A New Strategy of Admission Control for Video Transmission

Naveen Hemrajani
Suresh Gyan Vihar University , Jaipur
India

Dharm Singh
College of Technology and Engineering, MPUAT
Udaipur, India

Ekta Menghani
MGIAS, JECRC Campus
Jaipur

ABSTRACT
Admission control is a network Quality of Service (QoS) procedure. Admission control determines how bandwidth and latency are allocated to streams with various requirements. Admission control schemes therefore need to be implemented between network edges and core to control the traffic entering the network.
An application that wishes to use the network to transport traffic with QoS must first request a connection, which involves informing the network about the characteristics of the traffic and the QoS required by the application. This information is stored in a traffic contract. The network judges whether it has enough resources available to accept the connection, and then either accepts or rejects the connection request. This is known as Admission Control.
Here in our project we are concerning for the different measurement based admission control algorithms used for this purpose. For this we will discuss four measurement based admission control algorithms, such as Hoeffding Bounds, Acceptance Region and Measured Sum. Further a new algorithm is proposed using priority measurement based on the round trip times of the nodes. we will show the output of the algorithms using the NS-2 simulator and compare the simulation output of the four measurement based admission control algorithms using Nam. We will simulate using X-graph to find the throughput and utilization of the bandwidth. Further these parameters are compared performing the video transmission on all the four admission control algorithms,

General Terms
Video Communication, Load Balancing, Network Simulator, Wireless Network communication et. al.

Keywords
Video Transmission, Admission Control, bounded delay, link utilization, intricate collisions.

1. INTRODUCTION
Admission control is a network Quality of Service (QoS) procedure. Admission control determines how bandwidth and latency are allocated to streams with various requirements. Admission control schemes therefore need to be implemented between network edges and core to control the traffic entering the network.

An application that wishes to use the network to transport traffic with QoS must first request a connection, which involves informing the network about the characteristics of the traffic and the QoS required by the application. This information is stored in a traffic contract. The network judges whether it has enough resources available to accept the connection, and then either accepts or rejects the connection request. This is known as Admission Control.
Admission control is useful in situations where a certain number of connections (phone conversations, for example) may all share a link, while an even greater number of connections causes significant degradation in all connections to the point of making them all useless such as in Congestive collapse

Admission control is one of the main tasks that a Bandwidth Broker has to perform, in order to decide whether an incoming resource reservation request will be accepted or not. Most Bandwidth Brokers use simple admission control modules, although there are also proposals for more sophisticated admission control according to several metrics such as acceptance rate, network utilization, etc.
The role of any admission control algorithm is to ensure that admittance of a new flow into a resource constrained network does not violate service commitments made by the network to admitted flows. The service commitments made could be quantitative (e.g., a guaranteed rate or bounded delay), or it could be more qualitative (e.g., a "low average delay"). There are two basic approaches to admission control:

1) The first, which we call the parameter-based approach, computes the amount of network resources required to support a set of flows given a priori flow characteristics;
2) The second, the measurement-based approach, relies on measurement of actual traffic load in making admission decisions.

There have been many proposals for supporting real time applications in packet networks by providing some requests real-time service, it must characterize its traffic so that the network can make its admission control decision. Typically, sources are described by either peak and average [1] or a filter like token bucket [2] these descriptions provide upper bounds on the traffic that can be generated by the source. The traditional real time service provides a hard or absolute bound on the delay of every packet; in [1,3] this service model is called guaranteed service.

Admission control algorithms (ACA's) for guaranteed service use the a priori characterizations of sources to calculate the worst case behavior of all the existing flows in addition to the incoming one. Network utilization under this model is usually acceptable when flows are smooth; when flows are bursty, however guaranteed service inevitably results in low utilization [4,5]

Admission control is a network Quality of Service (QoS) procedure. Admission control determines how bandwidth and latency are allocated to streams with various requirements [6]. Thus this scheme needs to be implemented between network edges and core to control the traffic entering the network.

For the purposes of this study, we assume that applications use a signaling protocol, such as Resource Reservation Protocol (RSVP), to make their requests for service to the network. These service requests contain a traffic descriptor describing the worst case behavior of the application traffic. The traffic descriptor takes the form of a token bucket with parameters r and b denoting the token rate and bucket depth, respectively. We measure the quality of the service delivered in terms of packet drops.

Soft real-time services are typically intended to be scalable, therefore we only consider MBACs that require no per-flow state; that is, and the measurements are taken on the aggregate traffic, not on individual flows. Since measurement is done on the aggregate and admission control decisions are made on per flow, rather than a per packet basis, implementation overhead is not critical. Some admission control algorithms do not fit within the framework we consider and are excluded from our study.

For example, we do not include one of the MBACs described in because it depends on per-flow (rather than aggregate) measurements. In addition to excluding algorithms that require per-flow measurements, we also do not consider algorithms that make any assumptions, either implicitly or explicitly, about the average behavior of flows.

For example, we do not include the MBAC presented in because it computes a per-flow average estimate and assumes that all arriving and departing flows conform to that average. We only consider algorithms that make no assumption about what a flow's contribution will be to aggregate load beyond the worst case parameters supplied by the flow. Similarly, when a flow departs the network, its prior contribution to aggregate load can only be determined by measuring subsequent aggregate load.

There is a rapidly growing literature on MBAC's; before proceeding, we briefly review some of the more relevant work. The controlled-load service specification uses a very parsimonious source characterization; sources are described only by a peak rate and a token-bucket filter.

The performance of four admission control algorithms- one parameter based and three measurements based (measured bandwidth, acceptance region, and equivalent bandwidth) - for controlled load service is compared.

The main results of the comparisons are summarized below:-

1. In the operating region where losses occur under all MBAC's, they can all be induced to give the same loss-load curve by tuning their measurement parameters.
2. All the MBAC's studied perform similarly because they are all based on admission equations of the same form:

   $$V' < f(\cdot)\mu - g(\cdot)$$

   Where,
   V' is the measured load,
   μ is the link bandwidth, and
   f(-) and g(-) are functions of the source's reserved rate and number of admitted sources.
3. For immediate implementation of MBAC for controlled-load services, we recommend the following algorithm:

   $$V' < v\mu - kr$$

   Where,
   V' is a utilization factor,
   μ is the link bandwidth,
   k>0 a constant,
   r, the reserved rate of an incoming flow.

The performances of these algorithms, while somewhat insensitive to the form of the admission control equations, appears rather sensitive to changes in the parameters controlling the measurement process.

2. MEASUREMENT BASED ADMISSION CONTROL ALGORITHMS (MBAC

Types of Measurement Based Admission Control algorithms (MBAC):-

1. **Measured Sum**:- It uses measurement to estimate the load of existing traffic. This algorithm admits the new flow if the following test succeeds:

$$V' + r^{\alpha} < v\mu$$

where v is a user-defined utilization target as explained below, and V' the measured load of existing traffic.

2. **Acceptance Region tangent at origin:-** It computes an acceptance region that maximizes the reward of utilization against the penalty of packet loss.
Given link bandwidth, switch buffer space, a flow's token bucket filter parameters, the flow's burstness, and desired probability of actual load exceeding bound, one can compute an acceptance region for a specific set of flow types, beyond which no more flow of those particular types should be accepted.

3. **Acceptance region tangent at peak:-** A new flow is admitted by the network if the condition stated under satisfies :-
$$\eta p(1-e^{-sp)} + e^{-sp}v' \leq \mu$$

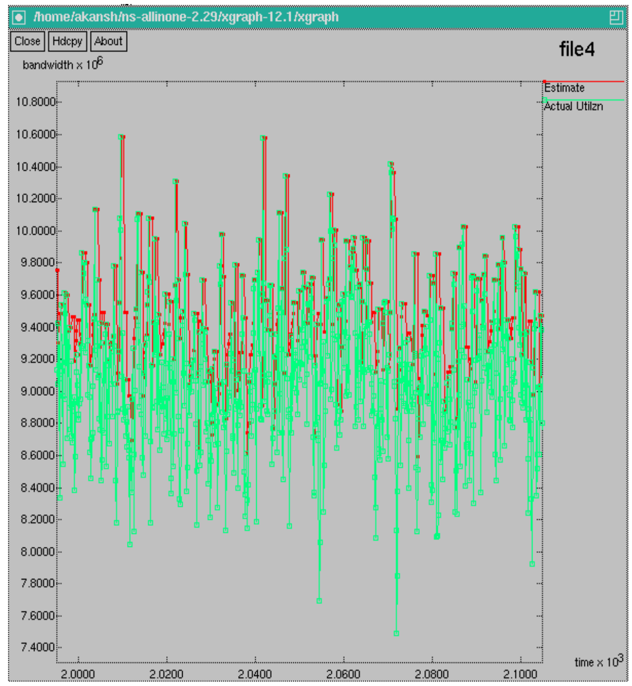
4. **Hoeffding Bounds(HB):-** It computes equivalent bandwidth for a set of flows using the Hoeffding bounds. The equivalent bandwidth of a set of flows is defined in references as the bandwidth C(£) such that the stationary bandwidth requirement of the set of flows exceeds this value with probability at most £(called as loss rate in this paper).
In an environment where large portion of traffic is best-effort traffic, real time traffic rate exceeding its equivalent bandwidth is not lost but simply encroaches upon best-effort traffic. In reference the measurement based equivalent bandwidth based on Hoeffding bounds (C_h) assuming peak rate (p) policing of η flows is given
by:
$$(C_h)\ (v,\{p_i\}\ 1\leq i\leq n,\ £\ ) = v' + \sqrt{}\ (\ln(1/£)\ \textstyle\sum(p_i)^2)\ /2$$
Where,
V' is the measured average arrival rate of existing traffic,
£ is the probability that arrival rate exceeds the link capacity.
It indicates that the measured average arrival rate may be approximated by measured average load.

3. EFFECT OF VARIOUS PARAMETERS ON RESULTS

1. The graph was drawn for the algorithm measured sum. Here, the parameter S i.e. sampling period is set to T/20, T was taken as 3 and S as 15e2. This increases the utilisation. This case is considered as the best utilization value for the algorithm MS. Specified parameter for simulating MS algorithm over ns2 is used. The graph specifies the packet drops and bandwidth utilization.

Packet drop = 2.74735e-05

Utilization = 0.908722

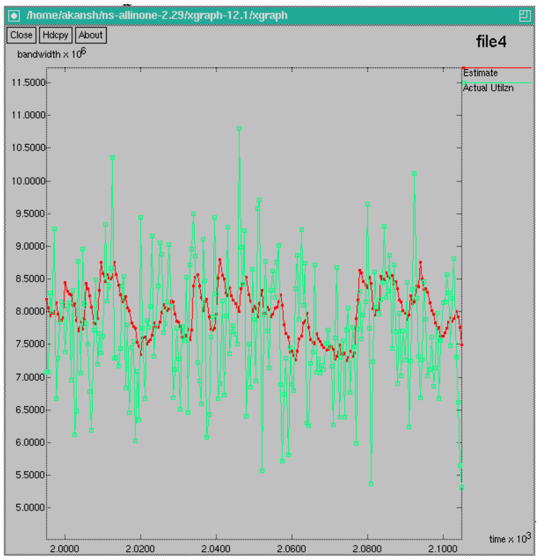


2. The graph was drawn for the algorithm Hoeffding Bounds. The rate of token bucket was changed to 512k from 64k and 2 token buckets were taken. This decreases the utilisation but the packet drops decreases by a great rate.

Specified parameter for simulating HB algorithm over ns2 is used. The graph specifies the packet drops and bandwidth utilization.

Packet drop = 0.001346

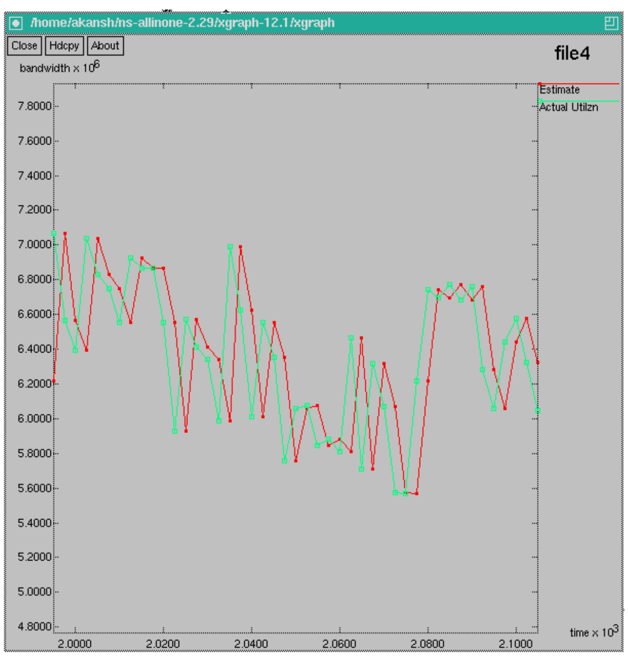
Utilization = 0.786782



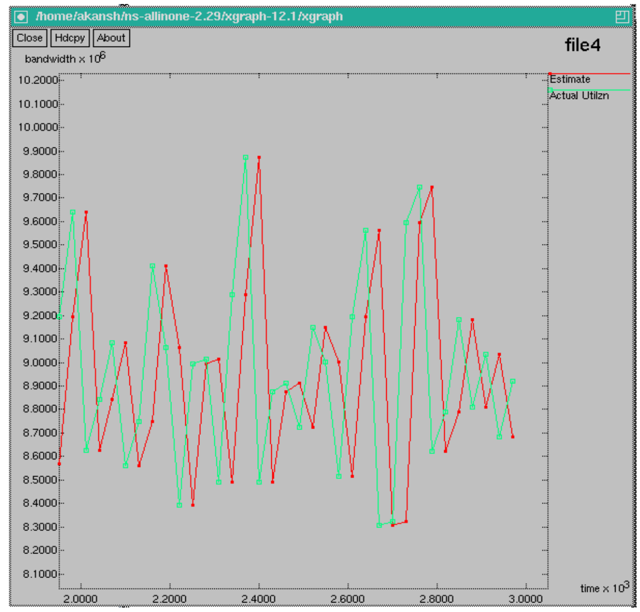
3. The graph was drawn for the Acceptance Region Tangent at Origin. The rate of token bucket is set to 256k instead of 64k and the burst time is changed to 0.1562 from 0.3125. Due to this the packet drops declines to 0 but degrades the band width utilization to a greater extent. Specified parameter for simulating ACTO algorithm over ns2 is used. The graph specifies the packet drops and bandwidth utilization.

Packet drop = 0.0

Utilization = 0.626694



4. The graph was drawn for the Acceptance Region Tangent at Peak. The sampling time parameter is increased to 3e5 from 2.5e4. This decreases the packet drop and bandwidth utilization increases. This gives a result for the transmission of packet in the admission control.



Specified parameter for simulating ACTO algorithm over ns2 is used. The graph specifies the packet drops and bandwidth utilization.

Packet drop = 0.001896

Utilization = 0.8973361

4. COMPARISONS OF ALGORITHMS

ADC 125Bytes	**Drops**	**Utilization**
MS	4.5927e-06	0.889605
HB	7.5492e-05	0.920020
ACTO	1.3510e-06	0.881026
ACTP	5.7212e-06	0.892432

Table 5.1: Comparison for packet size of 125 bytes

ADC 1250Bytes	Drops	Utilization
MS	0.0	0.872529
HB	0.0	0.756894
ACTO	0.00180	0.908220
ACTP	0.000621	0.906202

Table 5.2: Comparison for packet size of 1250 bytes

5. PROPOSED ALGORITHM

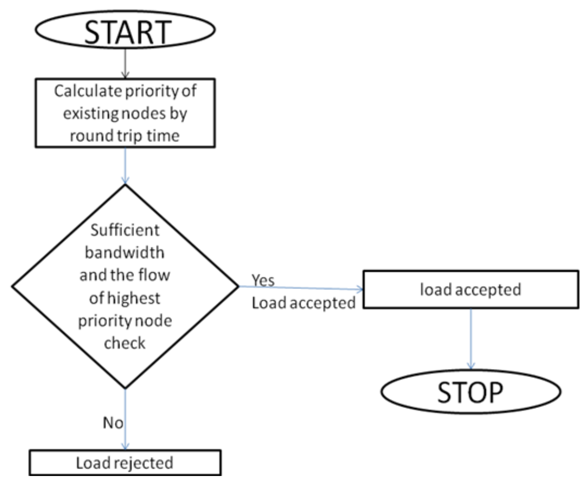
In this priority based algorithm, priority is decided using the ROUND TRIP TIME (RTT) of various nodes. Node with highest priority i.e. least RTT is allowed to admit flow first. To calculate RTT a source node sends a ping packet to a receiver and track the sending time. The receiver sends a packet back and the sender calculates RTT from tracked sending time. The changes in certain files are

- packet.h
- tcl/lib/nsdefault.tcl
- tcl/lib/nspacket.tcl
- ping.tcl

The file ping.tcl is executed which contain the procedure to calculate the round trip time of each node. The four node architecture is created in the file and the ping packets are sent to each node one by one. The procedure recv is defined which calculate the RTT of each node. Now in a network a new flow is admitted, first priority of the nodes is checked for the conflicting flows and node having the highest priority i.e. the minimum RTT of the conflicting flow is admitted to the network given that the bandwidth is available to admit the flow.

To deal with the problem of starvation in the network time stamping can be used. In this if a unique time stamp is attached to each flow. And if the flow is rejected then its time stamp is increased by 1. So all the flow can be admitted and no flow can go with starvation.

Flowchart [7]:-



Algorithm:-

1. Select the new admission control algorithm.
2. Round Trip Time(RTT) for all entering nodes calculated.
3. Compute the priority of those entering flows on the basis of RTT values.
4. Check for the bandwidth requirement of the highest priority flow and if the required bandwidth is available then that load is accepted else the flow is rejected.
5. If the flow is admitted, then create a new TCL object using a tcl class.

Video transmission on the Network using Admission Control:-
First the packet of 125 bytes was sent on the network. Then instead of the packets the video packets were inserted in the network. For this purpose the video foreman_qcif.yuv was used. The file intserv.tcl was changed and the video packets were inserted.

6. CHALLENGES AND RECOMMENDATIONS FOR FUTURE ENHANCEMENTS

The new proposed algorithm can be a recommendation for further enhancement implementing a hybrid model i.e. model can be made by the combination of priority based algorithm over the measurement based algorithms.
We succeeded in the video transmission on the network using admission control, but this degraded the utilization of the bandwidth, thus it can be a further challenge to implement the video packets without the degradation of the bandwidth utilization.
Admission control algorithm can be used on the cellular networks. The voice and data traffic can be controlled by

them. Different criterion can be made for handling the flow of new call and handoff call by degrading the bandwidth used by the current network by not compromising the QoS to a greater extent. And when free bandwidth is available the system can automatically upgrade the bandwidth provided to the current flow in the network to give them a better QoS. For this a bandwidth broker is used. Thus the throughput of the overall system can be increased to a greater extent. These admission control algorithms can be applied to VoIP (Voice over Internet Protocol) for the efficient traffic transmission over the internet.

7. CONCLUSION

Comparing actual and estimated bandwidth utilization,graphs were drawn. Estimated utilization is shown by red line and green shows the actual utilization. All the four graphs were prepared to justify the algorithms prepared and compared them for the different video packets. The utilization of bandwidth and the packet drops were checked and compared.

Admission control algorithms above performance is calculated by measuring the actual link utilization and the drop rate.

The HB algorithm gives the best utilization for smaller packet size and the drop rate of packet is minimum for ACTO algorithm.
For the transmission of packet of size 1250 Bytes the ACTO algorithm give the best bandwidth utilization .The ACTP algorithm gives near about the same utilization. But the packet drop rate in MS and HB algorithm is near to zero.
So finally we conclude that the algorithm HB is best for smaller packets but as the packet size increases and the algorithms ACTO and ACTP gives the best bandwidth utilization.

8. REFERENCES

[1]. Ferrai D. and D. C. Verma. “A scheme for real time channel establishment in wide area networks. “IEEE. J. Select. Area. Commun. Vol. 8 no. 3. Pp. 368-379. 1990.
[2]. Ohnishi, H, T. Okada and K. Noguchi. “Flow control schemes and delay/loss tradeoff in ATM networks “IEEE J Select Areas Commun. Vol. 6 no 9 pp. 1609-1616. Dec. 1988
[3]. Clark, D. D. S. J. Shenker and L. Zhang. “Supporting real time applications in an integrated services packet network; architecture and mechanism” in Proc. ACM SIGCOMM’ 92, pp. 14-26.
[4]. Zhang, H. and D. Ferrari. Improving utilization for deterministic service in multimedia communication, presented at the IEEE Int. Conf. Multimedia Computing and Systems, 1994.
[5]. Duffy D. E. A. A. Mclntosh, M. Rosenten and W. Willinger. Statistical analysis of CCSNSS7 traffic data from working CCS subnetworks. “IEEE J Select. Areas Commn. Vol. 12. No 3. Pp. 544-551. 1994.
[6]. Hiroshi Saito (1993). *Teletraffic Technologies in ATM Networks*. Artech House. ISBN 0-89006-622-1.
[7]. N. Hemrajani, Dharm Singh, A. Rawal and E. Menghani, Generation of a New Algorithm using Priority Measurement Based Routes, Global Journal of Computer Science and Technology (GJCST) Volume 11 Issue 13 Version 1.0 August 2011, pp24-29, Online ISSN: 0975-4172 & Print ISSN: 0975-4350.