Combining Transport Layer and Link Layer Mechanisms for Transparent QoS Support of IP based Applications

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Abstract

Providing IP services with QoS support in heterogeneous networks involving wired and wireless sections has been identified as a challenging task [6]. The goal of this paper is to show how link level mechanisms can be enhanced by transport layer information in order to meet transport layer reliability and delay requirements within wireless scenarios. Error control mechanisms in Layer 2, Layer 4, and a combination of them are compared. The benefit for TCP-based and RTP/UDP-based applications of using a CDMA based MAC protocol that employs "**S**imultaneous **M**AC **P**acket **T**ransmission" (SMPT) are demonstrated. It is investigated how the capability of SMPT to use one or several wireless channels can be exploited to stabilize jitter and reduce losses caused by wireless error control. Schemes are presented that allow to control the number of channels used by SMPT depending on Layer 4 timing information.

1 Introduction

Mobile terminals with wireless access links may show dissatisfying network performance due to limited wireless bandwidth in combination with high bit error rates as well as high burst error rates. As wireless bandwidth is a scarce resource, reduction on needed wireless bandwidth is an important issue. At the same time, end-to-end performance crucially depends on error recovery mechanisms selected [10, 1, 2]. Without link level error control for the wireless link, end-to-end error control mechanisms cannot recover errors efficiently. Where link level error control schemes are applied, their poor adaptation to end-to-end requirements may lead to poor throughput or delay. When performing a large number of retransmissions, a wireless link layer protocol may introduce large delays to IP packets that are affected by the retransmissions, and also to subsequent IP packets.

2 Requirements Of The Transport Layer

In this paper we demonstrate how layer 4 information of transport layers with **T**ransmission **C**ontrol **P**rotocol (TCP) and **U**ser **D**atagram **P**rotocol (UDP) can be exploited in order to influence layer 2 mechanisms at the wireless host and at the base station.

2.1 TCP Based Applications

TCP provides a reliable transport layer by calculating timeouts, in the following also called **R**etransmission **T**ime**O**ut (RTO), for each segment [16]. If the segment is not acknowledged before the timer is expired, TCP will retransmit the segment once again. The **R**ound **T**rip **T**ime (RTT) for each segment is measured by the sender side TCP entity and will be used to calculate the RTO. Calculation of the timeouts is described in [5, 15]. Obviously timeouts lead to throughput degradation and increased delay due to the retransmissions that are performed, and due to reduction of the congestion window.

2.2 RTP/UDP Based Applications

UDP, in contrast to TCP, offers an unreliable service. UDP packets not transmitted successfully over the wireless link are lost and will not be retransmitted by the transport layer. On top of UDP, RTP [11] plays the role as a protocol to transport continuous media streams, providing sequence numbers and time stamps with each packet.

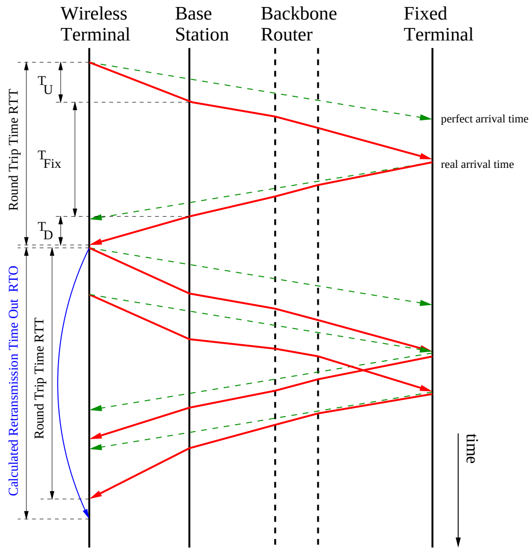

Figure 1: Message Sequence Chart for Calculation of the Retransmission Time Out (RTO)

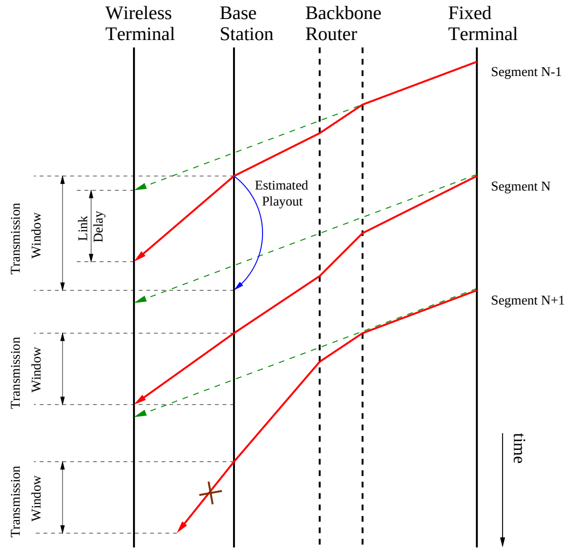

Figure 2: Message Sequence Chart for RTP traffic flow evaluating time stamps

3 Stabilisation Of The Wireless Link

A stabilization of the wireless link in terms of throughput and variation in delay can be achieved by the use of the "**S**imultaneous **MAC** **P**acket **T**ransmission (SMPT) approach, as shown in [4] (c.f. Figure 3). The idea behind this approach is that CDMA based systems allow the use of multiple channels in parallel as long as the total number of channels used does not exceed certain limits [8, 7]. Transmitting MAC packets sequentially lead to a higher delay for all other stored MAC packets. Therefore SMPT uses parallel channels to retransmit the corrupted packets, while subsequent packets are transmitted on the initial channel (see Figure 4). Using SMPT, segments consisting of a certain number of MAC packets can be transmitted with reduced delay and jitter.

In case of no errors on the wireless link, the time needed to send a whole transport segment is given by $t_{segment}$. A segment is transmitted successfully if all its MAC packets are transmitted via the wireless link within the **T**ransmission **W**indow (TW) which has the length $t_{segment} + t_{jitter}$. t_{jitter} can be set by the application, or may be determined by the MAC layer of the wireless terminal, based on transport layer state. When several wireless terminals use the SMPT approach to support the requested QoS the Signal to Noise Ratio (SNR), and therefore also the Bit Error Probability (BEP), could be become worse if all WTs at the same time uses parallel channels. Thus this instability has to be avoided.

3.1 TCP and SMPT

As mentioned before TCP performance degrades for increasing delay variation of the segments. In order to investigate improvements of TCP performance by SMPT we distinguish the following two

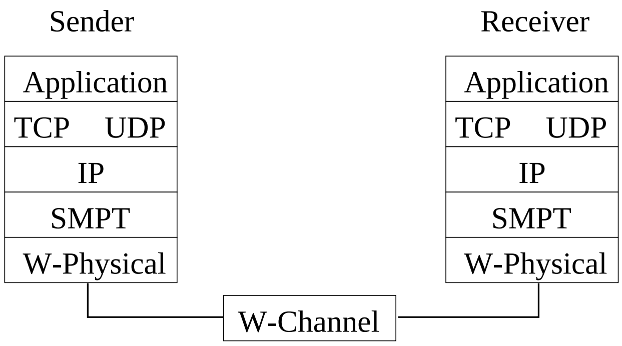

Figure 3: Protocol Stack

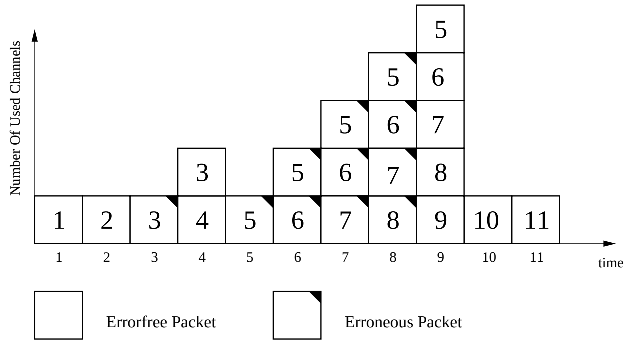

Figure 4: SMPT Approach

cases.

Mobile oriented TCP connection (uplink) For the mobile host acting as transmitter, the MAC layer may obtain knowledge of the RTO timer. If the MAC layer has full knowledge of this timer[1] it can dimension the layer 2 transmission window for transmitting the packets of one corresponding TCP segment. Figure 1 shows the approach how the MAC layer will adjust its TW under consideration of the calculated RTO after the first measured RTT timer. Because of the missing a priori knowledge for the time duration T_{Fix} and T_D, T_U will be set to $T_U = RTO_{estimated} - max\,\{T_{Fix}\} - max\,\{T_D\}$. $max\,\{T_{Fix}\}$ and $max\,\{T_D\}$ are know from the previous measurements.

Mobile terminated TCP connection (downlink) In this case the following alternatives for dimensioning of TW can be distinguished: using a fixed window, or dynamically adjusting TW based on an estimation of the RTO timer of the transmitter. Estimation of the RTO timer could be performed within the wireless terminal, or within the base station. As estimation of the RTO timer is not trivial, it appears to be a suitable approach to exploit the performance improvement possible by using a fixed layer 2 transmission window (TW). Even in cases where the base station has not the full knowledge of the most suitable TW, the fact that the number of used channels for the downlink are controlled by the base station can be used to decide on the number of CDMA channels used. This leads to a moderate BEP fully under the control of the base station. As shown in [3], a fixed TW leads to small segment delay variation, thus stabilizing quality of the wireless link and reducing the probability that the TCP RTO timer expires.

3.2 RTP/UDP and SMPT

Like in the case of TCP, also for RTP/UDP we differentiate between the uplink and the downlink case.

Mobile oriented RTP connection (uplink) In this case a transport layer information within the wireless terminal may be used for scheduling at layer 2. A fixed TW is used for stabilizing delay in the transmission over the wireless link.

Mobile terminated RTP connection (downlink) For this case we consider the following alternatives: using a fixed TW, controlling TW by functions located within the wireless terminal, within the base station, or a combination of both. Within the mobile terminal, filtering can be used to detect RTP segments upon arrival of the first MAC packets. The SMPT layer of the wireless terminal can determine a suitable TW, based on RTP timestamp information (which is also available before complete reception of the RTP segment), together with information on the delay budget for the RTP flow and the actual arrival time (see figure 2). The value for TW determined by the mobile terminal is signaled to the base station. Performance improvements

[1] This knowledge can be obtained either by accessing TCP state, or by recalculation within the MAC layer.

can be obtained by stopping the transmission of RTP packets that would arrive too late at the wireless terminal, and by signaling to the base station which TW is most suitable for a given segment, and for which segments the service quality benefits significantly from using parallel communication channels for their transmission.

Under certain conditions (e.g., no IPSEC is used), an alternative scheme can be used, where a transparent RTP agent is located at the base station. This agent contains informations about RTP connections by collecting state information for a number of RTP sessions up to a given maximum. RTP timestamp information, together with the actual arrival time of a packet can be used to dimension the TW for SMPT dynamically. By evaluating time stamps, information can be obtained about whether an individual packet is delayed more or less than the average delay calculated over a given window. Segments that arrive at the base station with a small delay have more time to be transmitted over the wireless link (allowing a larger TW), while segments with a large delay have to be transmitted using several CDMA channels in parallel. The RTP agent can be described as a QoS balancer, improving last hop performance of RTP packets that have been delayed higher than average within the fixed network, at the cost of RTP packets that have been delayed less. The advantages of this approach will be shown in the next chapter by simulations.

4 Performance Evaluation

4.1 Simulation Scenario

The chosen scenario consists of 18 wireless terminals using SMPT. In addition to these terminals 5 further terminals, using the sequential transmission method, will perform some *background noise*. Each WT generates a stream of transport units (like UDP segments and therefore called *segments*) with a specific load and pass these segments with length L to the MAC layer, where each of them is divided into a group of packets. To each packet a header ζ is added. This header ζ is used to identify MAC packets in the right order and to assign the MAC packets to the appropriate segment and means for error detection. The frame, which is composed by one MAC packet and the header ζ is called a **M**ac **P**acket **D**ata **U**nit (MPDU). The length of a MPDU is denoted as L_{MPDU}. All MPDUs are stored in an infinite queue and will be sent with different ARQ based transmission methods over the wireless link. Each terminal *see* his own wireless channel, which will be influenced by other active terminals. The wireless channel was modeled by a multilayered two state Markov chain (see also [4]), considering *good* and *bad* channel states. Within the *bad* channel states the error probability is assumed as ρ times higher as within the *good* channel state. The bit error probability itself can be calculated by equation 1.

$$BEP_{AWGN} = \frac{1}{2} \cdot erfc\left(\sqrt{\frac{N_{spreading}}{(k-1)}}\right) \tag{1}$$

For all simulations, we used the parameter values summarized in table 1. As mentioned before we will investigate the improvements adjusting the TW for a RTP stream within the base station for the downlink case in contrast to assume a fixed TW. Therefore we perform different kinds of simulations:

- Sequential Transmission and SMPT in combination with
  - a TW which is uniformly distributed $0 - T_{max}$
  - a TW which is set to T_{max} (no knowledge of the TW)

Application	L	530 byte
MAC	L_{MPDU}	106 bit
	ζ	40 bit
	Number of background channels	5
Channel	$N_{spreading}$	128
	ρ	10
	P_{good}	0.90
	P_{bad}	0.10

Table 1: Simulation Parameters

Transmission Method	**#Codes**	**Transmission Window**	**Segment Loss Probability**	**Late Loss Probability**	**Total Loss Probability**
Sequential Transmission	1	42	0.0775733	0.272740	0.3503133
		45	0.0116615	0.182667	0.1943285
		40-42	0.2041182	0.0	0.2041182
		40-45	0.1278665	0.0	0.1278665
Simultaneous Mac Packet Transmission	2	42	0.0021532	0.043823	0.0459762
		45	0.0002276	0.020461	0.0206894
		40-42	0.0424912	0.0	0.0424912
		40-45	0.0183086	0.0	0.0183086

Table 2: Simulation Results for 18 wireless terminal using the sequential transmission method or SMPT with a background noise of 5 other wireless terminals

4.2 Simulation Results

Table 2 shows the results of the simulations. For the sequential case in combination with a fixed TW $(42, 45)$ we receive the worst results. Adapting the TW within the base station lead to better results (*e.g.* the total loss will decrease from 35% to 20%). In equation 1 it can be shown that the error probability depends on the number of used channels. Thus having full knowledge of the TW will lead to a smaller number of used channels, because some MAC entities will abort earlier to transmit if they have the correct value of the TW.

A further degradation of the SLP can be achieved by the usage of SMPT. This time the SLP will amount to 5%. With knowledge of the TW the SLP averages in 4.2% in contrast to 4.5% with a fixed TW. The gain which can be achieved by the combination of layer 4 and layer 2 error control mechanisms is much higher for the sequential case, because SMPT itself offers already a stabilization of the variation of segment delay inherently. It has to be mentioned that SMPT for this simulation lead to good performance results because of the high number of MAC packets that represent one segment. SMPT has enough time to adjust the influenced jitter. For a smaller number of MAC packets representing one transport segment the gain having full knowledge of the TW is much higher.

5 Summary

We conclude that QoS for IP based applications over wireless links can be improved by Simultaneous MAC Packet Transmission (SMPT) using transport layer information for dimensioning of layer 2 Transmission Window (TW) and the number of CDMA channels used in parallel. It was shown for RTP traffic that the combination of layer 4 and layer 2 error control mechanisms and the use of SMPT that the total loss probability of layer 4 segments can be decreased from 35% to 4.2% under certain conditions.

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