QoS Support for VoIP Traffic to Prepare Emergency

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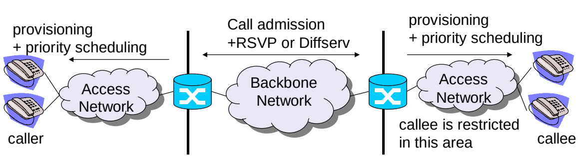
Background

- Necessity of QoS for VoIP communications
  - Emergency communications (E911)
  - Service for special user (government etc.)
- Over provisioning is not enough for emergency
  - Probability of calls were more than 50 times.
    (ex. Earthquake at 1995 in Japan)
- Why SIP is our target protocol?
  - SIP is a candidate for IP based cellular phone system
  - SIP is applicable to many communication media
    (ex. text, IM, VPN etc)

QoS scheme for SIP system

- Requirement
  - network resource must be reserved before the “Bell” rings
- Problem
  - destination IP address and port are unknown until the “Bell” stops
    ⇒ end to end resource reservation model is not applicable
- Possible solution
  - introduce some restrictions to callee’s IP address
  - priority packet forwarding + provisioning in access network



QoS requirements to prepare for emergency situations

- What happens in an emergency situation.
  - call probability is very high
  - system failure easily occurs because of high load
- requirements for system availability
  - avoid single point of failure
  - service must not be stopped under very high load
- service policy for emergency communication service
  - quality of high priority call must be high anytime.
  - quality of normal communication can be minimize to support as many communication sessions as possible.

QoS schemes

	Resource Reservation	Main function of call admission
Conventional Scheme	Reserve resource on each call by RSVP	Monitoring call status and update reservation
Conventional Scheme (variation)	Statically reserved for Diffserv class	Monitoring call status and calculate bandwidth
Proposed Scheme	Statically reserved for Diffserv class	Monitoring network and logging SIP message

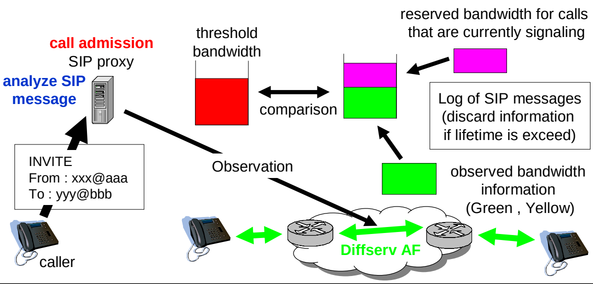
- Conventional schemes
  - Monitoring status of each VoIP call
- Proposed scheme
  - Monitoring network
  - Logging ***only*** “INVITE” messages

Problems of conventional schemes

- Cost of resource reservation with RSVP.
  - reserving resource with RSVP makes load of routers very high.
- How to monitor status of each call using UDP messages.
  - terminals report current status periodically.
  - lack status report = call is terminated
- Problem of call admission under emergency condition.
  Loss of packet becomes very large in an emergency.
  - Many calls lost network resources because of loss of signaling packets.

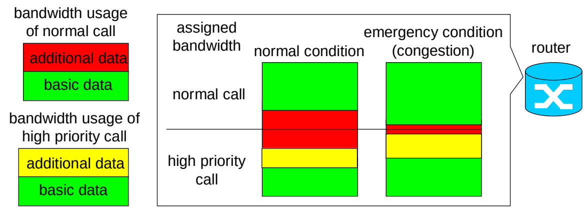
Proposed scheme

- SIP call admission based on traffic information and log of SIP INVITE messages



Mapping VoIP data into Diffserv AF class

- AF service has 3 preference category
  - Green , Yellow , Red
- VoIP data traffic divide into 2 service levels
  - basic data : data for minimum quality communication
  - additional data : data for high quality communication



Call admission method

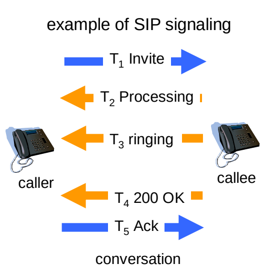
- criterion to accept new call
  - threshold > bandwidth of current call + bandwidth of new call
- two kinds of current calls
  - calls currently exchange data traffic.
    - it is observable on the router.
  - calls are currently signaling.
    - Call admission system calculates this from log of INVITE messages.

How to calculate bandwidth for signaling calls

- calculate method
  - Total bandwidth= sum(bandwidth of each log entry)
- structure of LOG
  - entry of log message = (SIP call ID, TTL , bandwidth)
  - $TTL = T_{arrived} + T_{lifetime}$
  - $T_{arrived}$: when INVITE message arrived at SIP proxy server
  - $T_{lifetime}$: Predefined value for SIP signaling
- Management method of the LOG
  - discard log entry if TTL is exceeded

Lifetime of SIP INVITE message ($T_{lifetime}$)

- Definition of Lifetime
  - $T_{lifetime} = \text{sum}_{K=1 \cdot\cdot n}(T_k)$
- Character of SIP signaling
  - T_3 is unlimited by any standard
- $T_{lifetime}$ must be defined from statistical data of real network service



Complexity and communication cost

	Complexity	Communication cost
Conventional scheme	O(n)	O(n)
Conventional scheme (variation)	O(n)	O(n)
Proposed scheme	O(n)	O(n)

- n : number of calls

Availability

proposed scheme > conventional scheme (variation) > conventional scheme

Effect of failure	router	SIP proxy server
Conventional scheme	Active calls loses reserved resources (recoverable)	Active calls loses reserved resources (unrecoverable)
Conventional scheme (variation)	No effect	Quality gets worth until current calls terminate
Proposed scheme	No effect	Quality gets worth during $T_{lifetime}$

Effect of heavy calls

- Quality of active calls get worth in the conventional schemes
- all schemes exhaust resources by heavy redial

	Overload of signaling traffic	Starvation caused by heavy redial
Conventional scheme	Active calls loses reserved resources	Yes
Conventional scheme (variation)	Quality get worth until current calls terminate	Yes
Proposed scheme	No effect	Yes

Conclusion and future work

- Proposed scheme
  - Diffserv AF service + Call admission method (Soft state)
- characteristic
  - high availability.
  - quality of each call loosely managed.
- Future work
  - evaluate characteristics of schemes under emergency conditions.
  - evaluate bandwidth usage of all schemes.