Effect of RED and different packet sizes on Multimedia performance over wireless networks

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Abstract. We consider the adaptation of random early detection (RED) as an active queue management algorithm for multimedia traffic in wireless networks. We studied features of RED, multimedia data and point out a weakness in both. We show that an appropriate RED algorithm can improve multimedia data performance.

1 Introduction

The future mobile wireless networks are expected to merge most types of services including Internet and multimedia applications, based on Internet protocols toward IP. We can see the explosive growth in the use of wireless computer equipped with wireless network interfaces conversing with each other using IP. Nevertheless, current TCP/IP protocols do not perform well in wireless environment because of mobile environment's nature.

The main reason for the degradation of QoS in the Internet is congestion. Several works proposed to use resource management in combination with fair queueing disciplines, with traffic shaping methods and admission control mechanisms at different levels. Another approaches are adaptive (i.e. network-aware) applications at application level. On the other hand, many new protocols such as Loss-Delay Adjustment Algorithm LDA [1], Rate Adaptation Protocol RAP [3] and TCP Friendly Rate Control Protocol TFRC[4] have been developed for multimedia applications.

In contrast to the main reason of the QoS issues in wired networks, the QoS degradation in wireless networks occurs not only caused by the congestion, but also due to possible high Bit Error Rate (BER) of the radio channels, fading and interference between channels etc. Some suggestion is to use small packet size at wireless path, which help to reduce the possibility of packet errors caused by bit errors. The packet size should be flexible, because BER is not a stable factor in wireless networks.

In this paper, we study the effect of dynamic packet sizes on random early detection, which is considered as active buffer management, support QoS of multimedia traffic. We also argue that, appropriate RED algorithm can improve multimedia traffic over wireless environment.

2 What is RED

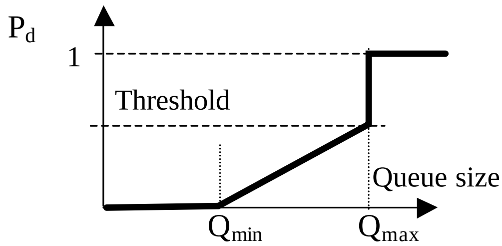
The random early detection (RED) algorithm is becoming a *de-facto* standard for congestion avoidance in the Internet and other packet switched networks. As a consequence of the incremental deployment of RED, several algorithms based on RED have been and are still being proposed to improve its performance.

RFC2309 [2] states that RED should be used as the default mechanism for managing queues in routers unless there are good reasons to use another mechanism. To this end, strong recommendations for testing, standardization and widespread deployment of active queue management in routers, to improve the performance of today's Internet are made.

How does RED operate?

The RED scheme operates as follows. RED computes the average queue size Q_{ave} and a drop probability P_d based on the instantaneous queue size and a weight factor ω. In addition, RED maintains two thresholds of queue size Q_{min} and Q_{max}. The algorithm of RED is as follows:

If $Q_{ave} < Q_{min}$, no packet drop
If $Q_{min} \leq Q_{ave} \leq Q_{max}$, drops each packet with the probability P_d.
If $Q_{ave} > Q_{max}$, drop every arriving packet.



RED is excellent in function as a traffic regulator. However, it is difficult in practice to define and to tune the parameters of RED scheme due to its dynamic nature and the problems related to synchronization of flows together with TCP. For smooth function, an adjustable factor ω is needed to determine the responsiveness of the average queue length regarding the instantaneous changes. If ω is small, RED responds very slowly to changes of queue size. In this case, the congestion may occur before any congestion control can be taken. If ω is close to 1, RED becomes more responsive but cannot be able to filter out the transient incipient congestion. Some authors (e.g. [5],[6]) suggested designing such parameters with regard to delay and load level.

3 Effect of Red on Multimedia traffic

In IP networks, Packets that bigger than MTU value of out going interface, will be broken to some smaller packet before sending. Each of these small packet has its own IP header and offset information in order to help IP layer at receiver to reunite.

Even MTU is bigger than the suggested value of packet size mentioned in [8] for wireless networks, but Multimedia data frames still be broken, because multimedia data is usually very big, such as video frame, voice, music, etc.

There is a problem with fragmentation at IP layer: when one piece of those fragments can not reach to the destination, which is often in error channels like wireless networks, the whole data packet will be discard at IP layer, without any inform to higher layer. In its turn, Multimedia application hardly recover the lose packet. Some existing solutions suggest using FEC for some data packets, so that the higher layer can recover loss packets from successful ones. However it cost much CPU resource and times delay, especially when data packet size is big such as in multimedia case.

With RED, new incoming packet will be dropped probability P_d to avoid congestion. This means the newer packets, which have more valuable for Multimedia data will be discarded, instead of older packets. More over, RED influents multimedia performance, since dropped packet causes whole data frame at application layer become useless, or wasting more resource to recover the error.

We suppose a method to help RED working more efficiently. RED should recognize fragment from one data frame and when it needs to drop one fragment it also drops others from the same frame. RED should also drop old packet in queue with probability Pd, so that sender can detect faster when some dropping occurred.

In order to achieve those features, not only RED getaway must be changed, but also it needs support from transport protocol.

4 Design Issues

RED should recognize fragment from one data frame and when it needs to drop one fragment it also drops others from the same frame. We suggest that the transport protocol of multimedia mark frame ID at IP packet header, so RED gateway can recognize fragments, which belong to the same data frame.

The RED is modify as following:

```
Initialization:
        avg ← 0
        count ← -1
for each last packet of a data frame arrival
        calculate new avg. queue size avg:
                if the queue is nonempty
                        avg ← (1-ϖ)avg + ϖ.q
                else
                        m ← f(time-q_time)
                        avg←(1-ϖ)^m .avg
                if Qmin<= avg< Qmax
                        increment count
                        calculate Pa:
                                Pb ← max(avg -  Qmin)/(Qmax - Qmin)
                                Pa ← Pb/(1-count.Pb)
                        With Pa:
                                If Pa< avg/q
                                Drop packets belong to old frame, in queue
                                count ← 0
                else if Qmax <= avg
                        Drop the arriving packet
                        count←0
                else count←-1
when queue becomes empty
        q_time <time
```

Variable:

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avg        : average queue size
q_time     : start of the queue idle time
count      : packets since last dropped packet
ϖ          : queue weight
Qmin,Qmax: min and max of threshold for queue
q          : current queue size
time       :current time
f(t)       : a linear function of the time t
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A transport Protocol support dynamic data packet size for multimedia

We propose a new protocol, running over UDP, which allows application layer protocols to classify data depending on data's dependency of time sensitive, loss sensitive. With help of Proxy at Base station, wireless errors can be distinguished with congestion errors. Packet size is calculated to optimize through put, depend on BER level.

There is a Base station (BS) which is connected to the wired network on one end and to the wireless network on the other end. We consider the transmission in both directions separately: transmission from mobile host(MH) to fixed host(FH), bandwidth compensation in scheduling at BS is used for error control behavior.

Wireless Bit Error rate and Dynamic packet length

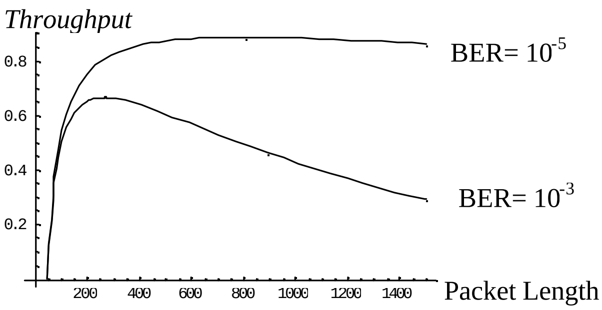

Illustration Throughput with BER » 10^{-4}

However, BER changes frequently and hardly to be measured. We supposed that errors occur in wireless path caused by BER. Thus, when wireless errors detected Sender at MH or BS will reduce packet length to ½ to reduce possibility of Packet corruption.

To detect quickly wireless Error, besides containing sequent number, packet header also contains packet predictor number, which informs BS a number of packet will be sent next time.

Error and Congestion control

Wireless error can be detected by ACK packets send back to sender from receiver when data packet is successfully received. When RED discards packet in it queue, sender will faster detect the error.

The congestion occurred when buffer at BS is overflow or when after some time out but no more packets (data or ACK) arrives from FH. BS will inform MH and FH in every ACK packet about status of Congestion. Sender at FH will reduce sending rate to ½ according to TCP behavior. When the link is good, the sender can increase sending rate according to EIMD mechanism.

Support RED gateway to detect frame in the same data frame

TOS is one unused field at IP layer, this field can be used to stored data frame ID. This field can be read by RED gateway at basestation, so that BS can manage buffer actively.

Order control

Packet in one reliable frame can be arrived in different order because of error and retransmission. The frame identifier and sequence number is enough information for the receiver recreates the correct order. Packets of Normal type will be discarded after packets from newer frame arrived.

5 Conclusion

We study the effect of dynamic packet sizes on random early detection, which is considered as active buffer management, support QoS of multimedia traffic. We also studied features of RED, multimedia data and point out a weakness in both. We then propose modification in RED algorithm, so that it can improve multimedia traffic over wireless environment. A transport protocol supports multimedia data by using dynamic packet size is also introduced. This protocol works with new RED algorithm, to help BS get better active buffer management.

6 References

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