Dynamic Packet Size Mechanism (DPSM) for Multimedia in Wireless networks

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Abstract. Due to the characteristics of mobile networks such as low speed, high error bit rate, variable link throughput etc, the applications over the wireless channel face with many challenging problems, especially multimedia applications, where the volume of data is huge. Without a suitable control, the Quality of Service (QoS) of such applications is adversely affected and the connection quality may become unacceptable. In this paper, we propose a dynamic Packet size mechanism(DPSM) adapting with a rate control proactive protocol at the base station of the wireless network. We argue that adaptation is necessary in a changing environment where static configuration of error control does not yield the best performance for many situations.

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 current Internet protocols offers only best-effort service, i.e. no Quality of Service (QoS including delays, jitters and losses), thus there have been considerable works in the research community in order to provide QoS in the Internet. There are many commercial media streaming players that are currently developed for Internet, such as Realplayer, Vxtreme and Microsoft Netshow. Although they claim to be adaptive, full QoS support, but no analysis is available to prove any claims. In order to extend service models for Internet, two major models, Integrated Services [1] and Differentiated Services [2] have been proposed in recent years. In these models, the IP layer is modified for providing better QoS to various applications or classes of applications.

The main reason for the degradation of QoS in the Internet is congestion. Several works proposed to use resource management in combination with fair queuing 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 [3], Rate Adaptation Protocol RAP [4] and TCP Friendly Rate Control Protocol TFRC[5] 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. In this case, the usual retransmission for recovering lost packets is not suitable and not efficient any longer. The architectures and mechanisms developed for wired networks can not be directly applicable to mobile networks without considerations.
In this paper, we propose a proactive QoS scheme based on a Rate Control proActive Protocol (RCAP) with Dynamic Packet Size Mechanism which will help application working in wireless network getting better performance. We argue that beside the proactive rate control, tuning to suitable packet size will keep wireless network to distinguish from QoS degradation caused by Bit Error Rate remarkably.

2 Background and Related Works

Congestion control and the principle of additive increase/multiple decrease AIMD have been well studied in the past. In wired networks packet losses are mainly due to congestion, thus TCP schemes and other TCP-like schemes such as LDA[3], RAP[4], TFRC [5] are based on this principle in order to adjust the transmission rate of the sender. All these schemes use closed-loop control, i.e. the feedback signal (acknowledgement ACK) from the receiver. When a packet loss is detected by a gap in the packet sequence numbers or by timeout, all previous schemes decrease the transmission rate. If no congestion, i.e. no packet loss occurs, the transmission rate is increased after every round-trip time (RTT). In fact, these schemes function effectively in the context of wired networks such as in the Internet.
However, the wireless networks have different characteristics. Link bandwidth of wireless networks depends on several factors including environmental conditions, location of the mobile hosts, etc. Moreover, the wireless links are characterised not only by a high link error rate but also by a highly time-varying bandwidth.
Under high bit error environments like wireless networks, it is possible that the performance is almost zero, regardless of congestion. The higher the packet error rate, the more “holes” in the stream of data. To recover from the holes, TCP either needs to wait for timeouts, which reduce throughput dramatically, or wait for duplicate ACKs. The high packet error rate also reduces the congestion window size (there are fewer packets in flight) and decreases the chances to get duplicated ACKs because there are few packets in the window in the first place. So, if the packet error rate is higher than a specific value, it is possible that the throughput is almost zero.
This problem is well studied in numerous works. A summary can be found e.g. in [6]. In [7], a snoop protocol has been developed to improve the performance of TCP flows over wireless links. The snoop agent located at the base station caches every packet, suppresses the duplicate acknowledgements (ACKs) and triggers the retransmission of packets accordingly. However, TCP is not suitable for delivery of multimedia as argued in several works such as [5,8]. In [8], a rate control scheme was proposed for TCP-friendly connections over high bit error networks such as satellite networks. This scheme uses dummy packets to probe the available network resources.

The previous schemes have not addressed the problems due severe channel impairments and the simultaneous occurrence of congestion and wireless losses. In addition, there is a doubt if congestion or link error interval is behind the range of the round-trip time (RTT). Dummy packets proposed by [8] may be lost and bring also many overheads.
Other works proposed are compensation mechanisms and channel predictions at lower time scales (see e.g. [9] and other references herein). Channel prediction was proposed to allow the sender to probe the state of wireless channel before any packet transmission. As indicated in [9] and other schemes, a handshake mechanism using RTS/CTS (Request to send/clear to send) signal pair is applied which usually operates at the MAC layer of the base station and mobile hosts.

Our work differs from previous works, as we propose an adaptive Dynamic Packet Size Mechanism (DPSM) in combination with a QoS watchdog at the networking layer of the Base Station, which takes into account the bandwidth estimation and the channel prediction as well as high bit error detection. Our proactive scheme attempts to take the advantage of cross-layer information by combining this QoS watchdog with a Dynamic Packet Size Mechanism and a rate control protocol.

3 Design Issues

We focus our scheme on two parts. We believe that the combined solution can effectively improve network Multimedia application performance in the wireless networks.
The first part is RCAP cored protocol (see [12]), a TCP friendly end-to-end scheme which was built from the idea of AIMD principle ([10]) and wireless bandwidth estimation ([11]) in heterogeneous wired and wireless environment.
The second part is Dynamic Packet Size Mechanism, which will optimize the size of packet to best value according to status of wireless network.

3.1 Dynamic Packet Size Mechanism (DPSM)

In wireless environments, packets transmitted over the air may have bit errors introduced. If we call that the bit error rate (BER) is the possibility for one bit to be corrupted, then the packet error rate (PER) grows as following formula when the sender increases the packet size:

$$PER = 1 - (1 - BER)^{(\text{length of packet in bits})}$$

According to that formula, we can reveal the relationship between PER and BER. In this situation, we assume that bit error rate is constant over time, so the distance between two consecutive bit errors is exponentially distributed over the bit stream.
The bit error rate (BER) experienced in wireless networks may also change over time, like the available bandwidth above. Because of the high packet loss rate, TCP may suffer under the change of bit error rate over time. TCP cannot efficiently recover from high packet loss rates.
We propose to change packet sizes dynamically and to probe for bit error rates to avoid the effect of high bit error rate in wireless environment.

Our DPSM algorithm is as following:

- Discover an incoming packet's corruption by the failure of the link-level checksum. If the IP packet is not corrupted, the router simply forwards the packet toward the destination.
- Notification by the failure of the Packet checksum or by an optional flag in the packet header. The receiver then sets the echo flag in the corresponding acknowledgment packet. This separate and explicit flag is necessary since the ACK packet itself may experience corruption during its return.
- Sender Reaction: after receiving the marked ACK packet, The sender then reduces the packet size by an adaptive factor (multiplicative decrease, but additive decrease may also apply).
- Sender Action: When it receives ACK packets for new data, the sender opportunistically increases the packet size by a second adaptive factor (additive increase, but multiplicative increase may also apply).

3.2 Adaptive DPSM with RCAP Protocol

In this section, we present a TCP-like rate control scheme for real-time constrained traffic such as streaming multimedia. Our scheme consists of a QoS watchdog for QoS monitoring (therefore we call "proactive" scheme) at the base station and the protocol agents at source and destination as depicted in Figure 1.

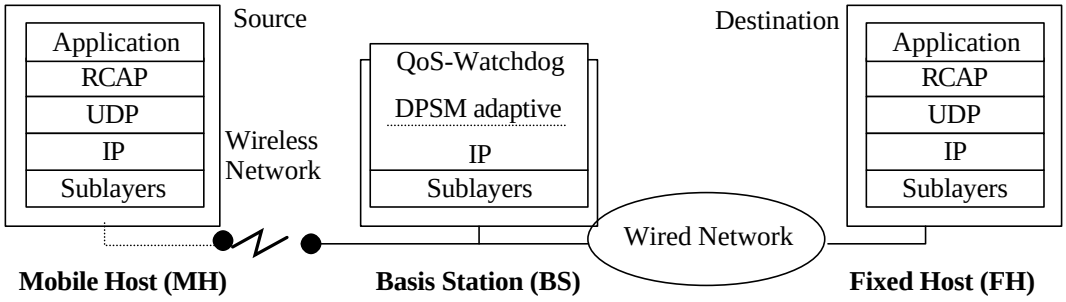


Figure 1. Network model of the proactive scheme

The RCAP agents operate in the application space, and run on top of UDP. No modification of the TCP/UDP/IP stack is needed at the endsystems.

The wireless link should be especially considered because it is more critical than the wired links. In our opinion, QoS Watchdog at Base Station (BS) is responsible for the sender at mobile host (MH) to quickly know the wireless losses, in order to make the relevant decision. Nevertheless, our QoS watchdog has additional function of bandwidth estimation as well as Bit Error detection (with DPSM). The watchdog agent is depicted above the IP layer (Figure1). However, this agent only carries out the support function to the QoS control and is in fact not an additional layer (such as transport layer) at the base station. In practice, the IP layer at the base station is to be modified to add this agent for QoS control.

3.3 The QoS watchdog

Qos Watchdog can detect Bit Error in a packet by the use of checksums. Corrupted packets are useless for the end hosts because TCP/IP has no mechanism to recover from individual bit errors. Thus, routers or hosts sensibly drop a packet silently. If PER is bigger than the specific value, Qos Watchdog will inform the sender.
Normally, there is reasonable level of Bit error in transmission that Wireless channel can accept. QoS watchdog will detect Bit Error only when PER is higher then that level.
We have to consider the transmission in both directions separately: transmission from the mobile host (MH) to a fixed host (FH) and transmission from the fixed host to a mobile host.

- Transmission from MH to FH:

The QoS watchdog has to estimate the wireless link bandwidth R1. We take the bandwidth estimation method from our earlier works [10] as follows:

$$R_{est} = \frac{L^k}{T}\left(1 - e^{-KT}\right) + R_{old}.e^{-KT} \quad (1)$$

The estimated rate Rest is periodically updated by arriving of each packet based on an average rate Rold*e-KT and a variance, where Lk is the size of arriving packet k. K is a constant, and T is the arrival interval between two consecutive arriving packets. This term allows us to achieve the asymptotic established value for the bandwidth estimation.
Receiving packets from the sender without any corruption, the QoS watchdog inserts the value R1 in the packet headers, and forwards the packets onto the receiver. At the receiver, the bottleneck bandwidth of the connection along the path is estimated as R_2 (using Jacobson's Pathchar). Thus, the initial sending rate of the connection is calculated as $R_{init} = \mathbf{min}(R_1, R_2)$ which is then sent back to the sender using an ACK packet. It also informs sender to increase packet size if the size is reduced before.
The RCAP source sends data packets with the sequence numbers. The QoS watchdog monitors the gap in the sequence of packets to detect losses due to wireless channel errors.
If packet corruption and losses are detected from packets sent from the sender, the QoS watchdog silently discards the damaged ones and informs the sender with marked ACK to ask it to reduce the packet size.
Loss detection is also assumed if a timeout happens at the QoS watchdog. If a loss is detected, the QoS watchdog sends an explicit wireless loss indication to the RCAP source/destination to inform it about the situation. We emphasize here that the sending of loss indication is piggybacked (inband), no additional connection is needed. Also, the RCAP source starts a backoff period when a wireless error is detected during which it defers the transmission of further packets. It tries to communicate with the QoS watchdog to periodically probe the wireless channel. When the wireless channel becomes good again, a positive feedback is sent from the QoS watchdog to the sender.
The algorithm of QoS watchdog in this transmission direction is as follows:

If packet arrival

- If (data packet from MH sender)
  - Estimation of rate
  - Forwarding packet to the FH receiver
  - Updating sequence number
- If (Bit Error detected)
  - Inform MH to reduce packet size
- If (gap in sequence number)
  - Sending wireless loss indication to the FH receiver
  - Inform MH to reduce packet size
- If (ACK from FH)
  - Checking the gap of ACK packets
  - Forwarding packet to the MH sender
  - Updating ACK sequence number
  - Inform MH to resume packet size

If (timeout)

- Sending wireless loss indication to the FH receiver
- Waiting for the probe packet from the MH sender or the connection close indication

- Transmission from FH to MH

In order to quickly know about the wireless channel state, we claim that the RCAP receiver sends back an ACK for any round-trip-time (RTT). Nevertheless, the size of ACK packet is small in comparison to data packets, thus the overhead is kept small. The bandwidth estimation is done as in the transmission from MH to FH. The sender sets an initial sending rate accordingly.

By monitoring the sequence number of packets, the QoS watchdog detects the congestion loss in the path from the fixed host to the base station, if a gap in packet sequence is found. In this case, the QoS watchdog sends a congestion loss indication message to the sender at FH. This indication is piggybacked in the ACK packet which will be sent from the MH to FH. After forwarding each packet to the MH, QoS watchdog waits for the ACK packet from the MH receiver. If a timeout, Bit Error Flag on ACK packets or a gap in sequence number of ACK packets is detected, the wireless channel is assumed as bad. The QoS watchdog asks MH sender to reduce packet size first to see any effecting. If the problem can not be fixed, it informs the FH about that and the RCAP sender at the FH side defers the transmission (within the deadline time of packets) by entering a backoff interval waiting for a positive acknowledgement from the QoS watchdog.

The algorithm of QoS watchdog in the transmission from the fixed host FH to the mobile host MH is as follows:



If packet arrival

- If (data packet from FH sender)
  - Estimation of rate
  - Forwarding packet to the MH receiver
  - Updating sequence number
- If (gap in sequence number)
  - Sending congestion loss indication to the FH sender
  - Ask FH to Optimize Packet size.
- If (ACK from MH receiver)
  - If gap in sequence number of ACKs
    insert wireless loss indication in ACK
    reduce packet size in buffer or ask FH reduce packet size
  - If Bit Error Flag in ACK is on
    reduce packet size in buffer or ask FH to reduce packet size
  - Forwarding ACK packet to the MH receiver
  - Updating ACK sequence number

If (timeout by waiting ACK from MH)

- Sending wireless loss indication to the FH sender
- Waiting for wireless channel probe until a positive ACK from MH, then sending indication to FH

The function of the QoS watchdog can be enhanced by combining with the fair queueing scheme at the base station. No reservation is then needed. The weight of connection can be set by the estimated bandwidth, which is usually the rate according to the minimum quality requirement of the connection. Furthermore, the bad period (i.e. error interval) of a wireless channel can be credited and the connection can be compensated later when the channel becomes good again. In addition, the QoS watchdog may work together with the buffer management and a proxy at the higher level in the base station. One such high-level proxy is described in [10]. According to the estimation of wireless bandwidth, some low priority packets (for instance low priority video packets of layered coding scheme) can be discarded. A transcoding of data (e.g. different colors, picture sizes, resolutions etc.) is then useful, thus the wireless channel is not overloaded while maintaining an acceptable QoS to connections. However, this is the topic of further work.

4 Conclusion

The paper proposes an adaptive dynamic packet size mechanism with a rate control proactive scheme based on link bandwidth estimation and TCP-friendly congestion control for multimedia flows in wireless networks. With the bandwidth estimation method, not only the link state is predicted but also the wireless link available bandwidth can be estimated, thus the wireless link can be efficiently exploited. In case of a bit error, packet size will be reduced to avoid possibility of corruption. This proposal is simple but rather effective since the possibility of packet corruption depends much on size of packet. If this can not help, the transmission is temporally suspended in order to reduce the loss rate and the power use of wireless terminals.

After the link is recovered, the previous packet size and rate is resumed, thus the performance is improved. By congestion, the control scheme behaves in a TCP-friendly fashion thus a fairness of different flows of different protocol types can be gained without starvation of other flows. The new scheme shows the comparable throughput with the TCP protocol. We are currently investigating the adaptation of the new scheme since there is a level of packet size under which packet sent is not effective due to the unacceptable ratio between bits of packet headers and real data in packets.

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