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A Study on Bandwidth and Buffer Management Mechanisms of IP Networks

IP 네트워크의 대역폭 및 버퍼 관리 메커니즘에 관한 연구

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요 약 대역폭 및 버퍼는 엔드-투-엔드 품질 서비스를 결정하기위한 중요한 네트워크 리소스이다. 본 논문에서는 대역폭과 버퍼 관리에 관한 몇 가지 기법에 대해 조사 및 TCP / IP 네트워크의 처리량을 제어에 사용되는 유형에 따라 분류 하고자 한다. 뿐만 아니라, 본 논문에서는 모델링 활성 대역폭을 위한 새로운 접근법을 제시 및 TCP / IP용 네트워크에 대한 버퍼의 제어 메커니즘에 대한 새로운 접근법을 제시한다.

Abstract Bandwidth and Buffer are critical network resources to determine the end-to-end quality of service. In this paper, we investigate several techniques on bandwidth and buffer management and classify them according to the types they used for controlling the throughput of a TCP/IP network. Moreover, in this paper, it present a new approach for modeling the active bandwidth and buffer control mechanisms for TCP/IP networks.

Key Words : TCP/IP Network, Bandwidth, Throughput

I. Introduction

Quality of Service (QoS) often occurs due to the shortage of bandwidth and buffer in the network nodes. For example, when a link bandwidth or the buffer of a network node exceeds the capacity (bandwidth, buffer) of available nodes, the result is a long delay of data delivery that may cause losses or drops of packets. This results again in degradation of the end-to-end quality of services. In this context, the performance or throughput degradation is closely related to the congestion control for the network.*

It was well known from various studies in the past that quality of service control mechanisms focus mainly on two main concepts at different levels, namely queue management and scheduling at lower level and end-to-end transportation at higher level[1-3]. Queue management mechanisms manage the length of packet queues by dropping packets whenever necessary whereas scheduling mechanisms determine which packets to be sent next[5-8]. These mechanisms are used primarily to manage the allocations of bandwidth and buffer among various flows. The transport protocols like TCP or TCP-like use congestion control mechanisms to react to packet losses, to reduce the network congestion and to enhance the throughput performance[9-12].

The convergence of communication networks enables the deployment of new applications and services like multimedia applications. This new trend increases the percentage of non-TCP traffic in networks. Many TCP-like protocols have been

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proposed in literature to address this new trend. Like TCP, TCP-like protocols can detect packet drops and interpret them as indications of congestion in the network. The sender will react to these packet drops by reducing their sending rates. This reduction in sending rate translates into a decrease in the incoming packet rate at the network node, which effectively allows the node to clear up its buffer queue. When the incoming packet rate is higher than the outgoing packet rate of a node, the buffer queue size will gradually increase and queue becomes full at one stage.

There is a necessary trade-off between bandwidth, delay, buffer and performance in the network. Traditional methods for buffer management used the technique of "tail drop" that monitors the queue length and accepts packets until the maximum queue length is reached. When the queue is full, the most recently arrived packets will be dropped. However, this technique did not reflect the real state of the queue length, thus results in poor performance of the network. In IP networks, dropping packets is considered as a critical mechanism for congestion control as well as quality of service control. Suitable solutions for buffer management are necessary. New approaches such as active buffer management (or active queue management), adaptive bandwidth management have been discussed in several research works[3]. Adaptive bandwidth management often uses adaptive scheduling mechanisms to allocate bandwidth to different application flows according to their needs. Bandwidth management is treated in closed relation to flow and congestion control as well as buffer management. By keeping the buffer size of a node at a reasonable level, buffer management mechanisms will reduce the delays seen by packets of application flows, thus, a better performance can be achieved. Active buffer management can ensure that there will be almost always a buffer available for an arriving packet and can prevent the issue of low bandwidth for highly bursty flows.

On the other hand, the poor dimension of buffer size at the network nodes may cause issues for TCP and TCP-like protocol throughput. Previous research works in buffer dimension mainly focused on network performance metrics such as link utilization or loss rate. However, as we indicated above, bandwidth and buffer management relate closely together. In this paper, we address several techniques on bandwidth and buffer management and classify them according to the types they used for controlling the throughput of a TCP/IP network. From this investigation, we present a new approach for modeling the bandwidth – buffer control mechanisms for TCP/IP networks. The paper is organized as follows. In section 2 we investigate the concept of active buffer – bandwidth management and classify their mechanisms based on metrics of congestion measure. In section 3 some of the active buffer management mechanisms based on different metrics was discussed. In section 4, a new approach for modeling the active bandwidth – buffer control mechanisms for TCP/IP networks is presented and the paper is concluded with section 5.

Ⅱ. Conception of active buffer

It is widely believed that a major reason for the relatively low end-to-end network throughput is that the sender adjusts its transmission rate based on its congestion algorithms according to the congestion measure of the underline networks or because of buffer size of the network nodes.

Transport protocols like TCP and TCP-like have slow performance despite of available bandwidth of high performance networks, mostly because of two reasons:

1. Small buffer sizes at the network nodes limit the effective window of the transfer, and thus the maximum throughput.
2. Packet losses cause large window reductions, with a subsequent slow (linear) window increase rate, reducing the transfer's average throughput.

There are two approaches to accomplish this. The first one is to dynamically adjust the transmission rate of the senders in response to the congestion along its path; the second one is to use a node mechanism that implicitly or explicitly conveys information about the current congestion measure of the network to sources using that link. In the current IP network, the first approach is carried out by congestion control mechanisms for TCP and TCP-like protocols, and the second approach is carried out by active buffer management and bandwidth scheduling at network nodes.

As you can be seen figure 1 it used to measure congestion, active buffer management mechanisms can be classified into three catalogs: queue-length based, load-rate based, and combination of them. In queue-length based mechanisms, congestion is measured using the average or instantaneous queue length and the control is aimed to stabilize the queue length. The drawback of queue-length based mechanisms is that a backlog is remaining a complex issue of design and it is difficult to exactly define the thresholds for the queue lengths. Load-rate based mechanisms use the prediction of the link utilization to determine the network congestion state and take actions based on the packet arrival rate. Rate-based mechanisms can provide early feedback for congestion. Bandwidth estimation methods [4] can provide a mean for link utilization prediction. Other mechanisms deploy a combination of queue length, load and rate to measure congestion and try to achieve a trade-off between queues stability and responsiveness.

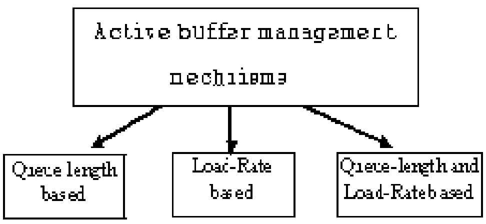

그림 1. 활성 버퍼 관리기법
Fig. 1. Classification of active buffer management

Bandwidth is an important network resource to ensure QoS, thus, many research works already focused on investigation of bandwidth management mechanisms.

As you can be seen figure 2, it used for bandwidth allocation, bandwidth management mechanisms can be classified into two main catalogs: flow-based or quantitative allocation mechanisms and aggregate-based or qualitative mechanisms. Typical for these mechanisms is Integrated Services (IntServ) and Differentiated Services (DiffServ) [6].

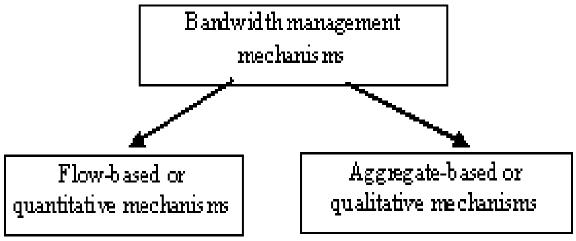

그림 2. 대역폭 관리기법
Fig. 2. Classification of bandwidth management

Quantitative allocation mechanisms [5] are usually for individual flows and guarantee bandwidth allocation based on common parameters such as: bandwidth, latency and jitter. The quantization is done by limited defined or statistical parameters. The general method is a simulation of circuit switches, that each traffic flow can be handled separately. Incoming packets should comply with a condition of rate and maximum amount of bits. Beside, it is supposed that there is no error in the transmission links. With this supposition, a quantitative allocation mechanism can reserve bandwidth for each traffic flow and guarantee necessary delay. Quantitative allocation mechanism is usually based on rate schedulers. The schedulers are designed by simulating fluid models of circuit switching, that can guarantee bound of delay by maintain a minimum rate during application's time.

Advantages of these mechanisms are ability to separate each traffic flow to serve, guarantee quantitative QoS for each traffic flow. Effectiveness of the mechanism and complexity are contradictions required to be solved in these mechanisms. Qualitative mechanisms allocate bandwidth based on classification and assign priority level for each traffic flow aggregate. That mean higher priority level flows will be served before lower priority level flows. Service model of these mechanisms considered the change of traffic, the difference of traffic flows (based on class levels), and unpredictable transmission channel condition. Based on that basis, bandwidth is allocated for each flow. These mechanisms are not clearly reserved resource for each traffic flow. There are two general methods: relative classification and ratio classification [3]. Relative classification permits to guarantee relative quality between two different flows, it mean quality of a flow is always better than the one of the other flow. Ratio classification guarantees a flow priority compared to the other and the ability to transform the ratio to control the flows according to the current circumstance.

Ⅲ. ACTIVE BUFFER MANAGEMENT MECHANISMS

RED is an active queue management scheme, called Random Early Deduction (RED) [7]. RED tries to alleviate congestion by early detecting incipient congestion and providing congestion notification to the senders, allowing them to reduce the transmission rates before buffer overflow occurs.

In fact, RED acts in response to congestion, and does not consider the issues of “Full queue” as in the widely deployed drop tail mechanisms. By maintaining the average queue size, RED can reduce the delays experienced by packets of flows. The effectiveness of RED depends mainly on the appropriate selection of the RED threshold parameters. This selection is often very complicated and it is difficult to determine the suitable RED parameters for buffer management in high speed networks. RED is design for aggregate of flows and is not suitable for unresponsive flows. This shortcoming was seen by Lin and Morris, thus they proposed a scheme, called Flow Random Early Detection (FRED) [8]. FRED addresses a fair buffer allocation between flows and attempts to provide fair buffer allocation between flows by isolating each flow from the effects of misbehaving or non responsive flows. During times of congestion FRED constraints all flows to occupying loosely equal shares of the queue’s buffer. Flows that repeatedly exceed an average fair share of the queue’s capacity are tightly constrained to consume no more than their fair share. Statistics must be maintained for every flow that currently has packets in the queue of the network node. By this way, “active flows” are allocated an equal share of the queue, which is determined by dividing the current queue size by the number of active flows. The number of packets a flow has enqueued is compared to the product of the flow’s share value and a constant multiplier factor. This factor can allow bursty arrival patterns among flows. A flow that exceeds the threshold including the multiplier is considered as unresponsive and is constrained to its share until it has no more packets in the queue. However, FRED still has a weakness of the overhead of tracking active flows. Class based Thresholds (CBT) [9] is an active buffer management mechanism that has the positive features of RED, but is possible to limit the influences of unresponsive flows and allows them to share a configurable amount of the link bandwidth. In comparison to FRED, CBT does not need to maintain the state per flow in the node. CBT uses drop thresholds of RED and buffer allocation method of FRED to provide throughput and delivery latency by isolating the active flows from the effects of other unresponsive flows.

Examples for active flows are continuous multimedia streams that need to be isolated from other flows. For these flows, statistics are maintained and their throughput is constrained during the times of congestion. Packets are classified into tagged flows and untagged flows. Tagged flows are streaming flows and should be isolated from other flows. Untagged flows are constrained by a threshold calculated based on the average number of queueing packets of untagged flows. In CBT, the classification of flows (packets) is complicated depending on the number of active flows. Stabilized Random Early Drop (SRED) [10] uses the same mechanism as RED by pre-emptively discarding packets with a load dependent probability when buffer in the node seems to be full. SRED has an additional feature that it has a wide range of load levels that can be used to stabilize its buffer occupation at a level independent of the number of active connections. In order to determine these load level, SRED estimate the number of active flows. In fact, the estimate is obtained without collecting or analyzing state information of individual flows. SRED uses the same mechanism to identify misbehaving flows. There is no need to compute the average queue length. The calculation of packet loss probability is based on the instantaneous buffer occupation and on the estimated number of active flows. While in RED the buffer occupancy increases in proportion with the number of connections, the buffer occupancy in SRED is almost always at least B/3 where B is the buffer occupancy in case of RED. That means buffer occupancy in SRED is independent of the number of connections. However, the impact of packet drop is very high when the bottleneck buffer occupancy is dominated by a few active flows with large windows and is very little when the bottleneck buffer occupancy is caused by a large number of connections with small windows. Although SRED can stabilize over a wide range of load levels, it can cause low throughput for the active flows.

Ⅳ. A new APPROACH FOR ACTIVE BUFFER

Figure 3 illustrates the basic idea of our proposed approach for active buffer-bandwidth management.

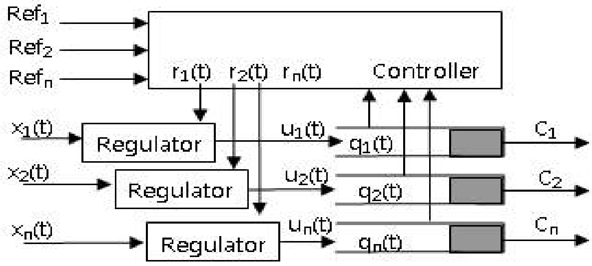

그림 3. 활성 버퍼 대역폭 관리를 위한 새로운 모델
Fig. 3. New Model for active buffer-bandwidth management

The notations we use in this figure are as follows:

$x_i(t)$:the incoming rate of class i.

Ref_i:the reference value for class i.

$q_i(t)$:the queue length of class i.

$u_i(t)$:the actual queue filling rate of class i.

$r_i(t)$:the control rate of class i.

C_i:the outgoin grate of class i.

The term of class is corresponding to the AF classes in the DiffServ model [5].

The concept of regulators can be followed by the principle of leaky-bucket or token-bucket. That is:

$$u_i(t) \leq r_i(t)+b_i/(t_2-t_1) \quad (1)$$

Where

b_i is the maximum bucket size of class i, and

T = (t_2-t_1) is the observe interval of congestion.

The incoming rate of each class i obeys the constraint of leaky-bucket condition indicated in the equation (1).

Using the fluid model for the class queues, we can have as follows for each class i:

$$\frac{dq_i(t)}{dt}=u_i(t) - \rho_i.C_i \quad (2)$$

Where r_i is the utilization factor of the link i.

Without loss of generality, we can assume that T is the considerable time interval of any congestion time. Moreover, we consider the worse case, where $u_i(t)=u_{imax}(t)$. That ist he case of maximum amount of packets can be accepted by the regulator. Let $v_i=b_i/(t_2-t_1)$, we can have from (1) that:

$$u_{imax}(t)=r_i(t)+v_i(t) \quad (3)$$

for any congestion time interval.

We can rewrite the equation (2) as follows:

$$\frac{dq_i(t)}{dt} = r_i(t) + v_i(t) - \rho_i.C_i \qquad (4)$$

This equation (4) describes the dynamic of the active buffer – bandwidth management for a class AF of DiffServ model, whereas $q_i(t)$describes the buffer occupation of class i, C_i describes the link bandwidth for class i, $r_i(t)$describes the regulation rate for the class i using the corresponding reference Ref_i.By giving various configurable parameters Ref_i to the class i, we can give different treatment to the corresponding class I.

V. CONCLUSION

This paper investigated several techniques on bandwidth and buffer management. We classified them according to the types they used for controlling the throughput of a TCP/IP network and discussed their advantages and weaknesses in controlling congestion. From this investigation, we claim about a possible combination of active buffer management mechanisms and buffer management. To this end, we proposed a new approach for modeling the active bandwidth – buffer control mechanisms for TCP/IP networks.

With this new approach, it is possible to describe the dynamic behaviors of the buffer size in relation with the bandwidth allocation. This approach promises a better presentation for the relationship between active buffer management and bandwidth management in order to achieve better throughput for TCP and TCP like protocols while keeping controllability of the end-to-end quality of services in term of throughput, packet loss, delay, link utilization. Further works will focus on the investigation of the model.

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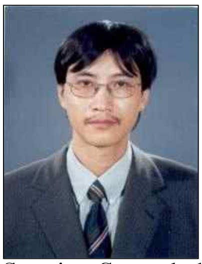

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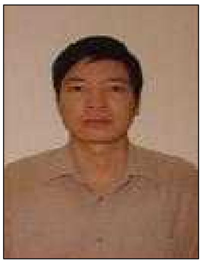

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