Comparison of Multi-Code Link-Layer Transmission Strategies in 3Gwireless CDMA

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ABSTRACT

Within this article we investigate the possibility of using multiple codes within a code-division multiple access system to reduce the losses in a delay bounded transmission for 3G multimedia applications over wireless links with fading. Multiple codes are used to recover from gaps on the wireless link after a fading period by means of ARQ-type retransmissions. One key feature of our work is that even a small number of multiple codes leads to a dramatic reduction of losses regarding delay bounds of multimedia applications.

INTRODUCTION

Third-generation (3G) wireless communications will be dominated by multimedia applications. Multimedia applications can be subdivided into data applications, real-time applications, and transaction applications. Data applications such as FTP, Web, and telnet are based on the Transport Control Protocol (TCP), while real-time applications, supporting audio and video, are based on the User Datagram Protocol (UDP). The quality of service (QoS) requirements for multimedia applications differ in terms of bandwidth, delay, delay jitter, and loss.

Until recently, transport protocols that support multimedia applications were designed for wireline networks; but the rapid increase in wireless devices requires solutions suitable for wireless networks. In contrast to wireline networks, wireless links are unreliable; they exhibit varying bit error probability and temporary outage periods, during which the bit errors are strongly correlated. Packets sent by the data link layer are affected by errors on the wireless link and might be corrupted. In order to reduce the loss rate seen within the data link layer, error correction schemes such as automatic repeat request (ARQ) and forward error correction (FEC) are used. The cost of static FEC within the data link layer is reduced throughput because of FEC overhead. Improvements in throughput at the expense of increased delay variation can be achieved if the FEC code is chosen *dynamically*, dependent on the channel state. The cost of ARQ is significantly increased delay variation, because stored packets have to wait until successful retransmission of the corrupted packet. Both reduced throughput and increased delay jitter lead to performance degradation of multimedia applications. Therefore, we identify the support of multimedia applications over the wireless link as an emerging area of research.

Code-division multiple access (CDMA) is a key feature of 3G wireless communications. In addition to ARQ and FEC, other CDMA-based mechanisms that reduce losses, like a varying spreading factor or changes in the power control scheme, are possible and described later. The main idea behind our approach is also CDMA-based and exploits the capability of wireless devices to transmit on multiple CDMA channels. The capability to send and receive on multiple channels is made possible by multicode CDMA (MC-CDMA). MC-CDMA was primarily introduced in [1] to increase capacity for wireless terminals (WTs) within a CDMA system.

We first introduced our approach, called simultaneous MAC packet transmission (SMPT), in [2] to support multimedia applications. SMPT is located within the data link layer, stabilizing the QoS parameters for the application. In our approach, the sender transmits a higher protocol data unit within a given delay bound. The segment is fragmented into data link packets. We assume that the allowed transmission time for a higher protocol entity is a multiple of the data link packet transmission time. As long as the wireless channel is error-free, packets can be sent sequentially on one channel. In case of an error-prone link, ARQ retransmissions of data link packets are done within the code domain using multiple channels. Furthermore, if we assume that fading durations on the wireless link are smaller than the transmission time of an entire transport segment, we perform medium access control (MAC)-level retransmissions on

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multiple channels only if the wireless link is less error-prone to achieve a high spectral efficiency. The advantage of SMPT is that it can recover from gaps caused by errors on the wireless link within a given TCP or UDP segment. The proposed scheme can be implemented in a decentralized fashion; thus, no extra signaling is necessary to use multiple channels.

Using multiple channels in parallel, however, can degrade the overall system capacity, because each additional active channel will lead to a degradation of the signal-to-noise ratio (SNR), which results in high bit error probabilities. There are many possible ways to use multiple channels. In this article we compare multicode-based link layer transmission strategies and demonstrate which strategies lead to performance improvements for multimedia applications within a 3Gwireless CDMA environment.

MODEL DEFINITIONS

We consider a wireless network where multiple WTs operate within one wireless cell. All WTs communicate with one central base station (BS), whose coverage defines the cell boundaries. We assume a slotted MC-CDMA-based mobile communication system with time-division duplex (TDD) and a frame length of τ_{TDD}. The TDD structure provides alternating downlink (BS to WT) and uplink (WT to BS) transmission slots. As depicted in Fig. 1, original MC-CDMA splits a serial source rate s into small basic rates on parallel channels. These basic rates are spread with the same spreading gain, but different code sequences $c[i]$, $i = 1 \cdots k$ over the entire transmission bandwidth, where k denotes the maximum number of parallel channels per WT. k is limited by WT hardware. Thus, each WT is capable of transmitting and receiving on multiple channels. All active channels of one WT are superimposed and modulated afterward. In our model, we assume that the bit rate to be supported by a single WT is exactly equal to the nominal bit rate achievable by a single code. Phenomena investigated in this article can be supplemented with the effects of supporting higher bit rates by multiple codes. This is beyond of the scope of this article. For SMPT, we assume that pseudo-noise (PN) code sequences or code sequences obtained by the subcode concatenation scheme [1] are used within the uplink. The number of possible CDMA codes within one cell can be assumed to be much higher than the number of active used channels. After a WT has registered at the BS, the BS provides the WT with a unique set of PN codes. This set of codes can be used by the WT at its own discretion. Perfect power control within the WTs is assumed.

The application of each WT generates an information stream. This information stream is segmented into transport units (e.g., UDP segments; referred to as *segments* in the following) within the transport layer. The length of the segments is denoted by L_S. A segment is passed to the data link layer via the network layer (e.g., Internet Protocol, IP. At the data link layer a segment is divided into N packets, $N \geq 1$. To each packet a header is added. The header is used to order data link packets, to assign them to the appropriate segments, and for error detection.

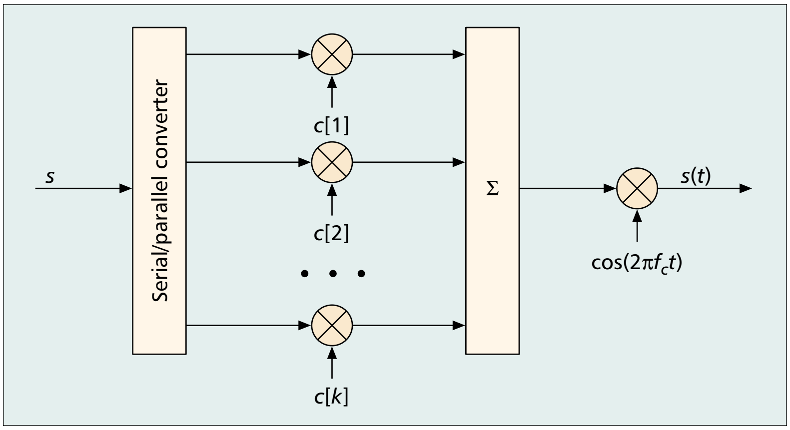


Figure 1. *The multicode CDMA sender/receiver.*

One data link packet and a header are called a link PDU (LPDU). The length of an LPDU is denoted by L_{LPDU} and fits into one transmission slot on the wireless link. All LPDUs are stored in a transmission queue with a fixed buffer capacity, L_Q, within the sender side data link layer and are to be sent to the BS by different ARQ-based transmission methods over the wireless link.

The wireless uplink is considered unreliable with a varying bit error probability (BEP). For the downlink, we assume error-free transmission. The basic bit rate of one CDMA channel is R_C. The wireless link is characterized by periodic outage periods with a varying BEP. Thus, there are situations on the wireless link that can be considered *good* or *bad* states. The value of the BEP in the *good* channel state, which has an average sojourn time of τ_{good}, depends on the number of used channels and spreading gain G_S [3]. In the *bad* channel state, which has an average sojourn time of τ_{bad}, no communication is possible. There is no error correction scheme assumed, so data link packets with one or more bit errors will not be decoded successfully on the receiver side and will be considered lost.

The sender side transport entity passes one segment to the IP-based network layer at time T_0. If an error-free sequential transmission is assumed, the segment will arrive at the receiver side network layer at time $T_1(L_S)$, which depends on the segment length L_S. Thus, the minimal delay is given by $D_{min}(L_S) = T_1(L_S) - T_0 = N \cdot \tau_{TDD}$. Because of the error-prone wireless link, retransmission of corrupted packets is required, which increases the segment's delay. Suppose that the receiver side transport entity accepts only segments that arrive with delay no higher than a maximum permissible delay $D_{max} = \min\{\tau_{delay}, D_{min}(L_S) + \tau_{jitter}\}$, where τ_{delay} denotes the deterministic delay bound and τ_{jitter} the deterministic delay jitter bound [4]. τ_{jitter} and τ_{delay} are predefined by the application or, if possible, gathered from the transport layer segment. If a segment arrives at the receiver side at time T_2, such that $T_2 - T_0 \leq D_{max}$, the segment was transmitted successfully. For successfully transmitted segments the delay jitter $J = T_2 - T_1(L_S)$ is measured. Note that the delay jitter of successful segments is never larger than τ_{jitter}. In addition to the delay jitter, the delay and segment loss probability, resulting in the goodput value, are measured.

The SMPT approach transmits LPDUs representing a higher protocol segment within a given delay bound using multiple channels to overcome the variations on the wireless link.

To maximize the SNR value, the sender side data link entity aborts transmission of LPDUs of a segment if the segment is likely to fail to meet the prespecified delay bounds. All corresponding packets are removed from the queue, and the segment is lost. This leads to improved overall system performance. The sender side data link entity will proceed with the next stored segment. For fair comparison of the studied SMPT approaches, we apply the same baseline rule throughout to decide whether to abort or continue transmission of the LPDUs of a given segment. If strictly sequential transmission (using only one CDMA code channel) of the remaining LPDUs over an error-free channel is not able to meet the segment's delay bound, the transmission is aborted. We note that for the different SMPT approaches, further QoS improvements can be achieved by optimizing the abort criterion. If all LPDUs of a segment are transmitted successfully, the receiving data link layer reassembles the LPDUs into the segment. The successfully received segment is passed immediately to the application via the network and transport layers.

Simultaneous MAC Packet Transmission

Errors on the wireless link reduce the throughput, because some LPDUs are prone to errors and cannot be decoded correctly. Due to the variation on the wireless link, the throughput becomes variable over time. With Send and Wait, the simplest ARQ mechanism, as discussed in [4], each erroneous LPDU is retransmitted. The subsequent packets in the transmission queue have to wait until the corrupted packet has been transmitted successfully. Due to retransmissions, the delay jitter for a single LPDU as well as for segments of higher protocol layers increases. High variable throughput and increased delay jitter are not acceptable for many multimedia applications [5, 6].

The SMPT approach transmits LPDUs representing a higher protocol segment within a given delay bound using multiple channels to overcome the variations on the wireless link. The delay bound is a specific value by the application. Multiple channels are used by the WT at it's own discretion. Which policy on adding additional channels to apply is definitely an open question. In our earlier papers [2, 7] we used a very simplified approach, adding additional channels each time a packet was not transmitted successfully. We demonstrated that even such a simplified schema might be attractive. This would, however, increase the level of interference in the bad state of the channel. Therefore, we believe we should avoid using multiple channels as long as the state of the channel is bad and start increasing the number of used channels only after the state of the channel improves.

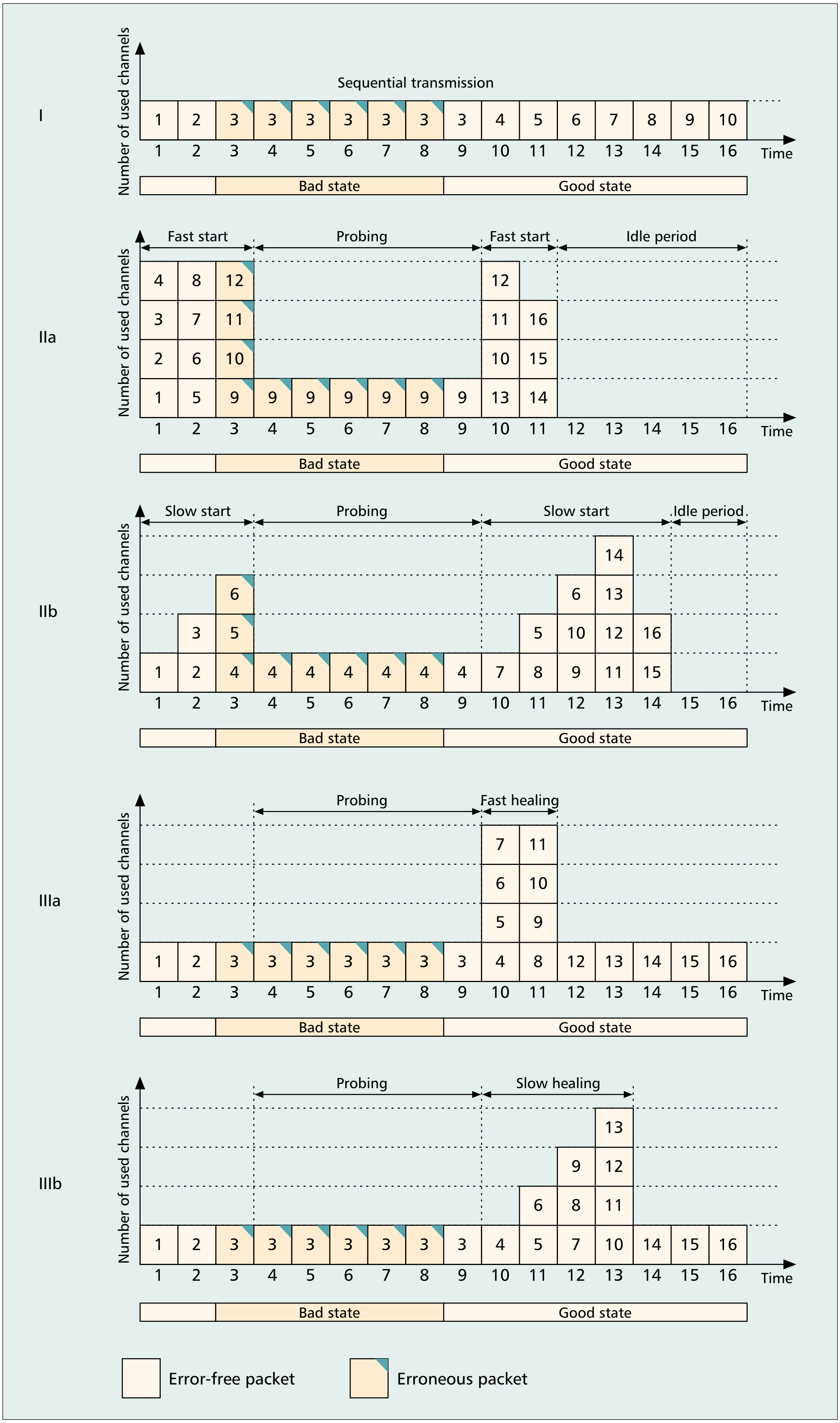
We profit from considerations presented by Zorzi and Rao, who claimed in [8] that classical ARQ strategies lead to considerable waste of energy due to the large number of retransmissions within the outage period of the wireless channel. Therefore, they introduced a new protocol which switches between *probing* and *normal* mode depending on the channel state. As soon as the sender side data link entity realizes that the channel state has changed from *good* to *bad*, it sends probing packets. The probing mode continues until one probing packet is transmitted successfully. The probing packet may be a *normal* LPDU (e.g., the last corrupted LPDU). We assume in this article that a successfully transmitted packet is equivalent to an indicator of a good state, while a lost packet is equivalent to an indicator of a bad state. To achieve a high SNR, multiple channels are used only in the *good* channel state. Once a *bad* channel state is detected, the protocol will probe out the channel as proposed in [8] on the initial channel. In fact, after recognizing a good state of the channel, we have multiple options to proceed in assigning additional channels. Several of these options are pointed out in Fig. 2.

The investigated SMPT approaches, namely Slow Start, Fast Start, Slow Healing, and Fast Healing, work as follows. Figure 2 illustrates the different SMPT transmission strategies as well as sequential transmission. As an example we demonstrate the transmission of 16 LPDUs. We distinguish between the Self Healing and Start Up mechanisms. Self Healing is only used if sequential transmission (using one channel) falls behind due to channel errors. The Self Healing mechanism reduces the accumulated delay jitter using the capability to send on additional channels. There are two main approaches to the Self Healing process. Fast Healing (IIIa) uses all available resources immediately after detecting a good channel state, while Slow Healing (IIIb) uses the resources incrementally (Fig. 2). The Start Up method sends on additional channels whenever the channel is in a good state. While Fast Start (IIa) transmits on all channels, Slow Start (IIb) transmits incrementally. As illustrated in Fig. 2, the Start Up methods can lead to periods during which nothing has to be sent. We refer to these periods as *idle* periods. We note that for the sequential transmission (I), each retransmission has an impact on the following transmissions. While all SMPT approaches are able to send the 16 LPDUs within 16 time slots, sequential transmission is not. All transmission strategies are probing only on the initial channel after detecting a bad channel state.

In our previous work, we have demonstrated the performance improvement in TCP and UDP traffic for simplified SMPT scenarios. For TCP, which suffers by the delay variance resulting from reliable MAC, we have demonstrated in [7] that with SMPT a stabilized throughput can be achieved without changing the TCP protocol. In [2] we demonstrate the performance improvement for UDP-based applications.

Performance Evaluation

In this section we present the simulation results for different SMPT approaches and sequential transmission. We concentrate on the QoS parameters goodput (i.e., bit rate at the transport layer consisting of successfully transmitted segments), segment loss probability, and delay jitter. The transport-level delay jitter and delay bounds are set to τ_{jitter} = 240 ms and τ_{delay} = 600 ms. The segment length is uniformly distributed from 1 to



■ **Figure 2.** *Examples of different transmission strategies (top down): sequential transmission, Fast Start, Slow Start, Fast Healing, and Slow Healing mechanisms.*

The SMPT approach transmits LPDUs representing a higher protocol segment within a given delay bound using multiple channels to overcome the variations on the wireless link. The delay bound is a specific value by the transport layer. Multiple channels are used by the WT at it's own discretion.

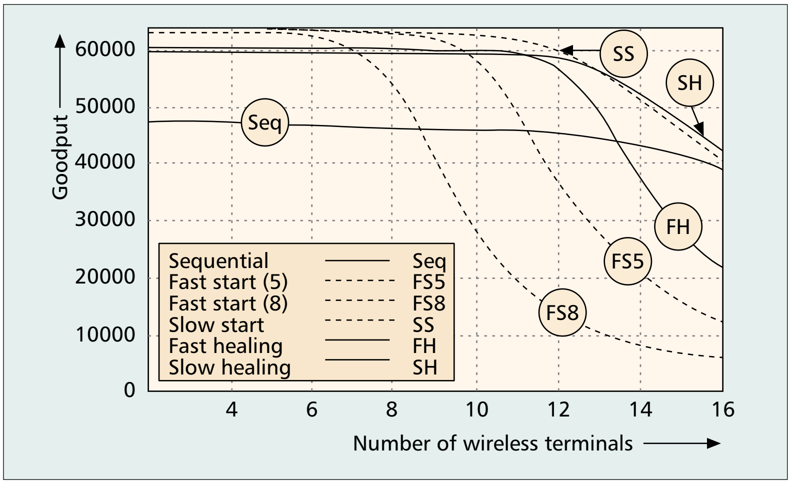


Figure 3. *Goodput vs. number of wireless terminals for different link-layer transmission strategies with a maximum number of eight channels per WT (Fast Start five channels).*

4000 bytes. Each WT is allowed to use up to k = 8 channels in parallel. The basic rate of one CDMA channel is fixed at R_C = 64 kb/s with L_{LPDU} = 192 bytes. The application of each WT generates a load with constant bit rate R_C; thus, each WT uses exactly one channel when the communication is error-free. The queue within the MAC layer can store L_Q = 4800 bytes. The simulations were performed with a confidence level of 95 percent for the goodput value.

In Table 1, the goodput and segment loss probability is given for a specific cell penetration of 10 WTs. The Slow Start, Slow Healing, and Fast Healing mechanisms yield high goodput values of roughly 60 kb/s, while Fast Start gives only 28 kb/s, which is even lower than the goodput of sequential transmission. The results for segment loss probability are similar. We note that the different transmission strategies were optimized to achieve high goodput. All delay jitter values are bounded by τ_{jitter}. The Slow Start approach leads to the smallest values, both in terms of mean (10 ms) and variation (32 ms). The precise value for the mean number of used codes to support 10 WTs is 19.74 for Fast Start and 10.59 for Slow Healing. As noted above, a high number of used channels leads to performance degradation.

In Fig. 3 the goodput vs. the number of WTs for different transmission strategies is depicted. For a small number of WTs the goodput for all SMPT strategies is much higher than the sequential case with 47 kb/s. Start Up mechanisms yield 63 kb/s, while Self Healing mechanisms result in 60 kb/s. With an increasing number of wireless devices some strategies lead to results even worse than the sequential case. Slow Start and Slow Healing always have better performance in terms of goodput values. As mentioned above, using channels in parallel can lead to intra/intercell performance degradation. Therefore, Fast Healing with a maximum number of five and eight channels is depicted. It can be demonstrated for this approach that a lower maximum number of parallel channels leads to a performance gain. The reason for this is the lower mean number of used channels within the cell. Allowing up to 5 channels/WT with 10 WTs within the cell results in an overall number of 13.47 instead of 19.74 for a maximum number of 8 (Table 1). Even with a small number of parallel channels such as 2 or 3, we observe very good results for all SMPT approaches. A higher number of codes leads only to performance improvements for a small load within the cell, which can be observed by goodput increased from 60 to 63 kb/s.

The resulting impact on power control and power saving is discussed in the following. Therefore, in Figs. 4 and 5 the conditional probability $p(\tau + 1, \tau)$ of sending with an overall number of codes during slot $\tau + 1$ after slot τ for two different SMPT approaches is depicted. The conditional probability $p(\tau + 1, \tau)$ reflects the dynamics of the number of used codes within the wireless cell. Fast timescale variations of code usage require high-performance power control able to deal with such large energy fluctuations. Figure 4 depicts the conditional probability $p(\tau + 1, \tau)$ for Fast Start. Figure 5 gives the conditional probability for Slow Healing. We observe that Fast Start has a significantly larger variance in code usage than Slow Healing. This conclusion is also supported by the values summarized in Table 1. The variance in the total number of codes used by Fast Start is 156.47, while Slow Healing only has a variance of 1.69. It must always be ensured that the system is protected against instability. Both the mean and variance in code usage have a large impact on intracell and intercell interference [9], and should therefore be kept small. If the overall number of codes within one cell has a large variance, the power control unit of the CDMA-based mobile terminal cannot recalibrate itself. Therefore, moderate dynamic of code usage must be guaranteed.

Parameter		Sequential	Simultaneous MAC packet transmission			
			Fast Start	Slow Start	Fast Healing	Slow Healing
Goodput	[bit/s]	43849.36	28208.3	61827.5	60213.3	59930.8
Delay jitter	E{} [ms]	90	54	10	22	27
	Var{} [ms]	148	175	32	31	45
Segment loss rate	[%]	24.03	44.25	2.28	3.75	4.30
Overall codes	E{}	8.98	19.74	11.16	10.81	10.59
	Var{}	1.154	156.474	12.510	3.524	1.691
Idle period	Length	8.49	6.75	5.75	1.98	2.63
	prob. [%]	11.11	32.38	38.35	3.30	3.82

Table 1. *Simulation results for 10 WTs using sequential transmission and the SMPT approach with eight parallel channels per WT.*

At last, the energy aspect has to be interpreted. We see from Table 1 that the sequential transmissions, Fast Start and Slow Start, achieve a large probability of having idle periods during which no sending takes place. The idle periods occur if no transport segment has to be transmitted at the moment and the sender has either already transmitted the last LPDU of a transport segment or dismissed transmission because of the delay bounds. Moreover, the average idle periods are long. Note, however, that sequential transmission and Fast Start mechanisms achieve long idle times because they fail to meet the prespecified QoS parameters and discard LPDUs, resulting in a high segment loss probability. Slow Start can achieve both energy saving as well as a small segment loss probability.

ENABLING TECHNOLOGIES AND RELATED WORK

Existing second-generation CDMA systems, such as the IS-95 (Rev. A) digital cellular system by Qualcomm, already support voice and data traffic at a basic rate. Further improvements in IS-95 (Rev. B) allow for use of MC-CDMA. In combination with load- and interference-based demand assignments (LIDA [10]), mixed traffic can be supported. A mobile capable of high speed can utilize up to seven additional channels, called *supplemental code channels* (SCHs), on top of the fundamental code channel (FCH), which results in eight usable channels per WT. The channels are controlled by the infrastructure and assigned to the mobile through a dedicated signaling channel.

As an example of 3G mobile communication systems, we have chosen the Universal Mobile Telecommunication System (UMTS). UMTS Terrestial Radio Access (UTRA) has two modes: frequency-division duplex (FDD) and TDD [11]. The frame length for both modes is set to 10 ms and consists of 15 time slots/frame. In the TDD-mode up to 15 codes within one time slot with a spreading gain between 1 and 16 can be allocated for the uplink. Unfortunately, the code sequence set is fixed. Code sequences cannot be used by the WT at its own discretion, but have to be controlled by the BS. In FDD mode, rate adaptation is achieved by varying the spreading factor within one time slot. For both modes only minor changes have to be made to employ SMPT.

Distributed queuing request update multiple access (DQRUMA) is a flexible multiple access protocol for broadband wireless ATM packet networks and was introduced in [12]. It supports uplink as well as downlink communications, separated by FDD. Use of MC-CDMA allows WTs to transmit at different data rates. The number of parallel channels per WT is under control of the BS and has to be signaled to the WT. The maximum number of usable channels within the cell depends on the SNR. The channels are distributed to the WTs using the *Bandwidth on Demand Fair Sharing Round Robin Scheduler* (BoD-FSRR).

Both DQRUMA and LIDA focus on MAC-layer performance only and support QoS in terms of bandwidth. Therefore, the scheduling decision is made only on the contents of their MAC queues. SMPT, on the other hand, schedules according to whether a higher protocol segment can be delivered within the required delay bound; that is, SMPT supports QoS in terms of delay, delay jitter, and segment loss probability. Moreover, multiple channels within the SMPT approach are used by the WT at its own discretion, while the former approaches assign multiple channels via a centralized entity. The distributed use of multiple channels saves a great amount of signaling time, but can lead to system instability if multiple channels are used excessively.

A power control approach to support QoS on the network level is presented in [13]. This approach dynamically adapts the SNR requirements with power control schemes. The advantage of such a system is that it can easily be integrated into existing CDMA systems. During the transmission process, the power is adjusted depending on the channel state to achieve a required SNR; for

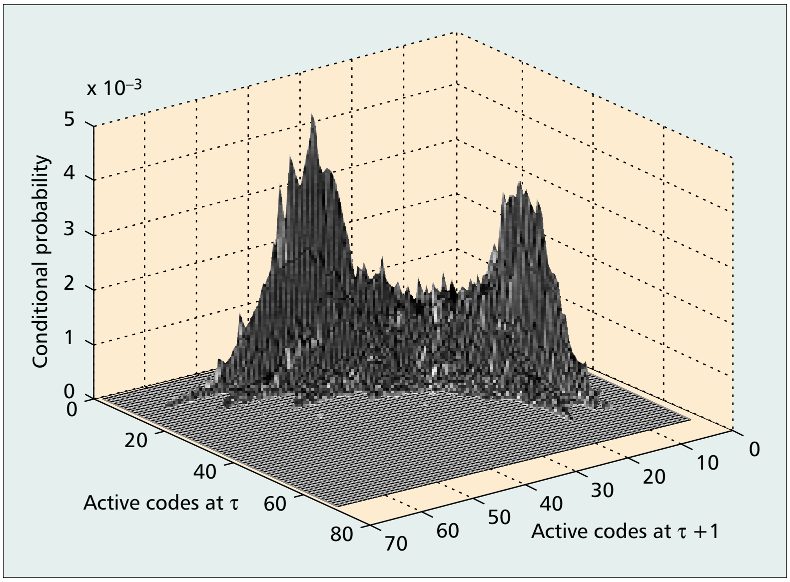


Figure 4. *Conditional probability sending with overall number of codes for the Fast Start approach.*

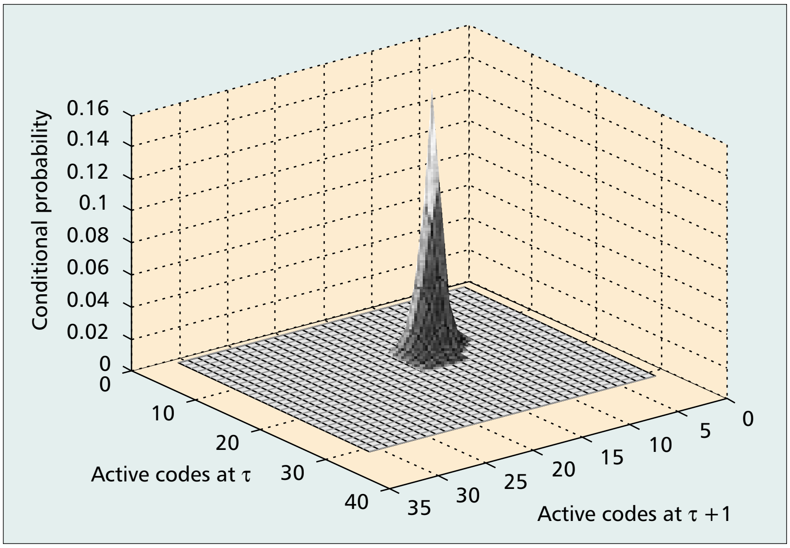


Figure 5. *Conditional probability sending with overall number of codes for the Slow Healing approach.*

Let us stress that in the expected SMPT usage scenario, a single code will be assigned to each terminal. Additional codes might be activated, if necessary, according to one of the strategies discussed in this article.

example, if the channel becomes worse (higher interference), the power level is increased to stabilize the SNR. But as already mentioned earlier, we assume that it makes no sense to transmit in a bad channel state. The SMPT approach instead probes the *bad* channel state and utilizes more resources if the channel switches back to the *good* state. Furthermore, if the delay jitter of a higher protocol segment is exceeded once, the power control approach has no ability to recover from the delay impact, which is a strong point of SMPT. Because of page limitations, similar approaches that adapt spreading codes are not explained fully, but they have properties similar to the power control approach.

Conclusion and Outlook

We have demonstrated that MC-CDMA-based 3G wireless communication systems can benefit from the introduced SMPT mechanisms. Our simulation results indicate that SMPT mechanisms improve QoS parameters in terms of delay, delay jitter, and segment loss probability over sequential transmission. Different SMPT approaches are introduced and evaluated. Mechanisms that use the wireless resources in a moderate way, such as Slow Start and Slow Healing, achieve the best results. For a typical scenario of 10 WTs, both mechanisms achieved a segment loss probability of less than 5 percent; the segment loss probability of sequential transmission is almost five times larger. While Slow Start achieves better power saving, Slow Healing has a smaller variance in the number of used codes; the variance is almost as small as for sequential transmission, which is very important for minimizing the impact on the power control entity. Moreover, we have demonstrated that using multiple channels can result in performance degradation if the mechanism (e.g., Fast Start) uses wireless resources excessively. We conclude that it is insufficient to merely equip the WTs with data-rate-adaptive procedures. Instead, mechanisms such as Slow Healing SMPT have to be implemented within the data link layer to protect system stability and enable heterogeneous QoS support for multimedia applications.

It can be observed that the improvement in throughput and delay within SMPT is achieved with a relatively low number of additional codes. Let us stress that in the expected SMPT usage scenario, a single code will be assigned to each terminal. Additional codes might be activated, if necessary, according to one of the strategies discussed in this article. As we can reasonably assume, individual WTs will experience bad channel states independently. We can expect a significant statistical multiplexing effect on the usage of additional codes.

In our ongoing work we study how to exploit the synchronization between parallel channels used by the same WT. Instead of using pure PN sequences, code concatenation, as proposed in [10], will lead to lower self interference among channels of the same mobile.

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