QoS Support In Wireless Networks Using Simultaneous MAC Packet Transmission (SMPT)

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Abstract—The goal of this paper is the adaptation of the "Simultaneous MAC Packet Transmission (SMPT)" approach to a CDMA based mobile communication system with a varying number of mobile users. SMPT attempts to stabilize the Quality Of Service (QoS) of a wireless CDMA system in terms of throughput, loss rate and delay even if the propagation conditions on the wireless medium change dramatically. In the future there will be highly heterogenous QoS requirements by the mobile users. To support these requirements the network has to maximize the spectral efficiency by statistical multiplexing of the used channels. We apply this mechanism to all mobile users within one wireless cell and investigate the spectral efficiency. The suitability of our approach is shown by observing its effects on typical QoS parameters of multimedia applications by simulations.

I. INTRODUCTION

Parallel with the dramatic growth of cellular telephony an increasing demand for different kinds of multimedia applications is notable. Until recently, multimedia protocols were designed for fixed networks. But the ongoing development of omnipresent mobile communication environment requires solutions that are also suitable for mobile and wireless networks. Multimedia applications typically require stable throughput, small jitter, bounded delay and a bounded loss rate. Therefore Quality of Service (QoS) support in wireless cellular networks is one of the most challenging questions today.

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Using the new degree of mobility the mobile user expects the same QoS comparable to the wired network. But in contrast to wired networks the wireless link oscillates between good and bad states. This means it has temporary outage periods where the wireless channel is more prone to correlated bursts of errors [10, 11]. From the application point of view a bad channel condition is recognized by a reduced throughput, higher loss rate, higher jitter and higher delay, which stands in sharp contrast to the requirements for multimedia applications. Instead of attempting to live with the unstable service (e.g. adaptive applications [8]) we have introduced a mechanism called Simultaneous MAC Packet Transmission (SMPT) in [4, 5] that stabilizes the service of a CDMA system by reducing the loss rate without increasing the jitter or delay.

The idea behind our approach is that CDMA systems allow multiple channels in parallel for one Wireless Terminal (WT) as long as the total number of channels used within one cell is not exceeded dramatically. For instance the simultaneous use of multiple channels is already realized in IS-95B (merging IS-95A, ANSI J-ST and TSB74), which specifies the high speed data operation using up to eight parallel channels, resulting in a maximum bit rate of 115.2 kbit/s. But in contrast to IS-95B, where parallel channels are used statically, our approach uses multiple channels only in case of bad channel conditions. Thus, assuming a sufficient number of parallel channels a bounded jitter can always be guaranteed. In contrast to fixed channel allocation SMPT offers the advantage to switch on channels dynamically in case of bad conditions on the wireless link without occupying bandwidth in good wireless conditions.

The scope of the following paper is to investigate the spectral efficiency of several WTs using the SMPT approach within the same wireless cell. Our previous investigations (see [4, 5]) considered only one WT with the SMPT approach on top of several static background channels. After the elaboration of the principals of SMPT, we now want to show that the overall perfor-
mance of the whole cell can also be improved by SMPT. Furthermore we work out the optimal number of maximum usable channels for each WT in dependency on the overall load within the cell. The results allows a central-
ized entity (e.g. base station) to guarantee high spectral efficiancy on the wireless link.

The paper is organized as follows. In chapter II. we give a brief introduction of the SMPT approach and describe the scenario under investigation. Chapter III. will conclude this paper and give a detailed performance evaluation.

II. SIMULTANEOUS MAC
Packet Transmission

A. Scenario

The chosen scenario consists of a specific number of WTs operating in a wireless cell. All WTs communicate with the base station. We assume a CDMA based mobile communication system with a number of codes higher than the number of active WTs. Each WT generates a stream of transport units (like UDP segments and therefore called segments) with a specific load and pass these segments to the MAC layer, where each of them is divided into a group of packets (see also figure 1). 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 Mac Packet Data Unit (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.

![Fig. 1. Simulation Model with Segmentation and Reassemble Entity for two WTs and a Bit Error Generator on the Wireless Link](image1)

The wireless link is considered to be unreliable with a varying Bit Error Probability (BEP). The value of the BEP depends on the number of used channels $k$. For the chosen scenario we assume an Additive White Gaussian Noise (AWGN) channel with Binary Phase Shift Keying (BPSK). We neglect the fading effects and assume an optimal power control in the WTs. Nevertheless using the wireless channel each WT will influence other WTs by an increased background noise. Assuming the general structure of a CDMA channel as depicted in figure 2, the signal $x(t)$ of one user transmitted with the power $P_s$ will be disturbed by other active users. A possible jammer with power $J$ and a wide-band additive white Gaussian noise $N$ will influence the wireless link. The BEP for an AWGN channel using $k$ channels is described in equation 1 under consideration of [9, 2, 3]. A friendly communication environment ($J = 0$) and a noise level smaller than the transmitting power of one WT ($N \ll P_s$) is assumed. $N_{\text{spreading}}$ describes the processing gain of the CDMA system$^1$.

![Fig. 2. General Structure of a CDMA Channel under Consideration of Other Active Wireless Terminals](image2)

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

The receiving MAC acknowledges each received MPDU and reassembles the received MPDUs to one segment, if all MPDUs of one appropriate segment are transmitted successfully. A successful received segment is then passed to the application.

B. The Sequential Transmission Method

At the beginning of the transmission only one CDMA channel is used and the transmissions of other wireless terminals are simply treated as noise. Thus, $B_{\text{good}}$ represents the mean available bit rate for the initial CDMA channel in case of a good channel state. In this case the

$^1$The Malekum Q function can be transformed in the erf function: $erf(\alpha) = 2 \cdot Q \left( \sqrt{\alpha} \right)$
QoS is adequate for the communicating applications. In the basic scheme each packet suffering a bit corruption is considered as loss. In addition to that a loss of a single packet implies the loss of a whole segment. Using the simplest ARQ mechanism Send and Wait, as it is discussed in [1] and suggested within the recent wireless LAN standards [15, 16], each erroneous packet is retransmitted while following stored packets have to wait until the packet has been transmitted successfully. The sequential transmission is shown in figure 3. So the effective bit rate decreases from $B_{good}$ to $B_{bad}$ and in the same time the jitter increases. We assume that a resulting bit rate $B_{bad}$ is not acceptable for the required throughput specified by the QoS parameters. Further the increased jitter is not acceptable for the application.

### C. The SMPT Transmission Method

SMPT is a method that overcomes the described problem. During bad channel condition we compensate the bit rate degradation $\omega = B_{good} - B_{bad}$ by using multiple CDMA channels. In order to recognize the changes of the link quality an information feedback is needed. In general a CDMA radio front-end can support several channels in parallel. It is obvious that a platform dependent number of channels exists which can be used in parallel by one mobile. Depending on the system properties this maximum number can even equal one, but a number of channels up to 5 seems realistic to the authors.

One possible SMPT transmission method is shown in figure 4. As long as no error occurs the packets are transmitted sequentially using one CDMA channel. When a packet is corrupted SMPT retransmits it via an additional CDMA channel, while the next packet is transmitted in parallel over the initial CDMA channel. If subsequent packets are also erroneous new CDMA channels are allocated. Here and in all further considerations we assume that in a single time slot $\tau_{air}$ all the parallel transmitted packets will either be corrupted or received properly. The additional CDMA channels are released, when all packets, which were influenced by the error prone link, have been successfully transmitted.

The joint bit rate $B_{joint, k}$ using $k$ CDMA channels of bit rate $B_{bad}$ is in general smaller than the sum of the bit rates of all channels, because the higher number of used channels results in a higher bit error probability [2, 7]. It has to be mentioned that the presented SMPT transmission method is only one possible scheme. Based on this scheme we have investigated other schemes that optimize the Segment Loss Probability (SLP) or minimizing the usage of the wireless link [6]. The following performance evaluation is done with the basic SMPT approach.

In opposite to IS-95B we assume however that a single wireless mobile terminal uses multiple channels only temporarily when the channel becomes worse since using channels in parallel decreases the Signal to Noise Ratio (SNR). This leads to a statistical multiplexing effect (total number of channels) among a set of terminals which helps to keep the SNR high (see figure 5).

With an increasing number of allocated channels the noise level will also increase resulting in a lower effective bit rate. The effect of this parameter depends on the overall system. Therefore mechanisms are needed which protect the system against bad influence due to the SMPT approach.

For example a base station, which has full knowledge about the dimension of the system might assign only an limited number of channels to each mobile. Alternatively a mobile might stop using an excessive number of channels if the throughput does not improve. This leads to the assumption that an optimal number of channels can be found, where the influence on other wireless mobile terminals is low and on the other side the stability of bandwidth is assured.

### Table 1: Number of Channels Used

<table>
<thead>
<tr>
<th>Time</th>
<th>Channel Use</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
</tr>
<tr>
<td>4</td>
<td>1</td>
</tr>
<tr>
<td>5</td>
<td>1</td>
</tr>
<tr>
<td>6</td>
<td>1</td>
</tr>
<tr>
<td>7</td>
<td>1</td>
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<tr>
<td>8</td>
<td>1</td>
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<tr>
<td>9</td>
<td>1</td>
</tr>
<tr>
<td>10</td>
<td>1</td>
</tr>
<tr>
<td>11</td>
<td>1</td>
</tr>
</tbody>
</table>

![Fig. 3. Sequential Transmission Method](image)

![Fig. 4. SMPT Transmission Method](image)
D. SMPT For Multimedia Applications

Multimedia application generally have high requirements on the QoS. Therefore we assume that we have a given jitter bound \( t_{jitter} \), defined by the QoS, for all segments. In addition to that we expect a further limitation of \( b \) usable channels for each WT using the SMPT approach, because of hardware limitations. The time needed to transmit one MPDU is \( t_{MPDU} \) within a time slot \( T_{slot} \) and the time needed to send the whole segment in case of no errors on the wireless link is given by \( t_{segment} \). The total number of retransmissions in the dimension of time can than be described as \( \beta = \frac{t_{jitter}}{t_{segment}} \). The number of MPDUs of one segment is denoted by \( \alpha \). While the jitter is not exceeded and all MPDUs belonging to one appropriate segment can be transmitted within the Transmission Window (TW) \( TW = t_{segment} + t_{jitter} \); see figure 5) the sender side MAC will proceed to transmit MPDUs. If the total number of retransmissions of MPDUs within a single segment exceed \( \beta \), the MAC will abort retransmitting and all the MPDUs of the corresponding segment are discarded. Whether this lost segment will be retransmitted depends only on the reliability of the transport layer.

III. PERFORMANCE EVALUATION

In order to achieve a first feeling how additional channels using SMPT will influence the overall performance within the cellular network, we have developed a simulation model corresponding to the system model described in chapter II. using the simulation tool PTOLEMY [12]. We will investigate how SMPT reduces the Segment Loss Probability (SLP) without exceeding the negotiated bounded jitter by allocating additional resources.

Our investigations concentrate on the uplink. The downlink is considered to be no problem because of the base station’s knowledge of how many channels will be used and the resulting influence on the SNR. All applications operating on one WT have the same homogenous requirements in terms of QoS (delay, bound jitter).

For all simulations, we used the parameter values summarized in table 1. For the calculation of the BEP we assumed ten terminals that transmit all the time, reflecting some background noise. The error model were fixed for all simulations. We studied the system behavior for an increasing number of WTs (10 - 28) on top of the background channels changing the Maximum Number of simultaneously usable Channels (MNSuC) from one to five.

Further we investigated different values for the jitter bound. The value of the jitter bound \( \beta \) was set to zero, three and six, reflecting situations where the requirements for a multimedia applications are high (\( \beta = 0 \)), medium (\( \beta = 3 \)) and low (\( \beta = 6 \)), respectively. The value of the jitter bound represents the number of retransmissions measured in MPDUs that can be done without exceeding \( t_{jitter} \). The segment length \( L = 530 \) bytes seems to be practically for video streams that are generated with common applications (e.g. Video Conferencing Tool (VIC) [14] using Real Time Protocol (RTP) [13] on top of UDP).

Figure 6, 7, 8 and 9 show the SLP versus the maximum number of parallel channels, that each WT uses for the SMPT approach, in dependency of the number of existing WTs in one wireless cell. The point of interest is the optimal number of parallel channels that one WT
will use to achieve a low SLP without disturbing itself or other WTs too much. We already mentioned that allocating additional channels will lead to an increased BEP. If such an optimal number of channels exists (where the SLP for all WTs reaches a minimum) a centralized entity (e.g. base station) has only to inform all WTs about the maximum number of usable channels.

The SLP for a required bounded jitter of zero \((\beta = 0)\) is depicted in figure 6 and 7. All available channels can be used to minimize the SLP if the number of WTs is lower than 18. If the number of WTs reaches 18 the optimal number of parallel channels will decrease from five to four in order to achieve the lowest SLP for all WTs. Increasing the number of WTs furthermore results in a smaller optimal number of usable channels. For 28 WTs the lowest SLP can be achieved if all WTs are sending with the SMPT approach and only two parallel channels.

Figure 6 shows the SLP in dependency of the WTs and the maximum number of parallel channels. The black line represents the decision that a centralized entity has to make to appoint the optimal number of channels to the WTs using SMPT.

The SLP of applications that allow a bounded jitter \(\beta\) of three or six is depicted in figure 8 and 9. Also in this case an optimal number of channels allowed in maximum for the SMPT approach can be found in dependency of the total number of WTs. Consider the situation were 16 WTs send their MPDUs with different transmission methods over the wireless link: The sequential transmission method results in a SLP of 15.7\% while the SMPT approach with an optimal number of four channels can offer the application a SLP of 0.038\%. This shows in an impressive way how application benefit from the SMPT approach.

**IV. CONCLUSION**

In summary we have shown that multimedia applications which require a bounded jitter benefit from the introduced SMPT algorithm. We presented simulation results showing that SMPT improves the SLP for a given jitter bound when the number of used channels is known. The optimal number of channels for the SMPT approach depends on the number of parallel existing WTs.

The simulation presented in this paper where limited to a SMPT algorithm which is only based on ARQ (e.g. retransmissions of the original packets). On the other hand it is generally accepted that FEC or a mixture of ARQ and FEC (Hybrid Type I,II) can improve the performance of wireless LANs.
<table>
<thead>
<tr>
<th>Application</th>
<th>( L )</th>
<th>530 byte</th>
</tr>
</thead>
<tbody>
<tr>
<td>load</td>
<td>SMPT stations</td>
<td>75%</td>
</tr>
<tr>
<td></td>
<td>background stations</td>
<td>100%</td>
</tr>
<tr>
<td>MAC</td>
<td>( L_{MPDU} )</td>
<td>106 bit,</td>
</tr>
<tr>
<td></td>
<td>( \zeta )</td>
<td>5 byte</td>
</tr>
<tr>
<td></td>
<td>( \beta )</td>
<td>0,3,5</td>
</tr>
<tr>
<td></td>
<td>( k )</td>
<td>1,2,3,4,5</td>
</tr>
<tr>
<td>Channel</td>
<td>( N_{spreading} )</td>
<td>128</td>
</tr>
<tr>
<td></td>
<td>Number of background channels</td>
<td>10</td>
</tr>
<tr>
<td></td>
<td>Maximum Number of Channels per WT</td>
<td>5</td>
</tr>
</tbody>
</table>

Table 1. Simulation Parameters

REFERENCES


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