Endpoint Admission Control Enhanced Systems for VoIP Networks

Abdulkhalig A. Bilhaj and Kenichi Mase
Department of Information Engineering, Niigata University
Niigata 950-2181, Japan
Phone: +81 25 262 7446, FAX: +81 25 262 7446, E-mail: mase@ie.niigata-u.ac.jp

Abstract

This paper presents QoS control enhanced architecture for VoIP networks. In this architecture we use both the probe flow delay and average loss rate measurement systems. First we apply the probability-based EMBAC scheme on our delay system. Then we propose a new probability-based EMBAC with a severe congestion consideration scheme to improve the admission control scheme in both measurement systems. We compare the performance of the enhanced systems in terms of blocking probability under the same condition of achieving average packet loss rate no greater than the certain target by setting an appropriate admission threshold in each system under each scenario. In this study, it is shown through simulations that the enhanced systems proposed in this paper can be a powerful and reliable EMBAC tool for VoIP networks with minimum blocking probability and minimum average loss rates.

I. Introduction

Ensuring high speech quality is the most important issue in Voice over IP (VoIP) networks. In order for VoIP to be commercially available as the circuit switched technology, the voice quality should be at least as good as the quality in today's telephony. Integrated services and differentiated services [1,2] are two technologies that address the issue allocating resources to various types of flows in any network. Integrated service architecture is based on per-flow resource reservation. At each hop, by using the signaling protocol, RSVP, admission control checks whether sufficient resources are available along the entire path to accept the new reservation. As a consequence, the integrated services scalability affects large-scale deployment. Rather than making per-flow reservations, differentiated services architecture uses a combination of edge policing, provisioning, and traffic prioritization to achieve service differentiation, so that it is more scalable and reliable. Of late, in differentiated service architecture framework, a different approach called an end-to-end measurement based admission control (EMBAC) has been developed to realize differentiated quality of service (QoS) in the Internet. In this research area, there have been many proposals for supporting admission control as in [3]-[4]. Also the proposals [5]-[7] introduced an EMBAC for VoIP network. Also, using the delay variations as a measurement tool for EMBAC mechanisms has been proposed, see for example [4].

In this paper, to carry out the active real-time end-to-end measurements, two types of probe measurement systems are considered, namely the *delay measurement system* and the *loss measurement system*. Based on both systems and to optimize the network performance, in this research we develop an EMBAC scheme based on the consideration of severe congestion.

In Section II we provide a description of EMBAC operation procedures for VoIP networks. In Section III we propose a new probability-based EMBAC with a severe congestion consideration to improve the conventional scheme for both delay and loss measurement systems. The first part of Section IV describes the network model for the end-to-end measurement adopted in this paper. The second part of the same Section shows the performance evaluation of the considered schemes in terms of blocking probability under the same condition of achieving certain target by setting an appropriate admission threshold in each system under each scenario. Section V discusses and concludes this article with a brief summary of our results.

II. EMBAC Operation for VoIP Networks

In our modeling, a VoIP network is composed of end nodes, routers, and links connecting them (see Fig. 3 in Section IV). In VoIP applications, voice packet flows may be carried in both directions. Thus, each incoming call requires end-to-end measurement in both directions to conduct call admission control (CAC).

The EMBAC procedure is divided into the probing packet-monitoring phase and the VoIP session packet transmission phase. Upon receiving a call request, two probe flows are independently carried out on the forward and backward paths between the pair of end nodes. Note that probe time required for sending probe flow increases the call-setup delay. Thus, the probe time should typically be set to 1 or 2 seconds at most. We use 1 second probe time throughout the paper.

The probing flow has the same transmission rate of the VoIP. The probe target end point set a probe monitoring window and measures the probe QoS (Quality of Service).

Proceedings of the 2004 International Symposium on Applications and the Internet (SAINT'04)
0-7695-2068-5/04 $20.00 © 2004 **IEEE**

The probe monitoring window starts at a time when the first probe packet arrives and continues 1+α seconds. The value α is the maximum possible queuing delay that may occur related to the probe buffer size and the lower priority of the probe flow. The probe QoS, $\boldsymbol{x}$, takes numerical values between 0 and 1. Particularly, we divide our measuring system for the admission control procedure into two systems. In the first system, the average packet delay that the probe flow experiences in the considered path is used for probe QoS; the average delay is measured by counting the delay of all arriving probe packets only received within the monitoring window. In the second system, the average packet loss rate is used for probe QoS. In the loss system, average packet loss rate takes a value between 0 and 1 and it is directly used for the probe QoS. In the delay system, delay is measured in the unit of seconds. If no packet arrives during monitoring window, possibly because probe packets suffer from a delay higher than 1 +α seconds, $\boldsymbol{x}$ is simply set to 1. In this case, setting the value $\boldsymbol{x}$ to 1 is also hold in the loss system when no probe packet arrives. In both measurement systems a certain admission threshold is used for CAC. The admission control decision outcome is reported with call setup signal notification or call block signal notification to the sender at the end node. The basic framework described above of the call setup protocol of the EMBAC for VoIP networks has been proposed in [5] and [6]. These call setup functions may be incorporated in the standard call signaling protocol such as SIP and H.323 with slight modification.

III. Probability Based EMBAC with Severe Congestion Consideration

Suppose that a new call request arrived at an end node, we define the call originating node as node O, and call target node as node T. Now consider an admission test at node T, which is in charge of judging whether to accept the flow from the node O to node T or not. To do this, node T measures probe QoS for the probe flow from node O to node T. If probe QoS is greater than the admission threshold, $\boldsymbol{r}$, this flow is rejected with a probability p = 1-*f(r)*, where *f(r)* is a monotonous increasing function of $\boldsymbol{r}$. This probability-based scheme (EMBAC-P) is first introduced in [8] *for the loss system*. Note that the conventional EMBAC scheme with the deterministic policy (EMBAC-D) has p equal to 1. [8] discovered that EMBAC-D tends to over-control call admission. The parameter "*f(r)*", which is chosen to be equal to $\boldsymbol{r}$, is thus introduced in to relax the strength of control. If the admission threshold "$\boldsymbol{r}$" increases, the chance of success in the admission test also increases. As the result, more calls are accepted. We find that with a severe congestion condition in the network, where a contending traffic overload occurs for a period of time, the EMBAC-D blocks all the coming calls while the EMBAC-P have no choice only to accept those calls, which should be rejected, through its probability function. Thus the EMBAC-P gives high packet loss rates (PLR) comparing to the EMBAC-D. The above observations for EMBAC-D and EMBAC-P are summarized in Table I. Considering these facts we develop a new EMBAC scheme, which has a feature of EMBAC-P in the temporary congestion and that of EMBAC-D in the severe

Table I. Control operation management of EMBAC Schemes.

Conges-tion type	EMBAC-D	EMBAC-P	EMBAC-P-SCC
Temporary congestion	Unnecessarily block the calls.	Accepts the rejected calls in advance	Has the same property of EMBAC-P
Severe congestion	Block all calls	Accepts calls with its probability and gives high PLR	Has the property of EMBAC-D
Control function	p=1	p=1-*f(r)* (*f(r)* =*r*)	p=1-*f(r,x, z)*

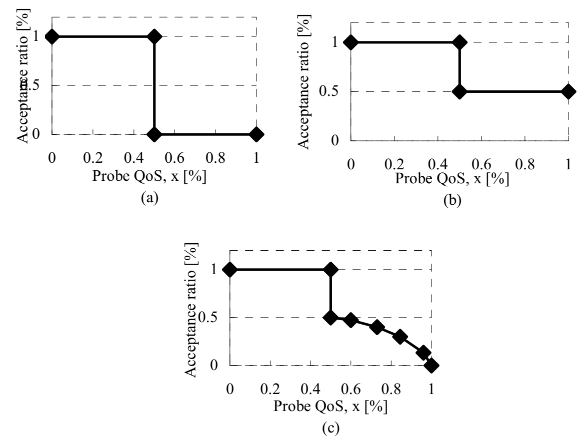

Fig.1-(a,b,c). Admission control operation concepts of EMBAC-D, EMBAC-P, and EMBAC-P-SCC respectively at $\boldsymbol{r}$=0.5.

congestion, termed EMBAC-P-SCC (Probability-based EMBAC with Severe Congestion Consideration) to optimize the network performance at any congestion level. Note that, the acceptance ratio (AR) is the probability of accepting the incoming call request according to the considered EMBAC scheme (*AR=1-p*). Figures 1-(a, and b) show a graphical representation of the control operation concepts of both schemes EMBAC-D, and EMBAC-P respectively. In these Figures, where the admission threshold, $\boldsymbol{r}$, is assumed to be 50%. If $\boldsymbol{x}$ is more than the value of $\boldsymbol{r}$=0.5 EMBAC-D blocks all calls *(p=1 and AR=1-p=0)*, while EMBAC-P accept all calls with the same probability of 50% *(p=1-r=0.5, and AR=1-p=0.5)*. In EMBAC-P-SCC control operation concept (Fig.1-c) the probability of every admission test to fail or succeed depends mainly on the probe QoS, $\boldsymbol{x}$, over the admission threshold.

To support and satisfy the concept and functionality of EMBAC-P-SCC, we developed an implicit asteroid function with the realization of the three parameters the admission threshold, $\boldsymbol{r}$, the probe QoS, $\boldsymbol{x}$ and the control strength factor, $\boldsymbol{z}$, to the following Equation:

$$f_{(r,x,z)} = -r\left(1-\left|\frac{x-1}{1-r}\right|^{\left(\frac{2}{z}\right)}\right)^{\left(\frac{z}{2}\right)} + r$$

$$0 < z < \infty$$

$$r \le x \le 1 \quad (1)$$

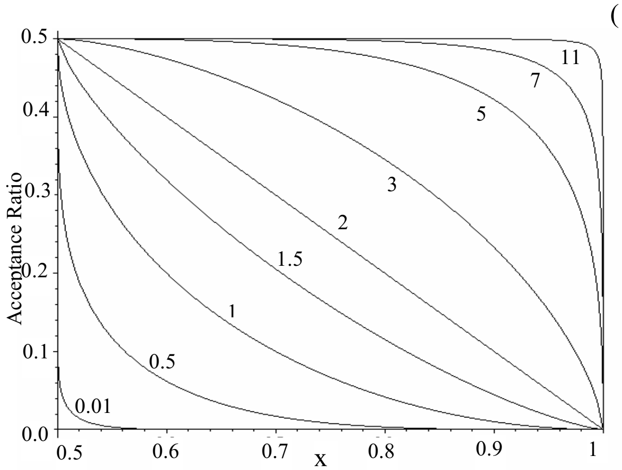

Fig. 2. Control operation of EMBAC-P-SCC related to the average delay/loss, ***x***, over an admission threshold, *r=0.5*, at different control strengths, ***z***.

Over the admission threshold, ***r***, and at any given value of ***x***, the admission control operation can be managed through the factor ***z***. The control hardness increases as the control operation gets close to EMBAC-D ($z \rightarrow 0$). Also, with a large value of ***z*** ($z \rightarrow \infty$), EMBAC-P-SCC is equal to EMBAC-P for all ***x*** values except ***x***=1 where EMBAC-P-SCC is equal to EMBAC-D. Fig. 2 illustrates the acceptance ratio, with the EMBAC-P-SCC operation under admission threshold of 0.5 with control strength factor of 0.01, 0.5, 1, 1.5, 2, 3, 5, 7, and 11 respectively. Using simulation we will prove that the EMBAC-P-SCC performance behavior is as designed in Table I.

IV. Performance Evaluation

A. Simulation Methodology: To evaluate the performance of EMBAC for VoIP networks, simulation is used. In the simulation methodology we consider the network model of Fig. 3-1. This model consists of end nodes and intermediate link with two intermediate routers and other links connecting the intermediate routers to the traffic end nodes. In this model we assume that the links connecting intermediate routers A and B to end nodes are lightly loaded (with no or ignorable congestion and negligible latency), while the main congestion occurs at the intermediate link routers, which is the common and the shared link between all end nodes. This situation may happen when local area networks (LAN) in distant places are connected by direct links through long-distance networks (WAN). It is natural to assume that each end node is a traffic source and a traffic sink simultaneously, where calls originate at the traffic source and terminate at the traffic sink. Nevertheless, we assume that end nodes at the left-side are traffic sources and those at the right-side are traffic sinks for simplicity. Of course, each node is a voice flow source and a voice flow sink simultaneously. Also we assume that the calls arrive at the traffic source end nodes according to Poisson arrival process. With this assumptions, the call arrivals at the intermediate link edge router A will be equal to the summation of call arrivals at all traffic source end nodes with Poisson arrival process. From this aspect, in our simulation we focus on the intermediate link A-B as the bottleneck link in the network, where node A and node B represent aggregation of traffic source and sink end nodes at both sides respectively as shown in Fig. 3-2. This simple simulation model enables us to investigate the ability range of different EMBAC schemes to ensure adequate and desired QoS for the VoIP flows. The amount of the signaling flows is not as significant compared with that of voice flows and probe flows. For simplicity we ignore the signaling traffic. Voice flow is given higher priority in packet scheduling than probe flows and it is served based on a non-preemptive priority scheduling. Also, we do not consider best-effort traffic. Accordingly, The probe flow can use only the unused bandwidth over the voice flow. Particularly, the bandwidth and the traffic intensity of the bottleneck link are set as a parameter ranging from 825 to 925 kbps with a range of traffic intensities.

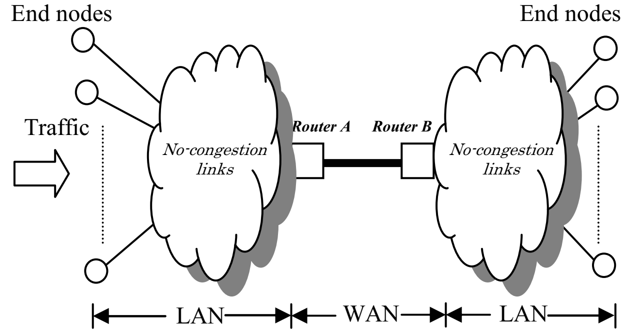

Fig. 3-1. *Simulation Methodology*

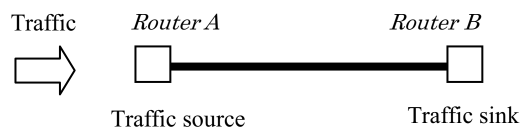

Fig. 3-2 Simulated model

The call holding time is based on exponential distribution with the average being three minutes. We consider active/silent states in the voice flow. For an established call, every active period of speech (talkspurts) is an exponential distribution of 3 seconds, and the overall voice activity p is assumed to be at 30%. Each packet has 40 bytes overhead composed of 20 bytes IP header, 8 bytes UDP header, and 12 bytes RTP header. The maximum length of a packet is 60 bytes with one packet generating every 20 ms during active periods of each direction for each call. Thus, the maximum rate for a VoIP call is 24 kbps. We also assume probing duration as 1 sec as mentioned above. A size of a probing packet is always 60 bytes. The buffer size for the voice flow and the probe flow are set at 40 and 20 packets respectively at each router. Satisfying different PLR targets, we use the blocking probability as the performance parameter. Each simulation executed for five hours in simulation time and the performance data are obtained in the last two hours to eliminate the effect of initialization and transitional conditions. Five runs with different seed numbers are conducted for each system and the collected data are calculated and averaged over those runs.

B. Simulation Results and related viewpoints: In the EMBAC-D performance of the delay system, the delay system is very sensitive to the initial changes of the delay admission thresholds and it is not sensitive at high admission thresholds. Note that in the loss system, similar problem exists at high admission thresholds, which is solved for the loss system through the EMBAC-P [8]. To avoid redundancy, we did not cover the detail parts of the performance with EMBAC-D and also of EMBAC-P for both delay and loss systems. Also we evaluated the performance of EMBAC-P-SCC using control strength factor z of 3 and 11. As a result, the EMBAC-P-SCC

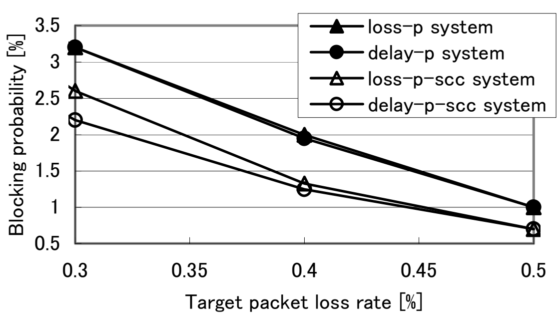

Fig. 4. Blocking probability vs. target loss rate for EMBAC-P and EMBAC-P-SCC

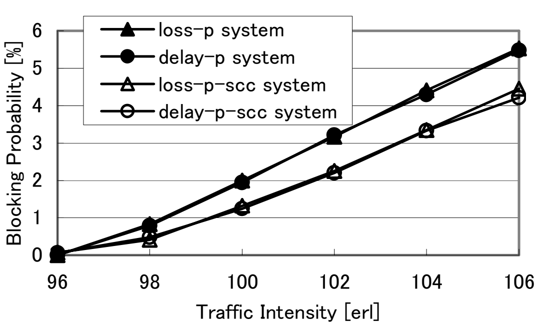

Fig. 5. Blocking probability under a 0.4% packet loss rate target.

with a control strength factor z of 11 performs higher than 3. Therefore, to compare the results of our enhanced scheme to those of EMBAC-P scheme, we will consider only a z value of 11 for the performance of EMBAC-P-SCC. Fig. 4 compares the performance for different PLR targets ranges between 0.3% and 0.5% at a traffic load of 100 erlangs and a bandwidth of 900 kbps. The optimal admission threshold is obtained as the admission threshold that satisfies the Target PLR with minimum BP. Fig. 5 illustrates the performance of these schemes at a PLR target of 0.4%. It is clear that the EMBAC-P-SCC is more effective than EMBAC-P. The reason is that, the EMBAC-P accepts the coming calls with the same probability whatever the congestion level, while EMBAC-P-SCC regulates and balances the admission process by accepting the calls that should be accepted in temporary congestion and blocking the calls that should be blocked in severe congestion.

V. Conclusion

This paper is a study of the main issues of EMBAC for VoIP networks using both delay and loss measuring systems. A probability based EMBAC with severe congestion consideration scheme has been proposed to consciously control and optimize VoIP networks' QoS. Using simulation we compared the performance of EMBAC schemes in terms of blocking probability under the same condition of achieving average packet loss rate no greater than the certain target by setting an optimum admission threshold in each scheme under each scenario. It is shown that the enhanced systems can be a powerful and reliable EMBAC tool for VoIP networks with minimum blocking probability and minimum average loss rates.

References

[1] R. Braden, D. Clark, and S. Shenker, "Integrated services in the Internet architecture: an overview," RFC 1633, Internet Engineering Task Force, June 1994.

[2] S. Blake et al. An architecture for differentiated services, 1998. Internet RFC 2475.

[3] Lee Breslau, Edward W. Knightly, Scott Shenker. Et "Endpoint Admission Control: Architectural Issues and Performance", *SIGCOMM'00*, Stockholm, Sweden.

[4] G. Bianchi, F. Borgonovo, A. Capone, L. Fratta, and C.Petrioli, "PCP-DV: an end-to-end admission control mechanisms for IP telephony," IWD2001, pp.470-480.

[5] Abdulkhalig A. Bilhaj, Yuichiro Toyama, Kenichi Mase, Masakazu Sengoku "Admission Control for VoIP Networks using Edge-toedge QoS measurements," APCC2001, 2001.

[6] K. Mase, Y. Toyama, A.A. Bilhaj, and Y. Suda, "Qos management for VoIP networks with Edge-to-edge admission control," Globecom2001, 2001.

[7] K. Mase and Y. Toyama, "End-to-end Measurement Based Admission Control for VoIP Networks", ICC2002.
framework," Globecom 2001.

[8] Hisato Kobayashi and Kenichi Mase, "Probability-based End-point Admission Control for VoIP networks.", In the Proceeding of the 2003 IEICE General Conference, March 2003.