Abstract

In this paper, we address the problem of packet scheduling for high speed CDMA, which can provide high speed data service for mobile users. We first investigate the appropriate multiple access schemes in high speed CDMA. We find through mathematical analysis that TDM is more efficient than CDM for downlink in high speed CDMA networks in some general situations. Based on this finding, we propose and evaluate an efficient scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC), for packet scheduling in high speed CDMA networks. CSDFSC tries to maximize channel throughput under fairness and transmission power constraints. This scheme differentiates best-effort and guaranteed services by assigning different queue weights to various traffic classes. By considering the difference between the pre-defined and actually occupied bandwidth proportion in the priority computation, CSDFSC exhibits good fairness properties for best-effort flows. We compare the performance of CSDFSC with that of WISPER and WRR by using ns-2 simulations, where VBR video and TCP traffic flows are considered. The numerical results show that CSDFSC outperforms them in terms of throughput, link utilization and average end-to-end packet delay, while retaining a low implementation complexity.

Keywords: CDMA; Packet scheduling; Fair service; Channel states

1. Introduction

Code-division multiple access (CDMA) is the multiple access standard of 3G wireless networks and will be widely used in future communication infrastructure [18]. CDMA systems have been
In this paper we address the packet scheduling problem in high speed CDMA networks. We first summarize the problems faced by fair queuing scheduling in wireless networks and review several known scheduling algorithms for CDMA networks, including the wireless multimedia access control protocol with BER scheduling (WISPER) [9]. Based on the CDMA downlink scheduling model we formulate, we theoretically prove that TDM is more efficient than CDM multiple-access scheme for downlink in high data rate CDMA network under certain conditions. In other words, base station should schedule only one user to transmit in a time-slot. Our finding coincidentally is the same as CDMA/HDR and HSDPA (High Speed Downlink Packet Access).

Based on this finding, we propose a new scheduling algorithm especially for high speed CDMA networks, namely Channel States Dependent Fair Service for CDMA (CSDFSC), which makes decision in each time-slot and chooses the user with highest priority to transmit data in one time-slot. Since bandwidth is the most scarce resource in wireless networks, the objective of our proposal is to maximize channel throughput and minimize packet loss while supporting multiple classes of traffic and providing fair service for best-effort flows. In order to achieve these objectives, CSDFSC computes the priority of packets according to their queue weights, fairness between flows and current channel states. By using different queue weights for best-effort and guaranteed traffic flows, CSDFSC differentiates the services and provides higher priority and lower packet delay for the latter. Another advantage of CSDFSC is that it tries to provide fair service for best-effort traffic by measuring the difference between actual occupied bandwidth and predefined proportion of bandwidth, and computing the priority of flows based on this difference. In the design of CSDFSC, multiple states Markov model is used to describe the CDMA link states and the maximum transmission rate in CSDFSC. The flow perceiving channel error will have lower priority and may give up their transmission opportunity to those without channel error. Furthermore, the scheduling process is much simpler in CSDFSC than that in WISPER, because of the lack of packet allocation.
procedure in CSDFSC. As a result, its implementation complexity is significantly lower than that of WISPER. For a more comprehensive evaluation for the performance of our proposed algorithm, we have conducted extensive simulation experiments to compare CSDFSC with both WISPER and Weighted Round Robin (WRR). The numerical results demonstrate that CSDFSC can achieve not only higher channel capacity, but also higher link bandwidth utilization and lower packet loss ratio, compared with WISPER or WRR. Moreover, CSDFSC can achieve lower average packet delay for the guaranteed traffic and “fairer” service for best-effort traffic compare to WISPER.

The rest of this paper is organized as follows. In Section 2, we summarize the major issues which need to be addressed for wireless scheduling. Section 3 introduces several known scheduling schemes for CDMA networks. We present the system model and elaborate the theoretical analysis on the multiple access scheme for high speed CDMA in Section 4. We present the proposed CSDFSC in Section 5. The simulation results and discussions are presented in Section 6. We conclude this paper by highlighting our contributions in Section 7.

2. Major issues in wireless scheduling

As wireless links are subject to bursty and location-dependent errors, packet scheduling in wireless networks is link state dependent. Other key characteristics of the wireless transmission influencing packet scheduling include: (a) The wireless channel states and capacity vary randomly in time on both slow and fast time scale [15]; (b) There is contention in the channel among multiple mobile hosts; (c) The bandwidth is scarce resource in wireless networks; and (d) Mobile hosts always have limited processing power and battery power.

Due to the above mentioned wireless channel characteristics, existing packet scheduling algorithms which have been proved efficient in wireline network are not directly applicable to wireless networks. The design of wireless scheduling algorithms needs to address a series of issues discussed below.

1. Location-dependent channel state: the channel of wireless network is error-prone and suffers from interference, fading and shadows. This channel variability is time-dependent and location-dependent as well i.e. users at different location may receive different SNR and different service quality. Unfortunately, traditional fluid fair queuing schemes can guarantee neither fairness nor packet delay in wireless channel.

2. Fairness between flows: in wireline networks, scheduling fairness is usually achieved by guaranteeing a certain service rate to a flow. The fairness issue in wireless networks is much more complicated. When the channel of one mobile terminal is in bad state, and a packet is transmitted to this terminal, this packet will be corrupted and the transmission will waste transmission resources. For better channel utilization, deferring transmission of this packet until the channel recovers from the error state is obviously a reasonable choice. As a result, the flow with error channel state temporarily loses its share of the transmission bandwidth and another flow with good channel state “steals” more share of bandwidth. For ensuring fairness, the flow that lost share of bandwidth should be compensated later when the link recovers. However, determining how to compensate for it is not trivial task. There already exist some fairness definitions for wireline scheduling algorithms available in [7,8]. However, the definition and objectives of fairness guarantees in a wireless communication network become more ambiguous, since the appropriate interpretation of fairness should depend on the service model, traffic type, and channel characteristics.

3. QoS for heterogeneous classes: in future broadband wireless networks, heterogeneous classes of traffic with different QoS requirements should be supported. Thus, flow class differentiation and QoS guarantees must be provided by wireless scheduling algorithms. IntServ [4] and DiffServ [5] have different requirements and constraints for corresponding wireless scheduling scheme. Per-flow-based QoS requirements should be guaranteed for IntServ, while for DiffServ, prioritized scheduling service for
aggregated traffic is the minimum requirement for scheduling algorithm. The variability of wireless channel state will make it difficult to guarantee QoS in wireless network. Nevertheless, QoS should be guaranteed for the flows, either deterministically or statistically, on those links that have bounded errors.

4. Throughput and link utilization: bandwidth is the most precious and scarce resource in wireless networks. An efficient wireless scheduling algorithm should guarantee short-term throughput for flows with error-free channels and provide enough long-term throughput for all flows. Furthermore unnecessary transmission loss should be minimized while efficient data transmission should be maximized.

5. Power constraint: mobile terminals are often constrained in terms of process power and battery power. An efficient wireless scheduling algorithm should have low implementation complexity and utilize control information from mobile terminals as little as possible.

The above peculiarities distinguish the wireless scheduling from the conventional wireline scheduling. It is therefore important to make use of the information on channel states in the design of wireless scheduling algorithm.

3. Related work

CDMA is the most promising multiple-access technique for the future wireless network, and is chosen to be standard of 3G wireless networks. One of the greatest differences between CDMA and other multiple access modes is that in CDMA a mobile terminal may use multiple channels at the same time. In other words, the channel capacity in CDMA is “soft”. This feature makes scheduling problem in CDMA networks more complicated. In order to make use of this soft capacity feature of CDMA, an accurate power control mechanism is necessary.

There has been much work addressing the scheduling problem for CDMA networks [9,12,13,15]. Wireless multimedia access control protocol with BER scheduling (WISPER) [9], proposed by Akyildiz et al. is one of the first proposal on packet scheduling for CDMA networks. In future wireless network, there will be a mixture of different traffic classes that have different QoS requirements, such as BER, transfer rate and delay bound. For example, voice service can tolerate BER’s of up to $10^{-3}$, but data packet transmission requires BER’s below $10^{-9}$. It will be wasteful to transfer voice and data packets in same channel, since the channel must satisfy the most stringent BER specification of data packets. WISPER incorporates a novel packet scheduler, which performs the selection and efficient accommodation of the packets to be transferred in frames. For each frame, scheduler computes the packets’ priorities, accommodate packets with highest priority to the frame, and try to arrange packets with same or similar BER requirements in the same slot. Thus the BER requirements are satisfied and at the same time channel throughput is maximized. However, WISPER does not consider the fairness and error-prone channel issues.

Arad et al. proposed a hybrid CDMA/TDMA scheduler—Scheduled CDMA [12], which is mainly focused on uplink scheduling with power constraint. In SCDMA, data is exchanged between the Base-Station and Mobile-Stations in a fixed-size unit called capsule. When an MS has new packets to transmit, it sends a capsule transmission request (CTR) to BS. The scheduler in BS places CTR in a global queue, when there is a time slot to be filled, scheduler chooses CTRs according to their priority level, delay requirement and the time spent in queue. The choice procedure continues until the sum of all CTRs’ power index reach power constraint. Then BS sends transmission permission capsules to MSs to inform them the transmission time and power level.

Based on SCDMA [12], Gurbuz et al. proposed Dynamic Resource Scheduling (DSR) [13], which is a modified SCDMA without the TDMA aspect. DSR is a centralized and adaptive framework for providing resource scheduling through optimal power assignment and code hopping in wideband CDMA (WCDMA). The advanced feature of DSR is that it classifies flows to two queues, guaranteed and best effort, according to their characteristics of requested service. The CTRs in
guaranteed queue receive service before those in best effort queue, thus predefined rate of sessions are guaranteed. However, delay bounds are not guaranteed in both SCDMA and DSR.

Besides above proposals, an “online” scheduling algorithm, namely Modified Largest Weighted Delay First (MLWDF) is presented in [15]. One of the advanced features of MLWDF is that it chooses queue to serve based on not only current channel state but also current queues’ states. And it is very easy to implement. Another advanced feature of MLWDF is its ability to adjust delay bound for flows of different class and the delay distributions can be shaped by choosing suitable parameters.

In [6], author proposed a MAC protocol with efficient fair scheduling for multi-code CDMA (MC-CDMA) networks. Similar as WISPER, it can maximize throughput by allocating packets with similar BER requirement into the same time-slot. Beside this desirable property, it also can provide fair service to multiple flows similar as GPS with less overhead information.

However, the above mentioned work on the subject of this paper address some what different problem. For example, all above proposals do not guarantee fairness between best-effort traffic flows. SCDMA cannot support multiple classes of traffic. The objective of our proposal is to maximize channel throughput and minimize packet loss while supporting multiple classes of traffic and providing fair service, while taking into account the channel states.

4. System model and theoretical analysis

4.1. Proposed CDMA scheduling model

In this section, we present the proposed packet scheduling model for downlink in CDMA networks. We consider a CDMA system with a single cell, where there are \( N \) users and each user has one flow that needs to be served. We denote the set of users by \( N = \{1, \ldots, N\} \). Each flow has its own queue in the base station. The channel transmitting time is divided into some intervals with fixed duration. A time interval \([t, t + \delta]\), where \( t = 0, \delta, 2\delta, \ldots, \) is called the time slot. For simplicity, the channel state is represented in terms of the noise level, including fading and interference, experienced by the users. \( C = \{s_1, s_2, \ldots, s_C\} \) is the set of possible channel states, where \( C \) denotes the number of the channel states, and the channel state is invariable within one time slot. The random channel state process \( c \) is assumed to be a discrete time Markov chain with the finite state space \( C \), an efficient model to describe CDMA link [17]. Associated with each state \( c \in C \) is a maximum feasible data rate \( r_c \in R \). This model is denoted from [10]. \( R \) is the finite set of feasible rates of CDMA users, \( R = \{r_1, r_2, \ldots, r_C\} \). Let \( c_{ik} \) be the channel state of user \( i \) at slot \( k \), which is assumed to be known by base station or predictable by some algorithms such as one-step prediction [8]. Let \( p_{ik} \) be the normalized power allocation of user \( i \) at time slot \( k \).

In CDMA, base station can assign different signal power level by choosing spreading code for mobile user, the actual transmission rate is determined by SIR (Signal Interference Ratio) experienced by mobile user. Then the transmission rate of user \( i \) at time slot \( k \), represented by \( r_{ik} \), is given by

\[
g(c_{ik}, p_{ik}) = r_{ik} \in R,
\]

where \( g() \) expresses the transmission rate as a function of channel state and transmission power level. Moreover, let \( Y_i(t) \) be the throughput of user \( i \) from time 0 to \( t \). \( t = M \delta \), where \( M \) is an integer. Thus we have

\[
Y_i(t) = \sum_{k=1}^{M} r_{ik} \cdot \delta
\]

The scheduling objective is to maximize the total throughput

\[
\sum_{i \in N} Y_i(t) = \sum_{i=1}^{N} \sum_{k=1}^{M} g(c_{ik}, p_{ik}) \cdot \delta
\]

subject to two constraints:

1. Fairness constraint

\[
\frac{Y_i(t)}{\phi_i} = \frac{Y_j(t)}{\phi_j} \text{ for each } i, j \in N,
\]

where \( \phi_i \) represents the assigned proportion of bandwidth of user \( i \).
2. Power constraint

\[ \sum_{i \in N} p_{ik} \leq 1 \quad \text{and} \quad p_{ik} \geq 0, \]

where \( p_{ik} \) is normalized by total transmission power, since total transmission power of the base station is limited [19,20].

In the following, we analyze the multiple access method for high speed CDMA by using the proposed system model.

4.2. Multiple-access schemes in high-data rate CDMA: TDM vs CDM

As the CDMA packet scheduling issue is related with the multiple access scheme, we first compare the existing multiple access methods to find the appropriate one for downlink in high speed CDMA. TDM (Time Division Multiple access) and CDM (Code Division Multiple access) are two types of widely deployed multiple access schemes for communication networks. In TDM, only one user can transmit at a time and it only transmits in a short time scale. In CDM, multiple users transmit data simultaneously in a long time scale but with a lower transmission rate. Both schemes have their advantages and disadvantages. Interference between users exists in CDM, while TDM needs precise and complicated control of transmission time. In the following, we employ a mathematical model to examine TDM and CDM for high speed CDMA networks to choose an appropriate one for high speed CDMA networks.

First, we examine the transmission rate in one time-slot. Our objective is to maximize the sum of all users’ transmission rates in this time-slot. If the transmission rate is maximized in every timeslot, the total throughput can be also maximized. In a time-slot, user \( i \) receives transmission power \( P_i \) with SIR \( \gamma_i \) and transmission rate \( R_i \). Hence the optimal problem becomes Maximize \( \sum_{i \in N} R_i \). First, let us consider a special case where \( R_i r_0 = \gamma_i / \gamma_0 \), in which \( r_0 \) is the minimum feasible transmission rate, and \( \gamma_0 \) is the minimum SIR which can provide \( r_0 \) for user \( i \). We have the following theorem.

**Theorem 1.** If the maximum transmission rate is in proportion to SIR, \( \sum R_i = \sum \gamma_i (r_0 / \gamma_0) \) reaches maximum value when only one user transmits at full power.

**Proof.** For simplicity, we express \( P_i \) in unit of total interference \( I + \sigma \), where \( \sigma \) is the thermal noise and \( I \) is the total interference from other sources, including the other cells in networks [16]. To meet SIR requirement, the following condition should be satisfied:

\[ \frac{P_i}{\sum_{j \neq i} P_j + 1} \geq \frac{\gamma_i}{\gamma}, \]

Obviously, for the optimal situation the solution will be \( P_i / (\sum_{j \neq i} P_j + 1) = \gamma_i / \gamma \), or \( P_i / \gamma_i = \sum_{j \neq i} P_j + 1 \). Thus we have

\[ P_i \gamma_i + 1 = \sum_{j \neq i} P_j + 1 = \text{Const}, \]

where Const is a constant. Const can be resolved as \( \text{Const} = 1 + \text{Const} \sum_{i \in N} (\gamma_i / (\gamma_i + 1)) \), or \( \text{Const} = 1 / (1 - \sum_{i \in N} (\gamma_i / (\gamma_i + 1))) \). Defining \( x_i = \gamma_i / (\gamma_i + 1) \), from Eq. (5) we have

\[ P_i = \frac{x_i}{1 - \sum_{i \in N} x_i}. \]

Considering the power constraint \( \sum P_i \leq 1 \), in optimal situation,

\[ \frac{\sum_{i \in N} x_i}{1 - \sum_{i \in N} x_i} = 1 \Rightarrow \sum_{i \in N} x_i = \frac{1}{2} \Rightarrow \sum_{i \in N} \gamma_i / (\gamma_i + 1) = \frac{1}{2}. \]

Our objective is to maximize \( \sum R_i = \sum \gamma_i (r_0 / \gamma_0) \), while satisfying Eq. (7). Using \( x_i = \gamma_i / (\gamma_i + 1) \), the problem becomes:

\[ \text{Maximize } f(x) = \sum_{i \in N} x_i, \quad \text{s.t. } x_i = \frac{1}{1 - x_i}, \]

where \( x_i = \gamma_i / (\gamma_i + 1) \), subject to constraint \( \sum_{i \in N} x_i = \frac{1}{2} \).

According to Lagrange multiplier, let

\[ F(x) = f(x) - m \left( \sum_{i \in N} x_i - \frac{1}{2} \right). \]
The extremum of \( f(x) \) subject to the constraint given by Eq. (7) should satisfy:
\[
\begin{align*}
\frac{\partial f}{\partial x_i} - m &= 0, \\
\sum_{i \in \mathbb{N}} x_i - \frac{1}{2} &= 0.
\end{align*}
\]  
\tag{9}

From Eq. (9), it is easy to find that
\[
F''_{x_i x_i} = \frac{2}{(2N - 1)^3} > 0, \quad i = j,
\]
\[
0, \quad i \neq j.
\]  
\tag{11}

We can find that the objection function is a convex function. We also have
\[
b_{ij} = \frac{2(2N)^3}{(2N - 1)^3} > 0, \quad i = j,
\]
\[
0, \quad i \neq j.
\]  
\tag{12}

Obviously \( B \) is positive definite (\(|B| > 0\)) and hence \( M_0 \) is a minimum extremum point. Then the maximum value of \( f(x) \) should exist at the boundary. The boundary of \( f(x) \) is given by
\[
x_i = \begin{cases} 
\frac{1}{2}, & i = j, \\
0, & i \neq j,
\end{cases} \quad i = 1, 2, \ldots, N; \quad j \in \mathbb{N},
\]  
\tag{13}

and the value of \( f(x) \) at boundary is
\[
f\left( x_i = \frac{1}{2} \right) = 1
\]  
\tag{14}

In conclusion, we should choose only one user in a time-slot and exploit total available power to transmit data as fast as possible, when the maximum transmission rate is proportional to SIR.

Next, we consider a looser condition. Suppose \( R_i = g(\gamma_i) \) and \( g() \) satisfies \( r_0 = g(\gamma_0) \). We have the following theorem.

**Theorem 2.** Suppose \( g() \) is an increasing function and the increasing rate of \( g() \) is high enough, such that \( g(1) > N \ast g(1/(2N - 1)) \). \( \sum R_i \) reaches the maximum value with TDM deployed.

**Proof.** In this case, since \( g() \) is an increasing function with \( g'(\gamma_i) > 0 \), we have \( \partial f / \partial x_i = g'(\gamma_i) \) \((1 - \gamma_i)^2 > 0 \). Similar to Eq. (8), \( M_0 = \{ x_i \}, \quad x_i = 1/(2N), \quad i = 1, 2, \ldots, N \) is also the extremum point. Then the extremum is
\[
f(M_0) = \sum_{i \in \mathbb{N}} g\left( \frac{2N}{2N - 1} - 1 \right) = \sum_{i \in \mathbb{N}} g\left( \frac{1}{2N - 1} \right)
\]
\[
= N \ast g\left( \frac{1}{2N - 1} \right).
\]  
\tag{15}

The value of \( f(x) \) at the boundary is given by
\[
f\left( x_i = \frac{1}{2} \right) = g(1).
\]
\tag{16}

Regardless that \( f(M_0) \) is the maximum or minimum extremum, if \( g(1) > N \ast g(1/(2N - 1)) \), the maximum value of \( f(x) \) exists at the boundary shown in Eq. (16). Obviously we have the same conclusion as Theorem 1. Unlike in Theorem 1, we do not consider convexity of object function here.

Let us use an example to illustrate Theorem 2. Table 1 shows the relationship between transmission rates and SIR value (in dB) in CDMA 1xEV-DO system [14]. For instance, \( N = 5 \), according to Eqs. (15) and (16), we have
\[
\gamma_{M_0} = \frac{\gamma_{M_1}}{2N - 1},
\]
\tag{17}

where \( \gamma_{M_0} \) is the SIR of a user at \( M_0 \) point, and \( \gamma_{M_1} \) is the SIR value at boundary \( x_i = \frac{1}{2} \). If we measure SIR value in dB, Eq. (17) becomes \( \gamma_{M_0} = \gamma_{M_1} - 10 \log 9 = \gamma_{M_1} - 9.5 \). For example, with \( \gamma_{M_1} = 3 \) dB, \( \gamma_{M_0} = -6.5 \) dB. According to Eqs. (15) and (16),
\begin{align*}
  f(M_0) &= 5 * g(-6.5 \text{ dB}) = 5 * 153.6 = 768 \text{ Kb/s} \\
  f(x_i = 1_2) &= g(3 \text{ dB}) = 1228.8 \text{ Kb/s}
\end{align*}

(18)

Obviously, TDM for high speed CDMA has higher throughput in this case and hence Theorem 2 holds.

5. Channel states dependent fair service for CDMA (CSDFSC)

Motivated by the above finding, we propose a new packet scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC). CSDFSC is designed for high data rate CDMA wireless network to achieve service differentiation, fairness among best-effort traffic flows, and high channel throughput with low implementation complexity. In CSDFSC, we assume that system operates in the hybrid TDMA/CDMA mode, in which TDM is deployed in downlink and CDM is deployed in uplink. The downlink scheduling decision is made at each time slot and only one user can transmit data in each time slot.

We also assume that the channel state follows the multi-state Markov channel model, and the base station is aware of the state information from the channel feedback. Two-state Markov channel model is widely used in previous research for its simplicity and efficiency to describe wireless channel characteristics. However, in CDMA the situation is more complicated since the user’s SIR is related to its transmission rate in CDMA network. With different code and higher power allocation, a mobile user can transmit in lower but not zero rate under “Bad” channel state. Thus, multiple states Markov chain model [21] is more accurate for describing CDMA channel characteristics. In CSDFSC, we employ a three-state Markov chain for modeling channel states.

As shown in Fig. 1, CSDFSC consists of two main components: Flow Classifier and Scheduler (FCS), and Frame Structurer (FS). The arriving data are classified at the base station and put into different queues. FCS chooses the transmission flow according to the channel states, bandwidth allocations and flow weights. FS organizes the head of line (HOL) packets of selected flows into a data frame, which may have different size in different systems, and sends them out.

With the channel states, weight of flows and bandwidth allocation, the priorities of flows are calculated as:

\[ \text{prio} = W_i * (Z_{\text{max}}(t) - Z_i(t)) * R_i(t), \]

where \( W_i \) is the predefined weight of flow \( i \), \( Z_i(t) = Y_i(t) / \phi_i \), and \( Z_{\text{max}}(t) = \text{Max} \{Z_i(t)\} \), and \( R_i(t) \) is the maximum feasible transmission rate of flow \( i \) at time \( t \). In our implementation, \( Y_i(t) \) is calculated at the end of each time slot. Suppose flow \( j \) is selected to transmit data in this time slot, then only \( Y_j(t) \) will be updated after the transmission completed. It’s necessary and efficient since scheduling procedure make decision each time slot. Another way is to update \( Y_i(t) \) packet by packet, which needs more computation.

<table>
<thead>
<tr>
<th>SIR</th>
<th>Rates (Kb/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>-12.5</td>
<td>38.4</td>
</tr>
<tr>
<td>-9.5</td>
<td>76.8</td>
</tr>
<tr>
<td>-8.5</td>
<td>102.6</td>
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<td>-6.5</td>
<td>153.6</td>
</tr>
<tr>
<td>-5.7</td>
<td>204.8</td>
</tr>
<tr>
<td>-4.0</td>
<td>307.2</td>
</tr>
<tr>
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</tr>
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<td>1.3</td>
<td>921.6</td>
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<tr>
<td>3.0</td>
<td>1228.8</td>
</tr>
<tr>
<td>7.2</td>
<td>1843.2</td>
</tr>
<tr>
<td>9.6</td>
<td>2457.6</td>
</tr>
</tbody>
</table>

Table 1

Transmission rate vs SIR value
In this formula, the term $Z_{\text{max}}(t) - Z_i(t)$ is the difference between the service received by a specific flow and the maximum service, which reflects the fairness among flows. The current channel states are reflected by $R_i(t)$. The predefined weight is used to differentiate guaranteed flow from best-effort flow.

When the next frame needs to be scheduled, CSDFSC assigns the flow with the highest priority in the first time-slot and updates this flow's priority after allocation. The above procedure is repeated until all time-slots in this frame are fully occupied, as shown in Fig. 2.

In order to achieve low implementation complexity, CSDFSC uses a simple priority equation to determine the sequence of transmitting packets. As a result, the implementation complexity is much less than that of WISPER. In WISPER, the number of packets transmitted and the number of mobile users in a time-slot are not fixed; scheduler should compute them according to the BER requirement of flows. Furthermore, the procedure of allocating packets into a frame is also very complicated. To avoid the complex virtual service time computation which is essential in GPS and similar scheduler in wireline networks, CSDFSC tries to emulate GPS by calculating relative difference, represented by $Z_{\text{max}}(t) - Z_i(t)$, of received service between two flows in computing the flows' priorities. The flow that has obtained more transmission opportunity will have lower priority than a flow that has shared less transmission opportunity. Then the fairness between flows will be achieved to a certain extent.

Now we use an example to illustrate the scheduling procedure in CSDFSC. Let there be three flows $f_1$, $f_2$, and $f_3$. The queue weights and assigned proportion of each flow are listed in Table 2. A t time 0, there are 14, 6 and 6 packets belonging to $f_1$, $f_2$, and $f_3$, respectively. The maximum transmission rates, in unit of packets, of each flow from time slot 1 to 8 are listed in Table 3.

The process of making scheduling decisions in CSDFSC is interpreted as follows. At time 0, all flows' priorities are equal to zero, as shown in Table 4. The scheduler chooses the flow with the largest number of packets, $f_1$, and the maximum transmission rate of $f_1$ at time slot 1, $R_1(1)$, equal to 4 packets, and thus at time slot 1 flow $f_1$ transmits 4 packets, as shown in Table 5. Table 6 shows the $Z_i(t)$ of each flow during the scheduling process. After time slot 1, the throughput of flow $f_1$ is changed to 4. According to Eq. (19) the priority of flow $f_2$ is changed to 16 because $R_2(2) = 4$, $Z_{\text{max}}(2) - Z_2(2) = 4$ and $W_2 = 1$. The priority of $f_3$ remains 0 because $R_3(2) = 0$. Hence flow $f_2$ is

![Fig. 2. Frame allocation procedure in CSDFSC.](image-url)
chosen to transmit at time slot 2. Repeating this procedure, all packets of flows are transmitted after time slot 8 and no packet is lost during transmission.

6. Simulation results

In this section, we compare the performance of CSDFSC with that of WISPER and WRR by using ns-2 simulations on a typical network topology. WISPER is a known CDMA scheduling algorithm, the comparison results will be used for verifying Theorems 1 and 2. Moreover, WISPER is proved to be efficient in terms of throughput. It would be convincing to claim our CSDFSC is efficient if CSDFSC outperforms WISPER. In some recent relevant papers, such as [6], the authors also compare their schemes with WISPER. WRR assigns different queue weight value for different flows, which is similar as CSDFSC. In wireline networks, WRR is proved to be efficient in throughput and fairness. The performance metrics we evaluate include the number of users the system can support, packet loss, link utilization, fairness, etc. The traffic in the simulations experiments include VBR video and TCP flows.

6.1. Network topology

Fig. 3 depicts the network topology used in our simulation experiments. There are two cells in the network and each cell has a base station and multiple video and data users. One cell is “destination cell” which contains a base station and some destination mobile stations while the other is used as “source cell” which contains a base station and

![Network topology for simulation experiments.](image)

Table 4

<table>
<thead>
<tr>
<th>Slot 1</th>
<th>Slot 2</th>
<th>Slot 3</th>
<th>Slot 4</th>
<th>Slot 5</th>
<th>Slot 6</th>
<th>Slot 7</th>
<th>Slot 8</th>
</tr>
</thead>
<tbody>
<tr>
<td>f1</td>
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<td>0</td>
<td>32</td>
<td>0</td>
<td>16</td>
<td>0</td>
<td>96</td>
</tr>
<tr>
<td>f2</td>
<td>0</td>
<td>16</td>
<td>0</td>
<td>0</td>
<td>32</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>f3</td>
<td>0</td>
<td>0</td>
<td>32</td>
<td>0</td>
<td>0</td>
<td>16</td>
<td>0</td>
</tr>
</tbody>
</table>

Table 5

<table>
<thead>
<tr>
<th>Slot 1</th>
<th>Slot 2</th>
<th>Slot 3</th>
<th>Slot 4</th>
<th>Slot 5</th>
<th>Slot 6</th>
<th>Slot 7</th>
<th>Slot 8</th>
</tr>
</thead>
<tbody>
<tr>
<td>f1</td>
<td>4</td>
<td>0</td>
<td>0</td>
<td>2</td>
<td>0</td>
<td>0</td>
<td>4</td>
</tr>
<tr>
<td>f2</td>
<td>0</td>
<td>2</td>
<td>0</td>
<td>0</td>
<td>4</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>f3</td>
<td>0</td>
<td>0</td>
<td>4</td>
<td>0</td>
<td>0</td>
<td>2</td>
<td>0</td>
</tr>
</tbody>
</table>

Table 6

<table>
<thead>
<tr>
<th>Slot 1</th>
<th>Slot 2</th>
<th>Slot 3</th>
<th>Slot 4</th>
<th>Slot 5</th>
<th>Slot 6</th>
<th>Slot 7</th>
<th>Slot 8</th>
</tr>
</thead>
<tbody>
<tr>
<td>f1</td>
<td>0</td>
<td>8</td>
<td>8</td>
<td>8</td>
<td>12</td>
<td>12</td>
<td>12</td>
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<tr>
<td>f2</td>
<td>0</td>
<td>0</td>
<td>8</td>
<td>8</td>
<td>8</td>
<td>24</td>
<td>24</td>
</tr>
<tr>
<td>f3</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>16</td>
<td>16</td>
<td>16</td>
<td>24</td>
</tr>
</tbody>
</table>
some source nodes. Traffic flows are transmitted from source nodes to destination nodes, thus in destination cell there is only downlink traffic. Since our theoretical analysis is concerned with downlink traffic in a single cell, we focus on the analysis of the traffic transmission within the destination cell. The link bandwidth between two base stations is 4 Mb/s, and all link bandwidth between mobile users and base station is 10 Mb/s. Multi-state Markov model is used to define wireless channel in our simulation. The parameters of the link model are listed in Table 7. The same traffic weight values in this table are also used in WRR.

Two types of traffic are used in our simulation. One is VBR video traffic, which is modeled as three-state on-off source. At different state, packets are generated at different data rate. Another is TCP data traffic, which simulates the transmission of a very large file via FTP. The VBR video traffic has higher traffic weight than TCP traffic. The parameters of two types of traffic are listed in Table 8. We use the same WISPER parameters (Table 9) of these two types of traffic as those in [9].

6.2. Number of users the system can support

In the first experiment, we compare the maximum number of video and TCP data users which the forward link is able to support under CSDFSC with that under WISPER and WRR (Weighted Round Robin) schemes. In this experiment, the link with packet loss ratio of video users larger than 1% is said to be “out of capacity”. Otherwise the link is said to be “in capacity”. The maximum number of video and data users when the link remains “in capacity” is called “maximum traffic load”. Fig. 4 shows the relationship between the number of TCP users and the number of video users the system is able to support simultaneously. For example, CSDFSC can support nine TCP users in addition to two video users. In comparison, WISPER and WRR can support only additional five TCP users as shown in Fig. 4. With seven video users, CSDFSC can support another two TCP users while WISPER and WRR cannot support any more TCP user. Obviously, the link can support more users with CSDFSC than with WISPER and WRR. This is because that CSDFSC addresses the location-dependent and burst error, while WISPER and WRR do not consider the channel state.

Table 7
Parameters of channel model

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of states</td>
<td>3</td>
</tr>
</tbody>
</table>
| Transition matrix        | \[
|                          | [0.9995, 0.005, 0]         |
|                          | [0, 0, 1]                  |
|                          | [1, 0, 0]                  |
| State 1 average duration time | 0.1 s                      |
| State 2 average duration time | 0.01 s                    |
| State 3 average duration time | 0.01 s                    |
| Transmission rate in state 1 | 4 packet per slot         |
| Transmission rate in state 2 | 2 packet per slot         |
| Transmission rate in state 3 | 0 packet per slot         |

Table 8
CSDFSC parameters of traffic

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video State 1 transmission rate</td>
<td>500 Kb/s</td>
</tr>
<tr>
<td>State 2 transmission rate</td>
<td>400 Kb/s</td>
</tr>
<tr>
<td>State 3 transmission rate</td>
<td>200 Kb/s</td>
</tr>
<tr>
<td>State 1 probability</td>
<td>0.5</td>
</tr>
<tr>
<td>State 2 probability</td>
<td>0.3</td>
</tr>
<tr>
<td>State 3 probability</td>
<td>0.2</td>
</tr>
<tr>
<td>Average on-time</td>
<td>1 s</td>
</tr>
<tr>
<td>Average off-time</td>
<td>0.1 s</td>
</tr>
<tr>
<td>Packet size</td>
<td>256 bytes</td>
</tr>
<tr>
<td>Traffic weight</td>
<td>2</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>TCP Type</td>
<td>Reno</td>
</tr>
<tr>
<td>Window</td>
<td>40</td>
</tr>
<tr>
<td>Packet size</td>
<td>256 bytes</td>
</tr>
<tr>
<td>Traffic weight</td>
<td>1</td>
</tr>
</tbody>
</table>

Table 9
WISPER parameters of traffic

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video Maximum tolerable BER</td>
<td>10^{-6}</td>
</tr>
<tr>
<td>Maximum number of packets per slot</td>
<td>4</td>
</tr>
<tr>
<td>Packet timeout value</td>
<td>3 frames</td>
</tr>
<tr>
<td>Maximum transmission capacity</td>
<td>4 packets/slot</td>
</tr>
<tr>
<td>TCP Maximum tolerable BER</td>
<td>10^{-9}</td>
</tr>
<tr>
<td>Maximum number of packets per slot</td>
<td>4</td>
</tr>
<tr>
<td>Packet timeout value</td>
<td>(2*total_packets) frames</td>
</tr>
<tr>
<td>Maximum transmission capacity</td>
<td>4 packets/slot</td>
</tr>
</tbody>
</table>
6.3. Packet loss

Next, we compare the packet loss ratio of flows in two schemes when the numbers of the video and TCP users equal to the values in Section 6.2. As shown in Fig. 5, the packet loss ratio of TCP users remains a low level in CSDFSC. In comparison, in WISPER and WRR TCP users experience significantly higher packet loss ratio, which will impede the transmission of TCP traffic. This is mainly because that WISPER and WRR do not use the channel state information. When the channel state is not “good”, all transmitting packets will be lost in WISPER and WRR. In CSDFSC, on the other hand, the scheduler will lower current user’s transmission rate or schedule another user with good channel state to transmit. Another observation is that the packet loss ratio of video users is below the minimum requirement 1%, in all schemes, as shown in Fig. 6. In CSDFSC, the packet loss ratio increases with the number of video users, while it is relatively stable in WISPER and WRR.

6.4. Link utilization

Next, we compare the link utilization between WISPER, WRR and CSDFSC schemes under “maximum traffic load” as Section 6.2. As shown in Fig. 7, CSDFSC achieves higher bandwidth.
utilization than WISPER, but slightly lower than WRR. Although WRR can achieve highest link utilization, it can support the smallest number of users. The difference of link utilization between WISPER and CSDFSC becomes more significant when the number of video users is greater than four. Higher packet loss ratio implies not only the waste of transmission bandwidth, but also the impairments of transmission rate of TCP traffic.

6.5. Fairness

Fairness between best-effort traffic is also an important performance metric of scheduling scheme. We evaluate the fairness performance of CSDFSC in the scenario of two video users and two TCP users. In Figs. 8 and 9, the transmission rates of two TCP users under two different schemes are described. Although the transmission rate of TCP users in WISPER are almost twice that in CSDFSC due to the lack of control on TCP traffic in WISPER, we only consider the relative rate difference since we focus on fairness between TCP flows under same scheme in this paper. The two straight lines, which represent the average transmission rates of two TCP traffic flows, are very close in CSDFSC scheme. However, these two lines are separated far away in WISPER scheme. This means that the CSDFSC scheme can provide better fairness performance for data users than WISPER.

Another difference between two schemes is that the TCP transmission rate is more stable in CSDFSC than that in WISPER. Moreover, TCP traffic flows are under the channel proportion limit in CSDFSC scheme. They do not try to occupy all available bandwidth as they do in WISPER.

6.6. End-to-end delay

Fig. 10 depicts the cumulative distribution function (CDF) of end-to-end packet delay for both video and TCP traffic under CSDFSC and WISPER, respectively. We can see that video users experience shorter delay in CSDFSC, at the expense of longer delay for TCP data users. In
WISPER, the service received by guaranteed (video) traffic has no much difference from the best-effort (TCP) traffic. However, using different queue weights in CSDFSC, guaranteed and best-effort traffic flows are well differentiated. These two kinds of traffic flows will be served at different levels. We can observe that the end-to-end delay of guaranteed traffic flow in CSDFSC is less than that in WISPER.

6.7. Channel utilization vs video users

Next we compare channel utilization of the two schemes when there are only video users in the system. Since the video model used in our simulation has rate limitation-500 Kb/s, the channel utilization increases with the number of video users. As shown in Fig. 11, the channel utilization ratios are very close between two schemes, while CSDFSC achieves higher utilization as WISPER experiences more packet loss. The channel utilization increases with the number of video users since each user needs limited bandwidth. However, when number of video users is more than the x-axis value of right end point, the packet loss rate of video users will be larger than 1%, while implies that the system cannot support them.

6.8. Channel utilization vs TCP users

Finally we compare the channel utilization of the two schemes when there are TCP users only.

TCP can adjust transmission rate according to the available channel bandwidth. Thus small number of TCP users can reach high channel utilization, as shown in Fig. 12. From the same figure, we can also find that channel utilization is quite stable while the number of TCP users increases in CSDFSC, although the utilization drops slightly when number of TCP users is more than 6. On the other hand, the channel utilization varies largely when number of TCP users increases in WISPER. The main cause is also the higher packet loss in WISPER. When the number of TCP users is small, each TCP user should have high transmission rate. Unfortunately, packet loss has more dominate influence on TCP users than on video users, especially on high transmission rate TCP users. Thus small number of TCP users cannot achieve high transmission rate and high channel utilization in WISPER. Furthermore, because of the same reason, the difference of channel utilization between two schemes with TCP users is larger than that with video users, as shown in Figs. 11 and 12. Because TCP users can adjust transmission rate according to available bandwidth, the right end points of curves in Fig. 12 are not the maximum number of TCP users which can be supported.

7. Conclusion

In this paper, we have investigated the appropriate multiple-access method for downlink in high
speed CDMA network. According to our mathematical analysis, TDM instead of CDM should be used in most situations. In other words, one user transmitting in a time-slot at full rate is more efficient than multiple users transmitting simultaneously in a time-slot, and each user has lower transmission rate.

Based on above the conclusion, we propose a new packet scheduling scheme designed particularly for high speed CDMA network, namely Channel States Dependent Fair Service for CDMA (CSDFSC). CSDFSC uses multi-state Markov model to define CDMA wireless link. We compare the performance of CSDFSC with that of WISPER and WRR via ns-2 simulations. The numerical results show that CSDFSC is more efficient in terms of channel capacity, link resource utilization, packet loss, fairness between best-effort traffic flows, end-to-end delay of guaranteed traffic. The good properties in providing fair service to TCP flows shows that CSDFSC emulates GPS quite well. In WISPER, guaranteed traffic experiences lower end-to-end delay than best-effort traffic but the difference is not significant. However, in CSDFSC guaranteed traffic experiences lower end-to-end delay than best-effort traffic and the average delay is much lower than best-effort traffic. From these results, we can draw the conclusion that the queue weight setting in CSDFSC is efficient and guaranteed traffic will obtain better service.

References


Fei Long received his B.E. and M.E. degrees in Electronic Engineering from the Tsinghua University, in 1999 and 2002, respectively. He entered the Information Communication Institute of Singapore, Nanyang Technological University in March 2003 to pursue Ph.D. degree. His research mainly focuses on traffic control for wireless networks.

Gang Feng received his B.E. and M.E. degrees in Electronic Engineering from the University of Electronic Science and Technology of China, in 1986 and 1989, respectively, and the Ph.D. degrees in Information Engineering from The Chinese University of Hong Kong in 1998. He joined the Information Communication Institute of Singapore, Nanyang Technological University in August 1999. Before that, he worked for about one year in the Department of Electronic Engineering, City University of Hong Kong as a postdoc. From 1989 to 1995, he was with the Research Institute of Information Systems, University of Electronic Science and Technology of China. His research interests include routing and performance evaluation for computer networks, Traffic control for TCP/IP networks. Recently, he branches out to work on reliable multicast, QoS provisioning, wireless networks, etc.

Chee Kheong Siew is currently the Head of Information Communication Institute of Singapore (ICIS), School of EEE, Nanyang Technological University. He obtained his B.E. in Electrical Engineering from University of Singapore in 1979 and M.Sc. in Communication Engineering, Imperial College in 1987. After six years in the industry, he joined NTU in 1986 and was appointed as the Head of the Institute in 1996. His current research interests include packet scheduling, traffic shaping, admission control, service curves and admission control for QoS provisioning, congestion control and multipath routing.