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PAPER

Endpoint Admission Control Enhanced Systems for VoIP Networks

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SUMMARY 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 for the same target voice average loss rate, the enhanced systems proposed in this paper outperform the conventional schemes in handling the network resources. Then we will seek to prove that, for extra traffic loads within a busy period of time and with an optimal admission threshold chosen in advance, the enhanced systems can be a powerful and reliable EMBAC tool for VoIP networks in achieving high network performance with minimum blocking probability and minimum average loss rates. Finally it is shown that the enhanced systems have reasonable scalability.

key words: *Internet QoS, VoIP, admission control*

1. 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. Bandwidth utilization is another challenge for VoIP as any similar IP network is. In VoIP networks, if new calls are accepted without limit in the VoIP network, packet loss rates for calls in progress may become excessive, because total bandwidth required for the calls exceeds the network capacity. Therefore, a mechanism called call admission control (CAC) is necessary to reject a new call when enough network spare capacity is not available. Integrated services and differentiated services [1], [2] are two technologies that address the issue of 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. Thus, the EMBAC scheme does not depend on hop-by-hop signaling such as RSVP nor does it require any additional functionality for intermediate routers in the backbone network. In this research area, there have been many proposals for supporting admission control as in [3]–[15]. Also the proposals [16]–[18] introduced an EMBAC for VoIP network, which is designed to achieve loss rate and blocking probability targets by admitting generated calls to the network depending on probe-loss-rate admission threshold. Also, using the delay variations as a measurement tool for EMBAC mechanisms has been proposed, see for example [5]–[7].

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 Sect. 2 we provide a description of EMBAC operation procedures for VoIP networks. In Sect. 3 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 Sect. 4 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 packet loss rate target by setting an appropriate admission threshold in each system under each scenario. In Sect. 5 we illustrate how our enhanced systems can support the network performance at extra traffic loads. Section 6 gives efficiency evaluation. Section 7 discusses and concludes this article with a brief summary of our results.

2. EMBAC Operation for VoIP Networks

In our modeling, a VoIP network is composed of end nodes, routers, and links connecting them (see Fig. 5 in Sect. 4).

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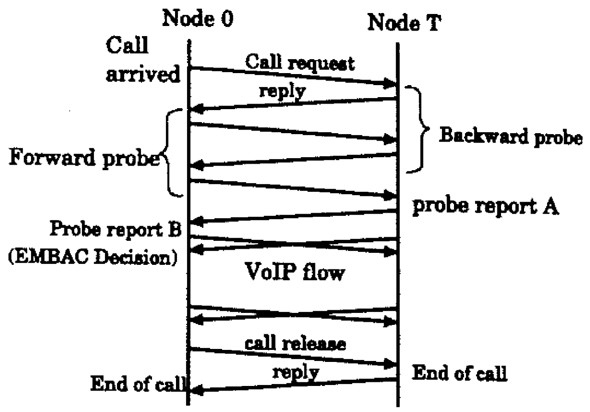

Fig. 1 Voice and probe flows with EMBAC Process, where the EMBAC decision depends on the applied EMBAC scheme.

An end node may be a router connected to a corporate LAN or a VoIP gateway, accommodating circuit switches or IP-based PBX servers in different locations. For simplicity we assume that calls originate and terminate at the end nodes, and according to the definition of EMBAC these end nodes are in charge of the admission control. 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 CAC.

The EMBAC procedure is divided into the probing packet-monitoring phase and the VoIP session packet transmission phase as depicted in Fig. 1. 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. Upon receiving a call request, two probe flows are independently carried out on the forward and backward paths between the pair of end nodes. Nodes O and T set a timer to receive the probe flow, respectively. 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 unless mentioned. We may have best-effort traffic. We use three priority classes. The highest priority is given to voice flow. Second priority is given to probe flow. The lowest priority is given to best-effort traffic. The maximum available bandwidth for voice and probe traffic is limited to a pre-determined value. The unused bandwidth, which is allocated to voice flows, can be used by best-effort traffic without affecting voice traffic. This priority is implemented by using three queues in parallel served in a priority order. The priority information may be carried in the IPv4 TOS or IPv6 DS field in the DiffServ framework, and is used by all nodes to forward packets according to priority scheduling.

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), which will be defined later, over the probe monitoring window. The probe monitoring window starts at a time when the first probe packet arrives and continues $1 + \alpha$ 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. We may set an appropriate upper bound on α, say 1 second, depending on the network and traffic conditions. To cope with the case when all probe packets are lost, the maximum time for waiting the last probe packet is determined based on empirical data and used for terminating the probe monitoring window. The probe QoS, 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. Note that every probe packet has an arrival time stamp that allows the delay system to run its measurements. For the time stamps the clock synchronization in the network can be guaranteed using the Network Time Protocol or other methods. The delay of a packet is the sum of the propagation delay and queuing delay. When the network is very lightly loaded (for example, in the midnight), there is almost no queuing delay and the propagation delay is dominant in the probe packet delay. Thus, propagation delay for a given path can be estimated by taking the minimum delay over some delay samples obtained by trial probe flow transmissions. This estimated propagation delay is deducted from the measured delay to be used for probe QoS evaluation. In the second system, the average packet loss rate is used for probe QoS; the average loss rate is considered as the ratio of the non-arrived packets within the monitoring duration to the total generated packets. 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. For example, 0.7 represents a delay of 700 ms. Due to the lower priority of the probe flows, the probe packet delay may be relatively longer than that of voice packet. But it is rare for the delay to be more than 1 second. If no packet arrives during monitoring window, possibly because probe packets suffer from a delay higher than $1 + \alpha$ seconds, x is simply set to 1. In this case, setting the value 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 will be done depending on admission test result. The success of the result means that probe QoS is no greater than the pre-determined admission threshold, and failure means otherwise. The 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 [16] and [17]. These call setup functions may be incorporated in the standard call signaling protocol such as SIP and H.323 with slight modification.

3. Probability-Based EMBAC with Severe Congestion Consideration

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, r, this flow is rejected with a probability $p = 1 - f(r)$, where $f(r)$ is a monotonous increasing function of r. This probability-based scheme (EMBAC-P) is first introduced in [19] for the loss system. Note that the conventional EMBAC scheme with the deterministic policy (EMBAC-D) has p equal to 1. [19] discovered that EMBAC-D tends to over-control call admission. The parameter "$f(r)$," which is chosen to be equal to r, is thus introduced in to relax the strength of control. If the admission threshold "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 each congestion type in Table 1. 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 congestion as shown in Table 1, termed EMBAC-P-SCC (Probability-based EMBAC with Severe Congestion Consideration) to optimize the network performance at any congestion level.

To give the EMBAC-P-SCC the management capability as in Table 1, we first summarize the control operation management concepts of both schemes EMBAC-D, and EMBAC-P and then we show how the required concept for EMBAC-P-SCC is explored using these concepts, and finally we introduce a new mathematical function which is created and developed to support the control operation management of this concept. For the control operation concept explanations we define the acceptance ratio (AR) as the probability of accepting the incoming call request according to the considered EMBAC scheme ($AR = 1 - p$).

A) EMBAC control operation concepts:

Figures 2(a) and 2(b) show a graphical representation of the control operation concepts of both schemes EMBAC-D, and EMBAC-P respectively, which are described in Table 1. In these figures, where the admission threshold, r, is assumed to be 50%. If x is more than the value of $r = 0.5$ ($x > 0.5$) 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* $AR = 1 - p = 0.5$).

B) EMBAC-P-SCC control operation concept (Fig. 2(c)):

Table 1 Control operation management of EMBAC schemes.

Congestion type	EMBAC-D	EMBAC-P	EMBAC-P-SCC
Temporary congestion	Unnecessarily block calls in the period of this type. After the end of this period the network will has a spare bandwidth unused because of the blocked calls	Probability based admission system accepts, with its probability, the rejected calls in advance so it is not affected by this type of congestion	Has the same property of EMBAC-P
Severe congestion	Blocks all calls	Accept calls with its probability even in this type of congestion resulting in high PLR within this period	Has the property of EMBAC-D
Control operation function	P=1	P=1-f(r) (f(r)=r)	P=1-f(r,x,z)

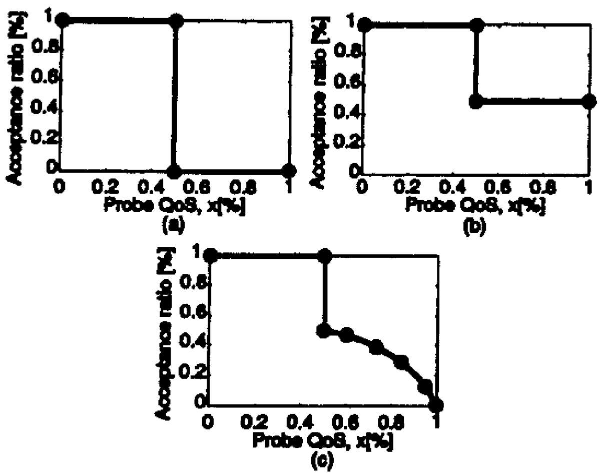

Fig. 2 Admission control operation concepts of EMBAC-D, EMBAC-P, and EMBAC-P-SCC respectively at r=0.5.

In this concept the probability of every admission test to fail or succeed depends mainly on the probe QoS, x, over the admission threshold.

To support and satisfy the concept and functionality of EMBAC-P-SCC scheme summarized in Table 1, there are specific requirements needed for the mathematical function of the control operation as the following:

- Unification: can unify the bases of both EMBAC-D, and EMBAC-P schemes including $f(x) = r$ at $x = r$ in temporary congestion and $f(x) = 0$ at $x = 1$ in severe congestion. An example of this control operation concept is graphically represented in Fig. 2(c).

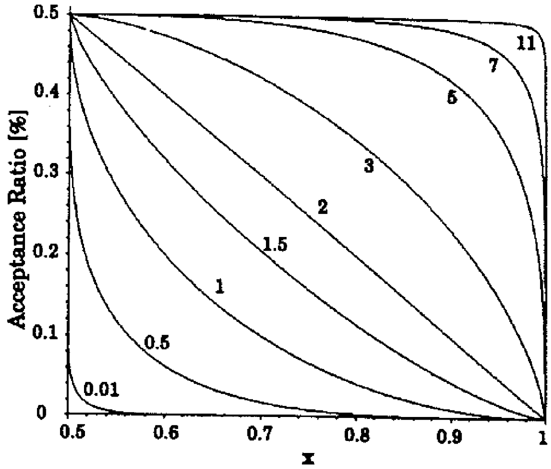

Fig. 3 Control operation of EMBAC-P-SCC related to the average delay/loss, x, over an admission threshold, r=0.5, at different control strengths, z.

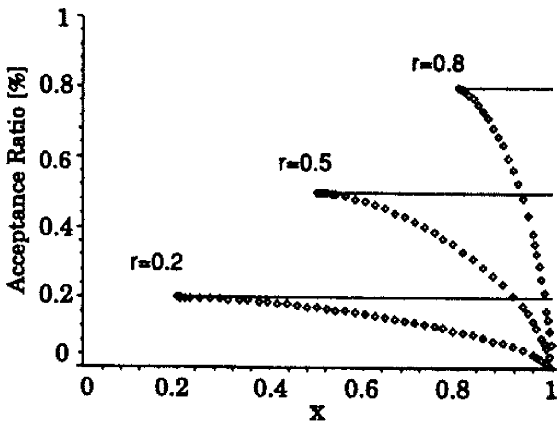

Fig. 4 Control operation of EMBAC-P with no action related to the changes of average delay/loss, x, and Control operation of EMBAC-P-SCC at control strength, z=3, over an admission threshold, r.

- Elasticity: convertible from high level of relaxation as in EMBAC-P to a high level of hardness as in EMBAC-D using one parameter.
- Implementation easiness: easy to implement in computer simulation and products.

After trying so many functions, a number of them could satisfy most of these requirements. Among these functions we selected the implicit asteroid function of the following equation:

$$x^{(\frac{a}{2})} + y^{(\frac{a}{2})} = 1, a > 0 \tag{1}$$

Consequently, to handle all mentioned requirements with the realization of the three parameters the admission threshold, r, the probe QoS, x, and the control strength (relaxation/hardness) factor, z, we developed Eq. (1) to the following equation:

$$\begin{aligned} & f(r,x,z) = -r\left(1 - \left|\frac{x-1}{1-r}\right|^{(\frac{2}{z})}\right)^{(\frac{z}{2})} + r \\ & 0 < z < \infty \\ & r \le x \le 1 \end{aligned} \tag{2}$$

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($z \to 0$). Also, with a large value of z ($z \to \infty$), EMBAC-P-SCC is equal to EMBAC-P for all x values except $x = 1$ where EMBAC-P-SCC is equal to EMBAC-D. Figure 3 illustrates the acceptance ratio, AR, 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.

Figure 4 shows the control operation difference between the EMBAC-P and EMBAC-P-SCC schemes. In both schemes the end of the line on the left side is the admission threshold r. If our admission threshold r is equal to 0.5, and the probe QoS, x, is equal to 0.8, EMBAC-P will accept the call with a probability of 0.5, while EMBAC-P-SCC will accept the call with a probability of 0.345. Besides, when no probe packets can arrive to the destination within the probe duration period the call will be naturally rejected in EMBAC-P-SCC.

Figures 3 and 4 give a good insight to both schemes. Using simulation we will prove that the EMBAC-P-SCC performance behavior is as designed in Table 1.

4. Performance Evaluation

4.1 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. 5. 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 (wide area networks, WAN), because LAN has large bandwidth enough to handle a number of calls larger than that of wide area networks in most real network situations. 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

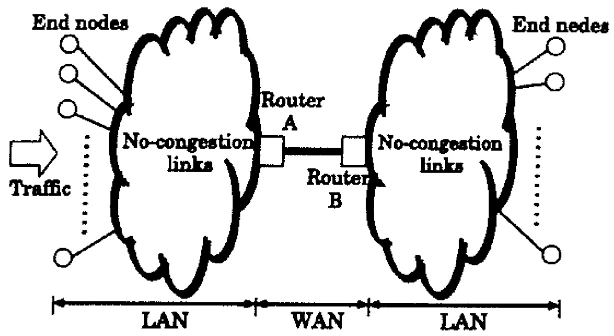

Fig. 5 Simulation methodology.

Fig. 6 Simulated model.

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. 6. 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. As mentioned in Sect. 2, voice and probe flows preferentially use a given bandwidth in each link. Therefore the effect of best-effort traffic is limited and there would be no essential difference in network performance between the case with best-effort traffic and that without best-effort traffic. For simplicity, we do not consider best-effort traffic. 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. Also we neglect the link propagation delay but the propagation delay can be estimated and deducted from the total measured delay, so that queuing delay is properly evaluated for the probe QoS in the delay system as mentioned in Sect. 2. As mentioned above voice flow is given higher priority in packet scheduling than probe flows and it is served based on a non-preemptive priority scheduling. Accordingly, The probe flow can use only the unused bandwidth over the voice flow. Particularly, the bandwidth of the bottleneck link is set as a parameter ranging from 825 to 3300 kbps with a range of traffic intensities.

The call holding time is based on exponential distribution with the average being three minutes. We consider active/silent (ON/OFF) 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%. The activity represents the average percentage of time during which speech

Table 2 The performance of the network under no control, when offered traffic is 100 [erl].

Link Bandwidth [kbps]	825	850	875	900	925
Average Loss Rate [%] (at 100 [erl])	2	1.24	0.89	0.65	0.38

Table 3 The performance of the network under no control, when link bandwidth is 900 [kbps].

Traffic Intensity [erl]	90	95	100	105	110
Average Loss Rate [%] (at 900 [erl])	0.16	0.33	0.65	1.19	2.09

is present. 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. Since the probe flow rate is selected same as the voice flow rate, the number of probe packets is 50 packets for each call request. A size of a probing packet is always 60 bytes. The buffer size for the voice flow is set at 40 packets at each router. From the importance of the probe flow as a network-controlling tool, we studied the performance of different buffer sizes through simulation; some of the results are shown in 4.2 and then considered for the rest of this paper. Satisfying a target packet average loss rate of 0.5%, we use the blocking probability as the performance parameter. Note that, in every simulation run, the average loss rate is calculated as the total lost voice packets divided by the total voice generated packets, and the blocking probability is calculated as the total rejected calls divided by the total call requests. 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.

4.2 Simulation Results and Related Viewpoints

The performance of the network without admission control is shown in Tables 2 and 3. In both tables the blocking probability is always 0, so the decrement of the bandwidth in Table 2 or that the increment of the traffic load in the network in Table 3 will result in degradation in the QoS, which is related to the voice average loss rate, beyond an unacceptable level.

Figure 7 illustrates the effect of probe buffer size on the network performance through the blocking probability satisfying an average loss rate target of 0.5% at probing durations (pt) of 1 and 2 seconds with traffic intensity and bandwidth of 100 [erl] and 900 [kbps] respectively. We found that in one hand, for a probe buffer size less than 20, less performance is obtained because the probe flow reflects only a

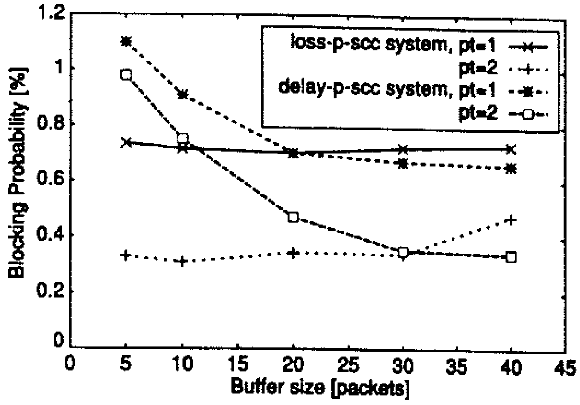

Fig. 7 Network performance at different buffer sizes and monitoring durations, at average loss target of 0.5%.

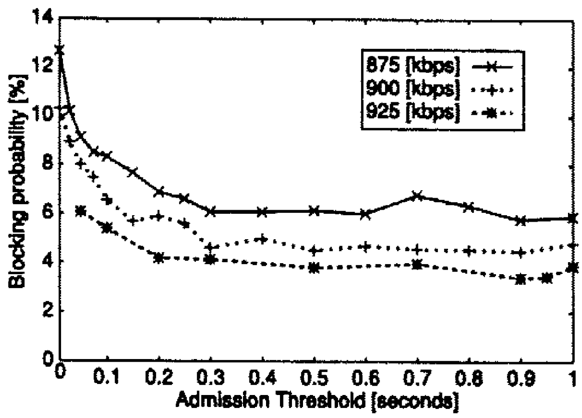

Fig. 9 Blocking probability vs. admission threshold in EMBAC-D of delay system.

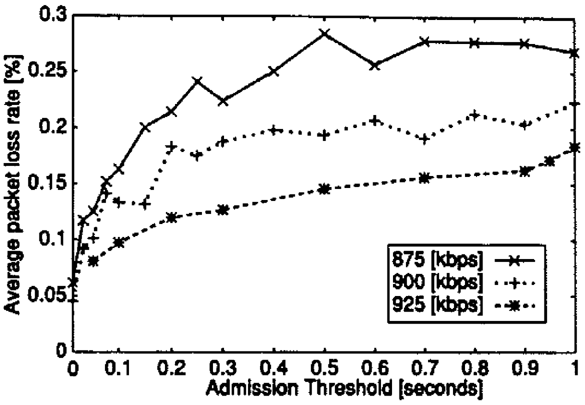

Fig. 8 Average packet loss rate vs. admission threshold in EMBAC-D of delay system.

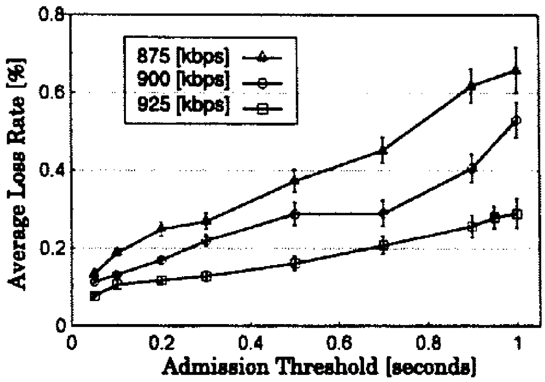

Fig. 10 Average packet loss rate vs. admission threshold in EMBAC-P-SCC of delay system with a confidential interval of 95%.

very temporal buffer state. On the other one, for long buffer the performance has almost no change in both measurement systems. Consequently, a buffer size of 20 packets gives a suitable probe average delay and average loss rates for the admission control (especially in our case where the probe flow duration per call unit is only 1 second fulfilling the on-demand method requirements). In probe duration of 2 seconds, similar trend are obtained with noticeable increase of the performance. This is related to the chance of the probe packets to reflect more precise network status information with longer monitoring period.

The EMBAC-D performance of the delay system through the average packet loss rate (PLR) and blocking probability (BP) versus admission thresholds are shown in Figs. 8 and 9 respectively, when 100 erlangs traffic is offered. With the increasing of the admission threshold, PLR tends to increase and BP tends to decrease as expected. Obviously, the delay system is very sensitive to the initial changes of the delay admission threshold, where the increasing of the PLR and the decreasing of the BP are very sharp at low admission thresholds. However, the effect of admission threshold on the performance is not sensitive at high admission thresholds (when the admission threshold increases from 20% to 100%). Note that in the loss system, similar problem exists at high admission thresholds, which is solved for the loss system through the EMBAC-P [19]. To avoid redundancy, we did not cover the loss system performance part with EMBAC-D and also the detail part 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 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 evaluation of EMBAC-P-SCC.

Our proposed scheme's performance is shown in Figs. 10 and 11. It is clear that the admission control scheme in EMBAC-P-SCC is very smooth comparing to that of EMBAC-D. For the confidence of the EMBAC-P-SCC, the confidential interval of 95% is also presented in the same figures; it is shown that the proposed scheme is accurate and reliable. The other cases have the same simulation conditions, so we omitted the confidential interval in the other figures.

To study the effectiveness of EMBAC-P-SCC, we first compare the performance of EMBAC-P with EMBAC-P-

SCC for the loss and delay system in Figs. 12 and 13. Figure 12 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. Figure 13 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.

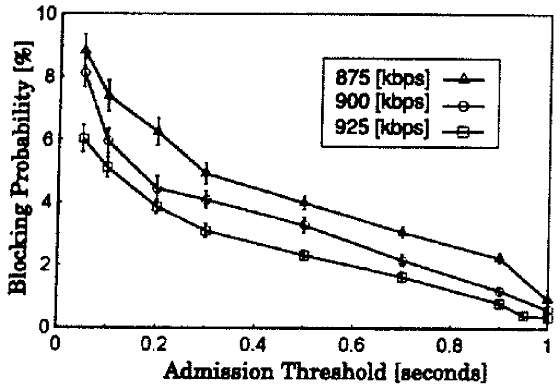

Fig. 11 Blocking probability vs. admission threshold in EMBAC-P-SCC of delay system with a confidential interval of 95%.

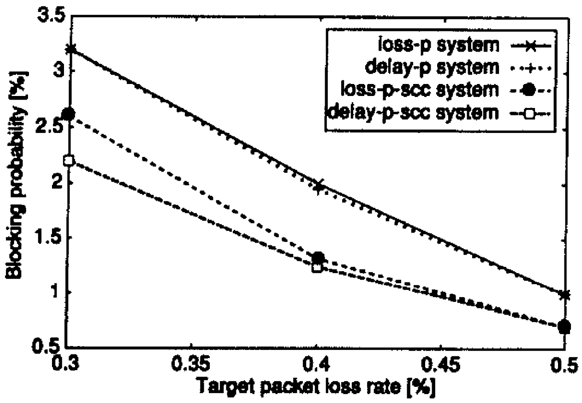

Fig. 12 Blocking probability vs. target loss rate for EMBAC-P and EMBAC-P-SCC.

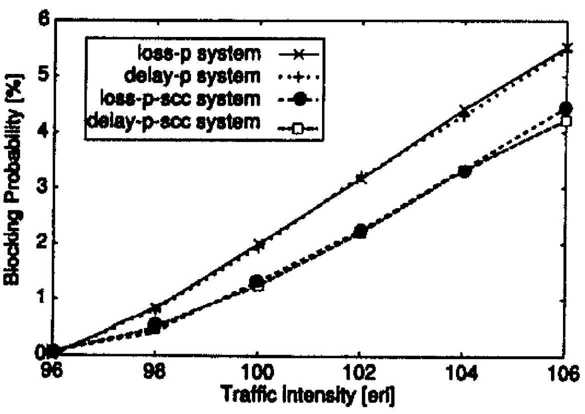

Fig. 13 Blocking probability under a 0.4% packet loss rate target.

5. Network Performance at Optimum Admission Threshold Chosen in Advance

The EMBAC-P scheme is based on a probability function that accepts calls, which should be blocked even at severe congestion states. To prove this, first we testify our network performance under a short to intermediate periods of fixed extra traffic load. Second, we study the network performance at different extra offered loads with a constant busy period of time. In real networks this case study happens when some network resources are unavailable related to link failure or traffic overload occurs in the network. The simulation run time, data collection, and the parameters are the same of the previous section unless mentioned (the bandwidth is set to 900 [kbps]). In both parts, the beginning of the extra traffic load period is selected randomly within the fourth hour of simulation.

In the first part, we study the increase of the offered traffic to 50% over the original considered traffic, from 100 [erl] to 150 [erl], for extra traffic periods of 5, 15, 30, and 60 minutes. The admission threshold chosen here is the optimal admission threshold that gives a 0.5% average loss rate target in the previous section with a constant traffic load of 100 [erl]. Figures 14 and 15 show the changes of the PLR target and the BP with the increment of the offered traffic load within the corresponding period in every scheme in every measurement system. At fixed extra traffic loads, we can see clearly that both EMBAC-P and EMBAC-P-SCC schemes worked as mentioned in Table 1. Although the EMBAC-P has lower BP in delay and loss measurement systems with intermediate periods, the total performance shows that the EMBAC-P-SCC scheme not only outperforms the EMBAC-P scheme but also tries to keep the average loss

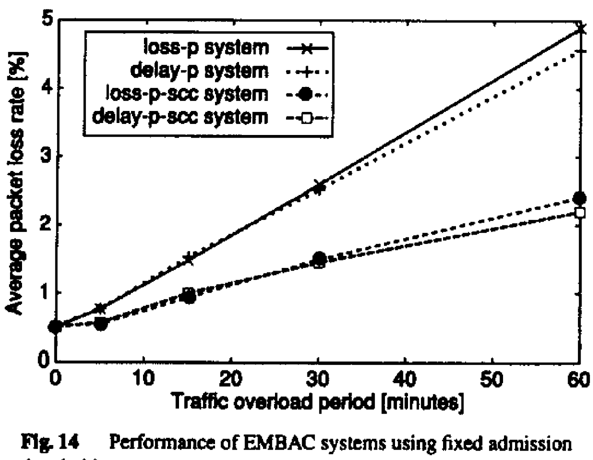

Fig. 14 Performance of EMBAC systems using fixed admission threshold.

rate close to the considered target.

In the second part, we consider a fixed extra traffic load period of 15 minutes and conduct our simulation study with different extra offered loads range between 30% and 75% over the original designed traffic of 100 erlangs. Figures 16 and 17 show the reliability of the EMBAC-P-SCC scheme with similar achievements of the first part.

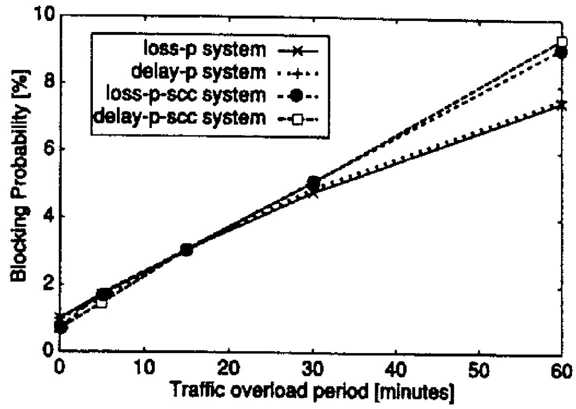

Fig. 15 Performance of EMBAC systems using fixed admission threshold.

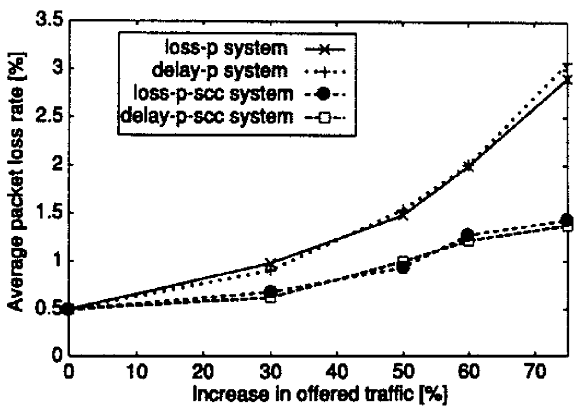

Fig. 16 Performance using fixed admission threshold for each EMBAC system.

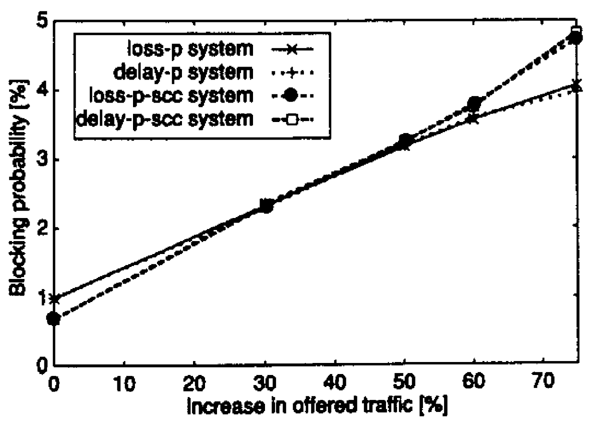

Fig. 17 Performance using fixed admission threshold for each EMBAC system.

6. Efficiency Evaluation

We calculate efficiency, which is defined as follows:

(1) EMBAC-P and EMBAC-P-SCC
Using delay and loss measurement systems, obtain the maximum value of offered traffic R, to meet average loss rate target of 0.15% and Blocking probability no greater than 1%, using optimum admission threshold for a given link bandwidth by simulation.

(2) Circuit-switch-based admission control
Obtain the maximum value of offered traffic S, to meet the same conditions of average loss rate and Blocking probability, using the optimum number of trunks N (maximum number of calls in progress) for a given link bandwidth by simulation.

For fair comparison we set the simulation conditions of the circuit-switch model as same as that of the EMBAC model, the only difference is that the network is controlled through delay and loss measurement systems in the case of EMBAC and through trunks availability in the case of circuit-switch model. Then, the efficiency of the EMBAC schemes and that of the circuit-switched model are defined as R/N and S/N, respectively. The result is shown in Fig. 18. Efficiency of the circuit-switched model increases with link capacity. Besides, The efficiency of EMBAC-P-SCC, for both delay and loss measurement systems, is closer to that of the circuit-switched model with the same trend and outperforms that of the EMBAC-P throughout the range of link capacity.

In circuit switch model, link-by-link admission control is performed. Hence, the number of calls in progress is measurable, which gives ideal and precise admission control. In EMBAC, the admission control is performed based on the probe flow to estimate whether there is enough bandwidth on the considered path. In actual fact, the EMBAC has no available information about the number of calls in progress. Nevertheless, EMBAC-P-SCC could achieve al-

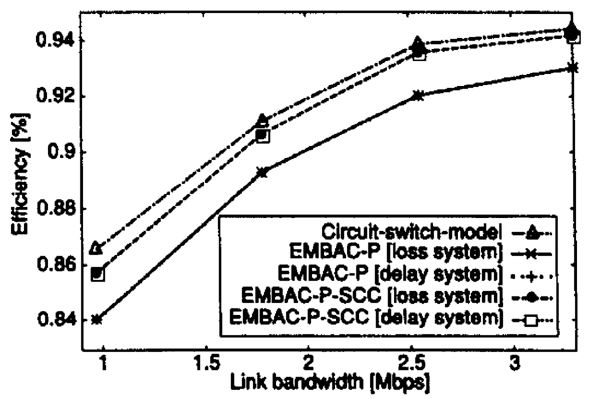

Fig. 18 Efficiency of circuit-switched model, EMBAC-P, and EMBAC-P-SCC.

most the same ideal efficiency of circuit switch based admission control.

This result suggests that EMBAC-P-SCC, for both loss and delay systems, is effective, scalable, and efficient for wider range of link capacity.

7. 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. Simulation was used to evaluate performance of different EMBAC schemes. 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 EMBAC-P-SCC scheme has improved the network performance. Also we could prove that, for extra traffic loads within a busy period of time and with an optimal admission threshold chosen in advance, the EMBAC-P-SCC can be a powerful and reliable EMBAC tool for VoIP networks and an important key factor in achieving high network performance with minimum blocking probability and minimum average loss rates. It is finally confirmed that EMBAC-P-SCC is scalable, effective, and efficient for wider range of link capacity.

As future challenges, our research investigation is continuing with many issues that are still open regarding the enhancement of VoIP systems. The issues that are currently under study are the following:

- We are planning to subject our enhanced scheme to heterogeneous traffic. We believe that the proposed EMBAC scheme could be modified to work well in much wider set of load patterns.
- A network with many congested bottleneck links was out of the scope of this paper, involving this scope is one of the important future works.
- Using Traffic Engineering and MPLS technology for the capacity allocation in VoIP networks for suitable average loss rate and BP targets is under investigation.
- Also, we would like to implement and develop our scheme for the dynamic routing.

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