ภาคผนวก

ภาคผนวก ก.

สมการของกระบวนการสุ่มที่ใช้ในโปรแกรมจำลองระบบเครือข่าย

1. Uniform Integer PMF

(uniform_int, a, b)

$$p_x(x_0) = 1/(1+b-a)$$

$$x_0 = 0, \pm 1, \pm 2, \ldots$$
$$a \le x_0 \le b$$

$$a = 0, \pm 1, \pm 2, \ldots$$
$$b = 0, \pm 1, \pm 2, \ldots \qquad a \le b$$
$$E(x) = (a+b)/2$$
$$\sigma_x^2 = \frac{1}{12}[(b-a)(2+b-a)]$$

Arg #0: min_outcome = a
Arg #1: max_outcome = b

2. Exponential PDF

(exponential, a, b)

$$f_x(x_0) = \begin{cases} ae^{-ax_0} & x_0 > 0 \\ 0 & \text{otherwise} \end{cases}$$

$$a > 0$$
$$E(x) = a^{-1}$$
$$\sigma_x^2 = a^{-2}$$

Arg #0: mean_outcome = 1/a
Arg #1: <ignored>

ภาคผนวก ข.

ผลงานวิจัยที่ได้รับการตีพิมพ์เผยแพร่

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A Call Admission Control with Call Request Delaying Method in IP Networks

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ABSTRACT

In this paper, we propose a new theme of call admission control mode called "Call Request Delaying Method" (CRDM) that could improve call blocking rate and utilization in QoS network, especially in saturated situation. The CRDM was designed to handle new incoming call requests instead of reject them when the request bandwidth is still not available. The call request holding time for a new released bandwidth could be estimated by traffic descriptor of present active connections. By applying this delaying and prediction technique, we can show the improvement of call blocking rate and call attempt waiting time through simulation results compared to the traditional call admission control (CAC) model on IntServ over DiffServ framework.

Keywords : Admission Control, QoS, DiffServ, IntServ

1. Introduction

The future trend of communication network is to integrate many services onto the single IP network. As we know, the traditional IP networks support only best-effort services, it is not possible to differentiate among them nor to assure specific QoS targets for a given service. But the need of QoS support in IP network has driven in many research works and also standard body to provide the QoS onto IP network.

One of the main elements required in the network to provide QoS is the CAC mechanism. If the network has no control on the number of active flows and their consuming bandwidth, the overall traffic demand may exceed the total provided and the QoS of every flow (e.g. the transmission delay, lost packets ratio) could be degraded With CAC support, once a new call request attempt to establish a connection, CAC would accept the new call request only if the available resources are sufficient or else just reject it.

Call request rejection is ordinarily raised immediately after CAC module find out that the requested bandwidth is not available. This induce more requests and tedious work for user. Moreover, in the saturate situation, the traditional CAC process may not be fair because a new incoming call request may be granted just after a previous call request was rejected. Consequently, besides of the QoS concerns, the awareness of network services access opportunity should be considered too.

If we consider on today applications and the future trend, the on-demand stored media applications and defined time period applications such as video on-demand, e-lecturing, Internet channel program is going to quickly consume the Internet bandwidth. By utilizing the traffic descriptor on allocating bandwidth and duration these applications, we could propose a new scheme of CAC process that could reduce call blocking ratio, amount of call request signaling message and enhance more fairness of network access in saturate network: the CRDM. The CAC scheme handles incoming call requests in FIFO queue and be waiting until the requested bandwidth is available or their timeout is reached. This scheme could reduce call blocking rate and help enhance the fairness of CAC process.

As to evaluate the proposed idea we applied the CRDM to Mei Yang's framework [11]. We have developed the CAC scheme onto OPNET model and the simulation results show the improvement in average call request retrying times significantly compared to the unmodified scheme. In this paper, the content was organized as follows, section 2 mention on a brief of existing admission control methods in [11] framework. The basic principles of our approach in detail was described in section 3. In Section 4, simulated results and analyses are provided, and finally the conclusion.

2. Call Admission Control Methods in IntServ over DiffServ

IETF has attempted to provide QoS on top of the current Internet and two approaches are introduced: IntServ and DiffServ. IntServ promotes end-to-end QoS guarantees by setting up an end-to-end connection for each flow and maintaining the states of all connections, whereas DiffServ promotes scalability by pushing flow classification to network boundary and offering QoS to service classes. Targeting at both end-to-end QoS guarantees and scalability, RFC 2998 [8] suggests a combination of the two architectures, an IntServ over DiffServ framework as to combine a good point of two architectures.

The CAC mechanism for Integrated Services (IntServ) [1] architectures was specified by IETF in the means of a signaling protocol called RSVP (Resource Reservation Protocol). [2] Using RSVP, IntServ faces to scalability problem since all routers need to maintain state information of every flow. As to solve this scalability problem DiffServ architecture was proposed [3]. In this architecture, flows are aggregated in classes according to their specific characteristics and to be treated differently according to their classes. Since DiffServ has no admission control and traffic policing mechanisms, therefore there is no certain QoS guarantee in DiffServ network.

As trying to take advantage of both two IETF architectures and reduce the scalability problem, several novel architecttures and algorithms have been proposed: [4], [5], [6], [7]. The proposed framework of [11] adopts overall best ideas of previous works, it assume that two types of routers, edge router and core routers are exist in a network. Function of edge router is making admission decision, mapping individual flows to different service classes and transmit packets to the network for the network objects (i.e., clients). Core routers are DiffServ routers that provide class-based service differentiation. In this framework, to provide class-based service differentiation and local admission control at network edge without hop-by-hop signaling, link bandwidth is organized in hierarchy. First, the physical link is statically divided into multiple Provisioned Links (PLs); and a PL is dedicated to only one traffic class. Each PL is further divided into multiple trunks; one trunk is assigned to a given traffic class of one incoming edge router irrespective to their destination. As to renegotiate trunks' bandwidth, source edge routers have to keep track of available bandwidth of their assigned trunks and performs admission control locally without hop-by-hop signaling. Figure 1 illustrates this hierarchical bandwidth organization.

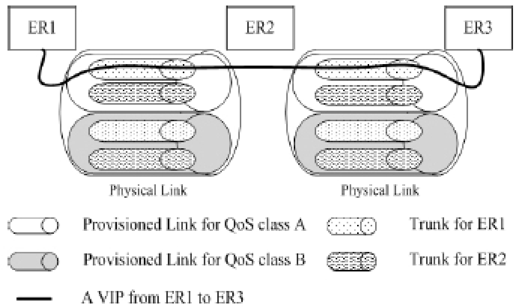

Figure 1 : Hierarchical bandwidth organizations.

3. Proposed Method

The proposed method is implemented by dividing the call admission control operation into two modes: normal mode and saturated mode. A network will be considered in saturated situation when there is not enough bandwidth for a new call request. Saturate network can not admit a new call request until some bandwidth would be released from the terminated connections. The traditional CAC responds to this situation by immediately performing call request rejection to deny new call request when the available resources is not sufficient. In order to access the network, user may try to send call requests iteratively and wait until the request would be granted. The call request iteration induces more signaling message and increase the connection establishing time. As to reduce the call request waiting time and call rejection, we proposed CRDM to delay call requests in FIFO queue and notifies the user later when the delayed request is admitted.

In order to make an admission decision, the CRDM determines the exist connections' traffic descriptors and their state information to examine an amount of bandwidth will being released in the near future. And to prevent excessive waiting time, we set a threshold on the wait time. If a request's lifetime is expire, the request will be dropped from the queuing buffer and CRDM will send a rejection packet to the user.

Figure 2.a shows the operation of the traditional CAC at edge router on the testbed framework. The CAC respond to a call request by sending an admit message or a reject message at once the resources determination is finished. In normal mode, the CRDM send admit message as same as the traditional CAC operation, but in saturate mode, CRDM operation is different as shown in figure 2.b – 2.d. CRDM immediately responds for every incoming call request by sending an acknowledge, this is to let the user know and

make their decision on the admission postponement. User may cancel the delayed request or just hold on a few seconds for the admission response. In case of accepting the admission postponement, CRDM would keep the call request for future resource reservation. In contrast, if user deny to postpone the request by sending cancel message, CRDM will delete the request information and user have to initiate a new call request again if he or she still need to access the network.

Figure 2.b and 2.c demonstrates success admissions for two delayed requests. Figure 2.b shows the shorter delay compare to figure 2.c, this is because the network got sufficient resources back from the terminated connection for just after the request in figure 2.b is received. In case that the network is still busy and some delayed requests lifetime are expire, the CRDM will send a reject message and clear the queued requests, as described in figure 2.d.

Other than, expecting CRDM to resolve call request iteration problems, we also utilized queue to improve the fairness of CAC process too. Even if the CRDM request queue was mentioned as FIFO but the CRDM operation detail in saturate mode is not truly FIFO. CRDM always admit the first queued request iff the current available resources are sufficient to provide the requested QoS level. In contrast, if the current resources is not sufficient to the first queued request, the second and the following requests will be determined. The first admissible request found by in sequentially scanning the delayed request queue, would be granted. This means, in some case the following request might be granted before of the head of queued request. In order to prevent call requests of moderate requirement application to be frequently rejected, CRDM was programmed to make an admission decision periodically. The admission period give more chance for the network to collect the released resources before determine the next admissible request.

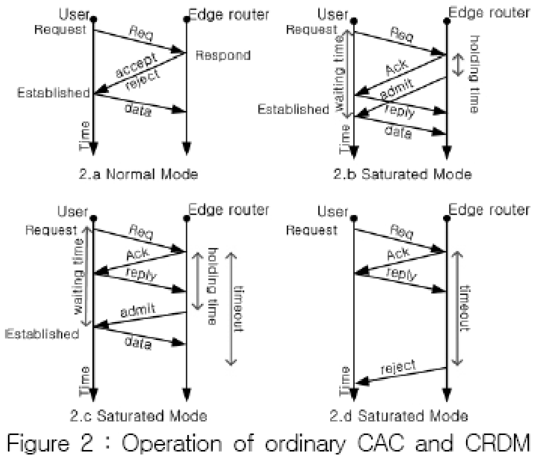

Figure 2 : Operation of ordinary CAC and CRDM

4. Simulation Results

This section show the simulation results of the traditional CAC over the reference framework [11] compared to the CAC modified by applying the CRDM. Since we concern on the services of predictable traffic then two traffic classes; video and audio are used as source traffics. Table 1 shows traffic characteristics of these two types of traffic. Descriptors of video and audio traffic are obtained from MPEG-4, VoIP [9][10] ITUT G.723.1 standard header correspondingly.

Table 1 : Traffic parameter values

Traffic Class	Bandwidth Requirement	Session Duration	Example
Audio	64 Kbps (CBR)	Exponential dist. with 300 seconds	VoIP ITUT G723.1
Video	1-6 Mbps	Exponential dist. with 1152 seconds	Video MPEG - 4

The network topology in our simulation is shown in Figure 3. As in the network, there are 15 core routers, and 10 edge routers are connected to each core router that support traffic from LANs. OC-48 fiber links are used as physical links between core routers and OC-12 fibers links were used to connect among edge routers and core routers. Therefore, there are 150 edge routers and 150 LANs in our simulation model. In the simulation, we defined the parameters as follows : packet length was fixed to 1024 bytes, the average inter-arrival time of call attempts was 50 seconds with exponential distribution. The simulation run time was set to 5000 seconds.

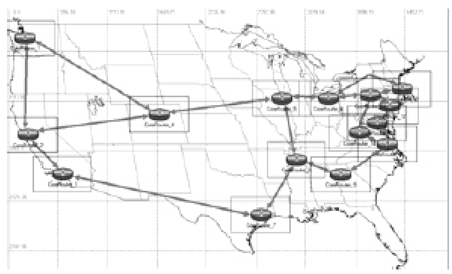

Figure 3 : Simulation topology.

Using the simulation model described above, the performance of the proposed method is evaluated and compared with the traditional admission control. The following three performance metrics are used in comparison:

- *Blocking Rate:* This is the ratio of the number of blocked user flows to the number of all user flows.
- *Average Waiting Time:* The average amount of time that an user attempt to access the network was requested until it was granted.

- *CAC Signaling Traffic:* A total number of CAC signaling messages counted in the simulation including all messages for call request, acknowledge, accept, cancel, reject, admit.

Figure 4 shows the blocking rate of traditional admission control and CRDM. It was shown that the proposed method could reduce call blocking rate significantly because we this method in saturated mode that was reduce rejection rate of the new incoming call request.

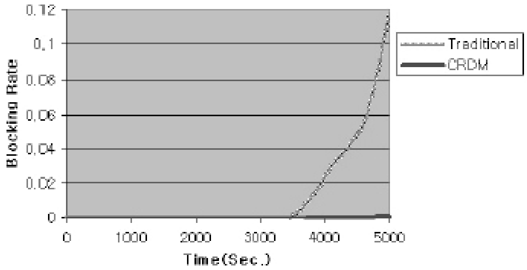

Figure 4 : Comparison on Call Blocking Rate

Figure 5 shows average waiting time of each call request. It was shown in this figure that the proposed method has extremely lower average waiting time for a call attempt. This is because waiting for the latest released bandwidth is quite shorter than making a new (or several) call request. As the result of decreasing call blocking rate, the total CAC signaling traffic are also reduced as shown in Figure 6.

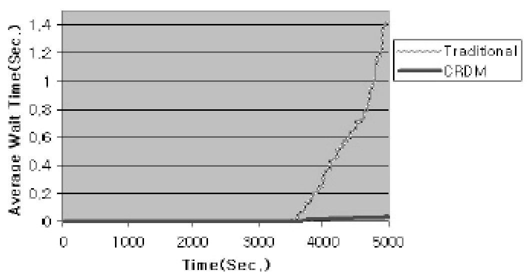

Figure 5 : Comparison on Average Waiting Time

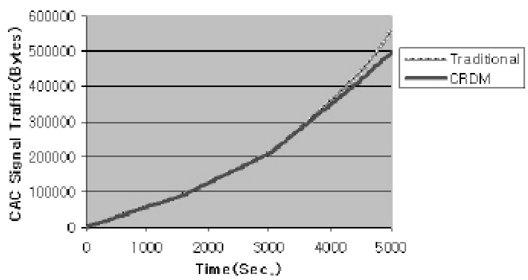

Figure 6 : Comparison on CAC Signaling Traffic

5. Conclusions

In this paper, we propose CRDM as a new CAC operation mode to solve the CAC problem in saturate network. The objective of CRDM is to reduce call attempt waiting time, call blocking rate and signaling traffic caused by call rejection and improve the CAC fairness in saturated situation. By extra delaying call requests and utilizing traffic descriptors, the CRDM could help improve the traditional admission control as show in simulation results. As result, this method offers user more ease and convenient access scheme to take place of the call request iteration and reduce signaling traffic when the network is busy.

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