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.

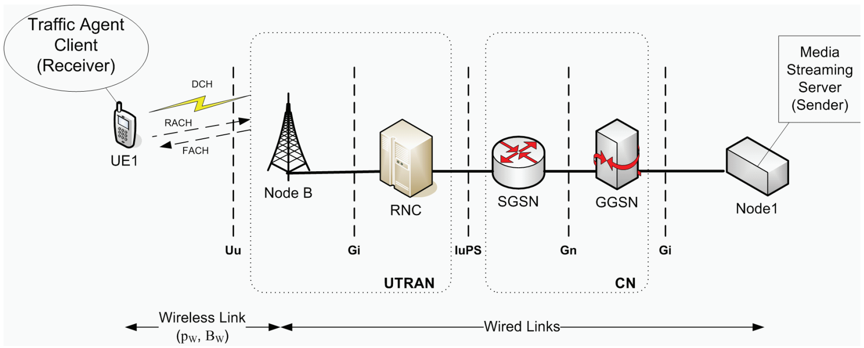
Chen, Low, and Doyle (2005) present two algorithms that formulate resource allocation in wireless networks. These procedures constitute a preliminary step towards a systematic approach to jointly design TCP congestion control algorithms, not only to improve performance, but more importantly, to make interaction more transparent. Additionally, Xu, Tian, and Ansari (2005) study the performance characteristics of TCP New Reno, TCP SACK, 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):

$$T = \frac{kS}{RTT\sqrt{p}} \tag{1}$$

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

B and hence, the round trip time for a given route has the minimum value (i.e., RTT_{min}). Thus, this assumption can be restated as follows: for a given route, $RTT = RTT_{min}$ if and only if $T(1 - p) \leq B_W(1 - p_W)$. This in turn implies that if $T(1 - p) > B_W(1 - p_W)$ then $RTT \geq RTT_{min}$.

3. The packet loss rate caused by wireless channel error (p_W) is random and varies from 0 to 0.16.
4. The backward route is error-free and congestion-free.

The communication between the sender and the receiver is based on RTP/RTCP sessions; and the sender, denoted by Node 1 (Figure 1), uses the RTP protocol to transmit the video stream. The client, denoted by UE1 (Figure 1), uses the RTCP protocol in order to exchange control messages. The mobile user in recurrent time space sends RTCP reports to the media server. These reports contain information about the current conditions of the wireless link during the transmission of the multimedia data between the server and the mobile user. The feedback information contains the following parameters:

- **Packet Loss Rate:** The receiver calculates the packet loss rate during the reception of sender data, based on RTP packets sequence numbers.
- **Timestamp of Every Packet Arrived at the Mobile User:** This parameter is used by the server for the RTT calculation of every packet.

The media server extracts the feedback information from the RTCP report and passes it through an appropriate filter. The use of filter is essential for the operation of the mechanism in order to avoid wrong estimations of the network conditions. On the sender side, the media server using the feedback information estimates the appropriate rate of the streaming video so as to avoid network congestion. The appropriate transmission rate of the video sequence is calculated from equation (1), and the media server is responsible for adjusting the sending rate with the calculated value. Obviously, the media server does not have the opportunity to transmit the video in all the calculated sending rates. However, it provides a small variety of them and has to approximate the calculated value choosing the sending rate from the provided transmission rates.

This extends the functionality of the whole congestion control mechanism. More specifically, the sender does not have to change the transmission rate every time it calculates a new one with a slight difference from the previous value. Consequently, it changes the transmission rate of the multimedia data to one of the available sending rates of the media server as has already been mentioned. In this approach, the number of the changes in the sending rate is small and the mobile user does not deal with a continually different transmission rate.

As mentioned above, it is essential to keep a history of the previous calculated values for the transmission rate. Having this information, the media server can estimate the smoothed transmission rate, using the m most recent values of the calculated sending rate from the following equation:

$$T^{Smoothed} = \frac{\sum_{i=1}^{m} w_i \cdot T_{m+1-i}^{Smoothed}}{\sum_{i=1}^{m} w_i} \tag{2}$$

The value m, used in calculating the transmission rate, determines TFRC's speed in responding to changes in the level of congestion (Handley, Floyd, Padhye, & Widmer, 2003). The weights w_i are appropriately chosen so that the most recent calculated sending rates receive the same high weights, while the weights gradually decrease to 0 for older calculated values.

Equivalently to the calculation of the transmission rate, the mobile user (client) measures the packet loss rate p_l based on the RTP packets sequence numbers. This information is sent to the media server via the RTCP reports. In order to prevent a single spurious packet loss having an excessive effect on the packet loss estimation, the server smoothes the values of packet loss rate using the filter of the following equation, which computes the weighted average of the m most recent loss rate values (Vicisiano, Rizzo, & Crowcroft, 1998).

$$p_1^{Smoothed} = \frac{\sum_{i=1}^{m} w_i \cdot p_{1,m+1-i}^{Smoothed}}{\sum_{i=1}^{m} w_i} \tag{3}$$

The value of $p_1^{Smoothed}$ is then used by equation (1) for the estimation of the transmission rate of the multimedia data. The weights w_i are chosen as in the transmission rate estimation.

FUTURE TRENDS

The most prominent enhancement of the adaptive real-time applications is the use of multicast transmission of the multimedia data. The multicast transmission of multimedia data has to accommodate clients with heterogeneous data reception capabilities. To accommodate heterogeneity, the server may transmit one multicast stream and determine the transmission rate that satisfies most of the clients (Byers et al., 2000). Additionally, Vickers, Albuquerque, and Suda (1998) present different approaches where the server transmits multiple multicast streams with different transmission rates allocating the client at these streams, as well as using layered encoding and transmitting each layer to a different multicast stream. An interesting survey of techniques for multicast multimedia data over the Internet is presented in Li, Ammar, and Paul (1999).

Single multicast stream approaches have the disadvantage that clients with a low-bandwidth link will always get a high-bandwidth stream if most of the other members are connected via a high-bandwidth link, and the same is true the other way around. This problem can be overcome with the use of a multi-stream multicast approach. Single multicast stream approaches have the advantages of easy encoder and decoder implementation and simple protocol operation, due to the fact that during the single multicast stream approach, there is no need for synchronization of clients' actions (as the multiple multicast streams and layered encoding approaches require).

The subject of adaptive multicast of multimedia data over networks with the use of one multicast stream has engaged researchers all over the world. During the adaptive multicast transmission of multimedia data in a single multicast stream, the server must select the transmission rate that satisfies most the clients with the current network conditions. Totally, three approaches can be found in the literature for the implementation of the adaptation protocol in a single stream multicast mechanism: equation based (Rizzo, 2000; Widmer & Handley, 2001), network feedback based (Byers et al., 2000), or a combination of the above two approaches (Sisalem & Wolisz, 2000).

CONCLUSION

An analysis of the TCP friendly rate control mechanism for UMTS has been presented. The TFRC mechanism gives the opportunity to estimate the appropriate transmission rate of the video data for avoiding congestion in the network. The three goals of this rate control could be stated as follows. First, the streaming rate does not cause any network instability (i.e., congestion collapse). Second, TFRC is assumed to be TCP Friendly, which means that any application that transmits data over a network presents friendly behavior towards the other flows that coexist in the network and especially towards the TCP flows that comprise the majority of flows in today's networks. Third, it leads to the optimal performance—that is, it results in the highest possible throughput and lowest possible packet loss rate. Furthermore, an overview of video transmission over UMTS using real-time protocols such as RTP/RTCP has been presented.

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KEY TERMS

Adaptive Real-Time Application: An application that has the capability to transmit multimedia data over heterogeneous networks and adapt media transmission to network changes.

Delay Jitter: The mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

Frame Rate: The rate of the frames, which are encoded by video encoder.

Multimedia Data: Data that consist of various media types like text, audio, video, and animation.

Packet Loss Rate: The fraction of the total transmitted packets that did not arrive at the receiver.

RTP/RTCP: Protocol used for the transmission of multimedia data. The RTP performs the actual transmission, and the RTCP is the control and monitoring transmission.

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