International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

Efficient bandwidth allocation scheme using enhanced AQM Technique

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Abstract: *A network bandwidth manager maintains optimal accessibility and usage of critical applications and information related to business. It is very much important to detect the performance of the network in any organization. If the available bandwidth is increased the speed enhancements for file based transmissions show a more effective performance. In some cases of streaming application the necessary bandwidth may not be achieved. Main objectives of Bandwidth management is improve the QOS of Networks*

INTRODUCTION

Bandwidth allocation is the process of assigning radio frequencies to different applications. The radio spectrum is a finite resource creating the need for an effective allocation process. Bandwidth can be defined as the rate of data transmission. It is measured in bits per second or hertz. The process by which bandwidth is controlled and measured is called bandwidth management. A bandwidth manager is simply a gateway to better management of your networkThus optimizing the Quality of Service for the network, while a shared internet connection is used. It should be capable of reducing the risk of lost information due to critical conditions. The responsibilities of the bandwidth manager are:Reducing bandwidth requirementsTrackingbandwidth usesEnsuring securityProtecting business by disabling access of blacklisted sitesEnsuring suitable allocation of resources among all workstations

Queue Management

Packets from different flows arrive at a switch (or) routes for processing. Scheduling is a basic component to control the bandwidth several scheduling techniques are designed to improve the quality of service.

FIFO queuing: In first-first-out (FIFO) queuing is first processed is a new packet arrives they are added at the Assigned to different classes and admitted to different queues. The queues, however, are weighted based on priority of the queues; higher priority means a higher Wight. The system processes packets in each queue in a round –robin fashion with the number of packets selected from each queue based on the corresponding weight. For example, if the weights are 3, 2, and 1, Three packets are processed from the first queue, two from the second queue, and one from the third queue, if the system does not impose priority on the classes, all weights can be equal, in this way, and we have fair queuing with priority. Figure 003 shows the technique with three classes.

Priority queuing: In priority queuing, packets are the first assigned to a priority class. Each priority class has its own queue. The packets in the highest-priority queue are processed first. Packets in the lowest-priority queue are processed last. None that the system does not stop serving a queue until it is empty. Figure 002 shows priority queuing with two priority levels (for simplicity) A priority queue can provide better Qos than the FIFO queue because higher-priority traffic, such a multimedia, can reach the destination with less delay. However, there is a potential drawback. If there is a continuous flow in a high-priority queue, the packers in the lower-priority queues will never have a chance to be processed. This is a condition called starvation.

Weighted Fair Queuing: A better scheduling method is weighted fair queuing, in this technique; the packers are still end of the packets wait in the queue until the node is ready to process.

Class Based Queuing: CBQ is queuing algorithm that divides a networks connections bandwidth among multiple queens or classes. A queue may optionally be configured to borrow bandwidth from its parent queue it the parent is being under utilized. CBQ arranged in hierarchical manner. A top of hierarchical is the root queue which defines the total amount of bandwidth available. Child queues are created under the root queue, each of which can be assigned some portion of the root queues bandwidth..

Objective

The idea of the proposed project was derived from the recent work done by Lu et al. by the title “CHOKeR: A Novel AQM Algorithm with proportional bandwidth

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

allocation and TCP protection." The idea of this paper was to accomplish proportional bandwidth allocation for TCP based connection by optimizing conventional AQM (Active queue management). However, there are certain issues working with AQM that has not been addressed in [X]. Using AQM leads to drop of packet even before the queue becomes full. In wireline router links, errors due to transmission are negligible. However, in wireless router links, packet errors due to transmission are inevitable. Therefore, in addition to congestion losses stemming from AQM drops, it also have non-congestion (or wireless) losses arising due to channel errors. TCP suffers substantially from non-congestion losses since it responds to all losses by invoking congestion control and avoidance algorithms which results in degraded end-to-end performance on paths with lossy link. Hence, the prominent issues addressed in definitely are not scalable in wireless network. Hence, the proposed project chooses to contribute by addressing the above mentioned issues in wireless network using a novel topology for bandwidth allocation and to ensure better queue management.

The main aim of the proposed project is to design a framework that can efficient perform bandwidth allocation scheme with optimal queue management exclusively for wireless networks. In order to accomplish the proposed aim, followed objectives are set:

To carry out review of literature for visualizing the prior techniques used for effectively managing the bandwidth.

To design a user interface for simulating the bandwidth allocation schemes on a multiple wireless zones.

To propose a novel topological approach for performing bandwidth allocation scheme.

To consider mobility factor and evaluate its effect on queue management.

To ensure the effectiveness in queuing management for heavy traffic by ensuring least call drops.

Problem Statement

A bandwidth allocation is the assignment of a bandwidth interval to at most one link in the communication system. An optimal bandwidth allocation is one in which some measure of total interference is minimized.

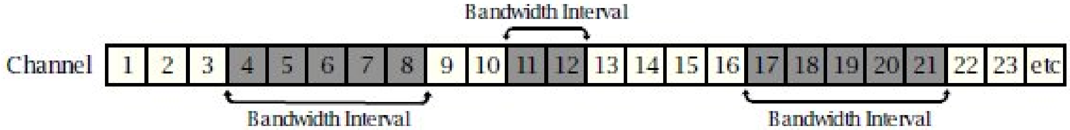

Figure 1.2: Channels and intervals in a bandwidth domain

Various Problems that has been identified in this process are as follows:

Available bandwidth is limited and insufficient to meet demand

Existing capacity is usually running at maximum capacity as a result it is often unusable. The cost of bandwidth is extremely high.

Expanding bandwidth capacity is limited due to finances, supply, technology. Existing bandwidth is often not managed

The main disadvantage of existing scheduling schemes is that they require constant per-flow state maintenance, which is not scalable in core networks.Less explored in wireless networks and challenges included in it.

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Solution to the Problem

The proposed project introduced the concept by representing the cluster as geometrical pattern. This is completely unique technique that was never experimented before. Majority of the work done in wireless network and their issues are designed in hexagon or pentagon way where the computation was quite time consuming and even difficult to show as a part of simulation too. But if the similar pattern of clusters and cells are design using more symmetrical pattern (like circle), it is expected that computation could be quite faster because of the assumptions that was considered during the experiment.

New Topology for bandwidth allocation: The proposed systems introduce a novel topology of wireless network that assumes that nodes moves around and relay each other packet in a circular environment. This circle is divided into equal sectors and cylinders. Each area has a unique id. It is assume that a central point in each area. Nearest nodes to this central points in each area are selected as local server. Each local server is responsible for response to location queries for all owner member nodes. Location update packages move alcylinder and location query packages move along the sector. These update and query propagation method decrease system response time, because the query packet traversed only one sector. Moreover impact of mobility will be studied considering multiple case scenario for much practical outcomes.

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

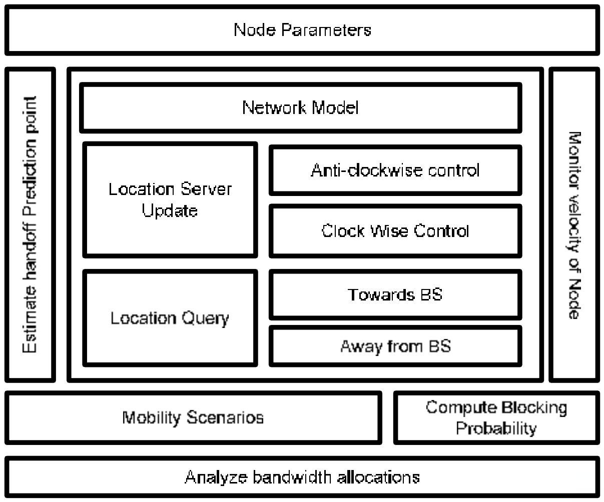


Figure 1 Architecture of Project

Novel Queue Management: As an substitute of AQM, the proposed system uses two module e.g. location server update and location query update.

Location Server Update: In the proposed work, servers are located in central areas. One sector considered as basic sector to start location server update process. Selection of basic sector is predefined in network model. Server update packet is sent periodically. These periods grows as the cylinder size grows. These update packets use synchronize aggregated method and we use this method with a little change. Location servers in each area in basic sector send two packets in clockwise and anti clockwise directions along on cylinder are on that. Each node that receive this packets if it is not server just forward packets to server on its area. Then apply member nodes location information changes and forward those to the next area. Update packets forward continues until packets reach to the basic sector again. Unlike conventional methods, so that once they attempted to gather information and then information collected will be sent. Receiving these update packets by server nodes in basic sector means that all location servers in cylinders are update and can respond to location queries.

Location Query: Location query process is done by sending a query packet in two different directions along the sector. When a source node wants to send the data to a destination node, if it doesn't become aware of destination's location, it should send a query packet to its location server. If the local server doesn't have destination position, had to find that. Intermediate nodes that receive the query packet, if they themselves have the destination's location, reply to query and send destination's location to source node, otherwise resend Query packet to previous direction. The destination node is located in one of the cylinder; therefore one of the local servers has information of destination location. So finally one of the two packages can be answered and the source node will be aware destination location. To prevent the exit of location servers from the sector the first cylinder (cylinder number zero) and the last cylinder when receive query packet, don't resend it and if it cannot respond to query packet, it sends back an error message to requested node.

Based on the data captured from the above operation, the system should be able to find hand-off prediction origination point, and target to accomplish better bandwidth allocation with guaranteed fairness and ensures minimized call drops.

Motivation

However, there are three problems that existing model cannot solved. First, the bandwidth differentiation at multiple priority levels becomes smaller after the number of flows increases. In particular, as the scale of the Internet becomes larger, all types of flows push into the Internet. In such complex environment, the drop probability of CHOKeW is close to CHOKe. The advantage of differentiation will be difficult to standout. Second, CHOKeW shows poor performance if the traffic is bursty. When the drawing factor increases quickly, the fairness diminishes. And CHOKeW cannot provide the assured bandwidth allocation for different priority flow. Third, when network congestion becomes worse, the bandwidth allocation in CHOKeW cannot cope with nonresponsive flows

Literature Survey

Barrera et al. [1] explored a statistical technique applied to AQM, namely, maximum likelihood estimation of congestion. The investigation of maximum likelihood estimation of congestion revealed a relationship between the observed marking rate, the observed queue occupancy and the likelihood of congestion.

Wen et al. [2] demonstrated that under a variety of congestion scenarios, CHOKeW is able to 1) support differentiated bandwidth allocation by affording a larger bandwidth share to higher priority flows, 2) provide the flows in the same priority with better fairness than conventional stateless AQM schemes such as RED and BLUE, 3) maintain high link utilization as well as short queue length, and 4) protect TCP flows by restricting the bandwidth share of high-speed unresponsive flows.

Floyd and Jacobson [3] Presented, Random Early Detection (RED) gateways for congestion avoidance in packet switched networks. The gateway detects incipient congestion by computing the average queue size. The gateway could notify connections of congestion either by dropping packets arriving at the gateway or by setting a bit in packet headers. When the

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

average queue size exceeds a preset threshold, the gateway drops or marks each arriving packet with a certain probability, where the exact probability is a function of the average queue size.

Mahajan et al. [4] illustrated a mechanism that combines simplicity and protection by keeping state for just the high-bandwidth flows. RED-PD uses the packet drop history at the router to detect high-bandwidth flows in times of congestion and preferentially drops packets from these flows. This paper discusses the design decisions underlying RED-PD. They show that it is effective at controlling high bandwidth flows using a small amount of state and very simple fast-path operations.

Pan et al. [5] proposed a packet dropping scheme, CHOKe, which aims to approximate fair queuing at a minimal implementation overhead. Simulations suggest that it works well in protecting congestion-sensitive flows from congestion insensitive or congestion-causing flows. Analytical models were derived for gaining insights about the algorithm and for understanding the simulations. Further work involves studying the performance of the algorithm under a wider range of parameters, network topologies and real traffic traces, obtaining more accurate theoretical models and insights, and considering hardware implementation issues.

Tang et al. [6] presented an equilibrium model of TCP/CHOKe. They prove that, provided the number of TCP flows is large, the UDP bandwidth share peaks at (+ 1) 1 = 0 269 when UDP input rate is slightly larger than link capacity, and drops to zero as UDP input rate tends to infinity. They clarify the spatial characteristics of the leaky buffer under CHOKe that produce this throughput behavior. Specifically, they prove that, as UDP input rate increases, even though the total number of UDP packets in the queue increases, their spatial distribution becomes more and more concentrated near the tail of the queue, and drops rapidly to zero toward the head of the queue.

Clark and Fang [7] proposed the “allocated-capacity” framework for providing different levels of best-effort service in times of network congestion. The “allocated capacity” framework-extensions to the Internet protocols and algorithms-can allocate bandwidth to different users in a controlled and predictable way during network congestion. The framework supports two complementary ways of controlling the bandwidth allocation: sender-based and receiver-based. In today’s heterogeneous and commercial Internet the framework can serve as a basis for charging for usage and for more efficiently utilizing the network resources. They focus on algorithms for essential components of the framework: a differential dropping algorithm for network routers and a tagging algorithm for profile meters at the edge of the network for bulk-data transfers.

Ramabhadran and Pasquale [8] illustrated a novel packet scheduler called Stratified Round Robin, which has low complexity, and is amenable to a simple hardware implementation. Stratified Robin Robin exhibits good fairness and delay properties that are demonstrated through both analytical results and simulations. In particular, it provides a single packet delay bound that is independent of the number of ows. This property is unique to Stratified Round Robin among all other schedulers of comparable complexity.

Padhye et al. [9] developed a simple analytic characterization of the steady-state send rate as a function of loss rate and round trip time (RTT) for a bulk transfer TCP flow. Unlike the models, their model captures not only the behavior of the fast retransmit mechanism but also the effect of the time-out mechanism. Their measurements suggest that this latter behavior is important from a modeling perspective, as almost all of their TCP traces contained more time-out events than fast retransmit events. Their measurements demonstrate that their model is able to more accurately predict TCP send rate and is accurate over a wider range of loss rates.

Schranzhofer et al. [10] demonstrated of the power-aware scenario-mapping problem to design embedded systems with applications specified by their execution probabilities. Multiple concurrently executing multi-mode applications result in a set of scenarios and each scenario can be attained by sequences of mode changes. They show that there is no polynomial-time approximation algorithm with a constant approximation factor and provide a polynomial-time heuristic algorithm.

Wu et al. [11] considered a novel server-based congestion control approach which is based on the well known AQM algorithm A-RED (Adaptive Random Earlier Detection) and is so called SF-RED. This approach can provide fair services to different servers without affecting congestion control performance in terms of packet loss rate and queue stability. They use NS2 simulator to prove the effectiveness of their approach.

Wang et al. [12] developed an analytical model for LRED, which demonstrates that LRED is responsive even if the number of TCP flows and their persisting times vary significantly. It also provides a general guideline for the parameter settings in LRED. The performance of LRED is further examined under various simulated network environments, and

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

compared to existing AQM algorithms. Their simulation results show that, with comparable complexities, LRED achieves shorter response time and higher robustness. More importantly, it trades off the goodput with queue length better than existing algorithms, enabling flexible system configurations.

Tan et al. [13] approached still faces challenges. First, it is hard for the wireless system to dynamically model the utility function of the users. Second, the utility function of soft QoS traffic is usually nonconcave, which brings the NUM optimization problem to be mathematically intractable. With deviation to the usual NUM theory, this paper proposes a novel optimization model and its algorithm to allocate bandwidth around the user's desired value to the soft QoS traffic in a wireless network. Their approach takes advantage of the basic feature of soft QoS traffic; that is, it demands a preferred amount of bandwidth but allows some flexibility during normal operation.

Bahl et al. [14] proposed a novel bandwidth allocation strategy which partitions the available bandwidth amongst the different traffic classes in a manner that ensures quality-of-service guarantees for digital video while minimizing the maximum blocking probability for voice and data connections. At the connection level, near-optimum utilization of the reserved bandwidth for video traffic is achieved through an intra-frame statistical multiplexing algorithm, while at the system level the delicate task of partitioning the bandwidth between voice, video and data is accomplished by developing an efficient algorithm which uses traffic parameters consisting only of the aggregate traffic load and the total available bandwidth.

Stoica et al. [15] proposed a novel scheduling algorithm called Hierarchical Fair Service Curve (HFSC) that approximates the model closely and efficiently. The algorithm always guarantees the service curves of leaf classes, thus ensures real-time and priority services, while minimizing the discrepancy between the actual services provided to and the services defined by the Fair Service Curve link sharing model for the interior classes. They have implemented the HFSC scheduler in Net BSD. By performing simulation and measurement experiments, they evaluate the link sharing and real-time performances of HFSC, and determine the computation overhead.

Harks et al. [16] presented a novel approach to the congestion control and resource allocation problem of elastic and real-time traffic in telecommunication networks. With the concept of utility functions, where each source uses a utility function to evaluate the benefit from achieving a transmission rate, they interpret the resource allocation problem as a global optimization problem. The solution to this problem is characterized by a new fairness criterion, utility proportional fairness. They argue that it is an application level performance measure, i.e. the utility that should be shared fairly among users.

Huang et al. [17] illustrated an energy saving scheduling scheme is proposed for OFDMA based two-hop relay systems. The novel scheme adjusts the modulation and coding (MC) mode and allocates the transmitting power dynamically according to the resource usage. It can also guarantee the QoS of different services by setting the scheduling priority. The simulation results show that the novel scheduling scheme can not only save system energy but also achieve higher throughput.

Meng [18] illustrated a novel scheduling algorithm for Orthogonal Frequency Division Multiplex/Time Division Multiple Access (OFDM/TDMA)-based systems, which extends the PF scheme to multiple service types with diverse QoS requirements. The design objective is to provide differentiated services according to their QoS requirements, while the objective can be achieved by adjusting only one unique parameter, the time window for evaluating the average throughput. By extensive simulation, it is shown that the proposed scheduling algorithm exploits the advantage of the PF scheme, enhancing the throughput, and distinguishes the services in terms of the average delay.

Todinca et al. [19] proposed a new session admission control algorithm that overcomes most of the difficulties encountered by the existing AC algorithms. The novelty of their approach is the use of fuzzy logic (fuzzy inference) for the AC. They demonstrate the efficiency of their AC algorithm by simulations.

Thing et al. [20] composed of a fractional bandwidth-allocation algorithm, taking into consideration the use of parameters with distinct differences. This proposed technique allows the computation of an optimal channel-allocation set, where degradation caused by interchannel interference and FWM is minimal. Simulation is carried out to show significant performance improvement, such as an average bit-error rate improvement factor of 1.336 for an eight-channel wavelength-division multiplexing system, without the requirement of increased bandwidth, unlike existing channel-allocation methods.

Saini et al. [21] studied different resource allocation and management schemes used in WiMAX networks. So the real time Quality of services (QoS) to the respective users can not be provided unless the system resources, like bandwidth and transmitter power which are limited, are intelligently used and properly optimized. In all the

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

schemes performance of the Wi-MAX network has been improved in terms of data rate, overall capacity, throughput etc.

Katabi et al. [22] developed a novel approach to Internet congestion control that outperforms TCP in conventional environments, and remains efficient, fair, scalable, and stable as the bandwidth-delay product increases. This new eXplicit Control Protocol, XCP, generalizes the Explicit Congestion Notification proposal (ECN). In addition, XCP introduces the new concept of decoupling utilization control from fairness control. This allows a more flexible and analytically tractable protocol design and opens new avenues for service differentiation.

Filali and Dabbous [23] focused on the inter-multicast fairness issue which addresses the way how the network resources are shared between competing multicast flows. They propose a simple and scalable single FIFO queue-based active queue management mechanism called MFQ (Multicast Fair Queuing) to achieve the desired inter-multicast fairness. MFQ interacts with an external multicast bandwidth allocation module which implements a pre-defined inter multicast fairness function. To guarantee a fine-grained packet queuing/dropping, MFQ uses a novel bandwidth sharing notion, called Multicast Allocation Layer (MAL).

Zhang et al. [24] designed and implement a novel scheduler, called Arbitrator, to maintain per-application performance no matter in terms of throughput or latency. In their scheduling framework, they introduce a factor to reflect how applications are sensitive to deadline missing. The scheduler employs a feedback mechanism to monitor latency guarantees and throughput allocation for each application, and compute how much applications deviate from their performance targets. Based on the estimation, Arbitrator makes the scheduling decision to achieve latency guarantee and proportional sharing of bandwidth.

Introduction about Existing system

The importance of TCP protection has been discussed by various authors. The problem of TCP Protection originates from TCP flows competing with unresponsive UDP flows in order to occupy scarce bandwidth. After the TCP flows reduce sending rates, the unresponsive UDP flows can seize the available bandwidth and cause starvation of TCP flows. This results in unfairness to large volume TCP flows. Conventional AQM algorithms such as Random Early Detection (RED) and BLUE cannot protect TCP flows. Per-flow schemes such as RED with preferential dropping (RED-PD) can punish non-TCP-friendly flows, but it requires reserved parameters for each flow, which significantly increases the memory requirement. CHOKe proposed by Pan et al. is simple and does not require per-flow state maintenance. However, CHOKe only serves as an enhancement filter for RED in which a buffered packet is drawn at random and compared with an arriving packet. If both packets come from the same flow, they are dropped as a pair (matched drops); otherwise, the arriving packet is delivered to RED. The validity of CHOKe has been explained using an analytical model by Tang et al. .

Advantage: Drop-tail queues have a tendency to penalise bursty flows, and to cause global synchronization between flows. By dropping packets probabilistically, AQM disciplines typically avoid both of these issues.

By providing endpoints with congestion indication before the queue is full, AQM disciplines are able to maintain a shorter queue length than drop-tail queues, which combats bufferbloat and reduces network latency.

Disadvantage: Early AQM disciplines require careful tuning of their parameters in order to provide good performance. Modern AQM disciplines are self-tuning, and can be run with their default parameters in most circumstances.

For AQM systems that drop, the result seems counter-intuitive to many network engineers

System Design

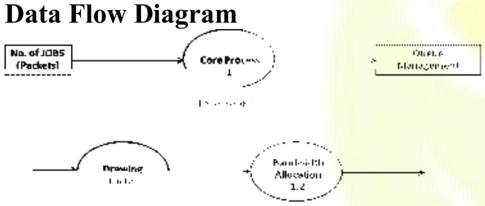
Although differentiated services (DiffServ) networks have been well discussed in the past several years, a conventional Active Queue Management (AQM) algorithm still cannot provide low-complexity and cost-effective differentiated bandwidth allocation in DiffServ. In this paper, a novel AQM scheme called CHOKeR is designed to protect TCP flows effectively. We adopt a method from CHOKeW to draw multiple packets randomly from the output buffer. CHOKeR enhances the drawing factor by using a multistep increase and single-step decrease (MISD) mechanism. In order to explain the features of CHOKeR, an analytical model is used, followed by extensive simulations to evaluate the performance of CHOKeR. The analytical model and simulation results demonstrate that CHOKeR achieves proportional bandwidth allocation between different priority levels, fairness guarantee among equal priority flows, and protection of TCP against high-speed unresponsive flows when network congestion occurs.

International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

AQM	Parameters	Value
CHOKeW [5]	L_{th}	100 packets
	L^-	125 packets
	L^+	175 packets
	p_w^+	0.002
	p_w^-	0.001
CHOKeR	L_{th}	100 packets
	L^-	125 packets
	L^+	175 packets
	p_r^+	0.002
	p_r^-	0.001
RED [6]	*min_th*	100 packets
	max_th	200 packets
	EWMA weight	0.02
	pmax	0.02
RIO [11]	*minth_out*	100 packets
	maxth_out	200 packets
	minth_in	150 packets
	maxth_in	250 packets
	pmax_in	0.01
	pmax_out	0.02
BLUE [7]	p_b^+	0.0025
	p_b^-	0.00025
	update interval	100 ms

Table -1

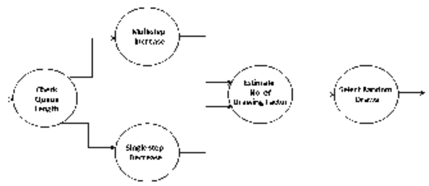
Data Flow Diagram



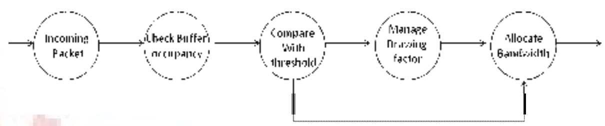
Level One

In Level 0, Core process checks for No. of JOB packets & process for Queue Management, in Queue management check for drawing factor & allocate bandwidth as shown in above diagrams.

Active Queue management (AQM) algorithms are among the most effective network control mechanisms and can achieve a satisfactory tradeoff between drop probability and throughput Different AQM parameters lead to different QoS performance. Moreover, the implementation of an AQM in DiffServ must provide bandwidth differentiation and TCP protection simultaneously. An important component of the many QoS architectures proposed is the packet scheduling algorithm used by routers in the network. The packet scheduler determines the order in which packets of various independent owes are forwarded on a shared output link. One of the simplest algorithms is First Come First Served (FCFS), in which the order of arrival of packets also determines the order in which they are forwarded over the output link.



In level-2 drawing factor, checks for queue length & further proceeds for multistep increase and single step decrease. Estimate no. of drawing factor & process for select randomly it will draws the packets to allocate the bandwidth as shown in level-2



Level -2(ii)

In level-2 bandwidth allocation, incoming packets are checked with buffer occupancy. The packets will be compared with incoming packets & buffered packets to check for threshold if, threshold meets it process to allocate band width, else it flows to mange drawing factor.

Algorithm 1 CHOKeR

1: {Initialization}
2:
3: {Update drawing factor }
4: for each incoming packet with priority do
5: if then
6:
7:
8: else if then
9:
10: else if then
11:
12: else if then
13:
14:
15: else if then
16: drop
17: end if
18:
19:
20: {generate random number }
21: if then
22:
23: end if
24: {Packet draw actions}
25: while do
26:
27: if then

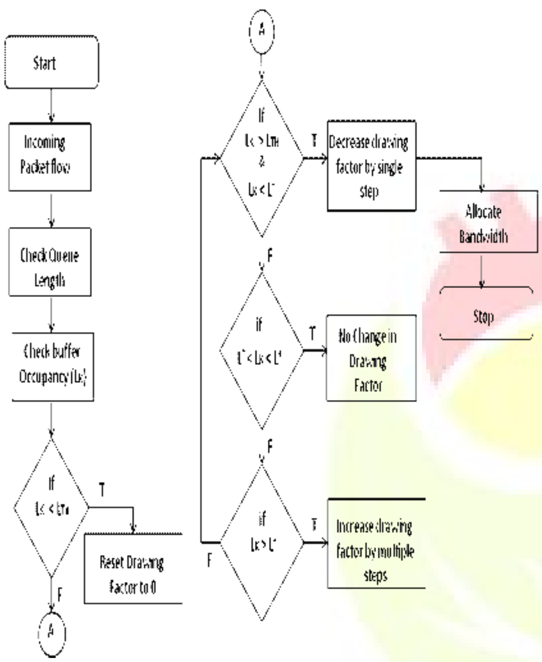
International Journal of Advanced Research in Biology, Ecology,Science and Technology(IJARBEST)
Vol. 1, Issue 2, May 2015

```
{Draw packets with matching flow ID}

break
else if and priority of drawn
packet pkt' is then
{Draw packets with matching priority}

end if
end while
```

Flow Chart CHOKeR



Conclusion

The proposed system introduces a cost effective AQM scheme. The framework uses multistep increase and single-step decrease to update the drawing factor, such that large-scale and burst data can be processed fast, and congestion can be avoided. When the number of flows increase abruptly, the framework's proportional bandwidth allocation with multiple priority is able to guarantee TCP protection. Both the analytical model and simulation results is expected to demonstrate that the framework achieved proportional bandwidth allocation for flows in different priorities, and fairness for flows in the same priority. With the increase of audio and video flows, framework reduces the UDP flows to protect TCP flows, and allocate a fair share of bandwidth to UDP flows.

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