New alternate routing scheme with endpoint admission control for low call loss probability in VoIP networks

New Alternate Routing Scheme with Endpoint Admission Control for Low Call Loss Probability in VoIP Networks

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Abstract: Call Admission Control (CAC) is an extension to Quality of Service (QoS) for voice traffic over IP and prevents the over subscription of voice traffic. For real-time delay-sensitive traffic such as voice, it is better not to initiate a new voice session rather than to allow voice sessions to be dropped or delayed later, causing intermittent impaired QoS and resulting in customer dissatisfaction. Dynamic routing mechanisms can be integrated with CAC to further extend the capabilities of QoS for VoIP traffic. In dynamic routing mechanisms, probe packets are sent to sense the congestion level of the network and according to the defined threshold, routes are chosen dynamically. CAC is therefore a deterministic and informed decision that is made before a voice call is established to provide suitable QoS for the new call.

All of the dynamic routing mechanisms have some issues in selecting random routes. This includes not considering number of hops, not doing admission threshold test, calculating all possible paths. In this research work, authors propose a new dynamic routing scheme, which considers a route history table with endpoint admission control. The main objectives of proposed approach are to increase the call admission probability, make call establishment time faster and save valuable CPU resources. Performance of the proposed scheme with respect to other dynamic routing schemes is studied using a mathematical/analytical model.

1. Introduction

Voice over Internet Protocol (VoIP) is a method for taking audio signals (audio) and converting them into digital signals which are transmitted over the Internet. VoIP does not provide guaranteed Quality of Service (QoS) as it suffers from packet loss due to route loss, congestion etc. in the network. Since IP does not provide a technique to ensure that data packets are delivered in sequential and proper order, VoIP implementation faces problems with latency and jitter. Call Admission Control (CAC) is an extension to Quality of service (QoS) for voice traffic over IP and prevents the over subscription of Voice traffic. Call admission control is basically used in VoIP to ensure quality of service and to prevent loss of packets.

Many different types of routing schemes have been proposed. Although these mechanisms provide an efficient service model and scalability, they require specific functionalities. Dynamic routing technique [1], [2], [3] is one of the new mechanisms which was proposed to achieve better performance for deploying call admission control, where condition of the network is learned by observing the condition of the probe packets and according to the pre defined threshold, we choose routes dynamically from the available routes to a particular destination.

2. Proposed Methodology/Algorithm

In one of the proposed schemes ***(Algorithm 4)***, probe packets are sent simultaneously along with two candidate routes to the desired destination. The shortest route is chosen as the default route and an alternate route is chosen as backup. This second route is chosen randomly from all available routes to the desired destination. Using a route table where the last rejected routes are memorized, none of these routes are used as the second route. So, after some time these rejected routes could provide enough performance to accept a new call since it can be safely assumed that the condition of the network changes with time.

The above proposed method selects two routes randomly, not considering the number of hops, calculates all the selected paths regardless of whether they will be selected or rejected, thereby wasting time by calculating the rejected path and also using precious CPU time. It is well known that the traffic volume of IP networks changes very frequently, so authors have proposed three schemes in this research work, taking three routes in to account to decrease the call loss probability, which is based on condition of the network

and predefined threshold, routes are chosen dynamically from the available routes to a particular destination. This results in more efficient use of the network resources, which ultimately leads to lower call-loss probability than the other existing mechanisms.

First scheme (Algorithm 1)

In this three routes are chosen, the first route (DR) is called shortest route and the other two are chosen randomly (TR1 and TR2). The Admission threshold (AT) for the DR is set a bit lower than the TRs, since DR is very likely to be congested most of the time and moreover to increase the possibility of a successful call. First, it checks for DR, then TR1, and then TR2. Routes that fail the AT test, are memorized for a certain period of time because of the changing nature of the network. After this fixed period of time, the memory is cleared so that the discarded routes can be considered again. Time to keep in the memory table is in the order *DR<TR*, since DR is supposed to change very rapidly (being the busiest route), than the TRs.

Second scheme (Algorithm 2)

DR is chosen as in the first scheme. But here, TR1 is chosen as the next shortest path and TR2 is randomly chosen. Here AT is different for all routes. The order is *AT<AT1<AT2*. If the call is through the DR or the TR1 route, then the efficiency increases. If not, then TR2 is always available for the call. In this same way, time to keep in the memory table is also in the order *DR<TR1<TR2*, whereby less time is kept for the busier routes because of their tendency to change frequently. Routes are tested in that same order as of first scheme i.e DR, TR1 and TR2.

Third scheme (Algorithm 3)

Same as the first scheme, but AT selected is same for all routes.

3. Mathematical Analysis and Results (Algorithm)

Here, we see how the algorithm 1 fares better among the algorithms that were proposed

3.1 Performance of four algorithms

During a call set up, predefined Threshold (Th) value defines whether a call is accepted or not. Probability that a call is accepted is given by

$$P_{me} + Z_R \sqrt{\frac{P_{me}(1-P_{me})}{N_{pr}}} \qquad (1)$$

where Zr=1-(1-R/2), R_{pr}= probe rate
***Th*:** Acceptance packet loss threshold
***Tpr*:** Length of probe, ***Rpr*:** Probe rate (bps)
***Lpr*:** Probe packet size (b)
***Npr*:** Number of probe packets per probing period
P_{me} : Measured value of congestion level
N_{pr} = probe packets / probing period

$$N_{pr} = \frac{R_{pr}.S_{pr}}{L_{pr}} \qquad (2)$$

S_{pr} = probe size, L_{pr}= probe length
$N_{min} = N_{pr} - [Pme.Npr]$
where P_{me} = measured probe loss probability
To pass our admission threshold test,

$$P_{me} + Z_R \sqrt{\frac{P_{me}(1-P_{me})}{N_{pr}}} < \text{Th} \qquad (3)$$

Algorithm 1 (proposed): Call will be accepted because of fixed path and random path 2; however, it will choose the least congested path.
Algorithm 2: Call will be accepted because of the random path 2; however, Algorithm 1 fares better as the congestion level of fixed path is less than other random path.
Algorithm 3: Call will be dropped.
Algorithm 4: Call will be accepted because of fixed path.

4. Conclusion

Based on the mathematical and analytical results, the proposed algorithm fared better than all the other schemes. It was observed that no calls were dropped in all the four cases considered for Algorithm 1, whereas at least one call was dropped for the other algorithms. The proposed scheme (Algorithm 1) also made efficient use of the network resources, leading to lower call loss probability than the other existing mechanisms.

[1] Xiang Kong and Kenichi Mase, Dynamic routing with endpoint admission control for VOIP networks, Department of Information Engineering, Niigata University, Japan, 2003.
[2] M Ivars and G Karlsson, PBAC: Probe based admission control, Department of Microelectronics and Information Technology, Royal Institute of Technology (KTH), 16440 Kista, Sweden, 2002.
[3] Ignacio M´as, Viktoria Fodor, and Gunnar Karlsson,The performance of Endpoint admission control based on packet loss, Department of Microelectronics and Information Technology, KTH, Royal Institute of Technology, Kista, Sweden, 2003.