Adaptive Multi-User Fair Packet Scheduling in HSDPA Network

Rola Naja[1,2], Jean-Pierre Claudé[1] and Samir Tohmé[1]

[1] PRiSM-CNRS,UMR 8114, University of Versailles- 45 avenue des Etats-Unis , F-78035 Versailles
[2] Faculty of Science–Lebanese University-BP 826-Lebanon
{rola.naja@prism.uvsq.fr,jean-pierre.claude@prism.uvsq.fr,samir.tohme@prism.uvsq.fr}

.Abstract—**Next generation mobile networks are expected to provide seamless personal mobile communications and quality of service (QoS) guaranteed services. In this paper, the focus is on the packet scheduling of data generated by exhaustive applications. The performance of a new packet scheduling called Adaptive Proportional Fair with User Multiplexing (APFUM) is proposed and compared to the proportional Fair scheduling. Bandwidth adaptation enables to increase network capacity while keeping a reasonable mean throughput of adapted users. Performance results show that the proposed algorithm is more effective in reducing the access delay and the rejection probability and in enhancing fairness. This benefit is obtained at the expense of introducing an adaptation rate.**

Keywords- Adaptation, Fairness, scheduling, HSDPA.

I. INTRODUCTION

In order to improve support for high data rate packet switched services, 3GPP developed an evolution of UMTS based on WCDMA known as *High Speed Downlink Packet Access* (HSDPA). HSDPA increases the UTRAN network capacity and the peak data rates and reduces the round trip delay. The key idea of the HSDPA concept is to increase packet data throughput with methods known already from Global System for Mobile Communications (GSM)/Enhanced data rates for global evolution (EDGE) standards, including link adaptation and fast physical layer (L1) retransmission combining.

Architectural changes are performed to bring scheduling and control for link adaptation closer to the air interface, more specifically at the Node B (Figure 1). The Node B estimates the channel quality of each active HSDPA user on the basis of ACK/NACK ratio, Quality of Service (QoS) and HSDPA-specific user feedback. Scheduling and link adaptation are then conducted at a fast pace depending on the active scheduling algorithm and the user prioritization scheme.

The link adaptation is carried out by using the Adaptive Modulation and Coding (AMC) technique. With this technique, users having favorable channel conditions are allocated higher order modulation schemes (16QAM instead of QPSK) [1]. Users will then experience different throughputs at different times depending on their location and instantaneous channel conditions.

To enable a dynamic range of the HSDPA link adaptation and to maintain a good spectral efficiency, a user may simultaneously utilize up to 15 multi-codes in parallel.

HSDPA link is expected to support multimedia applications that generate traffic having diverse QoS requirements. Scheduling algorithms must take into account the delay requirements of packets waiting to be serviced at the Node B, in addition to the need to maximise system throughput. Several packet scheduling algorithms have been proposed for HSDPA.

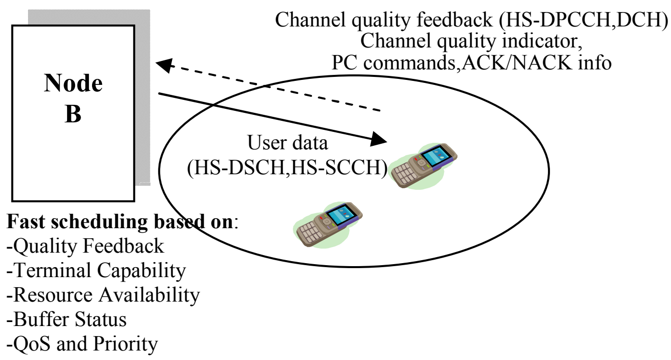

Figure 1:HSDPA Architecture Changes

Maximum Carrier-to-Interface Ratio (Max CIR) ([2], [3]) tends to maximize the system throughput by serving, in every time transmission interval (TTI), the user with the best channel quality. It can be seen that this algorithm provides high system throughput since only those with high current supportable data rates get served. However, this algorithm has an obvious drawback in that it ignores those users with bad channel conditions, which may lead to starvation. The unfairness issue in this algorithm has led to many proposals for scheduling algorithms that try to distribute resources evenly among users. One such proposal is Proportional Fairness (PF) [4]. The PF algorithm tries to increase the degree of fairness among users by selecting those with the largest relative channel quality. Relative channel quality is the instantaneous channel quality condition of the user divided by its current average throughput. Therefore, this algorithm considers not only those users with good channel conditions but also those with low average throughputs by giving them higher priority.

Bonald proposed the Score-Based (SB) algorithm in order to overcome this problem [5]. The SB algorithm computes the rank of each user's current channel condition among the past channel conditions observed over a window of size *W*. Then it selects for transmission the user with the lowest rank. Therefore, this algorithm selects the user whose current channel condition is high relative to its own rate statistics instead of selecting the one whose channel condition is high relative to its average throughput, as in the PF algorithm. SB is slightly more complex in terms of implementation than the PF algorithm. In addition, choosing the size of *W* might be another problem. Small values of *W* might not be appropriate to track the distribution of user channel conditions, while large values of *W* increase the time it takes to find the rank.

Users with Round-Robin (RR) algorithm are served in a cyclic order. This algorithm does not make use of information about the channel quality of users and therefore may offer lower system; it is fair in that it ensures that all users in the system get equal opportunity for transmission regardless of their channel quality conditions. Another scheduling algorithm is Fair Throughput (FT). The goal of this algorithm is to ensure that an equal number of bits is received by each user in the system regardless of their channel quality conditions [6]. This algorithm is fair in terms of the distribution of user throughput since each user gets the same amount of throughput regardless of its channel condition. Similar to RR, it suffers from lower system throughout than fast scheduling algorithms.

In this paper we propose a new algorithm that provides network adaptability and fairness. We strongly believe that in wireless mobile networks, the major issue to be addressed is the high level of fluctuation in resource availability due mainly to mobility. There is a growing consensus that adaptive QoS presents a viable approach to this issue.

Applications need to be adaptive, renegotiate the service and deal with changing conditions by accepting different QoS levels imposed by the network. End systems should be network aware as they take the network status into account and adapt application accordingly. Network must provide application aware-services and handle the adaptive QoS required by these applications.

As a result, the end-to-end QoS provisioning is no longer the sole responsibility of the application. Now, this responsibility is shared between the application and the network. They should together choose the QoS level to deliver multimedia content to a mobile terminal in the most acceptable form given the available resources in the network.

The Adaptive Proportional Fair with User Multiplexing (APFUM) introduces user adaptability to HSDPA network while maintaining fairness among users. In fact, APFUM enhances Proportional Fair algorithm with the introduction of adaptability concept. Users which are rejected with PF due to the lack of codes can be scheduled with APFUM if adaptation is performed.

Our approach consists of the following design principles: 1- Support for soft QoS. We think that the network should provide at least a minimum level of QoS to the users' applications. As for the applications, they should be able to adapt to the changing network conditions. 2- Serve terminals fairly. Terminals with bad channel conditions should be considered for scheduling when their average throughput significantly decreases. 3- Enable code multiplexing. We argue that self-interference can be as high as multi-user interference in a multipath fading environment, and based on this argument the throughput with code division multiplexing can be equal to that with Time Division Multiplexing in a fully loaded system without data rate limitation [7].

This paper is organized as follows. The HSDPA packet scheduler model is described in next section. The physical layer and the Channel quality indicator reporting are presented in section III. A simulation in section IV finalizes the study before concluding the paper.

II. HSDPA Packet Scheduler Model

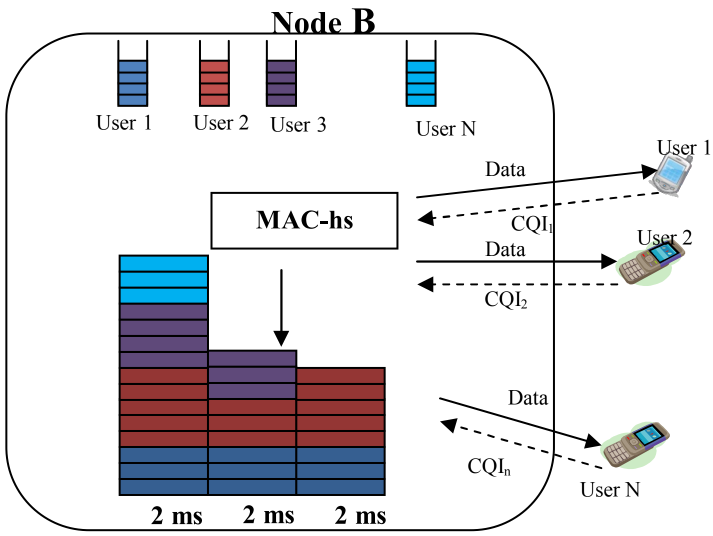

Figure 2:HSDPA Packet Scheduler Model

In this section we briefly describe the packet scheduler model and how it works in HSDPA. The packet scheduler for HSDPA is implemented at the MAC-hs layer of Node B. We assume that each user has one connection request. Thus, a Node B maintains one queue for every user, as shown in (Figure 2). Upon call arrival, the radio link controller (RLC) layer receives traffic in the form of IP packets from higher layers, which are segmented into fixed-size protocol data units (PDUs) or Transport Blocks (TBs). These TBs are stored in the transmission queue of the corresponding user in a first-in first-out fashion. Subsequently, the TBs are transmitted to the appropriate mobile user according to the adopted scheduling discipline. Next, we describe the implemented scheduling algorithm.

A. Scheduling Algorithm

As, our proposed algorithm, APFUM, is based on the Proportional Fair algorithm, we start first by describing the PF strategy. Then we exhibit APFUM algorithm.

Proportional fairness algorithm: To achieve a trade-off between fairness and efficiency, the PF strategy has been proposed. It consists of transmitting to the user with the highest Relative Channel Quality Indicator (RCQI). Relative channel quality is the instantaneous data rate of the user depending on the channel quality condition divided by its current average throughput. The latter is evaluated through an exponential weighted low-pass filter.

The Relative Channel Quality Indicator and average throughput are computed as follows (*Equation 1)*:

$$RCQIi(t) = \frac{Ri(t)}{Ti(t)}$$

$$Ti(t) = (1 - \frac{1}{Tc}) * Ti(t-1) + xi * \frac{1}{Tc} * R'i(t)$$

- *Ri* is the instantaneous rate
- *Ti* is the average throughput, get updated every time transmission interval (TTI) for all active users.
- *R'i* is the actual served rate when the user is scheduled.
- *xi* is equal to *1* if user is scheduled and *0* otherwise.
- *Tc* is a parameter varying between *800* and *1000*.

According to this scheduling scheme, if the average throughput of a user is low, the RCQI could be high and it might be granted the right of transmission even if its current channel condition is not the best.

We combine this method with the multi-user code multiplexing and call the resulting algorithm: the proportional fair with user multiplexing, PFUM. Algorithms outlined in section 1 perform one-by-one scheduling, i.e., only one user is allowed to transmit in any given TTI, even though HSDPA allows the code multiplexing of several users in the same TTI [8]. The primary reasons why many authors have thus far only considered one-by-one packet scheduling schemes are probably their simplicity and their assumption of negligible self-interference. In [7] and [9] it is argued that self-interference can be as high as multi-user interference in a multipath fading environment, and based on this argument the authors show that the throughput with code division multiplexing can be equal to that with Time Division Multiplexing in a fully loaded system without data rate limitation. Consequently with PFUM, we consider that the Node B can schedule at most four users given that the total number of codes does not exceed 15.

B. AFPUM Scheduling Algorithm

The HSDPA operation goes through the following steps:

a.The scheduler in the Node B applies the Adaptive Proportional Fair with user multiplexing, APFUM. Thus, it evaluates for different users the relative channel quality indicator.

b.The Node B determines then a list of four elected users (called *elected_list*), at most, to be served in a particular TTI according to the Proportional Fair with user multiplexing.

c.The Node B then proceeds to check the eligibility of each user: It checks whether the total number of assigned codes does not exceed 15 and whether the minimum inter-TTI interval is respected. The minimum inter-TTI interval defines the distance from the beginning of a TTI to the beginning of the next TTI that can be assigned to the UE. This parameter depends upon the terminal category [10]. Rejected users due to lack of codes are inserted in a list, *rejected_list.*

d.Once a user is selected for transmission, the Node B identifies the necessary HS-DSCH parameters [11]: modulation type, number of PDSCH codes (*nb_assigned_codes*), Transport Block size (*TBS_assigned*). These parameters depend upon the channel conditions of the terminal and its capability limitations. Users with good channel conditions will enjoy potentially higher supportable data rates by using higher modulation and coding rates, whereas users with bad channel conditions will experience lower data rates.

e.If the number of scheduled users is less than *4* and the total assigned codes less than *15*, then the Node B tries to *adapt* bandwidth of elected users in order to schedule other users rejected for lack of codes.

Adaptation proceeds as follows:

- The Node B computes the number of codes requested by the users in the *rejected_list.* It then tries to adapt the assigned TBS of the users in the *elected_list*, starting from the user with the *smallest* value of RCQI.
- Adaptation consists to find the greatest value of TBS, *adapted_TBS*, less than the *TBS_assigned* with number of codes, *adapted_codes*, strictly less than actual value of *nb_assigned_codes*. Consequently, adaptation permits to reduce the TBS of elected users and to release codes in order to schedule rejected users. This process reduces the delay for access to the HSDPA link.
- When the maximum number of permitted users per TTI is reached or when the available number of remaining codes can not be assigned to any user, upgrade operation is performed. If the available number of remaining codes is not zero, then users with highest value of RCQI are upgraded and allocated the initial value of TBS.

f. The terminal sends in the uplink direction an ACK/NACK indicator, depending on the outcome of the CRC check conducted on the HS-DSCH data as well as the monitored channel quality indicator (CQI). The latter is computed as stated in next section.

III. CQI REPORTING

Every *7.5 slots*, each user informs the Node B of its channel quality condition by sending a report known as a channel quality indicator (CQI) on the high-speed Dedicated Physical Control Channel (HS-DPCCH) to the Node B. Based on 3GPP standard [11], the CQI is related to the signal-to-interference ratio (SIR) for a Block Error Rate (BLER) of *0.1* according to the following relations (*Equation 2*):

$$CQI = \begin{pmatrix} 0 & if\ SIR \le -16dB \\ E[\frac{SIR}{1.02} + 16.62] & if -16dB < SIR < 14dB \\ 30 & if\ 14dB \le SIR \end{pmatrix}$$

The SIR (dBm) is computed as follows (*Equation 3)*:

$$SIR = P_{TX} - Ltot - 10\log_{10}(10^{\frac{I\,intra - Ltot}{10}} + 10^{\frac{I\,inter}{10}})$$

$$SIR = P_{TX} - 10\log_{10}(10^{\frac{I\,intra + Inter + Ltot}{10}})$$

Where

- P_{TX} *(dBm)* is the power allocated by the Node B on a channel HS-DSCH.
- *Iintra* is the intra-cell interference (dBm) taking into account the orthogonal factor.
- *Iinter* is the inter-cell interference.
- *Ltot* is the propoagation loss

The relative propagation loss is computed as follows (*Equation 4***)**:

$$Ltot = 10^{\xi/10}.K.r^{-\alpha}$$

Where

- ξ is a Gaussian variable with zero mean and standard deviation σ represented for shadowing effects. We assume that packets are transmitted without errors and that the channel loss caused by multipath fading is neglected.
- *K* is a constant factor.
- *r* is the distance between a mobile and a Node B.
- α is the path loss factor.

IV. SIMULATION RESULTS

In order to compare the different scenarios, the network depicted in Figure 2 was simulated. The adaptive modulation and coding scheme was implemented through the standard tables [11]. We simulated a one-cell case and, for simplicity, did not consider handoff. The cell radius is *500m*. The Node B is located at the cell centre. Users are connected to the Node B on the downlink by an HS-PDSCH, which is the actual physical channel for HSDPA and on the uplink by a HS-DPCCH channel, which is used to send the users' current estimates of their channel conditions to the Node B. The simulation time step is one TTI, which is *2 ms*, and the simulation time is *50000* TTI.

Users of category *8* and of minimum inter-TTI interval *1* are located on rings with radius *10, 50, 150* and 500 meters. The orthogonal factor and the path loss factor are assumed to be *0.1* and *2.7* respectively. We assume that users are receiving exhaustive traffic in a saturated cell. We consider the worst case in which there is always backlogged data ready to be transmitted due to the exhaustive nature of application traffic.

The maximum number of codes and the maximum number of users per TTI are *15* and *4*. For the PF scheduling, the parameter *Tc* is taken equal to *100.*

We conducted simulation runs in order to compare Proportional Fair Scheduling, the PF with user multiplexing and our proposed algorithm APFUM.

The performance parameters are depicted in the following figures. The mean access delay refers to the delay experienced by users before being scheduled. The average throughput increase is the gain achieved by the APFUM relative to the PFUM. The average throughput is computed according to Equation 1. Cell average throughput is the mean useful throughput per cell. The adaptation probability refers to the rate of adaptation. The rejection probability is the probability of not scheduling users when the number of parallel multiplexed users is less than *4*.

Figure 3 illustrates the mean access delay obtained with the three schemes. The mean access delay retrieved with the Proportional Fair and with the PFUM is higher than that with our proposed scheme for all active users. In fact, the APFUM permits to adapt bandwidth of elected users in order to schedule users which were rejected with PF and with PFUM: This leads to enhance the fairness and to lower the mean access delay with APFUM.

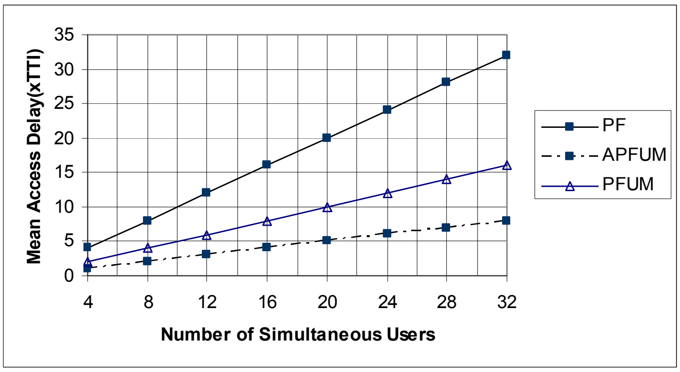

Figure 3: Mean Access Delay for PF, PFUM and APFUM

The rejection and adaptation probability are depicted in Figure 4. It can be seen that the APFUM keeps the probability of rejection *Prej* equal to zero and less than P*rej* with PF. With the PF case, the elected list will contain strictly one user, leading to a rejection probability of *75%.* The rejection probability is *0.5* with the PFUM. In fact, given the users' locations and the category *8*, the CQI reported by terminals will be *26, 22* or *10*. These CQI values correspond to either *10* or *3* codes. Thus, only two users can be scheduled per TTI. With our scheme, we were concerned about scheduling the maximum number of users per TTI: Elected users were adapted in order to satisfy other users.

Figure 4 shows as well the probability of adaptation *Pada* retrieved with the APFUM scheme. It can be seen that this probability is equal to *0,31*.

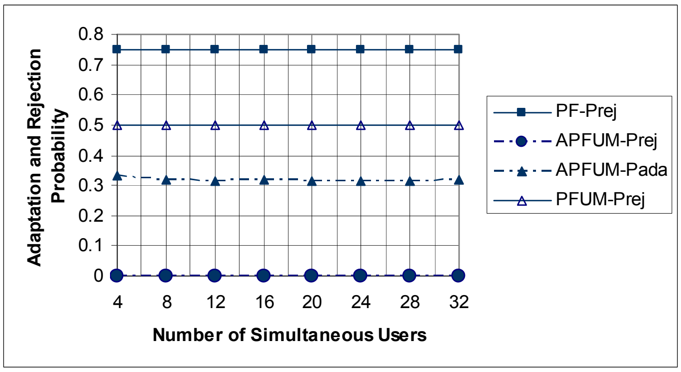

Figure 4: Rejection and Adaptation Probability for PF and APFUM

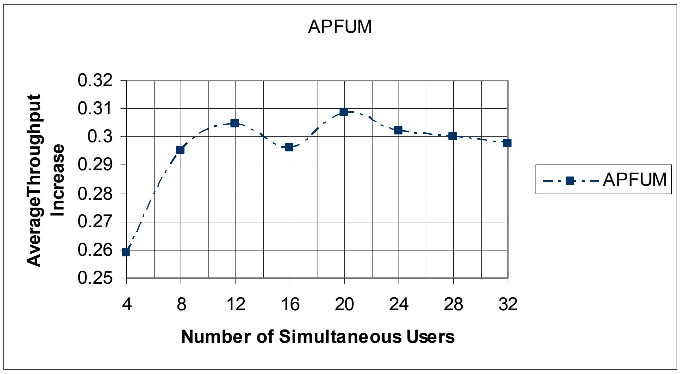

Figure 5: Average Throughput increase for APFUM compared to PF

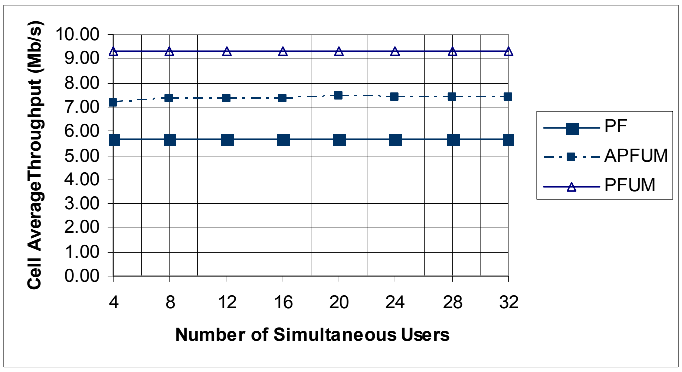

Figure 6: Cell Average Throughput for PF and APFUM

Figure 5 illustrates the average throughput increase of APFUM as compared to the PF scheme. It can be seen that the AFPUM increases the average throughput as it schedules more users per TTI than the PF scheme.

Figure 6 depicts the cell average throughput for each of the three schemes. As expected, APFUM achieves better throughput than the PF. However the PFUM provides higher throughput than with APFUM. In fact APFUM accepts more users at the expense of increasing intra-cell interference.

Consequently, the throughput will be reduced when compared to the APFUM. As a conclusion, the adaptive proportional fair scheduling enhances the fairness in serving users and reduces the mean access delays. This achievement is done at the expense of reducing the average cell throughput. One can see that the throughput achieved with APFUM is better than with PF. Thus there is a trade-off between fairness and cell throughput

V. Conclusion

In this paper, we proposed a scheduling algorithm APFUM that achieves fairness and adaptability in HSDPA networks. We compared the performance of APFUM with the PF and with PFUM algorithms. Simulations runs were conducted for different sets of active users. We showed that APFUM scheme lowers the mean access delay and the rejection probability and increases the average throughput when compared to the PF scheme.

APFUM succeeds to enhance the system fairness by scheduling the maximum number of users at the expense of lowering the cell throughput when compared to the PFUM scheme. This leads us to conclude that APFUM outperforms PF scheme, and that there is a compromise between fairness and throughput achievement.

In next generation of networks, users with different categories of terminals are expected. User categories will affect the coding and modulation scheme as well as the minimum inter-TTI interval. The impact of user categories on the APFUM will be the subject of our next work.

References

[1]. A. Toskala, H. Holma, T. Kolding, F. Frederiksen and P. Mogensen, "WCDMA for UMTS-Radio Access For Third Generation Mobile Communicaions", Wiley and Sons.

[2]. S. Borst, "User-Level Performance of Channel-Aware Scheduling Algorithms in Wireless Data Networks", Proc. IEEE INFOCOM, vol. 1, Mar. 2003, pp. 321–31.

[3].B.Al-Manthari, H. Hassanein, N. Nasser, "Packet Scheduling in 3.5G High-Speed Downlink Packet Access Networks: Breadth and Depth", in IEEE Network, Jan-Fev 2007, Issue 1, pp 41-46.

[4]. A. Jalali, R. Padovani, and R. Pankaj, "Data Throughput of CDMA-HDR a High Efficiency-High Date Rate Personal Communication Wireless System", Proc. IEEE VTC, May 2000, pp. 1854–1858.

[5]. T. Bonald, "A Score-Based Opportunistic Scheduler for Fading Radio Channels", Proc. Euro. Wireless, Sept. 2002, pp. 2244–2248.

[6]. P. Jose, "Packet Scheduling and Quality of Service in HSDPA", Ph.D. dissertation, Aalborg Univ., Oct. 2003.

[7]. Assen Golaup, Oliver Holland, Hamid Aghvami, "Multi-user Packet Scheduling for Delivery of Multimedia Traffic over the HSDPA Link of UMTS", to appear in Wiley's, J.Wirel.Commun. Mobile Comput.

[8]. 3GPP-TR 25.848-Physical Layer Aspects of UTRA High Speed Downlink Packet Access-Release 4.

[9]. J.Cho, D. Hong,"Throughput Comparison of TDM and CDM for Downlink Packet Transmission in CDMA Systems with a Limited Data Rate Set", IEEE Transactions on Wireless Communications 2004-pp.600-611.

[10].3GPP TS 25.306- UE Radio Access capabilities- Release 8.

[11].3GPP-TS 25.214-Physical Layers Procedures (FDD)-Release 8.