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A Policy-Based Admission Control Scheme for Voice over IP Networks

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Abstract: Problem statement: In Voice Over IP (VOIP) network, when more calls are admitted to the network, more voice packet traffic is created. Since bandwidth is always limited, this may result network congestion and/or may affect voice quality. Thus, we needed a mechanism for improving the Quality of Service (QoS) by controlling VOIP calls admission. **Approach:** Given a specified bandwidth and a constant background data rate, we attempted to explore the effect of Open Window and Leaky Bucket admission schemes on VoIP calls quality. These policy-based admission controls were simulated using NS-2 Simulator. The inter-arrival time distribution for the network background data traffic was assumed to be deterministic with a Constant Bit Rate (CBR). Voice packets traffic inter-arrival time is assumed to have an exponential distribution. Each voice call has a rate of 64 kbps for duration of 120 min. **Results:** Various performance measures of VoIP calls and packet traffic were evaluated including: packet loss, packet drop rate, delay, jitter and call rejection rate. Performance results of the experiment are summarized in a power ratio index which presented the impact of a collection of performance parameters on VoIP service quality. **Conclusion:** Implementing a policy based admission scheme on VoIP network will improve its QoS and the degree of improvement depends on the network setting parameters. If threshold rate for call admission is set above network ceiling bandwidth, leaky bucket will result a higher and unacceptable jitter. Overall, leaky bucket scheme was considered inferior when compared to open window for improving QoS of VoIP.

Key words: Voice over IP, Call admission control, Leaky bucket, Open Window Policy schemes

INTRODUCTION

Voice and video transmission over telecommunication networks requires specific performance quality. If such quality is not maintained, the receiving end will then suffer-e.g., the received video freezes or there will be unacceptable delay in voice. Similarly, transmitting voice over IP networks will have the same challenge. With this in mind, a call admission controller in VoIP networks is needed to maintain voice quality over a limited bandwidth link. Call admission control will determine if a call will be accepted or rejected based on network resource availability.

Several admission mechanisms are available for Call Admission Control (CAC) over the Internet. Example of these are IntServ architecture which uses RSVP signaling protocol for reserving resources in a router[1] and EMBAC protocol which use probes transmission to estimates networks state from sender to receiver[2-4]. Other techniques for conducting call admission control are based on diffusion approximation which calculates bandwidth for a number of connections with given cell loss requirement[5]. Various CAC schemes were also developed for ATM networks[6]. Besides, CAC mechanisms are considered for wireless networks and in IEEE 802.11e standard environment to enhance its performance[4,7].

In this study, we apply packet admission control schemes currently in use for ATM cell switching networks, to improve VoIP traffic QoS. Two schemes, namely leaky bucket and open window are simulated to evaluate their impact on VoIP performance in term of packet delay, jitter, call drop and VoIP packet loss.

VoIP policy scheme description: This study proposes a policy-based admission control scheme for VoIP traffic. With such a policy, a VoIP server will check the availability of bandwidth every time when there is a new call request. Two policy-based schemes were investigated namely: Open Window and Leaky Bucket. Both mechanism works as follows: First, they estimate the average network traffic rate and compare it to a threshold rate set by the server. If the average traffic rate is less than the threshold, the system will admit the

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call. Otherwise, the system rejects the call request. Clearly, if the call request rate increases, the voice packet traffic will also increase.

Background data traffic is assumed to exist in the system in addition to VOIP packets. The admission policy scheme is applied on both types of traffic.

The network traffic is estimated using a simple moving average method, whenever a new call request arrives. The average network traffic is then compared to the threshold rate to admit the call. The detailed scenario works as follows:

Voice call requests arrive with a rate of λ_{CR}. Upon its arrival, the call requests are queued and examined to determine whether it will be granted the right to use the network resource or not. When a call request is accepted, the call packet source generates voice traffic stream using voice packet communication. Thus, more voice traffic is generated as the number of accepted calls increases. The network traffic is the VOIP packets traffic in addition to background data traffic, generated from independent network data sources with overall rate of λ_{BD}. When the average network traffic λ_{AVG} which is the simple moving average of the sum of λ_{CR} and λ_{BD}) achieves a predefined threshold rate (λ_{THRES}) set in advance by the call network server (or gateway), the call server has then to reject the new incoming call requests.

A Packet Admission Controller (PAC) is applied to further process both background data and voice traffic. This PAC has limited queue and a tester. Both voice and data traffics are first queued in the controller and the specified policing mechanism-open window or leaky bucket- is then applied. After this, the shaped traffic is tested to determine the average traffic rate value λ_{AVG}. This rate value is fed back to the call server, where the decision is made-to accept or reject the call request.

MATERIALS AND METHODS

The system is simulated using NS-2 Simulator. The simulation method is modeled as depicted in Fig. 1.

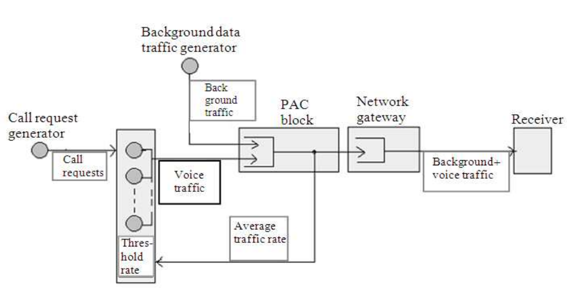

Fig. 1: Model of VoIP system

VoIP server generates both of voice traffic in addition to the background data traffic.

Call request is exponentially generated and is stored in the call-queue. After passing the queue a voice call server examines the call requests based on bandwidth availability. If call is admitted, the packet generator produces voice packets and transmits the packets to the receiving end. While being still in the PAC block, the flow packets are also marked to estimate the average traffic rate (λ_{AVG}). The PAC itself has limited queue and implements one of the policing mechanisms: Open window or the leaky bucket.

Traffic flows leaving the PAC and the network gateway is then forwarded to the receiver. The receiver also has a limited queue. The average traffic rate is also monitored at the receiving end.

Based on the above model, five nodes are defined in the NS-2 environment[8] as shown in Fig 2. These are described as follows:

- Node 0 represents both of the call request generator and the voice call server. The call server stores threshold value-λ_{THRES}. If the system average rate λ_{AVG} reaches the value of λ_{THRES}, the system will reject the incoming call request
- Node 1 is the background data traffic generator. This node generates background data packets with a fixed rate of λ_{BD}
- Node 2 and Node 3 implement the policy methods (Packet Admission Controller (PAC)), at the system input and output respectively. Node 2 inputs are the background data and voice call packet traffic. The PAC has a tester to estimate the average packet traffic rate λ_{AVG} which will be fed-back to voice call server
- Network gateway represents a transmission line with a defined amount of bandwidth and queue size
- Node 4 is for the receiver end. Here, the output average traffic is also monitored. The receiving end (N4) is merely a sink node to drop any packets originating from N0 or from N1

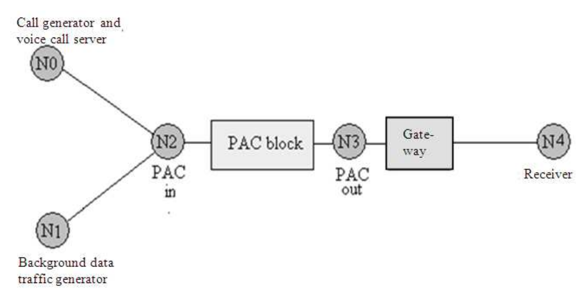

Fig. 2: Method Model representation in NS-2

Three simulation experiments are conducted. One is with no-policy applied in PAC and the other two are with the implementation of open window and leaky bucket schemes. The bandwidth and queue size for every associated node is kept the same for every experiment. We set the Constant Bit Rate (CBR) background data traffic to be fixed at 500 kbps at all times.

The generated voice traffic has an exponential inter-arrival distribution at a rate of 64 kbps for duration of 120 sec.

Using PAC, the system bandwidth has a ceiling of 3 Mbps and its queue is limited to 10,000 packets. The network transmission gateway has a bandwidth similar to the PAC ceiling rate and queue limit of 10,000 packets as well. Network delay is set to 0 to simplify the calculation of processing delay. We simulated every experiment for 500 sec with call request rate set to 1 sec.

With both open-window and leaky-bucket, we run the simulation experiment with threshold rate values in the range 2.90-3.10 Mbps. Open-windows has a ceiling bandwidth defined by the transmission link bandwidth of N2-N3 and equals to 3 Mbps. Besides, it has a queue size equals to 10,000 packets. While in leaky-bucket the ceiling bandwidth is determined by its token rate which is set to 3 Mbps. The experiment burst-rate is set to 100 kbps with a queue size of 10,000 packets.

To compare various policy CAC schemes, we introduced the power ratio measure, which describes mutual contribution from various performance measures of the VOIP traffic. The PowerRatio index is given by:

$$\text{PowerRatio} = \frac{(1-\text{pktLoss})(1-\text{pktDrop})(1-\text{callRject})}{(\text{AveragePktDelay} * \text{AveragrPktJitter})}$$

In the formula above, packet loss, packet drop and call reject rate is represented in percentage. While packet delay and delay jitter is normalized to their maximum allowable value which are of 150 and 40 ms respectively.

RESULTS

First experiment is conducted by just letting both background and voice traffic passes through the PAC without any control applied. This creates traffic flowing as depicted in Fig. 3.

In the no-policy experiment, the average power ratio is found equal to 0.029. This result indicates that, with no admission control for packets arriving at the network; the performance will be an unacceptable for VoIP traffic.

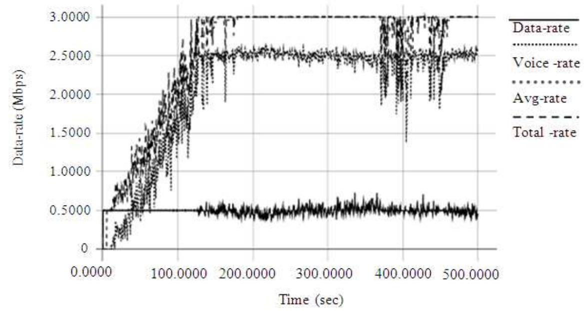

Fig. 3: Traffic flow from no-policy experiment

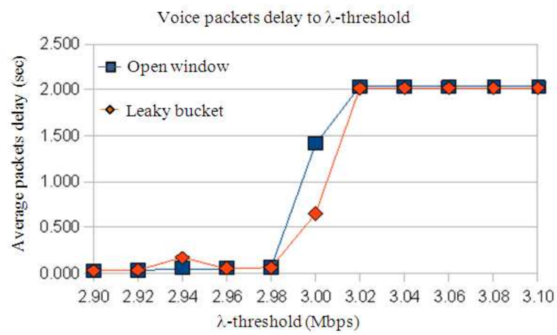

Fig. 4: Average voice packet delay versus threshold rate

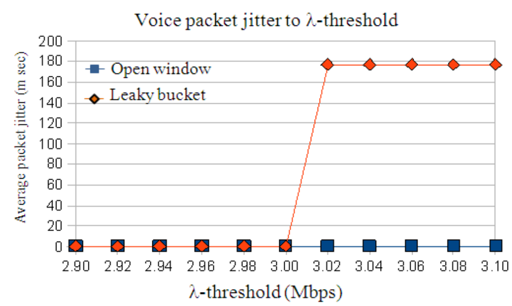

Fig. 5: Average packets jitter versus threshold rate

Figure 4 compare the delay performance results of open window and leaky bucket.

Voice jitter performance measure is shown in Fig. 5.

For packet loss, we found that leaky bucket reduces the loss to virtually zero, while open window still have a packet loss of 0.45% at 3 Mbps ceiling bandwidth. (Fig. 6).

Figure 7 shows the probability of voice packet drops. As shown, no significant difference exist at any threshold rate value. Minor variations however occur, but this does not reflect a significant packet drop performance difference.

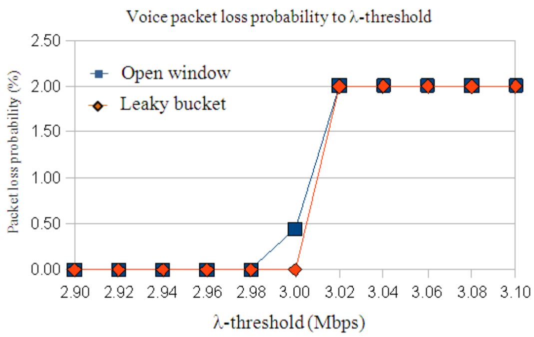

Fig. 6: Voice packet loss probability versus threshold rate

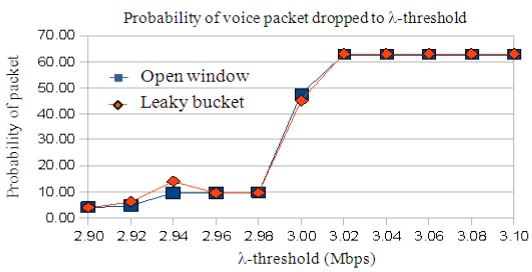

Fig. 7: Voice packet drop Probability versus threshold rate

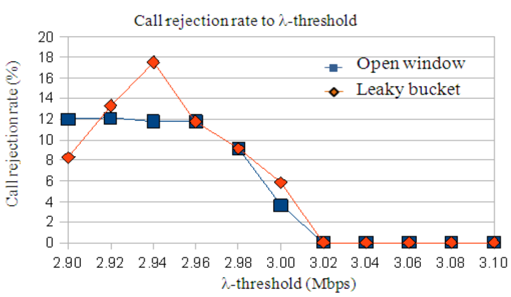

Fig. 8: Call rejection rate to threshold rate

Figure 8 shows the variation of the call rejection rate with the threshold rate. As shown, there is a clear variation in the total number of calls rejected as a function of the threshold rate, in both policies experiments.

Figure 9 and 10 show the total number of packets processed in PAC as well as total number of calls examined. As shown, in general, there is no significant difference between open window and leaky bucket methods regarding number of call processed by the schemes.

The power ratio of the two mechanisms is depicted in Fig. 11.

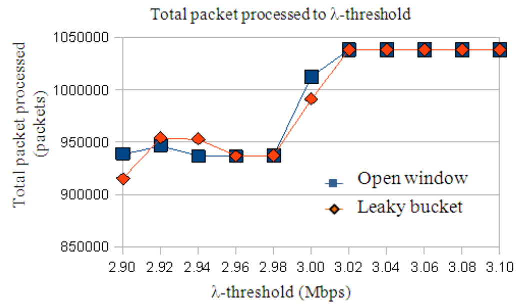

Fig. 9: Total packet processed versus threshold rate

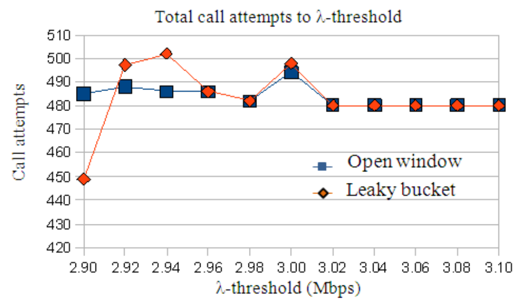

Fig. 10: Total call attempts versus threshold rate

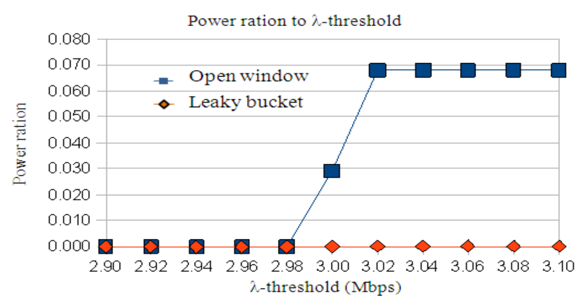

Fig. 11: Power ratio to threshold rate

DISCUSSION

With no-policy scheme in PAC, the average delay obtained is unacceptable and equal to (1.41) sec. Packets marked to have more than the maximum acceptable delay of 150 ms[11] represent 47.43% of all packets. This is considered unsatisfactory for a VoIP system. The average packet jitter is about 30 ms which is acceptable. However, the maximum packet jitter obtained in this experiment is extremely large (1613) sec. With no PAC policy, call reject rate is however acceptable with an average value of 3.64% for 494 call requests.

If threshold rate is set at the ceiling bandwidth, leaky bucket reduces packet delay by more than 50%

compared to the delay produced with open-window (from 1.41 sec to 646 m sec). Various threshold rate values do not have a significant impact on the leaky-bucket performance when compared with open-window.

When threshold rate exceeds ceiling bandwidth, the jitter increases to 176.95 m sec even if the leaky-bucket scheme is applied.

Both leaky-bucket and open-window has the same characteristics for packet loss probability when threshold rate exceeds its ceiling bandwidth

For other performance measures, we found that there is no significant difference between leaky-bucket mechanism and open window.

Also, from the power ratio measure result, we notice that, if the threshold rate is set below ceiling bandwidth, the total impact on the performance of various measures represented by the power index is non significant. This proves that with low threshold rate, the CAC scheme has no significant impact wither open-window or leaky-bucket is applied. The value of power ratio measure however, increase dramatically with open window when threshold rate is set to ceiling bandwidth or greater.

CONCLUSION

CAC policy is needed to manage VOIP traffic in order to achieve the required QoS, in terms of the average packet delay and jitter measure. With no-policy scheme implemented, the average packet delay is very high, about 1.2 sec and the voice packet drop rate is almost half of the voice packet generated rate. For this case, the voice packet drop probability equals to 47.43% at ceiling bandwidth. For the jitter measure, when threshold rate is set above ceiling bandwidth, leaky bucket results a high jitter which is not acceptable in voice calls.

Overall, using the power ratio measure, leaky bucket scheme is considered inferior when compared to open window for improving QoS of VoIP at variable threshold rate. We can therefore conclude that implementing of open window CAC scheme is preferred near system congestion state to significantly improve the QoS for VoIP traffic.

REFERENCES

1. Hill, R.L. and H.T. Kung, 2001. A DiffServ enhanced admission control scheme. Proceeding of the IEEE Global Telecommunications Conference, Nov. 25-29, IEEE Xplore Press, Washington DC., USA., pp: 2549-2555. DOI: 10.1109/GLOCOM.2001.966236
2. Mase, K. and H. Kobayashi, 2004. An efficient end-to-end measurement based admission control for VoIP networks. Proceeding of the IEEE International Conference on Communications, June 20-24, IEEE Xplore Press, Washington DC., USA., pp: 1877-1881. http://cat.inist.fr/?aModele=afficheN&cpsidt=17525071
3. Houck, D.J., E. Kim, H. Uzunalioglu and L.A. Wehr, 2003. A measurement - based admission control algorithm for VoIP. Bell Labs Tech. J., 8: 97-110. http://www3.interscience.wiley.com/journal/105557808/abstract?CRETRY=1&SRETRY=0
4. Khalifeh, A., 2008. An end-to-end measurement-based admission control policy for VoIP over wireless networks. Proceeding of the IEEE Wireless Communications and Networking Conference, Mar. 31-Apr. 3, IEEE Xplore Press, Washington DC., USA., pp: 2992-2996. DOI: 10.1109/WCNC.2008.523
5. Elsayed, K.M.F. and O.E. Hendy, 2003. A measurement-based call admission control scheme for ATM networks based on the diffusion approximation, Comput. Commun., 26: 622-630. http://www.ingentaconnect.com/content/els/01403664/2003/00000026/00000006/art00160
6. Bae, J.J. and T. Suda, 1991. Survey of traffic control schemes and protocol in ATM network. Proc. IEEE, 79: 170-189. http://ieeexplore.ieee.org/xpl/freeabs_all.jsp?arnumber=64405
7. Oh, S., J. Shin, D. Kwak and C. Kim,2008. A novel call admission control scheme for the IEEE 802.11e EDCA. Proceeding of the 10th International Conference on Advanced Communication Technology, Feb. 17-20, IEEE Xplore Press, Okinawa, Japan, pp: 1832-1835. DOI: 10.1109/ICACT.2008.4494141
8. NS-2 website, http://isi.edu/nsnam/ns/