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Extended Queue Management Backward Congestion Control Algorithms for Web Transfers in Wireless Environment

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Abstract — Wireless links are characterized by high error rates and intermittent connectivity. TCP congestion control has been developed on the assumption that network congestion is the only cause for packet loss. Upon detecting a packet loss, TCP drops its transmit window resulting in an unnecessary reduction of end-to-end throughput which results in suboptimal performance.The sender has to be made aware by some feedback mechanism that some of the losses reported are not due to congestion. The Active Queue Management algorithms (AQM) are used to reduce congestion, and in this paper, we have analysed four AQM algorithms, Random Early Deduction (RED), Wireless Explicit Congestion Notification (WECN), Queue Management Backward Congestion Control Algorithm (QMBCCA) and its enhanced version Extended Queue Management Backward Congestion Control Algorithm (EQMBCCA). WECN, QMBCCA & EQMBCCA algorithms make use of feedback mechanisms. WECN gives feedback using the CE bit. QMBCCA and EQMBCCA make use of ISQ notifications and also the CE bit whenever the average queue size crosses minimum threshold value. EQMBCCA reduces the reverse ISQ traffic by introducing a configurable intermediate threshold value IntThres. The comparison is made in terms of Delay, HTTP packet loss percentage and fairness for FTP flows in a wireless environment. It is found that the performance of EQMBCCA is almost equal to that of QMBCCA and better than RED and WECN.

Index Terms — Active Queue Management, Congestion Control, Delay, HTTP Packet Loss, Fairness

I. INTRODUCTION

The Internet has been in constant growth in the last two decades, reaching a greater number of users and supporting new applications. The wireless environment exhibits different characteristics compared to the wired one and TCP improvements are needed to assure reliable transport services to all the users regardless of the kind of links connectivity used. The nature of wireless links is quite different compared to wired networks for several reasons.

Wireless networks is subdivided into two main categories: Local area networks (LAN) and Wide area networks (WAN) on the basis of the dimension of the area they cover [1][2]. In order to understand future improvements of the TCP protocol, it is useful to summarize the most important properties of Wireless link [2][3].

Wireless wide area networks present limited data rates, around some tens of kilobytes per second. Moreover, sometimes bandwidth can be asymmetric, with more capacity on the uplink than on the down and also it can oscillate over time.Wireless links are characterized by higher propagation delays than wired links. A typical Round Trip Time varies between a few milliseconds to one second. This affects TCP throughput and dereases the user's possibility to work interactively. The latencyexperienced on the link may increase unexpectedly for several reasons, such as a temporal loss of radio coverage or handover of the mobile user. This is usually referred to as delay spikes. Some wireless links are characterized by higher error rates. The error rate is determined by the current radio conditions and they can vary considerably during the connection. Forward error correction (FEC) can be used to correct the bit errors on wireless links by using an encoding algorithm, but this involves a large use of bandwidth and is efficient only in some cases.

The performance of TCP in wireless networks suffers from significant throughput degradation and very high interactive delays. TCPs congestion controlmechanism is deficient in a wireless environment because packet loss does not always mean congestion but can be due to its characteristics. Thus, the reduction of congestion window, which can occur quite frequently in a wireless environment, can result in poor performance in terms of throughput and delay. Unfortunately, when packets are lost in networks for reasons other than congestion, these measures result in an unnecessary reduction in end-to-end throughput and hence, in suboptimal performance. Hence the sender has to be made aware that some of the losses reported are not due to congestion.

QoS (Quality of Service) has now become very important part of networking. Now there is a demand of high bandwidth and low latency traffic from sender to receiver but to provide this kind of request is not always possible because of low speed links at the wireless fusion point of interconnection that passes the packets. It becomes very important how different traffic is put in queue in different situation where some traffic needs to pass quickly and some needs greater bandwidth. When congestion occurs, the main difficulty is to send big amount of packets from one end to the other end of a network [4]. In the case of congestion, the competition of flows for obtaining network resources increases [5].

A network may be very much congested so that no data could effectively be transferred. A communiction link that uses only a tiny fraction of the bandwidth may be available at the bottleneck due to poorly designed protocol. Extensive analysis and testing play a crucial role in avoiding these and similar pitfalls in protocol design.

Internet is designed to allow for communication at reasonable rates. Congestion will occur if resources are over utilized. When there is more incoming traffic than network nodes can handle, congestion occurs. Congestion results in full queues, long delays, and dropped packets that have an adverse effect on throughput. Throughput may be increased if there is a reduction at the rate at which packets are sent [6].

Congestion control problem occurs when the demand on the network resources is greater than the available resources and due to increasing mismatch in link speeds caused by intermixing of heterogeneous network technologies. This congestion problem cannot be solved with a large buffer space. If packets stay too long in the buffers, the window will time out and TCP has to retransmit the packet adding to congestion. Clearly too much traffic will lead to a buffer overflow, high packet loss and large queuing delay. Furthermore, congestion problem cannot be solved by high-speed links or with high-speed processor, because the high-speed link connected via the high-speed switch with the low-speed links will cause congestion at the wireless fusion point of interconnection. The Usage of Queue Management mechanisms in an efficient way will avoid the congestion collapse and lead to high link utilization [7].

This paper is further organized as follows. Section 2 deals with various AQM related works in wireless networks. The QoS based Congestion Control algorithms RED, WECN, QMBCCA and EQMBCCA taken for analysis are discussed in detail in Section 3. Simulator, topology, parameters used and the simulation scenarios are discussed in Section 4. Simulation results of the four algorithms are given in Section 5. Section 6 concludes the paper.

II. AQM RELATED WORKS IN WIRELESS NETWORKS

Active Queue Management (AQM) is most commonly used in wired networks, principally in backbone routers where packet loss is due to network congestion [8].

The queue management algorithm, which is applied to a router, plays an important role in providing Quality of Service (QoS) [9]. Algorithms of queue management in IP routers determine which packet should be deleted when necessary [10]. AQM for wireless networks has been studied in [11]. The design of an AQM ensuring the stability of the congestion phenomenon at a router is proposed in [12]. It was subsequently recognized that TCP with an AQM router can be considered as a feedback control system. Thus control theoretic models of AQM have been developed in [13], [14], [15] among others and these models have helped in a better understanding of router queue dynamics in TCP networks. This has resulted in systematic and scalable techniques for controller design. See [16], [17] and the references therein for a detailed overview and analysis. In [18], a detailed study about AQM with respect to wireless links is given.

Hamann and Walrand [19] suggested an algorithm called "New-ECN", which works by preventing a fast connection from opening its congestion window too quickly, but instead enabling a slow connection to open its window more aggressively. When the marked packet is echoed to the sender, not only the congestion window size is decreased but also the rate of the congestion window increase is modified.

In [20], Xu et al. presented an AQM scheme for WLAN, but the performance analysis of the proposed AQM scheme is limited, because the authors only considered the delay from wired network to WLAN. On the other hand, fairness and percentage of loss are generally accepted as more important metrics in evaluating an AQM performance.

In [21], Pang et al. mainly focused on the comparative analysis of different versions of TCPs, particularly TCP Veno and TCP Reno under RED and Tail-Drop (TD) Queue. TCP enhancement for wireless access networks is done in [22].

III. ALGORITHMS ANALYSED

This section describes specifically about the algorithms are considered for the analysis.

A. RED

Random Early Detection [23] follows the concept of Early Random Drop by introducing the measure of average queue size and also dynamically changing the drop probability.

The use of dynamic drop probability ensures that the gateway reacts differently to different level of congestion expectation. Also the instantaneous queue sizes are not a correct indicator of congestion due to burst nature of Internet traffic. RED therefore uses average queue size measured over all times. The RED gateway finds the average queue size as exponentially weighted moving average of instantaneous queue size. The minimum threshold and the maximum threshold are compared with the average queue size.

Every arriving packet is marked when the average queue size is between the maximum threshold and minimum threshold. When the mechanism ensures that the marked packets are dropped, or if all source nodes are co-operative, then the average queue size does not exceed the maximum threshold.

B. WECN

Peng et al. [24] proposed the WECN algorithm which is taken for analysis. In WECN, upon noting the first lost packet due to buffer overflow in the router, CE bit in the headers of passing packets in the queue are marked. In this manner, the sender knows that a packet loss happened due to network congestion and not because of random loss due to degraded channel quality. This mechanism clearly gives the feedback that the packet loss was due to congestion only. The TCP responds by reducing the congestion window accordingly. If a packet is lost for reason other than congestion, the packets are not marked. The TCP times out and retransmits the packets. It reduces the window but the recovery time will be much faster compared to that of a loss due to congestion. Here the unnecessary reduction of window due to packet loss by any other reason is avoided.

C. QMBCCA

Queue Management Backward Congestion Control Algorithm (QMBCCA) [25] proposed by us is an alternative approach to WECN algorithm. It makes use of the existing Internet Control Messaging Protocol (ICMP) Source Quench (ISQ) signaling mechanism. Hence congestion notification is kept at the IP level.

ISQ is generated from the point of congestion and is send to the source. This approach reduces the reaction time to congestion in the network. The ISQ message can include information on the severity of the congestion. The end host can react and adjust the window accordingly, leading to maximal use of the resources and thereby maximizing network utilization [26]. This is termed as Multilevel Explicit Congestion Notification (MECN).

The ISQ messages will be generated from the period between the initial congestion detection and until the source end system adjusts. The ISQ messages can reach the source without complication since the back path from the point of congestion to the source is not really congested. The algorithm for QMBCCA is given next.

```
For each packet arrival
     calculate the average queue size avg
      if minth <  avg  <  maxth
        {
           Calculate probability pa
           With probability pa:
             if ECT bit is set and if the packet is
                        not already marked
                  {
                      Mark the arriving packet
                      Send an ISQ to the sender
                    }
             else
                      Drop the arriving packet
          }
         else if maxth <= avg
                      Drop the arriving packet
    Send an ISQ to the source.
```

A QMBCCA capable router implements the active queue management algorithm RED in it and hence requires few modifications. Both the CE and ECT bits are maintained as in ECN and is forwarded to the transport layer when the IP message is demultiplexed.

If the incoming packet causes the average queue to go between the minimum and maximum thresholds, RED probability may choose this packet to be marked. If the ECT bit of this packet is set and if it is not already marked (i.e. the CE bit is not set) then mark the packet (i.e. set the CE bit) and send back an ISQ to the source. If the ECT bit is not set and RED chooses this packet, then the packet is dropped. If the packets are marked previously by another router, ISQs are not generated even if the packet leads to congestion. If the packet has the ECT bit set in the header and is being selected to be dropped, then an ISQ is generated even if the packet is marked earlier by some other router. Upon identifying the flow causing the congestion, the source reacts by halving both the congestion window and the slow threshold value for that flow.

The following are the advantages of QMBCCA with respect to the drawbacks in ECN. QMBCCA uses existing network layer signaling the source quench mechanisms to notify the transport layer of congestion problems. Hence all IP transport protocols are notified in the same manner about network congestion. It does not require the use of any transport layer protocol for congestion notification. It is therefore protocol independent and can be used by other transport protocols such as UDP. Also, there may be value in providing a common mechanism for notifying all transport protocols about congestion. QMBCCA has the router send the congestion notification as soon as congestion occurs in the router to the sender. QMBCCA uses MECN and hence the network resources could be more effectively utilized since the feedback gives indication of the congestion level at the overloaded point in the network. When the congestion is between the thresholds, packets are marked probabilistically and not dropped. Such advantages include lower loss rates, reduction in number of TCP timeouts and retransmissions, faster congestion notification, and lower packet delay. QMBCCA has been analysed in wired and wireless networks for TCP bulk transfers and web transfers[27][28].

D EQMBCCA

QMBCCA makes use of both ECN and BECN mechanism. In case of congestion, it generates ISQ and also sets the CE bits. There is an overhead in the generation of ISQ messages even for lighter congestion. The performance of internet routers are affected due to the overhead required for generating ISQ. ISQ generation is carried by the control plane of the routers which is compared to be slower than the data plane. The transfer of packets between the control plane and data plane should be reduced in high speed routers.

ECN notifications are more reliable compared to ISQ notifications. When the destination receives the ECN notification, it generates ECN Echo acknowledgements continuously until the sender notifies the receiver. The sender reduces the congestion window and the threshold value and activates the CWR (Congestion Window Reduced) mechanism. Any loss of the ISQ will not be detected by the source and hence the congestion notifications cannot be reliably delivered.

Additional reverse traffic will be introduced when ISQs are sent to the source and is preferable to minimize such control traffic.

On the other hand the RTT taken by the CE mechanism will result in the delay of notifying the congestion leading to higher queue variance and reduced throughput.

EQMBCCA proposed by us tries to solve these issues by using CE mechanism for low congestion and ISQ in addition to CE for high congestion [29][30].

In order to differentiate between high and low congestions an intermediate value is chosen between the min and max values of RED called IntThres (Intermediate Thresh). IntThres is configurable and can be set anywhere between min and max values. The router behaves as an ECN when the queue is below IntThres and as QMBCCA when it is above it. Whichever notification reaches the sender first should cause a window reduction.

The end hosts perform end to end negotiations initially to establish ECN capability. The sender should be responsive to both ECE and ISQ notifications. The first packet to be send after window reduction will have Congestion Window Reduced (CWR) bit to stop the receiver from sending more ECE acknowledgements. The sender waits a full RTT before it starts increasing the window.

The algorithm of EQMBCCA is given below.
For each packet arrival

```
 Calculate the average queue size avg
  If (avg > maxth)
  {
     if (ECT  bit  marked in the chosen packet)
      {
    Drop the packet;
    Send an ISQ2  back to the source;
      }
     Else drop the packet;
  }
else if (avg < minth ) queue the packet;
       else if   (minth < avg < IntThres)
   {
     if (ECT  bit  marked in the chosen packet)
      {
        Mark the CE bit if not already marked;
        else Drop the packet;
      }
   }
Else if (ECT bit marked in the chosen packet)
      {
        Mark the CE bit if not already marked;
       Send ISQ1 to the Source;
      }
Else Drop the Packet; Send ISQ2 to the source;
      }
}
```

The EQMBCCA uses a single bit in the 32 bit unused field in the ICMP source quench packet to differentiate between an ISQ due to a marked packet and an ISQ due to a dropped packet. If the bit is set, it is due to a marked packet and the sender detecting it should wait a full RTT before increasing the window according to the TCP algorithm. If the bit is not set, it is due to a dropped packet and the source halves its congestion window and starts increasing the window according to the TCP algorithm.

IV. SIMULATION SCENARIO

The Simulations for the analysis between the algorithms were done using NS2 [31]. NS2 is a discrete event simulator developed by the University of California at Berkeley and the Virtual Inter network Tested (VINT) project. The NS2 support two languages, system programming languages C++ for detail implementation and scripting languages TCL for configuring and experimenting with different parameters quickly. The NS2 has all the essential features like abstraction, visualization, emulation, traffic and scenario generation. The Trace-graph draws a graph from the Trace file.

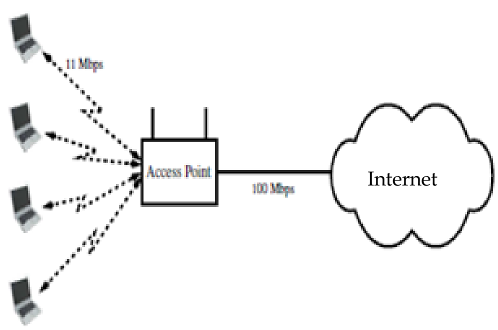

Fig. 1. Wireless Network topology for simulations

We model the TCP network by a simple topology as shown in Fig. 1 where TCP sources send data towards their destinations in Internet through a router which represents the bottleneck node in the network. With receipt of acknowledgement packet, each source increases its rate of transmission and this eventually causes the outgoing link capacity of the bottleneck node to be exceeded. When a packet drop is detected by the TCP source, it reduces its window size to half. In this way, each source attempts to determine the available capacity in the network.

The topology uses IEEE 802.11 Wireless Local Area Network (WLAN) scenario with channel data rate of 11 Mbps. The wired network LAN or Internet is connected to the access point with a link capacity of 100 Mbps.

There are TCP flows between the Internet and the wireless nodes. Lmax = 300 packets, Lmin = 100 packets, buffer size=600, IntThres=200, IEEE 802.11 propagation delay =0.8 ms, nominal round trip delay= 80ms, Probability based dropping pmax =0.2. FTP connections are varied using 5, 10, 15, 20, 25, 30, 35, 40, 45 flows to establish different levels of congestion. The FTP flows start randomly within the initial 5s of simulation while the web traffic connections start after 50s. The size of the HTTP object was 8KB and was the same for all the objects aiding in the calculation of fairness index for object transfer time. The intersession time = 0.5s, session size = 10 pages, inter page time = 10s, page size = 3 objects/page, inter object time = 0.1 s and the number of sessions is 1000. These values were chosen to ensure that several sessions were active throughout the simulation time.We consider fixed length packets, each of 1000 bytes. The maximum window size of the TCP flows is kep at the default ns-2 value of 20 packets. AQM is implemented at the access point (router). The channel between the wireless nodes and the access point is time varying due to fading. These capacity variations are performed using the model supported in ns-2 for wireless networks. The sampling time was taken as 40 ms.

The algorithms RED, WECN, QMBCCA and EQMBCCA were used for the analysis. The metrics used in this simulation scenario are ISQ reverse traffic, delay, HTTP packet loss, fairness and average web object transfer delay which is the interval between the time a web client makes an initial request and the time the server receives the acknowledgement to the last data packet for the object requested by the client

Percentage loss measures the ratio of the number of packets dropped at the bottleneck link to the total number of packets injected into the bottleneck link for a particular flow or set of flows.

V. Results and Discussions

The results of the simulation are given as graphs below for the comparative study between the algorithms.

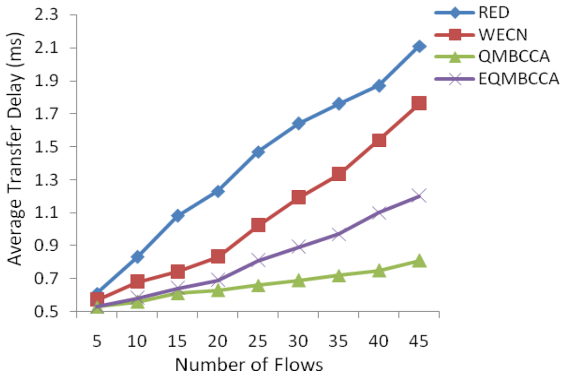

Fig. 2. Average Transfer Delay (ms)

Fig. 2. Shows the average transfer delay in ms by varying the number of flows. QMBCCA shows minimum delay compared to other algorithms. RED has the highest delay .

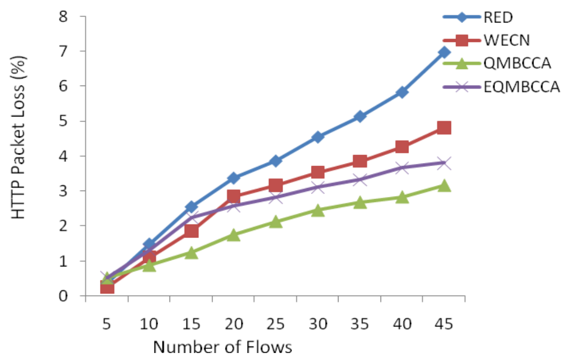

Fig. 3. HTTP Packet Loss(%)

Fig.3. shows the percentage of loss by varying the number of flows from 5 to 45. It is found that QMBCCA algorithm has the minimum number of packets dropped and when the number of flows increased; there was a constant increase in the number of packets dropped. EQMBCCA has less packet loss than WECN and RED. RTT time taken in WECN for the feedback to reach the TCP source was high. RED does not have any feedback mechanism to inform the source about the congestion and hence incurs highest packet drops.

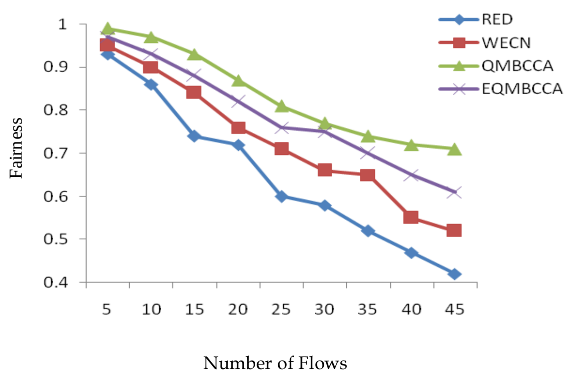

Fig. 4. Fairness among Flows

The fairness among flows for different algorithms is shown in Fig. 4. In QMBCCA, there is a two way feedback one in the forward direction using the CE bit and other in the reverse direction using ISQ leading to reliability and fast reaction. EQMBCCA makes use of ISQ messages only if the average queue size crosses IntThres. Hence the RTT taken by CE bit in EQMBCCA is high leading to slow reaction of the TCP. Hence QMBCCA outperforms EQMBCCA, WECN and RED in terms of fairness.

V. Conclusion

This paper discusses about characteristics of the wireless networks and the different AQM algorithms available in the wireless networks. The inconsistent nature of the wireless networks needs a feedback mechanism to inform the TCP whether the packet loss was due to congestion or not. The TCP responds by decreasing the window accordingly. The paper compares four QoS Congestion Control AQM algorithms namely RED, WECN, QMBCCA and EQMBCCA. The Comparisons are based on metrics gain in delay, loss of HTTP packets and fairness.

The Simulations reveal significant performance difference between the algorithms. As far as the performance is concerned QMBCCA gave better results. QMBCCA uses both ISQ and CE to inform the sender about the packet loss. The senders can react quickly and hence congestion can be rectified as early as possible. The RTT taken by EQMBCCA is more and hence its reaction time is high leading to more packet losses. It generates ISQ only after the queue size reaches the intermediate threshold value. WECN makes use of the CE bit only and hence performs lower than the proposed algorithms. RED has no feedback mechanisms at all and hence gives a overall poor performance. So it can be concluded that the proposed QMBCCA performs better than EQMBCCA in terms of deay, HTTP packet loss and fairness in a Wireless Environment.This work can be extended fot MANETs also.

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