

Adaptive Transmission of Multimedia Data over UMTS

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INTRODUCTION

As communications technology is being developed, users' demand for multimedia services raises. Meanwhile, the Internet has enjoyed tremendous growth in recent years. Consequently, there is a great interest in using the IP-based networks to provide multimedia services. One of the most important areas in which the issues are being debated is the development of standards for the universal mobile telecommunications system (UMTS). UMTS constitutes the third generation of cellular wireless networks which aims to provide high-speed data access along with real-time voice calls. Wireless data is one of the major boosters of wireless communications and one of the main motivations of the next-generation standards.

Bandwidth is a valuable and limited resource for UMTS and every wireless network in general. Therefore, it is of extreme importance to exploit this resource in the most efficient way. Consequently, when a user experiences a streaming video, there should be enough bandwidth available at any time for any other application that the mobile user might need. In addition, when two different applications run together, the network should guarantee that there is no possibility for any of the above-mentioned applications to prevail against the other by taking all the available channel bandwidth. Since Internet applications adopt mainly TCP as the transport protocol, while streaming applications mainly use RTP, the network should guarantee that RTP does not prevail against the TCP traffic. This means that there should be enough bandwidth available in the wireless channel for the Internet applications to run properly.

BACKGROUND

Chen and Zachor (2004) propose a widely accepted rate control method in wired networks which is the equation-based rate control, also known as TFRC (TCP-friendly rate control). In this approach the authors use multiple TFRC

connections as an end-to-end rate control solution for wireless streaming video. Another approach is presented by Fu and Liew (2003). As they mention, TCP Reno treats the occurrence of packet losses as a manifestation of network congestion. This assumption may not apply to networks with wireless channels, in which packet losses are often induced by noise, link error, or reasons other than network congestion. Equivalently, TCP Vegas uses queuing delay as a measure of congestion (Choe & Low, 2003). Thus, Fu and Liew (2003) propose an enhancement of TCP Reno and TCP Vegas for the wireless networks, namely TCP VenO.

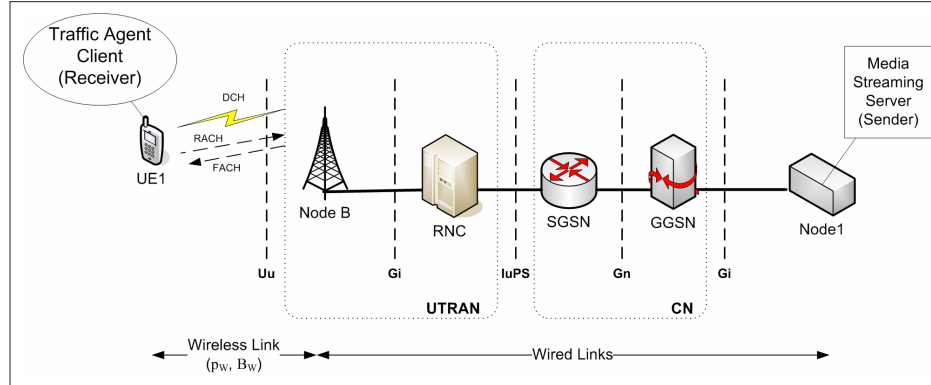
Chen, Low, and Doyle (2005) present two algorithms that formulate resource allocation in wireless networks. These procedures constitute a preliminary step towards a systematic approach to jointly design TCP congestion control algorithms, not only to improve performance, but more importantly, to make interaction more transparent. Additionally, Xu, Tian, and Ansari (2005) study the performance characteristics of TCP New Reno, TCPSACK, TCP VenO, and TCP Westwood under the wireless network conditions and they propose a new TCP scheme, called TCP New Jersey, which is capable of distinguishing wireless packet losses from congestion.

Recent work provides an overview of MPEG-4 video transmission over wireless networks (Zhao, Kok, & Ahmad, 2004). A critical issue is how we can ensure the QoS of video-based applications to be maintained at an acceptable level. Another point to consider is the unreliability of the network, especially of the wireless channels, because we observe packet losses resulting in a reduction of the video quality. The results demonstrate that the video quality can be substantially improved by preserving the high-priority video data during the transmission.

THE TCP-FRIENDLY RATE CONTROL PROTOCOL

TFRC is not actually a fully specified end-to-end transmission protocol, but a congestion control mechanism that is designed

Figure 1. Typical scenario for streaming video over UMTS



to operate fairly along with TCP traffic. Generally TFRC should be deployed with some existing transport protocol such as UDP or RTP in order to present its useful properties (Floyd, Handley, Padhye, & Widmer, 2000). The main idea behind TFRC is to provide a smooth transmission rate for streaming applications. The other properties of TFRC include slow response to congestion and the opportunity of not aggressively trying to make up with all available bandwidth. Consequently, in case of a single packet loss, TFRC does not halve its transmission rate like TCP, while on the other hand it does not respond rapidly to the changes in available network bandwidth. TFRC has also been designed to behave fairly when competing for the available bandwidth with concurrent TCP flows that comprise the majority of flows in today's networks. A widely popular model for TFRC is described by the following equation (Floyd & Fall, 1999):

$$T = \frac{kS}{RTT\sqrt{p}} \quad (1)$$

T represents the sending rate, S is the packet size, RTT is the end-to-end round trip time, p is the end-to-end packet loss rate, and k is a constant factor between 0.7 and 1.3 (Mahdavi & Floyd, 1997) depending on the particular derivation of equation (1).

The equation describes TFRC's sending rate as a function of the measured packet loss rate, round-trip time, and used packet size. More specifically, a potential congestion in the nodes of the path will cause an increment in the packet loss rate and in the round trip time according to the current packet size. Given this fluctuation, it is easy to determine the new transmission rate so as to avoid congestion and packet losses. Generally, TFRC's congestion control consists of the following mechanisms:

1. The receiver measures the packet loss event rate and feeds this information back to the sender.
2. The sender uses these feedback messages to calculate the round-trip-time (RTT) of the packets.

3. The loss event rate and the RTT are then fed into the TRFC rate calculation equation (described later in more detail) in order to find out the correct data sending rate.

ANALYSIS OF THE TFRC MECHANISM FOR UMTS

The typical scenario for streaming video over UMTS is shown in Figure 1, where the server is denoted by Node1 and the receiver by UE1. The addressed scenario comprises a UMTS radio cell covered by a Node B connected to an RNC. The model consists of a UE connected to DCH, as shown in Figure 1. In this case, the DCH is used for the transmission of the data over the air. DCH is a bi-directional channel and is reserved only for a single user. The common channels are the forward access channel (FACH) in the downlink and the random access channel (RACH) in the uplink.

The wireless link is assumed to have available bandwidth B_w , and packet loss rate p_w , caused by wireless channel error. This implies that the maximum throughput that could be achieved in the wireless link is $B_w(1 - p_w)$. There could also be packet loss caused by congestion at wired nodes denoted by $p_{node\ name}$ (node name: GGSN, SGSN, RNC, Node B). The end-to-end packet loss rate observed by the receiver is denoted as p . The streaming rate is denoted by T . This means that the streaming throughput is $T(1 - p)$. Under the above assumptions we characterize the wireless channel as underutilized if $T(1 - p) < B_w(1 - p_w)$. Given the above described scenario, the following are assumed:

1. The wireless link is the long-term bottleneck. This means that there is no congestion due to streaming traffic to the nodes GGSN, SGSN, and RNC.
2. There is no congestion at Node B due to the streaming application, if and only if the wireless bandwidth is underutilized—that is, $T(1 - p) < B_w(1 - p_w)$. This also implies that no queuing delay is caused at Node

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