

CAPACITY AND QOS FOR STREAMING VIDEO APPLICATIONS OVER TCP IN CDMA BASED NETWORKS

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Abstract

We study the capacity, i.e., the number of customers per cell, and the quality of service for streaming video in the uplink direction of a cellular **C**ode **D**ivision **M**ultiple **A**ccess (CDMA)-based wireless system. We advocate the use of TCP as the transport layer protocol for streaming video in a **M**ulti-**C**ode CDMA (*MC-CDMA*) wireless system. Our approach is to stabilize the TCP-throughput for video transmissions over the wireless links by employing a recently developed **S**imultaneous **M**AC **P**acket **T**ransmission (*SMPT*) approach. Our extensive simulations indicate that employing SMPT significantly improves both the video quality and the capacity when streaming video over TCP.

Keywords: Multi-Code CDMA, TCP, Video Streaming, Wireless Communication.

1 Introduction

Market research finds that mobile commerce for 3G wireless systems and beyond will be dominated by basic human communication such as messaging, voice, and video communication [1]. Because of its typically large bandwidth requirements, video communication (as opposed to the lower rate voice and the elastic e-mail traffic) is expected to emerge as the dominant type of traffic in 3G/4G wireless systems [2]. Video services are typically divided into real-time services (e.g., video conferencing) and streaming (e.g., web-based streaming of a news clip or the video feed from a surveillance camera). Our focus in this paper is on video streaming where the client may tolerate a small start-up delay before the playout commences. Video streaming schemes typically rely on the **U**ser **D**atagram **P**rotocol (*UDP*) as the transport protocol. As streamed video applications become more popular, the Internet will become dominated by UDP streams. UDP streams, however, can lead to instability in the Internet. This

is because UDP streams are not responsive to network congestion, as opposed to **T**ransport **C**ontrol **P**rotocol (*TCP*) streams [3]. With the explosive growth of multimedia applications, UDP-based multimedia streams have the potential to cause two major problems in the Internet: 1.) congestion collapse and 2.) unfair allocation of bandwidth among competing traffic flows [4]. Therefore, TCP friendly video streaming schemes are desirable. Jacobs and Eleftheriadis [5] introduced a TCP friendly approach for streaming video. In contrast, in this paper we advocate to use directly TCP as the transport protocol for video streaming. Besides ensuring the stability of the Internet and achieving fair bandwidth allocation, TCP has a number of important advantages. First, TCP is reliable and ensures the lossless transport of the video stream. This is important for video streams that do not tolerate errors or error propagation, such as surveillance video. Another advantage of TCP is that it ensures the in-order delivery of the video frames. We also note that there are a number of drawbacks to using TCP as the transport protocol for video streaming. First, TCP does not support multicast. Secondly, TCP's slow start mechanism and its **A**utomatic **R**epeat **R**eQuest (*ARQ*) based recovery from packet losses may interfere with the timely delivery of the video frames. Essentially, TCP trades off increased delay for lossless transport service. As is well known, using TCP as the transport protocol in wireless environments leads to the well documented performance problems of TCP over wireless links. Our approach is to employ **M**ulti-**C**ode CDMA (*MC-CDMA*) in combination with a recently developed **S**imultaneous **M**AC **P**acket **T**ransmission (*SMPT*) scheme [6, 7] to stabilize the data link throughput by reducing losses and delay variations. We demonstrate in this paper that by using SMPT at the link layer we can mitigate the interference of TCP's slow start and ARQ mechanisms with the timely delivery of the video frames. Our focus in this paper is on the streaming of rate controlled

encoded video in the uplink direction in a cellular wireless system, i.e., from wireless clients to a central base station. We focus on a reliable video streaming service that does not skip frames but instead suspends the playout at the receiver when the video consumption (temporarily) exceeds the video delivery. We provide extensive simulation results that demonstrate that our approach of combining SMPT at the data link layer and TCP at the transport layer supports video streaming in an efficient manner.

Related Work on Streaming Video over Wireless Links

The problem of efficient video streaming over wireless links has recently attracted a great deal of attention. Several works, see for instance [8, 9, 10] attempt to improve the video quality by employing adaptive video coding schemes. The basic assumptions shared by all mentioned works are that the traffic source is based on H.263 encoded video, real time services are applied, and the wireless link is modeled with a two state Markov-Chain. To bound the time delay within an acceptable range for real-time video services the allowed maximum number of retransmission attempts are limited. Some works employ hybrid error correction, see for instance [11], to make the video transmission more robust. We have recently developed a prefetching protocol for video streaming, which schedules the transmissions according to a Join-the-Shortest-Queue (JSQ) policy and the current channel conditions [12]. While the approaches pursued in this literature have made significant progress towards improving the efficiency of video streaming over wireless links, the issue of TCP friendliness has received very little attention. In fact, the proposed approaches rely largely on UDP as the transport protocol and typically have no mechanism to ensure TCP friendliness. In this paper we propose a video streaming scheme that uses TCP as the transport protocol and is therefore by default TCP friendly. We also demonstrate that our scheme has favorable performance characteristics.

Simultaneous MAC Packet Transmission

Due to the variation on the wireless link, the throughput becomes unstable, e.g. varies over time. With *Send and Wait*, the simplest ARQ mechanism, as it is discussed in [13] and illustrated in Figure 1 each erroneous MAC packet is retransmitted. The subsequent packets in the transmission queue have to wait until the corrupted packet has been transmitted successfully. In our example in Figure 1, packet number four can be

transmitted at time slot 10 for the first time. Even the subsequent packets are influenced by the retransmission process of packet number three. Due to retransmissions, the delay-jitter for a single MAC packet as well as for segments (e.g. TCP segment) of higher protocol layers increases.

The SMPT approach uses multiple channels to overcome the variations on the wireless link. One SMPT transmission mode called *Slow Healing* [7] is presented in Figure 2. After the error burst, packets are transmitted via multiple channels. After each successful usage of parallel channels one more channel is added until the influenced jitter is healed. In comparison with Figure 1 we see that a higher protocol segment consisting of 12 MAC packets is transmitted within 18 time slot using the sequential transmission and 12 time slots using the SMPT mechanism. The sequential approach would have used also 12 time slots if no error would have occurred on the wireless link. Therefore for the higher protocol layers such as TCP the variation on the wireless link is not noticeable anymore if SMPT is applied.

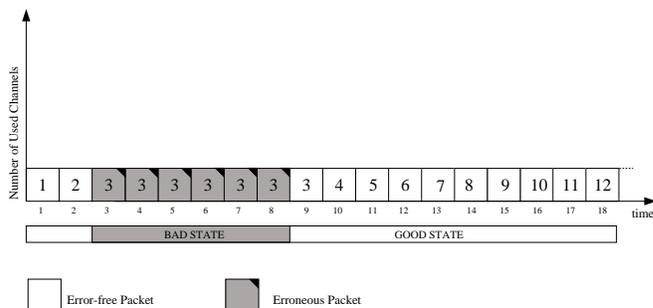


Figure 1: Sequential transmission mode with ARQ retransmissions in the time domain.

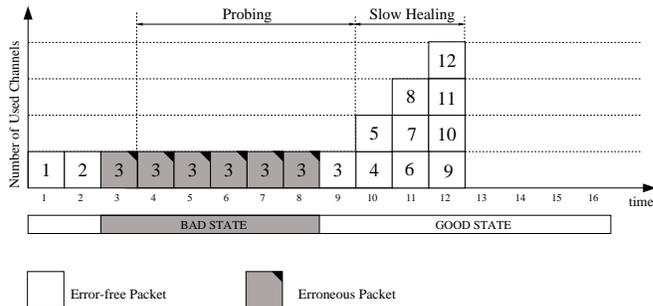


Figure 2: SMPT transmission mode with ARQ retransmissions in the code domain.

On the other hand we have to mention that multiple channels are used by the **Wireless Terminal**

(*WT*) at its own discretion. To avoid overwhelming signaling between the **Base Station** (*BS*) and the **Wireless Terminals** (*WTs*) we use **Pseudo-Noise** (*PN*) sequences. This means that by using additional channels we have an impact on the performance of all other *WT* which are active in the cell. The question is how this will influence the capacity of the cell.

2 TCP over Wireless Links

Typically, TCP is used in *wired* communication systems with very small errors probabilities. The error characteristics of wireless channels, however, differ significantly from that of wired channels. Therefore, TCP gives very poor performance if it is directly applied to a *wireless* communication system. Wired channels are characterized by miniscule packet loss probabilities and randomly spaced errors. In contrast, wireless channels are characterized by time-varying packet loss probabilities that are generally much larger than for wired channels. Also, the errors are typically bursty on wireless channels [14, 15, 16]. Moreover, wireless channels are distinct and time-varying between the **Wireless Terminals** (*WTs*), that is, the wireless link errors are location-dependent. The variability of the wireless channel quality is due to the mobility of the *WTs*, fading effects, interference from other *WTs*, and shadowing. All of these effects degrade the channel performance significantly and have a significant impact on higher protocol layers. Numerous studies have found that TCP supports wireless Internet access only very inefficiently [14, 15]. The key problems is that wireless channel errors lead to frequent expirations of the TCP retransmission timer, which are interpreted as congestion by TCP. In [7] a SMPT scheme for stabilizing the data link layer throughput over the wireless channel was introduced and studied for elastic data traffic. The SMPT approach is based on MC-CDMA.

We now give a brief discussion of the impact of wireless link error on the TCP performance for elastic traffic. We show the performance gain that can be achieved with SMPT as compared to a sequential transmission, we refer the interested reader to [7] for a detailed study. For illustration we consider the TCP performance over a wireless link for **Single-Code CDMA** (*SC-CDMA*) and MC-CDMA systems. For the SC-CDMA system we employ sequential transmission, while SMPT is employed in the **Multi-Code CDMA** (*MC-CDMA*) system. We consider one *WT* that is transmitting data to the base station using a **File Transfer Protocol** (*FTP*) application (note that in order to show the impact on TCP, FTP is the appropriate traffic source) without

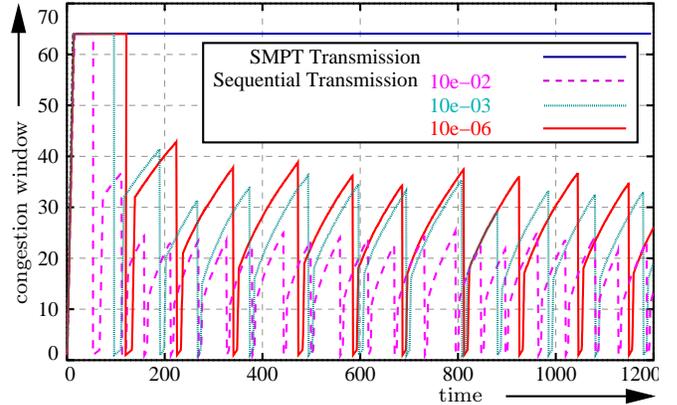


Figure 3: TCP congestion window size versus time for different packet error probabilities

Multiple Access Interference (*MAI*) interference, considering only a static channel error. The most important parameter for FTP applications is the TCP throughput.

Figure 3 shows the **Congestion Window** (*CW*) size over time for the SMPT and sequential transmission approaches for three different packet error probabilities. In this example we observe that the SMPT approach — in contrast to the sequential transmission approach — never shrinks its *CW*. With a small PEP the congestion window is quite large for the sequential transmission. However, an increased PEP causes the *CW* to shrink more often in the case of sequential transmission. The *CW* behavior of SMPT illustrates the stabilization of the throughput on the wireless link. There are two reasons for the stabilization of the TCP throughput when using SMPT. Primarily, the SMPT approach offers more bandwidth than the sequential case. The second reason is that *spurious* retransmission of TCP segments can be avoided. These spurious retransmission takes place every time the *CW* shrinks down. SMPT stabilizes the throughput by overcoming the fast-time scale variations (typically on the order of tens of milliseconds) of the wireless channel. This is important since video encoders can only react to the available channel bandwidth on a longer time-scale (typically on the order of hundreds of milliseconds or seconds). When SMPT is employed, the TCP segments are always transmitted within the pre-calculated **Retransmission Time Out** (*RTO*). For this illustrative example we assumed a *non-responsive* environment, *i.e.* that the background noise and therefore the BEP are constant, irrespective of how many channels are used. Using SMPT in the uplink forces

the usage of Pseudo-Noise (*PN*) spreading sequences [6, 7]. Due to the PN sequences the MAI increases if more channels are used within the cell. Henceforth, we assume an *all-responsive* environment, *i.e.* all active channels have an impact on each other.

3 Simulation Model and Scenario

In this paper we focus on the streaming of video over the wireless links in a single cell of a cellular wireless system. We consider the uplink streaming of rate controlled H.263 encoded video from J WTs to the base station. (We note that the streaming in the uplink direction is a particular challenge, as the WTs act in an independent, uncoordinated fashion; unlike the case of downlink streaming where the base station can coordinate the transmissions.) The base station acts as a receiver¹. At the receiver side (*i.e.*, base station) we assume a play-out buffer. In the simulations each WT randomly selects one out of 25 video sequences, which are obtained from [17] including sport, movie, and news video sequences. Also, each WT selects an independent random starting phase into the selected trace to ensure the statistical independence of the transported video streams. The WT commences the video streaming by filling the receiver-side play-out buffer to a pre-specified *offset* value τ_{off} (in units of time). The receiver side application starts to play out the video once the play-out buffer reaches the offset value. Under normal circumstances, for every frame period (which is typically an integer multiple of 40 msec for H.263 encoded video [17]) the receiver removes a frame from the play-out buffer, decodes it, and displays it. If at any of these epochs there is no complete video frame in the play-out buffer, the receiver experiences playback starvation, which we refer to a *buffer underrun*. When a buffer underrun occurs the receiver temporarily suspends the play-out of the video. The receiver waits until the play-out buffer is filled to the offset value, and then resumes the play out of the video. The average *interruption time* as well as the *buffer underrun rate* Ω depend on the *offset* value. While the interruption

time can be approximately calculated with

$$InterruptionTime = \frac{offset \cdot MeanFrameLength}{PhyLinkRate \cdot FrameSpacing}, \quad (1)$$

the buffer underrun rate Ω has to be evaluated by simulations. Note that no video frames are skipped when a buffer underrun occurs. This makes this reliable video streaming scheme well suited for applications that can tolerate short pauses in the video playback, but do not tolerate any loss of video frames, such as the video feed from a wireless surveillance camera. The duration of the buffer underrun (*i.e.*, the suspension of the video playback depends on the bit rate of the video and the throughput of the TCP transport protocol at that particular instant).

The wireless channel is modeled as a two state Markov chain with a *good* and a *bad* state, where no communication is possible in the *bad* state. The **Bit Error Probability** (*BEP*) in the *good* state depends on the total number of active channels. We use the *Improved-Gaussian-Approximation* [18] to calculate the BEP. As in [8] a **Bose-Chaudhuri-Hocquenghem** (*BCH*) FEC code is used and gives the **Packet Error Probability** (*PEP*) as a function of the number of active channels. We refer the reader to [19] for a detailed discussion of the parameter settings used in the simulation, which we can not include here because of space constraints. We note that all simulations were run until the 99% confidence interval was less than 1% of the corresponding sample mean.

At the data link layer we study two transmission approaches: (1) sequential transmission based on a SC-CDMA system with single and doubled bit rate, and (2) SMPT which is based on a MC-CDMA system. In our performance evaluation we study primarily the buffer underrun rate Ω (in buffer underruns per second) and the average time between buffer underruns T_{Ω} (in seconds). For higher values of T_{Ω} and lower values of Ω the systems performs better. We also investigate the inter-cell and intra-cell interference.

4 Performance Evaluation

In this section we discuss the system behavior for 64 kbps rate controlled H.263 encoded video. To accommodate the overhead of the upper protocol layers and some retransmissions we chose a bit rate of 72.8 kbps for the physical layer. (Note that the overprovisioning of the wireless channels allows even the sequential transmission scheme to perform retransmissions.) In Figure 4 we plot the buffer underrun rate as a function

¹As long as the wireless hop is the critical path we assume that the video streams are consumed within the base station. We note that the bandwidth of the back bones are normally that high, that all videos can be delivered to other terminals within the wired network in a timely fashion.

of the number of WTs for the SMPT approach ($R_{max} = 3$) and the sequential transmission approaches with single and double bit rate. The offset value τ_{off} is set to 0.5 sec and the spreading gain is set to 16. First,

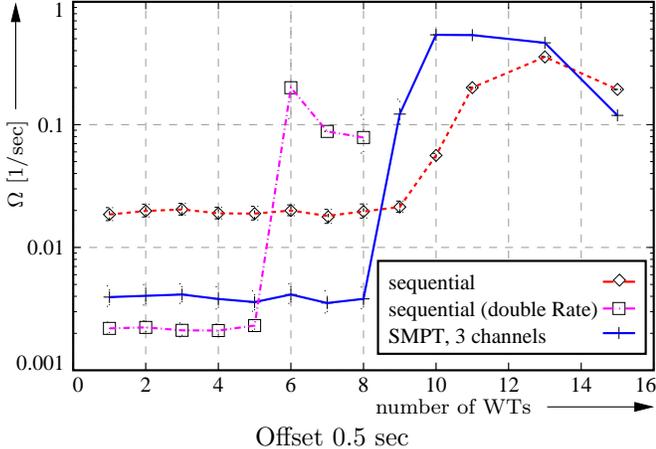


Figure 4: Buffer underrun rate Ω versus number of WTs for the SMPT approach ($R_{max} = 3$) and the sequential transmission approaches with single and double bit rate ($G_{Spreading} = 16$)

we discuss the differences between the sequential transmission and the SMPT approach. The buffer underrun rate for the sequential case is nearly one order of magnitude larger than for SMPT for a certain range of the number of WTs (from 1 to 9). Within this range the buffer underrun rate is almost constant. We refer to this range as *operational phase*. Note, that the operational phase of the sequential case contains one more WT (i.e., 9 WTs) than the operational phase of the SMPT approach (which is 1 to 8 WTs), but the user has to accept lower quality for the sequential case. For more WTs, i.e., to the right of the operational phase, the buffer underrun rate increases dramatically. The dramatic increase of the buffer underrun rate is the result of an increased usage of CDMA channels of all WTs, which results in higher BEPs. Increasing the number of WTs further leads to a small decrease in the buffer underrun rate. This is caused by TCP mechanisms, which try to adapt to the channel behavior. Henceforth, we concentrate on the operational phase, where both approaches give acceptable results. For the given scenario with 0.5 sec offset we observe for the operational phase that the average time between buffer underruns is $T_{\Omega,seq} = 50$ sec for the sequential and $T_{\Omega,SMPT} = 250$ sec for the SMPT approach.

To show that the SMPT gain is not only due to the higher bit rates (bundling of CDMA channels), we com-

pare the SMPT approach with a sequential transmission mode using double bit rate². The doubled bit rate is achieved by reducing the spreading gain G . Note, that this is still a SC-CDMA system. With the higher bit rate, we can send even two packets within one time slot. For each packet we use the same coding approach as before. We observe from Figure 4 that for this transmission mode with doubled bit rate, the average time between buffer underruns is $T_{\Omega,seq^2} = 500$ sec; thus the improvement in the buffer underrun rate is slightly higher than for SMPT. However, the operational phase (capacity of the cell) is much smaller than for the SMPT approach (five video streams for doubled bit rate versus eight video streams with SMPT). Thus, for a given buffer underrun requirement, SMPT increases the cell capacity by 60%.

Next, we investigate the capacity (i.e., the maximum number of supported WTs in the operational phase) and the impact of the spreading gain. With a spreading gain of 32 (instead of 16, as used before) the buffer underrun rate versus the number of WTs is given for an offset value of 0.5 sec in Figure 5. In this higher spreading gain scenario, SMPT achieves both a lower buffer underrun rate and a higher capacity (21 video streams with SMPT versus 19 with sequential mode, noting that the sequential mode has a buffer underrun rate that is one order of magnitude larger). The reason for this behavior is a larger multiplexing effect on the wireless link. With a spreading gain of $G = 16$ the BEP changes from 10^{-3} to 10^{-1} in the range of three active channels, while for $G = 32$ the range increases to five WTs. With this larger range, a higher multiplexing gain can be achieved.

5 Conclusion and Outlook

We have studied the streaming of video using TCP as the transport protocol in cellular CDMA-based wireless systems. TCP has a number of desirable properties, such as network stability and fair bandwidth allocation, for the future Internet (both wired and wireless). Video streaming over TCP, however, is generally known to give poor performance, especially in wireless environments. We have proposed a scheme that uses TCP to stream video in a MC-CDMA system. Our scheme employs **S**imultaneous **M**AC **P**acket **T**ransmission (*SMPT*) to effectively stabilize the wireless links. This stabilizing effect significantly improves the performance of TCP over the wireless links. We

²Even if we consider higher bit rates, we assume that the **I**nter **S**ymbol **I**nterference (*ISI*) will not change. This will lead to better results for the higher bit rate scenario.

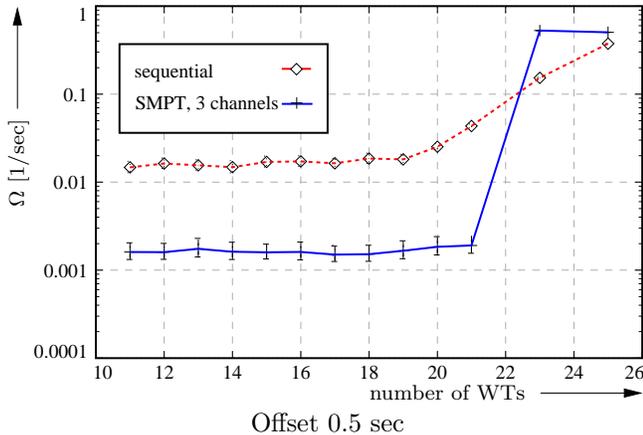


Figure 5: Buffer underrun rate Ω versus number of WTs for the SMPT approach ($R_{max}=3$) and the sequential transmission approach ($G_{Spreading} = 32$)

found that our approach gives good results for rate controlled encoded video. For a given buffer underrun requirement (which corresponds to a pre-specified client playback starvation probability), our SMPT approach increases the number of supported streams in a cell by 60% in typical CDMA scenarios. We also observed that SMPT performs better with higher spreading gains. We note that our streaming scheme preserves the isolation of the protocol layers. Specifically, SMPT does not require knowledge of any TCP parameters. Independently of the TCP operation, SMPT stabilizes the throughput over the wireless link and thus significantly reduces the probability that the TCP round trip time out is exceeded.

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