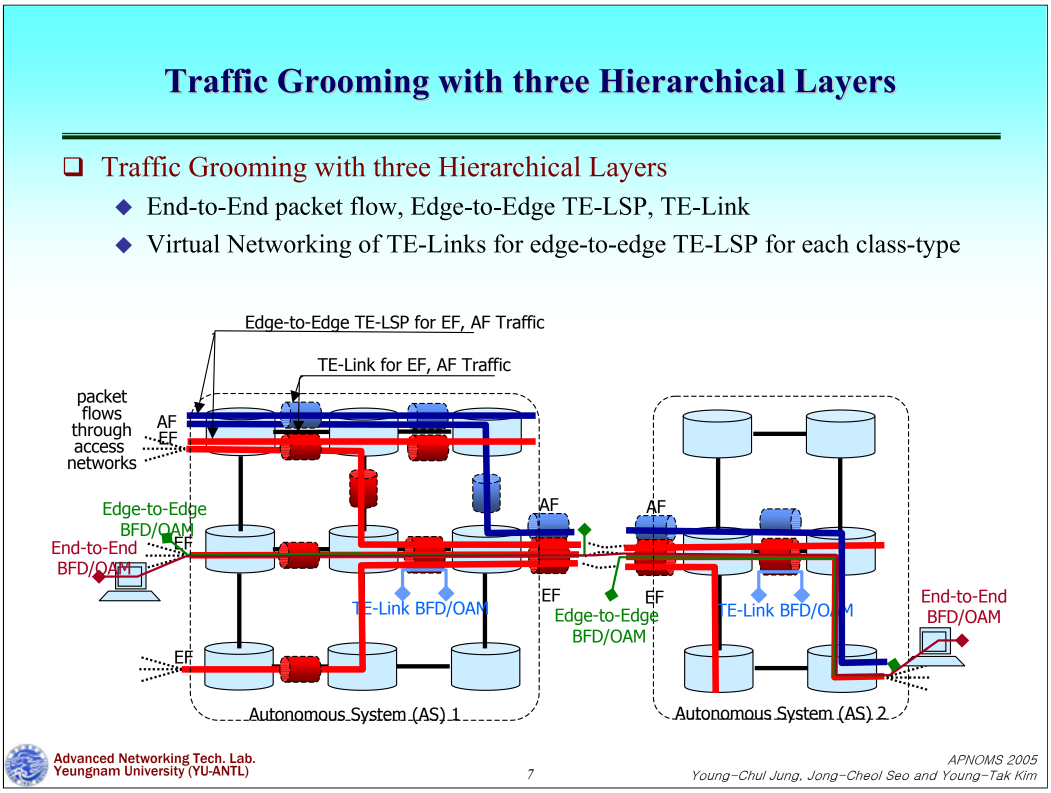
Bandwidth broker also discovers the QoS class-types that other AS-domain networks can provide, through web-service architecture based network management [8]; each AS-domain network advertise the available QoS class-type virtual overlay network services that it provides to others, and the reachability to other AS-domains. Based on the discovered QoS class-types of AS-domain networks, the bandwidth broker finds the CSPF (constraint-based shortest path first) route of end-to-end QoS-guaranteed connections across inter-AS networking.



For QoS-guaranteed realtime multimedia DiffServ provisioning, the SIP/SDP and RSVP-TE signaling modules are implemented in user multimedia terminals. This figure shows the message exchanges in SIP/SDP to configure multimedia session via SIP proxy servers. The media types and related attributes are offered by INVITE message, and accepted media list are acknowledged by Session Progress message.

The caller acknowledges the multimedia type and attribute by sending PRACK (provisional Acknowledge) message. When the called party receives PRACK reply, it sends 200 OK message for the PRACK reply, and the multimedia service coordinator in the user terminal starts to establish traffic & QoS guaranteed connection from the called party to the caller using the RSVP-TE signaling. The caller also starts connection establishment when it receives the 200 OK message from the called party.

When the two connections are successfully established for each direction, then the caller is sending UPDATE message to inform that the multimedia stream may be turned on to deliver multimedia audio/video data. By receiving the UPDATE message, the called party finally accepts the multimedia call / session by replying 200 OK (INVITE) message to the caller. The caller sends ACK message that informs the successfully established call/session. Bidirectional QoS-guaranteed multimedia communication can be provided at this moment. When either one of the caller / called party sends 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 [7].

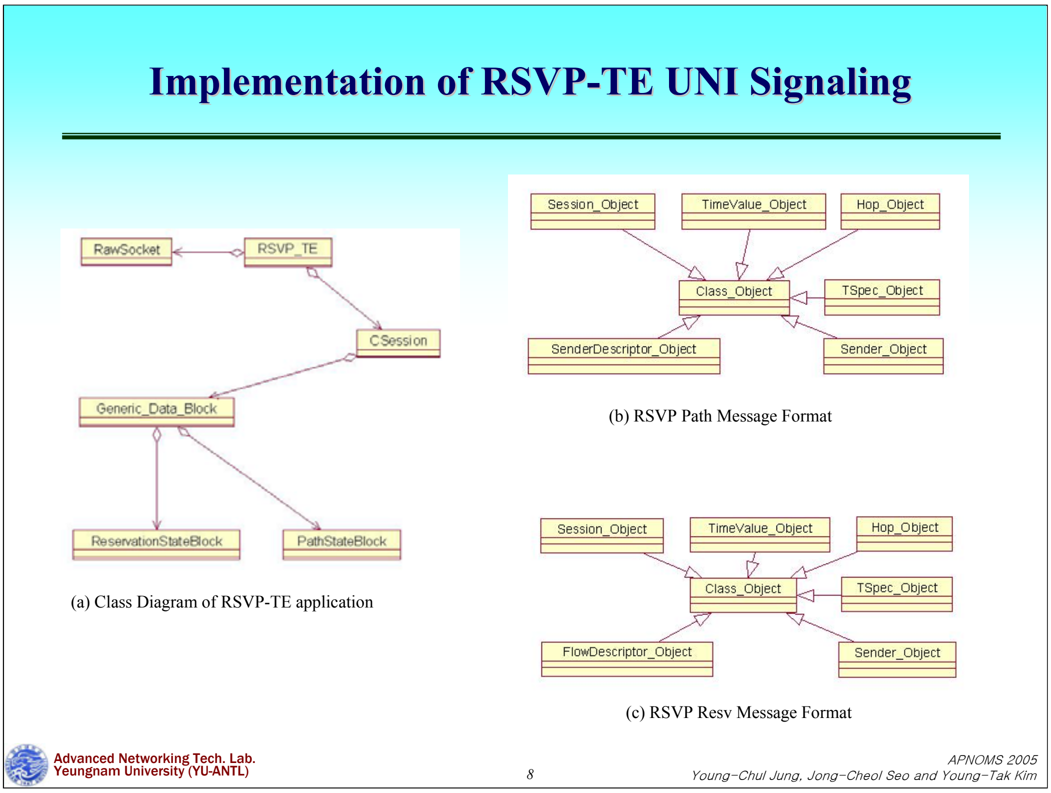


The configuration of per-class-type virtual overlay networks across multiple AS domain network is very important to support QoS-guaranteed DiffServ efficiently through multiple AS domain networks without scalability problem. In order to configure virtual networks for DiffServ class-types, the NMS establishes edge-to-edge TE-LSP in two-level hierarchy: (i) establishment of ASBR (autonomous system boundary router)-to-ASBR trunk TE-LSP as inter-AS transit tunnels, and (ii) establishment of edge-to-edge QoS-guaranteed TE-LSP through the transit tunnels among ASBRs. In order to establish QoS-guaranteed ASBR-to-ASBR transit TE-LSP, the network resource availability and traffic engineering parameters of each AS domain network must be collected. Current BGP (border gateway protocol), unfortunately, only provides reachability & route information, and does not provide traffic & QoS information of the route.

In order to guarantee the pre-configured QoS for DiffServ-over-MPLS, the virtual overlay network for the class-type must be continuously monitored, and the available bandwidth and edge-to-edge packet transfer delay must be analyzed. This slide shows the traffic grooming with hierarchical MPLS TE-LSP label stacking, and their associated OAM (Operation, Administration and Maintenance) functions. BFD (bidirectional forwarding detection) can be used for the OAM function for the edge-to-edge OAM and TE-Link OAM.

Each network interface module can implement the BFD function for edge-to-edge TE-LSP for each class-type, and for TE-Link for the aggregation of TE-LSPs of a class-type between adjacent IP/MPLS routers. BFD for each TE-LSP and TE-Link will send periodic monitoring packet with time stamp to measure the packet transfer delay, jitter (delay variation), and packet error or loss rate.

If RSVP-TE can be used as the NNI signaling on DiffServ-over-MPLS network, the inter-AS domain connection establishment will be much easier and fast. The RSVP-TE based NNI signaling, however, has not been fully implemented to be inter-operable among different AS networks. As an alternative solution, Web-service architecture can be used to implement the interactions among NMSes for per-class-type virtual overlay networking across multiple AS domain networks [8].



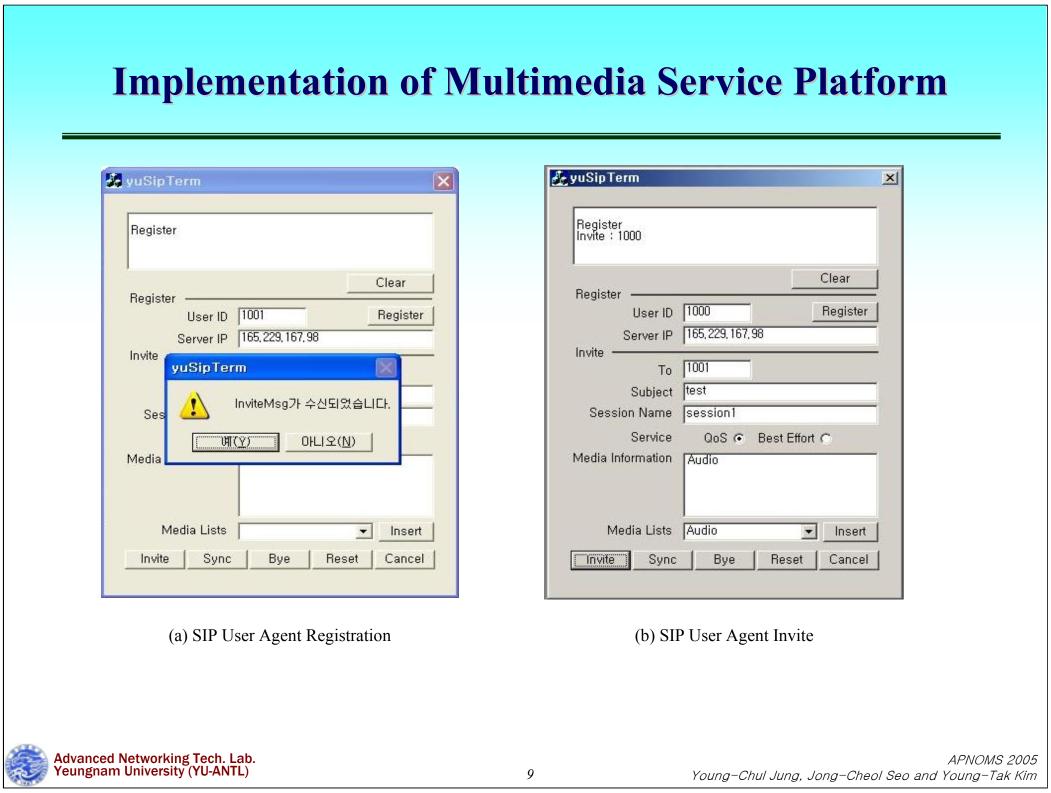
We have implemented the proposed architecture using SIP/SDP, RSVP-TE & CAC. By using this architecture, we can have the QoS guaranteed end-to-end service, even there is change in the network condition such as traffic congestion, Link failures, etc...

RSVP-TE is required as the UNI signaling to request QoS-guaranteed connection establishment. 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). PATH message is delivered to the destination, and the destination terminal sends RESV message to request resource reservation and connection establishment. Since SIP/SDP has been used to configure the bidirectional session, PATH / RESV message exchange is executed for each direction.

We assume that the QoS-parameters are defined in the SLA (service level agreement) / SLS (service level specification), and TE-LSPs are established among PE-pairs for simplified traffic grooming. A multimedia application may require multiple QoS-guaranteed connections for multiple media streams. In current implementation of multimedia terminal, the MM service coordinator is managing the overall session status and connection establishments.

Since RSVP-TE is using "soft state" connection management [5], the end user's terminal should periodically exchange 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 PATH/RESV message exchange.

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. 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

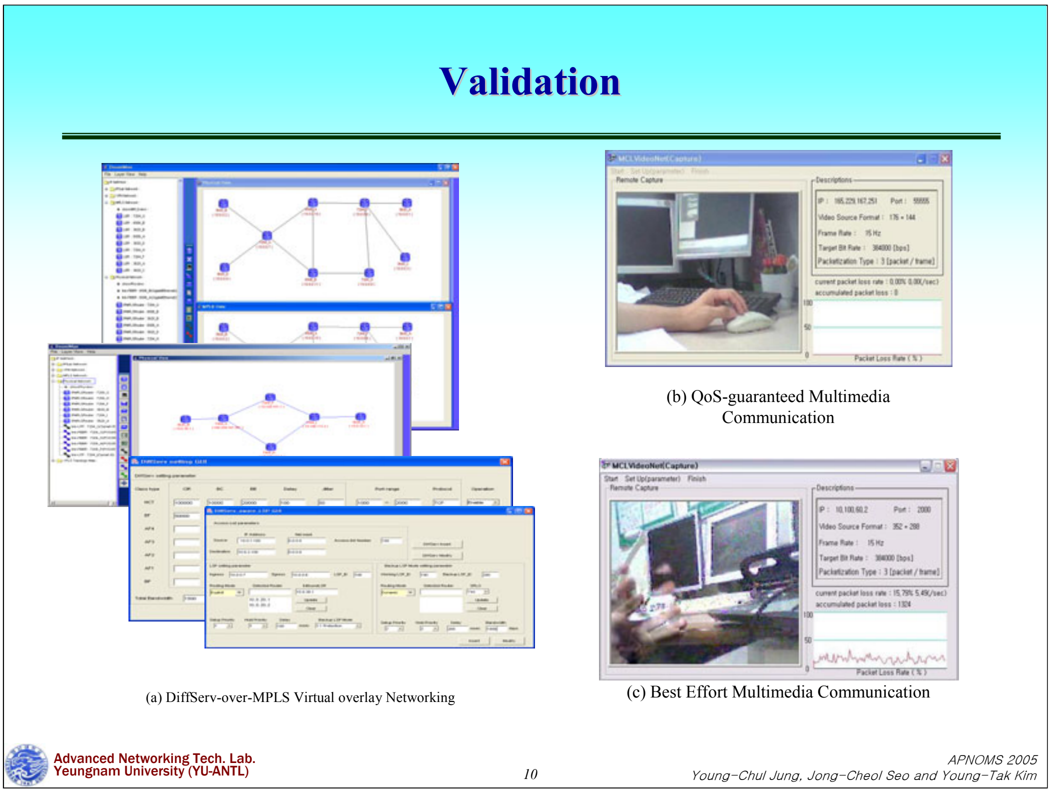


Multimedia coordinator in User agent manages SIP and RSVP-TE modules and Audio-Video connections. Media coordinator manages details of Session ID, source address, source port number, destination address, destination port number, protocol ID, service class, bandwidth etc. The above figure shows an SIP User Agent's Register Invite Message GUI and the User Agent of the receiver. Once invite message is accepted, the resource reservation will be carried out.

The SIP agent has been implemented with Object Oriented design concepts. The SIP agent design and implement for interaction with RSVP-TE. In this research work, we are putting more emphasis on the multimedia service platform with SIP/SDP and RSVP-TE signaling; so, we are using the SIP proxy server from VOCAL system [7] without any modification.

RSVP-TE signaling module has been implemented according to RFC 2205 and 3209. RSVP function modules has 2 parts : (i) management part, and (ii) message part. The RSVP protocol ID is 46 and it uses RAW socket to transmit data through RAW IP datagram. RSVP-TE Server handles RSVP resource reservation and resource allocation through the NMS. In each established session, the RSVP client sends one PATH message for resource reservation and one RESV message to allocate resource. To keep the reservation state persistent, a timer has been implemented which periodically sends PATH and RESV messages. For reservation tear down, the RESV TEAR and PATH TEAR messages have been implemented.

The video and audio data processing module has also developed that can support H.263 (CIF, QCIF) video format and G.711 (u-law, a-law) and G.723.1 audio format. The detailed functions of the video/audio processing module is out of the scope of this paper. The above figure shows the GUI interaction models of the multimedia service platform with SIP user agent, RSVP-TE and audio/video processing modules.

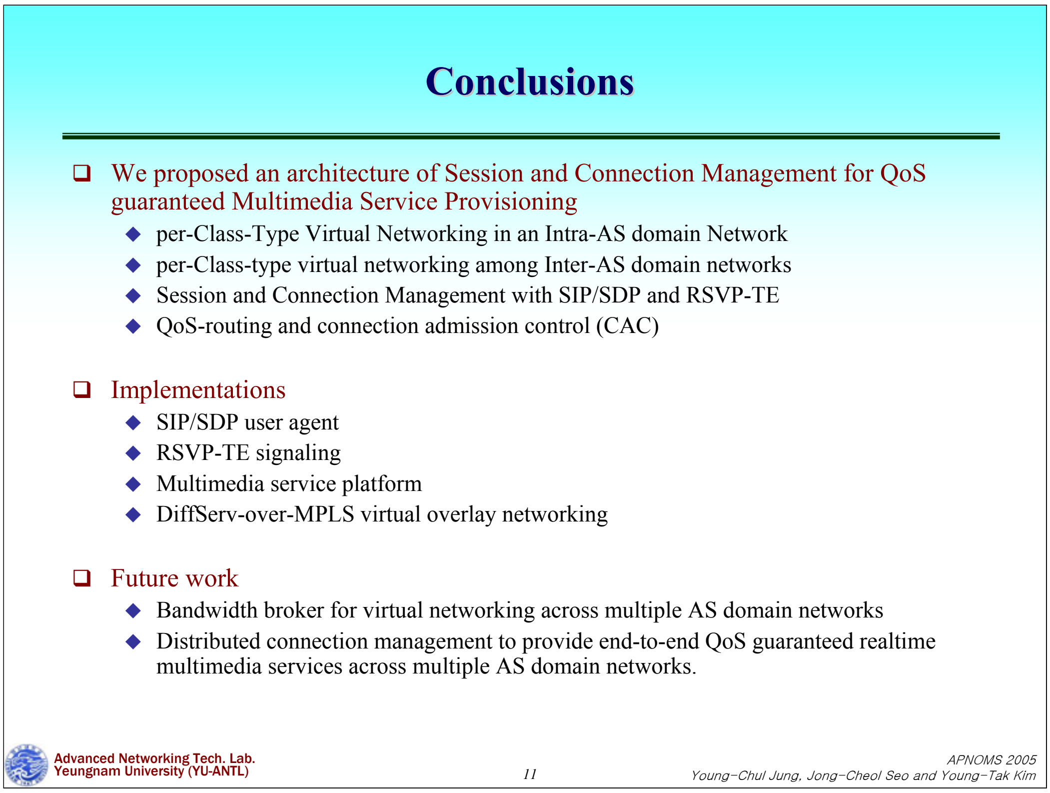


The QoS-guaranteed multimedia service platform with SIP/SDP and RSVP-TE has been implemented, and a QoS-guaranteed multimedia service provisioning with the DiffServ-over-MPLS virtual overlay networking has been tested. In the above figure, the DiffServ-over-MPLS virtual overlay networking with DoumiMan (DiffServ-over-universal multimedia internet Manager) is shown [9]. Our testbed network consists of four Cisco 7200 routers and three Cisco 6500 switches.

SIP/SDP has been tested for its detailed interaction to find the current location of the destination user, to offer and accept the available multimedia service type and related traffic & QoS parameters. The session configuration has been executed correctly, and the user can select different multimedia service configuration according to the capacity of the terminal.

The operation of RSVP-TE has also been tested to check the correct connection establishment with the selected traffic & QoS parameters. Also, the guaranteed bandwidth and QoS provisioning with the DiffServ-over-MPLS virtual overlay networking have been tested. DoumiMan [9] has been used to configure virtual overlay networks for NCT (network control traffic), EF (expedited forwarding), AF (assured forwarding) class-types. The audio traffic is transferred through EF virtual network, while the packet video traffic is transferred through AF virtual network.

In the performance analysis of the performance of multimedia service provisioning, we could verify the guaranteed QoS provisioning. In the case of QoS-guaranteed multimedia service, the pre-determined traffic parameters (committed bandwidth) and the QoS parameters (end-to-end delay, jitter, packet loss rate) have been guaranteed even in the network congestion status; while in the case of best effort multimedia service, massive packet loss and delay occurred under the network congestion.



In this paper, we analyzed the detailed operations of SIP and RSVP-TE for QoS-guaranteed multimedia service provisioning. The detailed procedure to select the QoS & traffic parameters using SDP has been explained, and the handshaking procedure was proposed for offer and selection model for the choice of proper multimedia service type for various terminal capabilities. We also proposed a multimedia service terminal platform and hierarchical traffic grooming for QoS-guaranteed service provisioning.

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Currently, we are updating the implemented SIP/SDP, RSVP-TE, COPS, bandwidth broker for virtual networking across multiple AS domain networks, and distributed connection management to provide end-to-end QoS guaranteed realtime multimedia services across multiple AS domain networks.

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