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ORIGINAL ARTICLE

End-to-end Measurement Based Admission Control VoIP protocol with loss policy

Sami Saleh Alwakeel *, Nasser M. Alotaibi

College of Computer and Information Sciences, King Saud University, Riyadh, Saudi Arabia

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Abstract This paper presents a study of the newly proposed End-to-end Measurement Based Admission Control scheme with Loss threshold policy (EMBAC-L). The proposed scheme allows automatic adjustment of the admission control scheme parameters using a feedback measurement of the network's active loss rate. If the loss rate is less than the predetermined threshold, then the call is accepted, otherwise the call is rejected, with probability *P* which is a function of the measured network loss. The scheme's call rejection probability is analyzed with various distribution functions in order to set the admission threshold and scheme parameters to the appropriate operational value. Simulation is used to evaluate the scheme's performance and to demonstrate its effectiveness. Our study shows that the proposed admission control mechanisms enhance the EMBAC mechanism's performance, and dynamically control the VoIP packet loss rate under various network operational conditions.

1. Introduction

Voice over Internet Protocol (VoIP) is a category of hardware and software that enables users to utilize the Internet as a transmission medium, i.e. using IP to send voice in packets rather than by the traditional circuit transmissions of PSTN (Public Switched Telephone Network). VoIP services can be offered over any data network that supports IP traffic, such as the Internet, enterprise IP networks, and Local Area Networks (LAN). Voice signals are digitized, compressed and converted into IP packets, and then transmitted over the IP network. Signaling protocols, such as Reservation Protocol (RSVP), are used to set up and tear down calls, carry the information required to locate users, and negotiate capabilities. Recently, VoIP data transport over WLANs has become a very attractive service, and now constitutes one of the fastest growing applications in modern WLANs. The main motivations for IP telephony are the very low relative costs, current demands for multimedia communication, and integration of voice and data networks (Tran et al., 2003).

* Corresponding author.
E-mail addresses: swakeel@ccis.ksu.edu.sa (S.S. Alwakeel), nassero@hotmail.com (N.M. Alotaibi).

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2. Quality of service for VoIP

The capacity of the Internet, and of WAN networks, is growing as it is used as a transfer medium for real-time media applications such as IP-telephony. Quality of service (QoS) is a challenge where voice calls are no longer using the old circuit switched network, and thus calls need to share bandwidth. Considerable research efforts have been expended during the last decade into providing QoS for these real-time applications over IP-based networks, and techniques and implementations have been documented in numerous old and recent publications. Researchers have proposed many techniques for improving quality of service (QoS) in the face of voice packet loss (Li et al., 2000; Markopoulou et al., 2003; Chen, 2003; Chua and Pheanis, 2006) over the Internet. Improvement of network capacity while providing QoS of voice service is also a big concern for wireless network planners (Zhang et al., 2008). Some of these techniques employ receiver-based packet-loss concealment (PLC) algorithms to synthesize audio when packets of audio data are missing. There are also sender-based loss-recovery techniques (SBLR), whereby the sender assumes an active role to help the receiver recover lost data or improve QoS when packet loss occurs (Chua and Pheanis, 2006). Another approach is to reduce network congestion due to voice packets using call admission mechanisms (Mase, 2004; Mase et al., 2001).

In what follows, we discuss EMBAC scheme, which is widely suggested as a call admission control in VoIP networks (Tran et al., 2003; Mase and Toyama, 2002; Bianchi et al., 2000; Mase and Kobayashi, 2004).

2.1. End-to-end Measurement Based Admission Control (EMBAC) schemes

To support real-time data delivery demands over IP networks with respect to QoS guarantees, various Call Admission Control (CAC) mechanisms have been developed to reject a new call when there is not enough spare capacity in the network (Mase, 2004). The EMBAC control mechanism reacts appropriately to the traffic in the network, e.g. minimizing delay by monitoring end-to-end data flows and collecting statistics on the behaviour of the network (Mase and Toyama, 2002; Bianchi et al., 2000). Based on measuring the loss in a stream of probe packets, admission control is applied to deliver acceptable QoS. Probes and voice packets of accepted flows are transmitted with low and high priority, respectively, in order to protect the accepted flows from the load of the probe streams. The probing packet's arrival statistics are collected over a fixed length measurement period, which is arbitrary and is set by the network administrator. At the end of the measurement period, on the basis of the statistics collected, the receiver estimates whether there are enough resources available along the connection path to meet a predetermined QoS requirement. The call is accepted if the probe packet loss rate is below a predefined threshold. This decision is notified back to the sender, which either switches from probing to voice data phase and starts transmitting high priority call packets, or aborts the call set-up (Bianchi et al., 2000; Mase and Kobayashi, 2004).

2.2. EMBAC with probability policy (EMBAC-P)

An enhanced EMBAC mechanism for VoIP networks, which contributes to increased benefits of EMBAC, is suggested by Mase and Toyama, (2002). This reference presents constant probing instead of conventional on-demand probing in order to eliminate call set-up delays due to probing. It also presents automatic adjustment of admission control parameters through feedback control. End-to-end Measurement Based Admission Control with Probability policy (EMBAC-P) in VoIP networks was presented by Mase and Kobayashi (2004), Bilhaj (2004), and this scheme provides a new admission control mechanism. The flow is rejected with probability $P = 1 - f(x)$, where $f(x)$ is a monotonous increasing function of a predetermined threshold x. When $f(x) = 0$, then we have a conventional EMBAC scheme, which is termed EMBAC with the Deterministic policy (EMBAC-D). The parameter $f(x)$ is thus introduced in this approach to relax the strength of the control. Increasing the admission threshold, x gives more chance of success in the admission test, and as a result, more calls are accepted. However, this in turn increases the packet loss rate for the calls in progress, and degrades voice quality. Decreasing x, on the other hand, has the opposite effect, which is obtained at the cost of resource efficiency. Thus, the admission threshold x controls the packet loss rate of voice flows. The EMBAC-P scheme shows that its performance is close to the ideal method of the virtual trunk-based admission control, where $f(x)$ is chosen to be equal to x, and so the new arrival call will be admitted if the probe loss is less than the predetermined threshold x, or another test is applied to accept or reject the call with probability $P = 1 - x$.

2.3. Research objectives and importance

The goal of this research is to design and study the performance of a loss based (instead of threshold based) EMBAC mechanism. In this protocol, the admission control is set based on the packet loss rate. A new call is rejected using a loss-based probability function. When implemented, the proposed mechanism uses various probability distribution functions to find a threshold value and rejection probability function that provides the highest call admission rate while maintaining the required QoS for the VoIP networks. Using these operational settings, the aggregate throughput of the network is then maximized.

It should be noted that network loss estimation through probing is critical in tracking the network's varying congestion status, and in improving transmission efficiency. However, network probing introduces additional overheads. For this, an alternative approach can be used to avoid implementing an imposed real-time measurement for the network loss measurement. VoIP RTP Control Protocol Extended Reports (RTCP XR), which is a standard protocol defined by IETF RFC 3611, may be used for VOIP call management (Friedman et al., 2003). This protocol allows deriving network loss-rate data for VOIP calls through voice packet traffic, which means we can implement our loss-based admission technique without imposing any new requirements on the existing VoIP system.

Another Alternative for end to end measurement is to use a policy-based admission control such as leaky bucket or a window-based mechanism for local measurement of the traffic.

Alwakeel and Prasetijo (2009) present a study for VOIP admission control using the leaky bucket scheme.

The importance of this research is based on the increasing demand for high-bandwidth applications, such as VOIP, video-on-demand, and grid computing, which are reviving interest in bandwidth reservation schemes such as the one proposed in this paper (Turner et al., 2010). In addition, it investigates an important concern for service providers deploying VoIP infrastructure, which is the provision and maintenance of high-quality voice services to their clients. This requirement becomes even more challenging when VoIP technologies are used to provide voice services to remote network sites over heterogeneous networks (Alawieh et al., 2008). Due to the inherently statistical nature of VoIP calls, it is usually difficult to provide QoS guarantees in random call-arrival scenarios. In order to provide QoS guarantees, there is a need to check on the flow load or the number of calls in the system. Call admission control is the only feasible solution that performs this function.

The rest of this paper is organized as follows: Section 3 presents a description of the EMBAC-L scheme, and discusses the rejection probability function. Section 4 presents the performance study and the performance results. Finally, conclusions are covered in the last section.

3. EMBAC-loss based (EMBAC-L)

In this section, we present the new loss-based approach (EMBAC-L) designed to enhance the End-to-end Measurement Based Admission Control scheme. The goal of EMBAC-L is to guarantee that the average packet loss rate of voice flows in progress is no more than a certain target. In the EMBAC-L mechanism, active end-to-end measurement is used to judge if the network can accept a new call, which arrives at an end node. Consider a probe packet flow that is transmitted from end node A (source) to end node B (destination). Assume that node A is in charge of the admission test, which judges whether or not to accept the flow from node A to node B. In order to do this, node A has a pre-determined admission threshold, and determines whether the admission test result is a success or failure, where success means that the measured packet loss rate for the probe packet flow is no more than the admission threshold, and failure means otherwise.

EMBAC-L implements the admission control using a function of the probe packet loss rate $f(y)$, where y is the computed probe packet loss rate. If the probe packet loss rate y is less than the predetermined threshold value x, then the new call request will be admitted, otherwise the new arrival call will be rejected with probability distribution function $f(y)$. Over time, more calls will be admitted into the network until the threshold value is reached, where more new calls will impact on the performance of the network.

The EMBAC-L scheme was studied with different functions for $f(y)$, as listed in Table 1. By changing this function, the scheme varies from a conventional EMBAC scheme to the deterministic policy (EMBAC-D), and to the loss-based control admission (EMBAC-L). These may implement various linear or non-linear distribution functions for $f(y)$ to provide the required QoS.

Various rejection probability functions are to be compared for the following reason. The main emphasis of this paper is to show the advantages of using a loss-based function instead of a threshold-based function. The VOIP admission control has to provide the best possible network performance that adapts to the varying network infrastructure and load. The EMBAC-L rejection probability function provides the desired bandwidth reservation that allows the provision and maintenance of high-quality voice services to the network users. To determine the set of scheme operational parameters, we, therefore, need to search for the probability function that maximizes the aggregate calls throughput of the network.

3.1. EMBAC-L queuing and network simulation model

In this section, we validate the performance of the EMBAC-L scheme used in the call source gateway. A simulation model was developed in C++ to evaluate the performance of the EMBAC-L scheme. In particular, we investigated the various threshold functions of $f(y)$ to find the scheme parameters that can be used to provide QoS acceptable for VoIP networks.

In the simulation study, two phases of Endpoint Admission Control were simulated: the probing phase and voice packets transfer phase. In the probing phase, the source gateway maps the QoS requirements to a network service, and starts a probing process to obtain information about its performance. During the probe phase period, the number of the probe packets lost is counted. If the probe packet delay exceeds 250 ms (any delay less than this value is not noticeable in human-to-human conversation (Tsetsgee and Lkhagvasuren, 2008), the probe packet is considered as a lost packet. Depending on the probe loss, the source gateway determines whether or not to admit the call into the network. A new call will be accepted or rejected based on the new proposed admission control. When a call is accepted, the voice packet phase is simulated and various performance measures are then collected.

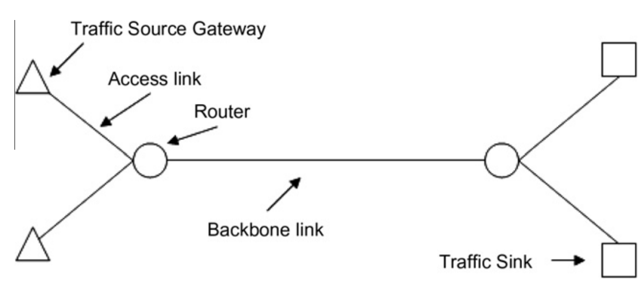

Figure 1 EMBAC-L network model.

Table 1 EMBAC probability functions.

$P(y)$	Scheme	Action
$P(y) = 0$	No admission control	Accept all calls
$P(y) = 1$	EMBAC-D	Reject all calls when threshold is exceeded
$P(y) = 1 - f(y)$	EMBAC-L	Reject with probability P (function of the loss y)

The network topology model used in this study is shown in Fig. 1. The model is composed of end nodes, intermediate nodes, and links connecting nodes. The model's traffic-source gateway generates three traffic service classes:

- Voice – This service class is meant for voice calls that require high QoS. Each voice call request accepted by admission control is guaranteed a high priority over the transmission line.
- Probe – This service class has a medium priority. The probe packets are generated during the probe phase period in order to do the active monitoring and provide link statistics.
- Best effort – This service category is for non real-time traffic. Applications, such as web browsing, email, FTP, remote login, database access, etc. fall under this category. In our model these data classes share the network bandwidth with voice and probe packets. At the edge source router, the voice, best effort, and probe packets are classified and served depending on their set priority.

The traffic queuing model used in this study is shown in Fig. 2, which shows the process of an incoming call (call-queuing model) as well as the packet queuing model.

Once a new call arrives, the probe phase starts, and if the call is accepted (based on the feedback of the loss rate), then the call will be moved to the voice gateway server and the voice packet phase will start. The voice packets are generated during the voice packet phase cycle, and continue to be generated until call termination. The source edge router transfers the arrival packet to the destination over the backbone link. Packets are serviced according to their priority, with voice packets having the highest priority.

Fig. 2 also depicts the traffic packets queuing for three different types of packet (voice, probe, and best effort) from source to destination. In this model, we assumed non-pre-emptive queuing with a different priority class. It is also worth noting that although our simulation model is based on a one-way VOIP call setup, EMBAC-L, however, can be implemented with a full duplex two-way VOIP call setting as well. The simulation study parameters are as follows:

Voice call arrivals are generated according to Poisson distribution, with a mean value equalling 30 ms. Call duration is exponentially distributed, with a mean value of 3 min.

The probe packets are generated deterministically every 70 ms. The probe phase cycle for each new arrival call is 1 s. Upon call acceptance, voice packets are generated according to the IPP process. In an IPP model, each voice source is characterized by ON and OFF periods (Alawieh et al., 2008). The mean ON talk duration is anywhere between 0.35 and 1.5 s, while the mean silent OFF period duration is anywhere between 0.65 and 2.25 s. During the ON period, voice packets are generated periodically at a constant rate of 64 Kbps. The link propagation delay is assumed to be equal to 5 μs/km, and the link length is 100 km. The voice packet has a length of 53 bytes, and has 3.3 ms transmission time.

3.2. EMBAC-L probability admission function

To evaluate the impact of the admission scheme proposed in this paper on the behaviour of the scheme, various functions

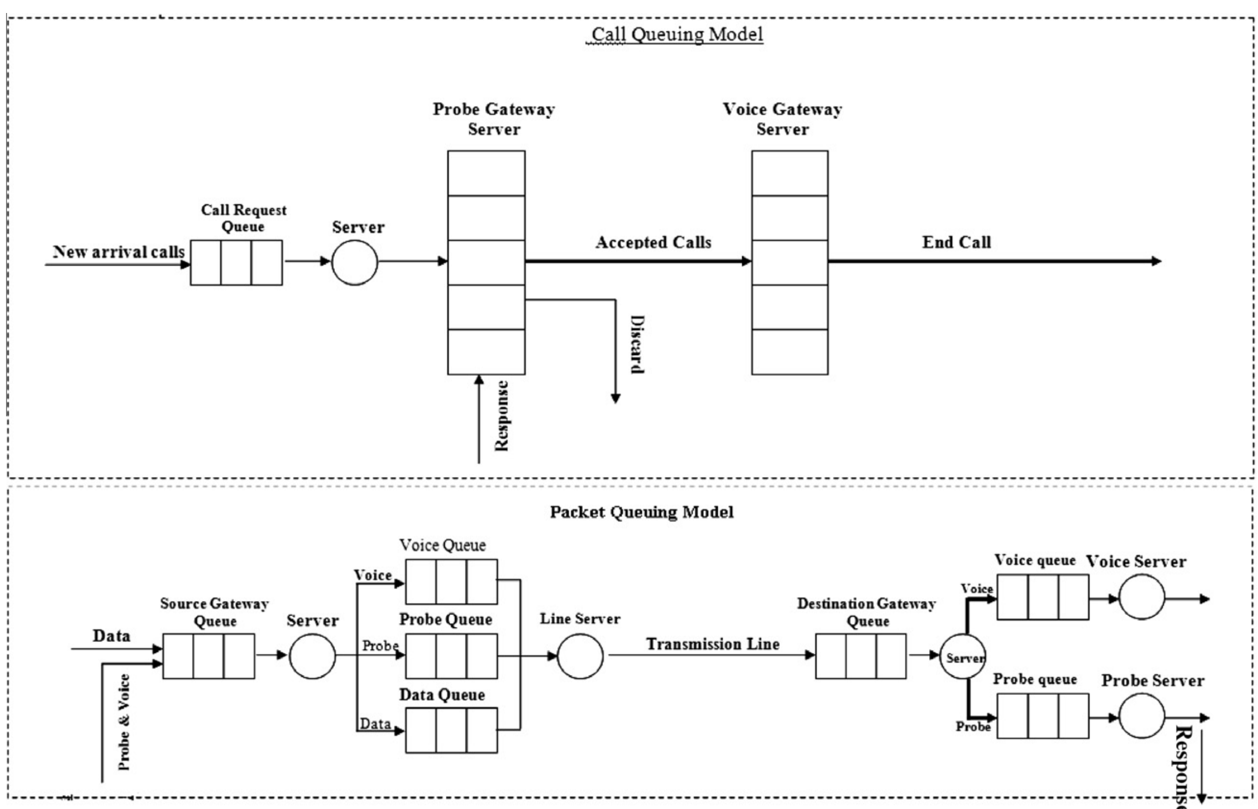

Figure 2 Traffic queuing model.

were used for the scheme's probability function. The criterion for selection is that the function should not exceed unity and cannot have negative values, thus it is necessary to set the value of probability between 0 and 1. Linear, non-linear, exponential and sinusoidal distribution functions are used during the simulation to obtain the operational setting for providing acceptable QoS.

The following distribution functions $f(y)$ were investigated, including:

– Linear function $f(y) = y$
– Square function $f(y) = y^2$
– Exponential function $f(y) = 1 - e^{-y}$
– Sinusoidal function $f(y) = sin(y)$

It should be noted that a linear probability function $f(x)$ is used with the EMBAC-P mechanism, with x being the threshold value, while $f(y)$ is used with EMBAC-L, which is a function of the packet loss rate. In our study, the EMBAC-L mechanism assumes two scenarios, one with a decreasing function equalling $1 - f(y)$, and the other one with an increasing function equalling $f(y)$. So, one scenario uses a probability function $P = 1 - f(y)$, and the other one uses $P = f(y)$. Besides, we are testing a non linear function for $f(y)$ to make the call blocking probability increment rate much higher than the loss increment rate for $f(y)$ scenario. For $1 - f(y)$ scenario the probability increment rate will be much smaller than the loss increment rate. Thus, with a non linear probability function, the rejection probability will have a higher sensitivity for loss rate change compared to a linear function.

4. Performance measures and results

In this section, the proposed approach's (EMBAC-L) performance is verified and evaluated. Also, a comparison is undertaken on its performance using various functions. The admission threshold and rate of traffic are set as input parameters, and during the simulation run, the probe packet loss is estimated and used as an input parameter for the admission control scheme.

4.1. Performance measures of EMBAC-L

Different measures were studied in the evaluation of the performance of the proposed scheme. The average call blocking rate is one of the more important factors, which gives an indication of how this scheme impacts on QoS. The average probe packet loss was also studied in our simulation. The probe packet loss rate is defined as the total number of probe packets lost divided by the total number of probe packets generated during the simulation run, averaged over all the calls.

The average voice packet loss is the most important factor for indicating whether or not the scheme has achieved its goal with respect to acceptable QoS. Voice packet loss occurs whenever there is a voice packet lost due to buffer overflow at the source gateway queue, or when the delay exceeds 250 ms across the network. The next section presents the performance results in terms of these measures.

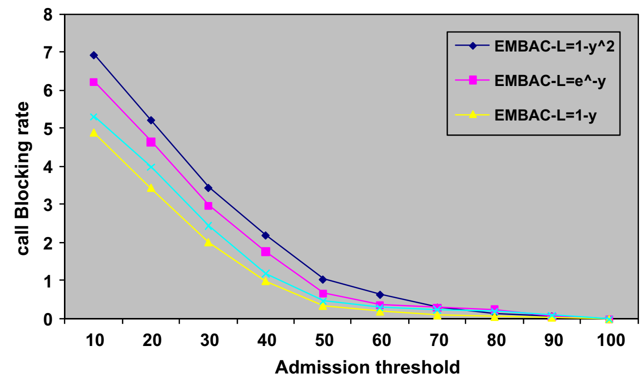

Figure 3 Blocking call vs. admission threshold.

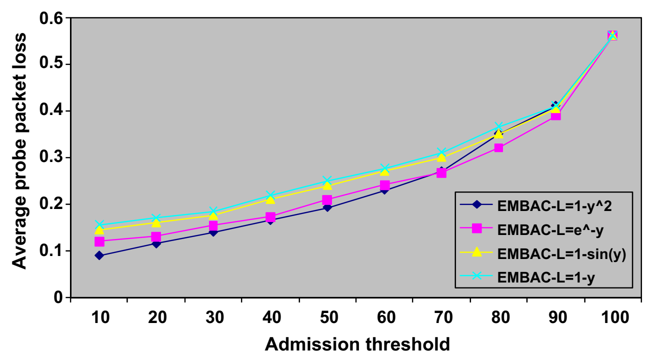

Figure 4 Average probe packet loss vs. admission threshold.

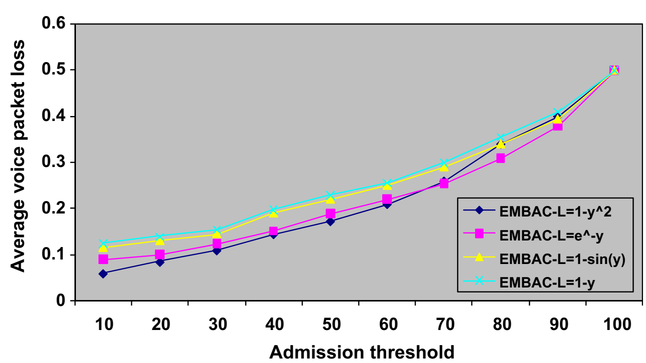

Figure 5 Average voice packet loss vs. admission threshold.

4.2. Performance results of EMBAC-L

Fig. 3 shows the call blocking rate versus admission threshold with different distribution functions. The admission threshold range is from 10% to 100%. As can be observed, the blocking rate tends to decrease as the admission threshold increases, as expected. The probability function with $f(y) = 1 - y$ gives the best results compared to other functions.

Fig. 4 shows the probe packet loss (in milliseconds) versus threshold. As shown in the figure, the probability of the rejection function, with $P = 1 - y$, results in the highest probe loss, while $P = 1 - y^2$ achieves the lowest probe packet loss.

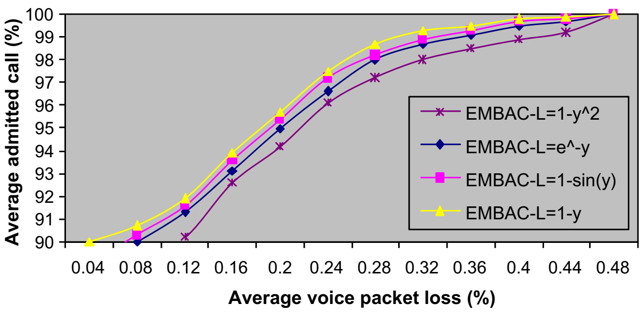

Figure 6 Average admitted calls vs. average voice loss rate for EMBAC-L ($P = 1 - f(y)$).

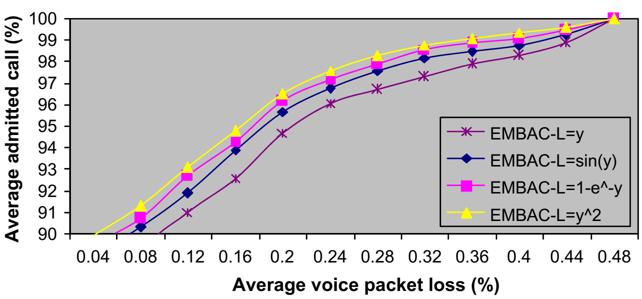

Figure 7 Average admitted calls vs. average voice loss for EMBAC-L ($P = f(y)$).

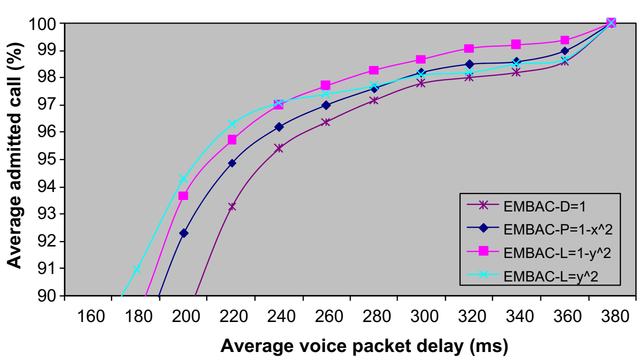

Figure 8 Average voice delay vs. average admitted calls.

In Fig. 5, we evaluate the performance of our scheme in terms of the voice packet loss (in milliseconds) for four types of probability rejection functions. This is another examination of the performance of the EMBAC-L scheme.

As shown in the figure, the highest voice packet loss is obtained when the probability of the rejection function is $P = 1 - y$.

In the previous figures, we presented the advantage of the scheme in providing a lower call rejection rate. In what follows, we examine the trade-off in the performance of the EMBAC-L in terms of network call admission rate and voice packet loss. A higher call rate and a lower packet loss rate mean better network performance.

Fig. 6 shows the EMBAC-L average call admission rate versus the voice packet loss rate, with various probability rejection functions P. As shown, the rejection function with $P = 1 - y^2$ admits fewer calls compared to other rejection functions at the same voice packet loss. The rejection function with $P = 1 - y$ has the highest admission call rate.

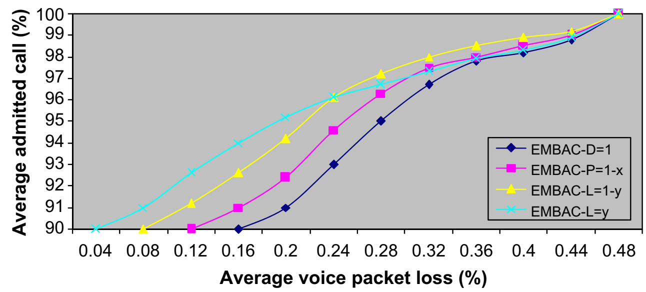

Figure 9 Average voice delay vs. average admitted calls for EMBAC-L and EMBAC-P.

In Fig. 7, EMBAC-L is evaluated with rejection function $P_y = f(y)$. As shown in the figure, the rejection function with $P = y^2$ admits more calls at all voice packet loss rates.

Fig. 8 shows a comparison between EMBAC-L and EMBAC-P. The figure plots the voice packet delay in milliseconds as a function of the admitted call rate. Both EMBAC-L with a probability function of $1 - f(y)$ and EMBAC-L with a probability function of $f(y)$ perform better when compared with EMBAC-P scheme. The square function (y^2) has been used for $f(y)$ in this figure.

The EMBAC-L performance with a probability function of $1 - y$ is shown in Fig. 9. Again, EMBAC-L has an improved performance when compared to both the conventional EMBAC-D scheme and EMBAC-P, with a threshold function of $1 - x$.

Based on these performance results we may state the following:

The linear probability function with $f(y) = 1 - y$ (y = loss rate) gives the lowest call rejection results compared to other functions. However, it also results in the highest voice packet loss of all functions. The non-linear probability function $P = 1 - y^2$ achieves the lowest probe and voice packet loss. Therefore, by using either a linear on a non-linear function, we have a trade-off in the performance of EMBAC-L in terms of network call admission rate and voice packet loss. A trade-off also exists between voice packet loss and voice packet delay. Regarding the EMBAC-L mechanism scenarios (using probability function $P = 1 - f(y)$, and $P = f(y)$), the first achieves better performance when the call admission rate is relatively small or moderate. At higher call admission rates, the $P = f(y)$ scenario should be used as it results in a better performance. Overall, the proposed EMBAC-L scheme has an improved performance when compared to both the conventional EMBAC-D scheme and EMBAC-P, for all probability functions. The size of the improvement achieved is relative and depends on the target performance measure. A 5% improvement in call admission rate over EMBAC-D and about 3% over EMBAC-P (see Figs. 8 and 9) can be achieved. This represents 50,000 more calls for a network with a million customers.

5. Conclusion

In this paper we have proposed the EMBAC-L scheme for QoS in VOIP networks. The main objective of this study has been to verify whether the new scheme can provide better QoS over IP networks, and to determine the various factors that impact on its performance. The performance of the scheme was evaluated using different distribution functions in order to find the admission scheme parameters that can provide the required acceptable QoS. Two different scenarios were studied using simulations, one with a decreasing rejection probability function, and the other with an increasing one. More distribution functions could be studied in future work.

This study of the EMBAC-L scheme shows that its performance provides an enhancement to the original EMBAC scheme. Compared to EMBAC-P and EMBAC-D, our scheme consistently achieves better performance in terms of call admission throughput, and also achieves lower voice packet delay for various call arrival rates.

References

Alawieh, B., Ahmed, R., Mouftah, H.T., 2008. Performance measurement for voice services in heterogeneous wired networks. In: Proceedings of the 4th ACM-sponsored International Conference on Testbeds and Research Infrastructures for the Development of Networks and Communities, Innsbruck, Austria.

Alwakeel, S., Prasetijo, A., 2009. A policy-based admission control scheme for voice over IP networks. Journal of Computer Science 5 (11), 817–821.

Bianchi, G., Capone, A., Petrioli, C., 2000. Throughput analysis of end-to-end measurement based admission control in IP. INFOCOM 2000 3, 1461–1470.

Bilhaj, A.A., Mase, K., 2004. Endpoint admission control enhanced systems for VoIP networks. In: IEEE Symposium on Applications and the Internet (SAINT '04), January 26, Tokyo, Japan, p. 269.

Chen, X., Wang, C., Xuan, D., Li, Z., Min, Y., Zhao, W., 2003. Survey on QoS management of VoIP. In: Proceedings of the IEEE International Conference on Computer Networks and Mobile ICCNMC.

Chua, T.-K., Pheanis, D.C., 2006. QoS evaluation of sender-based loss-recovery techniques for VoIP. In: IEEE Network.

Friedman, T., Caceres, R. Clark, A., 2003. IETF RFC 3611: RTP Control Protocol Extended Reports (RTCP XR). In: The Internet Society.

Li, B., Hamdi, M., Hou, Y.T., Jiang, D., Cao, X.-R., 2000. QoS enabled voice support in the next generation internet: issues, existing approaches and challenges. IEEE Communications Magazine 38, 54–61.

Markopoulou, A., Tobagi, F., Karam, M., 2003. Assessing the quality of voice communications over internet backbones. IEEE/ACM Transactions on Networking (TON) archive 11 (5), 747–760.

Mase, K., 2004. Toward scalable admission control for VoIP networks. IEEE Communications Magazine 42 (7), 42–47.

Mase, K., Kobayashi, H., 2004. An efficient end-to-end measurement based admission control for VoIP networks. IEEE International Conference on Communications 4, 1877–1881.

Mase, K., Toyama, Y., 2002. End-to-end measurement based admission control for VoIP networks. IEEE International Conference on Communications, ICC 2, 1194–1198.

Mase, K., Toyama, Y., Bilhaj, A.A., Suda, Y., 2001. QoS management for VoIP networks with edge-to-edge admission control. In: GLOBECOM 2001.

Tran, H.T., Ziegler, T., Ricciato, F., 2003. QoS Provisioning for VoIP Traffic by Deploying Admission Control. Springer Publisher, Berlin.

Tsetsgee, B., Lkhagvasuren, T., 2008. The study on quality requirements of interactive voice in IP network. In: Third International Forum on Strategic Technologies, IFOST 2008, Novosibirsk-Tomsk, June 23–29, pp. 314–318.

Turner, D.M., Prevelakis, V., Keromytis, A.D., 2010. A market-based bandwidth charging framework. ACM Transactions on Internet Technology (TOIT) 10 (1).

Zhang, L., Yu, K., Du, Y., Wen, X., 2008. Improving capacity and QoS of VoIP in IEEE 802.11e wireless LANs. In: Proceedings of the International Conference on Mobile Technology, Applications and Systems table of contents, Yilan, Taiwan.