A New VoIP Call Admission Control Scheme with Use of Alternate Routing for Low Call Loss Probability

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Abstract - Call admission control (CAC) is essential for the QoS guarantee communications. VoIP services require QoS guarantee as the existing telephony services. This paper proposes a new CAC scheme for VoIP to achieve efficient use of network resources by adopting a dynamic routing and a routing history table. This scheme chooses the better route out of two candidate routes where the admission conditions are satisfied. Two candidate routes are randomly chosen from all available candidate routes. The routing history table memorizes the resulting routing information for the fixed time interval for the succeeding calls. Within the fixed time interval, the succeeding calls will be assigned to the same route. After the fixed time interval, the memorized routing information is cleared, and the call admission control algorithm starts again from the beginning. It is confirmed by simulations that the proposed scheme achieves better performance than the existing CAC with alternate routing scheme.

Keywords-component: CAC, Routing, dynamic routing, QoS

I. INTRODUCTION

Call admission control is to resolve if a new coming call can be admitted. If QoS of a new call will not be guaranteed or the new call may affect the QoS of the existing calls when accepted, it is rejected. In the existing CAC schemes, probe packets are used to measure the packet loss ratio to estimate the network load [1], [2]. When the measured packet loss ratio is smaller than the threshold value, a new call is accepted. This scheme uses OSPF (Open Shortest Path First) as a routing protocol, and, therefore, cannot use another route even if such another route can provide better packet loss ratio.

The other CAC scheme deploying dynamic routing is proposed to achieve better performance [3]. In this scheme, probe packets are simultaneously sent along with two candidate routes, the shortest route as a default route and an alternate route that is randomly chosen out of all available routes. This adopts a routing table where the last unadopted routes are memorized. Any of these unadopted routes are not used as the second candidate route, because this unadopted route will be again very likely to be congested when the succeeding calls come. This observation will be valid when the average call duration is short enough, so that the unadopted route is very likely congested for the succeeding calls. VoIP traffic duration will be very similar to that of the traditional telephony. Therefore, after some time, the unadopted routes could provide enough performance to accept a new call.

This paper proposes a new call admission control scheme by introducing a new route selection scheme with the table of unadopted routes, which is reset after some fixed time interval. The succeeding call admission control will examine the other routes, which are not contained in the table of unadopted routes. This scheme enables more efficient usage of the network resources, resulting in the lower call loss probability than the existing schemes.

II. END POINT ADMISSION CONTROL

In the end point admission control schemes, call admission control is performed based on the traffic condition of the network estimated by the end systems, i.e., terminals. They are classified into two categories, active scheme [4] and passive scheme [5].

A. Active CAC scheme

Active CAC scheme uses a probe flow between originating end node and destination end node for estimating the traffic conditions of the route(s). CAC is performed based on the estimated results such as packet loss ratio and/or delay jitter. By using the same parameters of the probe packets as the real call, such as bandwidth and, packet size, the estimation will give accurate result, but they increase the traffic volume within the network. A sequence example is shown in Figure 1.

B. Passive CAC scheme

Passive CAC scheme is based on the direct measurement of VoIP packets themselves. It does not need probe packets, and therefore, it does not affect other traffics performance. However, the necessary functions of routers will be complicated. Moreover, the measurement may require higher performance end nodes.
III. DYNAMIC ROUTING CAC SCHEMES

A. The basic dynamic routing CAC scheme

A mesh type network model example is shown in Figure 2. The available routes from Node 1 to Node 2 are shown in Figure 3.

The basic dynamic routing CAC scheme sends probe packets simultaneously to the shortest route and one alternate route chosen randomly [3]. Packet loss ratios of the probe packets of these two routes are measured. If the measured packet loss ratios are higher than the admission threshold, a new call request is not accepted. If one of these measured loss ratios is lower than the threshold, the call is accepted. If both of two routes can accept the call, the shortest route is used.

B. The advanced dynamic routing CAC scheme

The advanced dynamic routing CAC scheme memorizes the routes that are used as the candidate routes for the last calls. In this scheme, probe packets are sent simultaneously to the shortest route and an alternate route chosen randomly. Loss ratios of the probe packets of these two routes are measured [3]. If the measured packet loss ratios are higher than admission threshold, a new call request is not accepted. The alternate route is memorized if it cannot accept the last call. This route is not used as a candidate route when succeeding calls are coming.

The basic dynamic routing CAC scheme may select again an unadopted route as an alternate route when the average occurrence interval between call is too short. The advanced dynamic routing CAC scheme stores the congested route information. The congested route at the last call attempt is not used as an alternate route. The exclusion of the congested route, i.e., unadopted route, at the last call request will exclude the possible rejection by call admission control for the next call. The exclusion enables more efficient usage possibility of other routes, which are very likely less congested than the unadopted route. Therefore, this CAC scheme will provide lower call loss than the basic dynamic routing CAC scheme.

C. Problems

The basic dynamic routing CAC scheme and advanced dynamic routing CAC scheme measure traffic congestions of the shortest route and an alternate route. Even when the shortest route is very likely to be congested, this route is used as a default candidate route. Therefore, both of these schemes have an efficiency problem of network resource utilization when the shortest route is congested. It will be highly possible that a new call is not accepted on the shortest route.

IV. PROPOSED SCHEME

The traffic volume of IP network is temporally changing. Therefore, if the occurrence interval between calls becomes long, the traffic conditions are easily changed in a network. The best QoS route will change with time. When occurrence interval of a call is short, the traffic volume of a network will change very slowly. The congested route will be free of congestion as the traffic condition is changed. The advanced dynamic routing CAC scheme does not use the route as an alternate route even after the route is free of congestion.

A flow chart of the proposed scheme is shown in Figure 4. In the proposed scheme probe packets are sent along with two routes which are selected randomly. If both routes satisfy the admission conditions, then the lower packet loss route is adopted. If either (or both) of these routes does (or do) not meet the requirements, the routing history table at the originating node temporarily memorizes such unadopted route(s) for the last call attempt. After some fixed time interval, memorized route information is cleared. This time interval is called ‘route information refresh time’ or RIRT in short.

A. Route Selection

In the proposed scheme, two candidate routes are chosen from all candidate routes, independently from number of hops. The shortest routes are used as one of alternate routes with the same importance.
B. Route Information Refreshing

The memorized route information is cleared, when there is no call request during RIRT. Any route is again used as a candidate route after the memorized information is cleared.

V. SIMULATION

A. Simulation Model

A simulation model is shown in Figure 5. In this paper, the routing table is assumed to be given. In case of OSPF, routing priority is determined by the route cost.

In the proposed scheme, priority is not used to select two candidate routes. The routes out of four are selected randomly. The four candidate routes are considered in the simulation. Other routing protocols also can be applied to the proposed scheme to get the routing table according to the requirements of call admission conditions.

The average holding time of the existing telephone call is 180 seconds. Each VoIP terminal generates the call of the exponential distribution where the average holding time is assumed to be 180 seconds.

The generating process of a call is the superposition process of call request from each independent user. Generally, it is known that the limit of the superposition process of an independent stochastic process with very low intensity will become Poisson distribution (Scale merit). Therefore, the average occurrence intervals of the call generation from each VoIP terminal are assumed to be Poisson distribution for 180 seconds.

B. Simulation Parameters

Simulation parameters are summarized in Table 1. The voice codec of VoIP assumes G.729. The quality objective value of VoIP assumes MOS (Mean Opinions Score) 3.0 (1-5). The packet loss ratio equivalent to MOS 3.0 of this voice codec is 2.0% [6]. Admission threshold and desired value of the packet loss ratio of VoIP call are set 2.0%.

C. Simulation Results

The call loss probability of the proposed scheme is shown in Figure 6. Average voice packet loss ratio of the accepted calls is shown in Figure 7. When RIRT is 10 seconds, call loss probability and packet loss ratio have very little difference as compared with those of the existing schemes. With RIRT of 100 seconds, call loss probability and packet loss ratio of the proposed scheme are better than those of the existing schemes. In the proposed scheme, the better performance is achieved by the selection of the route with better performance. While in the existing schemes, such flexible routing is not allowed.

The call loss probability over background traffic density in case of RIRT of 10 second and 100 second is shown in Figure 8. Introduction of the RIRT yields better call loss probability. The average packet loss ratio of the accepted calls over background traffic density is shown in Figure 9.

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**Table1. Simulation Parameters**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation Counts</td>
<td>500</td>
</tr>
<tr>
<td>VoIP flow’s activities</td>
<td>30%</td>
</tr>
<tr>
<td>VoIP flow’s Packet size</td>
<td>60 byte</td>
</tr>
<tr>
<td>VoIP flow’s send interval</td>
<td>20 milliseconds</td>
</tr>
<tr>
<td>VoIP flow’s average hold duration</td>
<td>180 sec (Exponential distribution)</td>
</tr>
<tr>
<td>Probe packet Size</td>
<td>60 byte</td>
</tr>
<tr>
<td>Probe packet send time</td>
<td>1 sec</td>
</tr>
<tr>
<td>Probe packet send interval</td>
<td>20 milliseconds</td>
</tr>
<tr>
<td>Background load traffic density</td>
<td>70% - 80%</td>
</tr>
<tr>
<td>Admission Threshold</td>
<td>2.0%</td>
</tr>
</tbody>
</table>
Very little difference is observed regarding average packet loss ratio among existing schemes and proposed scheme. The difference of route selection schemes by the proposed and the existing schemes are estimated with regard to the call loss probability over background traffic density, which is shown in Figure 10. The average packet loss ratio of the accepted calls associated with the conditions of Figure 10 is shown in Figure 11.

Figure 10 shown that call loss probability of the proposed scheme provides better performance than the existing schemes. The average packet loss probabilities of the accepted calls by the existing and proposed schemes in the same level as shown in Figure 11.

The better performance is achieved by the proposed scheme. It is because that in the existing schemes the shortest path is always the first candidate route even if the route is congested. In the proposed scheme, any route can be a candidate route unless the route was not adopted in the last call request. The routing selection of the proposed scheme is more flexible than the existing schemes. Such flexible routing enables the route usage with better performances.

The call loss probabilities over RIRT by the existing and proposed schemes are shown in Figures 12, 13 - and 14. Background traffic densities of Figures 12, 13 and 14 are 70 erl, 75 erl and 80 erl respectively.

Lower call loss probability is achieved by the proposed scheme for higher background traffic density as shown in Figures 12, 13, and 14. With RIRT of 100 - 120 seconds, lowest call loss probability is achieved by the proposed scheme for the background traffic density of 75 erl. When RIRT is set to 0[sec], it means no routing history table is used at the originating node, which temporarily memorizes the unadopted route for the last call request. This is equivalent to the basic dynamic routing CAC scheme. While if RIRT is long enough, it is equivalent to the advanced dynamic routing CAC scheme. The optimum performance can be achieved by dynamically setting RIRT according to the traffic conditions.
VI. CONCLUSION

This paper proposed a new CAC scheme that adopts new route selections and a routing history table. The simulation results confirmed that the proposed scheme has better performance with regard to call loss probability than the existing schemes, maintaining low packet loss ratio during the call.

Our future works includes VoIP performance evaluation of the proposed call admission control scheme for the mixed traffics including best-effort type of traffics. The scalability of the proposed scheme is also to be considered, including accuracy evaluation of practical end-to-end measurement method by probing packets. Finally, the impact of the routing to the Internet in the proposed scheme should be further evaluated.

REFERENCES


