Connection Admission Control in UMTS with respect to Network Capacity and Quality of Service

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***Abstract*—The increasing number of subscribers in third generation networks, such as UMTS, will push these networks to their capacity limits within the next years. Especially UMTS tends to become instable when operated close to the capacity limit. Therefore, to meet the subscribers' QoS requirements and maximize the operators' profit, a thoroughly parameterized Connection Admission Control (CAC) is indispensable in order to keep the system in a stable state.**

In this paper, we analyze existing CAC strategies. Furthermore, we offer new strategies to enhance the system's capacity without endangering network stability. This is of great interest for each UMTS network operator since a better CAC algorithm allows a higher network capacity without any additional hardware installation costs.

For this investigation a new simulation platform to evaluate the performance of mobile radio networks at a very detailed level has been designed and implemented. A UMTS Radio Interface Simulator (URIS), which emulates the standardized UMTS protocol stack and the TCP/IP protocol suite, is used together with a Radio Interference Simulation Engine (RISE), which is in charge of calculating the inter- and intra-cell interference, to evaluate the performance of the CAC.

I. Introduction

The delivery of high quality multimedia services to the mobile users is one of the major goals of 3rd generation mobile communication systems. UMTS as one of the 3G systems provides such services. The increasing need for high data rates coming along with these services will result in increasing traffic load for the networks, finally pushing them to their capacity limits. Two capacity limitations can be observed in UMTS. These are the transmission power, either at the Node B or at the User Equipment (UE), and the number of available codes at the Node B.

A well parameterized network will probably not run out of channelization codes but is likely to run out of transmission power resulting in dropped calls at the cell border. To counteract these effects UMTS offers different CAC strategies. This means new connections may be blocked, soft handover requests to a cell may be denied and the data rate of established connections may be downgraded to save transmission power and decrease overall interference.

This paper will examine the performance of different CAC strategies. In section II the CAC concepts are presented. Section III describes the Simulation Environment which is used to validate the concepts and evaluate their performance. For the simulations a UMTS Radio Interface Simulator (URIS) is used that models the radio interface protocols, the TCP/IP protocol stack and the traffic sources. A realistic calculation of the propagation and interence situation of the moving mobiles is done by a simulator named RISE (Radio Interference Simulation Engine). The scenario which is used for performance evaluation is discussed in section IV. Finally, the simulation results are presented in section V and a conclusion is made in section VI.

II. CAC Concepts

In UMTS two limitations have to be evaluated before allowing a connection to be established. The first one is the number of available channelization codes, which are organized in a binary tree. The second limitation is given by the interference, or rather the carrier to interference ratio ($\frac{C}{I}$) for each link. Depending on the amount of traffic in a given cell, either the uplink or the downlink limit the capacity. In case of a low amount of traffic, the uplink $\frac{C}{I}$ is the limitting factor. In case of a higher amount of traffic, the downlink $\frac{C}{I}$ is more critical.

According to [1], these limitations can be checked with 3 different strategies:

- BCAC – Busy Channel Admission Control
- TPAC – Transmitted Power Admission Control
- RPAC – Received Power Admission Control

The first algorithm controls the resources in the code tree, see figure 1, while the second algorithm checks for the total emitted power by the base station the mobile requests a connection to. The last algorithm checks for the total received power at the base station the mobile requests a connection to. All the values are compared to configurable thresholds which will be used to optimize the cell load.

In the URIS, a CAC has been implemented, which covers all of the base algorithms (figure 2). The algorithm which is derived from algorithms which are working in the currently deployed networks, has been enhanced to offer a higher capacity. The given CAC algorithm allows not only tracking of the channelization codes which are in use, but also admission of different spreading factors, which offers evaluation of different services with different QoS demands at the same time.

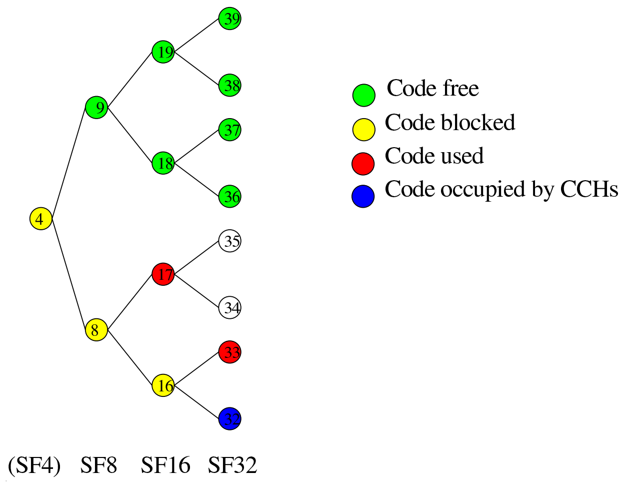

Fig. 1. Sample section of the code tree

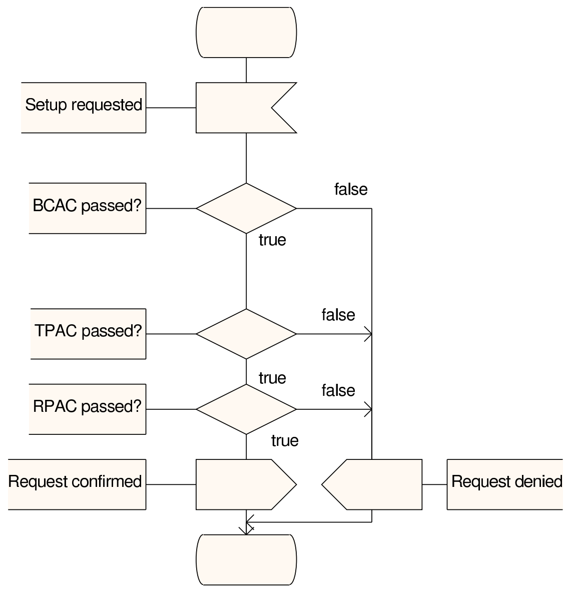

Fig. 2. Implemented algorithms

Evaluation of code tree fragmentation issues are covered as well.

Upgrading in terms of switching to lower spreading factors is triggered by the buffer occupancies of the MAC layer, downgrading can be triggered by different events. These include:

- low buffer occupancy
- CAC demanding a downgrade to stabilize cell load (graceful degradation)
- possibly failing soft handover with low spreading factor

A. *User classes*

During our analysis, we not only focussed on the overall bitrate, but also on different user classes, which have been defined to simulate different QoS parameters for different connections.

The three newly created user classes, share the same minimum (guaranteed) bit rate, but each class will be able to reach a different maximum bit rate:

Different priorities for signalisation of RRC protocol messages and user data (payload) have been chosen so a high amount of user data will not starve the signalling queue.

III. Simulation Environment

In this section the newly designed Simulation Environment (SE) as applied during the development phase of this investigation and finally used for the performance evaluation is presented.

TABLE I

User class definition

User Class	guaranteed data rate	maximum data rate
bronze	= 64 kbps	= 64 kbps
silver	= 64 kbps	$\leq$ 128 kbps
gold	= 64 kbps	$\leq$ 384 kbps

The performance of mobile radio networks of the third generation and beyond is difficult to estimate by analytical methods. This fact results from the complexity of the systems, trying to fully exploit the radio channel. Techniques that try to maximize the capacity of the radio channel like Fast Power Control (FPC), Hybrid Automatic Repeat Request (HARQ) schemes and Link Adaption (LA) normally have to be oversimplified for analytical models. In a simulator most of these techniques can be implemented in an exact manner.

CAC is one of the techniques that try to maximize the capacity at system level. Since UMTS is a complex system with feedback effects on various levels (FPC, CAC, Soft Handover (SHO), ...) the CAC algorithms discussed in this paper have been evaluated by means of simulation.

For this investigation a new stochastic, event driven, dynamic SE to evaluate the performance of mobile radio networks at system level has been designed and implemented in C++. Figure 3 shows the structure of this SE.

Some prerequisites had to be taken into account when designing the new SE. First of all and most important the already existing simulators should be integrated into the new SE. New tools for performance evaluation should be easy to integrate into the SE. Therefore, the SE has been designed to follow a modular plugin-based framework approach.

As visible from figure 3 the SE consists of the Simulation

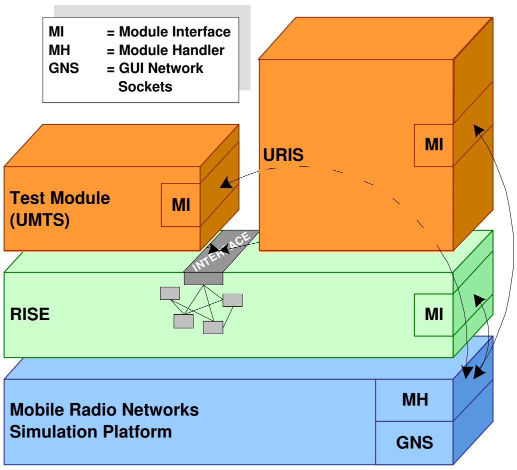

Fig. 3. Simualtion Platform with RISE, URIS and test module

Platform (SP) (at the very bottom of the figure) and various modules (green and orange cuboids) able to perform the necessary tasks for system level simulation of mobile radio networks, e.g. protocol stack emulation and interference calculation.

The SP contains a Module Handler (MH) and GUI Network Sockets (GNS), where the latter allow for transmitting data to Graphical User Interfaces (GUIs) that are connected to the SP via TCP/IP. The above mentioned modules are entirely handled by the MH. It is responsible for setting up and tearing down the modules (e.g. setting correct parameters for the module). The modules themself provide a so-called Module Interface (MI) to interact with the MH. Since modules may depend on other modules a sophisticated versioning system is provided to ensure that all dependencies among the modules are met.

Figure 3 shows three modules, namely RISE, URIS and a UMTS Test Module which have been used for this study. However, even more modules exist by now:

- RISE(calculates interference for various other modules)
- URIS (UMTS protocol stack, uses RISE)
- TETRA (TETRA protocol stack, uses RISE)
- S-WARP (802.11 a/e/g protocol stack, uses RISE)
- PRIME (HiperLAN/2 protocol stack, uses RISE)

Currently, a TCP/UDP module and an IP module is being developed, that might use one of the formerly mentioned protocol stacks.

The new modular design makes it easy to create, integrate and utilize modules for the performance evaluation of mobile radio networks. The collaboration of the modules and exploitation of the provided integration technologies enables communication engineers to leverage module reuse and concentrate on their core competencies to create new communication systems.

Two modules have been used for this investigation: URIS and RISE. A third module, the UMTS Test Module, has been used in the development phase.

The RISE module mainly features calculation of interference for the different stations avaiable in the scenario. This module has interfaces to each of the above listed protocol stacks in order to enable a sophisticated interference calculation for each of the modules.

The UMTS test module served as a URIS surrogate during the development phase. It used the RISE for interference calculation via the same interface than URIS. But the test module was not nearly as complex as URIS. As a result, the development and validation of new algorithms in RISE is a lot easier with this test module.

The UMTS Radio Interface Simulator (URIS), which emulates the standardized UMTS protocol stack and the TCP/IP protocol suite, is used to evaluate the performance of the CAC (figure 4).

URIS has been developed to perform capacity and QoS evaluations for various scenarios. The aim of URIS is to be a bit accurate implementation of the UMTS protocols which have an influence on the performance of the system [2]. Hence URIS uses a protocol emulation for performance evaluation.

Primarily the implemented protocols include the protocol layers of the Access Stratum (Physical (PHY) layer, Medium Access Control (MAC) layer, Radio Link Control (RLC) layer and Packet Data Convergence Protocol (PDCP) layer).

In order to perform realistic simulations in the context of CAC the relevant parts of the Radio Resource Control (RRC) [3] protocol are implemented as well. The complete signalling of management and measurement information in the Control Plane is done by this protocol.

In addition to the UMTS protocols, the user plane of the protocol stack is extended by protocols of the Internet Protocol Suite (Transmission Control Protocol (TCP), User Datagram Protocol (UDP), Internet Protocol (IP)). The actual traffic for the simulations is generated by protocols of the application layer (e.g. Hypertext Transfer Protocol (HTTP), File Transfer Protocol (FTP), Simple Mail Transfer Protocol (SMTP)) with their respective traffic models.

IV. Simulation Scenario

In our simulations, we simulated the download of a document from the Internet by the use of FTP. During the file transfer, the mobiles experienced several hand overs due to their mobility as well as upgrades and downgrades of the bitrate.

Blocked calls and dropped calls have been avoided. The average overall bitrate served as the user satisfaction criterion in this simulation series.

In our scenario, we simulated an environment with 9 base stations, which have been set up in an outdoor environment in a “quasi-smooth” topology. The base stations have been set up in a regular grid, with a distance of 800 meters to 1100 meters. The setup can be seen in figure 5. The mobiles moved with vehicular speed using a Brownian mobility model. Each mobile requested a connection to the network for circuit-switched data transmission to download a file of 1 MB by the use of FTP.

TABLE II

SIMULATION PARAMETERS

Traffic Generator FTP	64kpbs bearer	128kbps bearer	384kpbs bearer
TTI Length [s]	0.02	0.02	0.01
Transport Block Size [bit]	336	336	336
No. of Transport Blocks	0, 1, 2, 4	0, 1, 2, 4, 8	0, 1, 2, 4, 8, 12
Coding	Turbo Coding, coding rate = 1/3		
Max. Data Rate [kbps]	64	128	384
Spreading Factor	32	16	8
MLP		FTP: 3	
RLC Mode		AM	
Max. TCP Segment [byte]		512	
Max. TCP Window [kbyte]		16	
Min./Max. TCP RTO [s]		3 / 64	

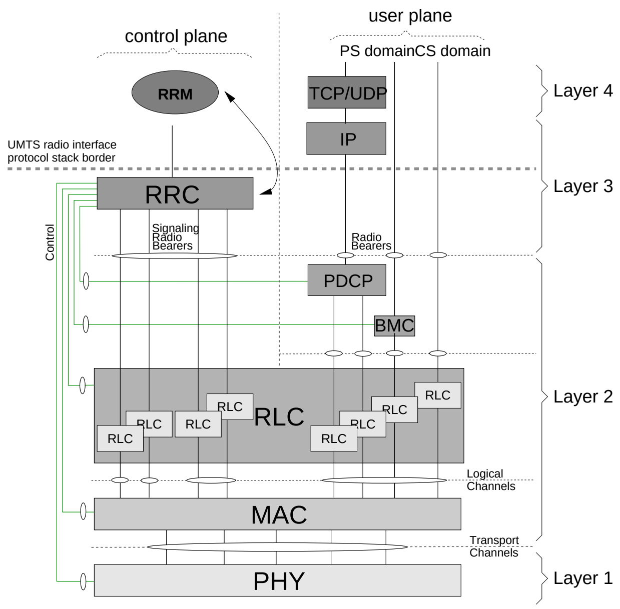

Fig. 4. UMTS protocol stack

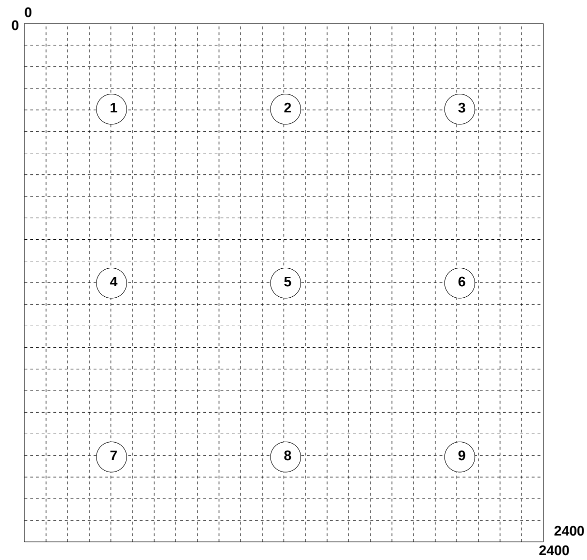

Fig. 5. Outdoor scenario: Base station setup

V. SIMULATION RESULTS

A. Verification of CAC Implementation

In this section, we prove that our implementation of a CAC algorithm is working correctly. We offer different amount of connection requests, starting from a number the scenario is able to bear up to more traffic than the base stations can bear and the CAC has to guarantee the network stability as well as the QoS parameters for each connection.

The peak capacity of the scenario can be roughly determined by assuming that half of the code tree is in use. As we use a spreading factor of 32 for each connection, and 9 base stations are involved, $16 \cdot 9 = 144$ connections can be handled.

To prove the functionality of our CAC implementation, we simulate scenarios with 50, 100, 150 and 200 users and compare the TX powers at the base stations as well as the CIR for each mobile and the BER for each mobile.

As one can see clearly from figure 10, the BER for about 90% of the mobiles is below 0.05, which is acceptable. The low BER for the simulation with 150 mobiles can be explained by the spreading gain. In this simulation, the mobiles are not able to upgrade their connections to a higher bit rate because of the high traffic. The spreading gain is able to reduce the BER in this case.

The gold user throughput on user data level, given in figure 12, is depending on the number of mobiles. In this setup, gold users are only able to receive a throughput higher than 100 kbps if there are no more than 50 mobiles.

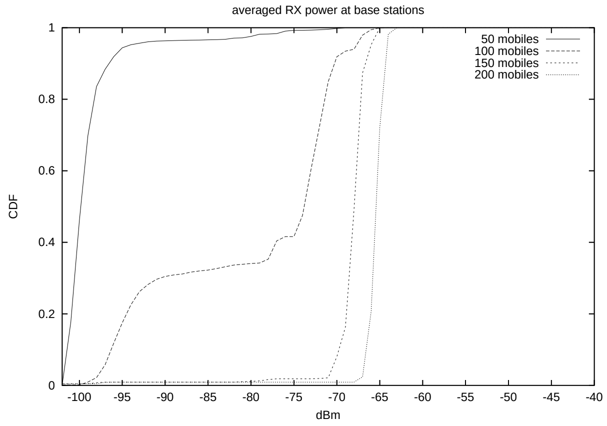

Fig. 6. RX Power at Base Station (Scenario A)

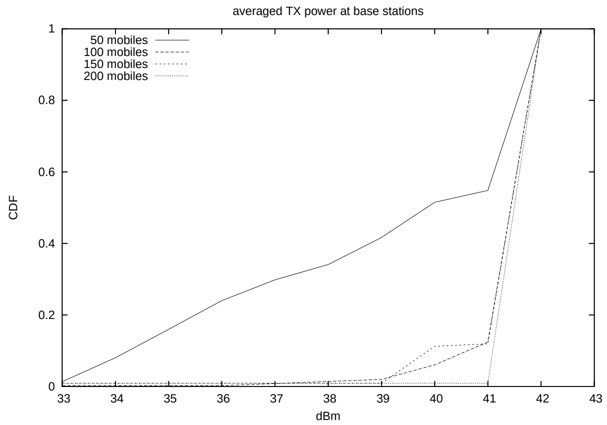

Fig. 7. TX Power at Base Station (Scenario A)

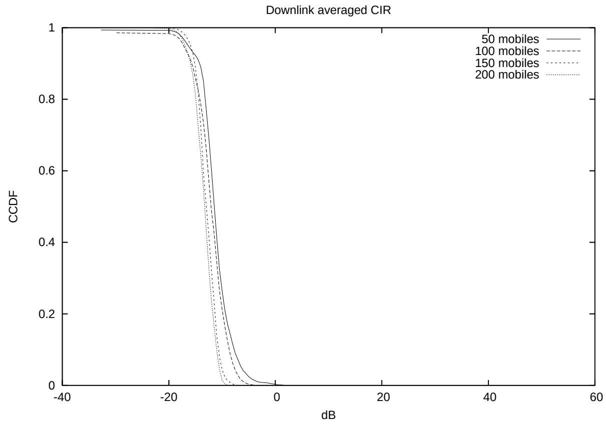

Fig. 8. Downlink CIR for Scenario A

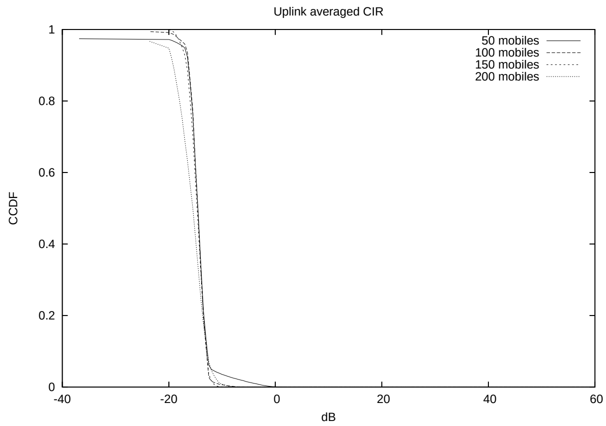

Fig. 9. Uplink CIR for Scenario A

B. User Class Concept

In this section, a simulation with 50 mobiles has been run. We assumed that every mobile should be able to receive its data with 384 kbps. This is the case if no user classes exist.

As one can see from figure 13, the overall bit rate is below 240 kbps. A goal of a network operator could be to increase the bandwidth of certain users without dropping the calls of others. This can be done by allocating spreading factors which correspond to higher bit rates to dedicated users.

C. Different User Class Shares

In this simulation, we simulated the same environment as before, but we changed the user class shares. In this case, the user classes have been split so that 50% of the users only get the lowest bitrate, 30% have access to the medium bitrate and 20% of the mobiles are able to receive data with a bitrate of 384 kbps. The values of this scenario are marked as "50_30_20".

A comparison of the user data bit rates per user class can be seen from figure 16.

Additionally, we analyzed a scenario, where 80% of the users are "bronze users" users, only 5% of the users are "gold users" and the remaining 15% are "silver users". These values are marked as "80_15_5".

The main strategy for this series of simulation was to satisfy the gold users, i.e. providing a high bit rate for these users but without dropping bronze users or sacrifying the network stability.

In the following figures, the results are given:

Comparing figure 17 and figure 18, one can see that the so-called "gold users" are able to reach a higher bit rate caused by the lower spreading factors allocated to them. This is even true although the load control algorithms deprives low spreading factors if the cell load raises above a certain level.

Although the "gold users" are not able to receive the full advantage of the lower spreading factor, both network efficiency and user satisfaction are goals which can be achieved at the same time.

Different user classes are able to receive their data with different bit rates, which can be seen from figure 16. Almost the full bitrate is available for each class, apart from a overhead of about 20%. For the user classes "silver" and "gold", this maximum bit rate cannot be guaranteed, because these classes have to downgrade their connections to lower bitrates to assure network stability.

Comparing the results of this series with section V-B, we can see that the "gold users" gain about 70 kbps and 210 kbps, respectively.

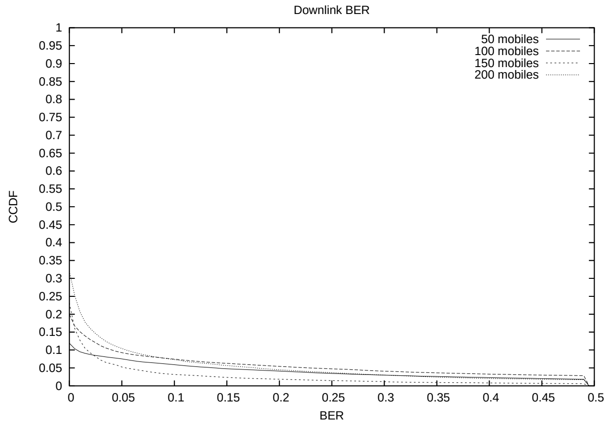

Fig. 10. Downlink BER for Scenario A

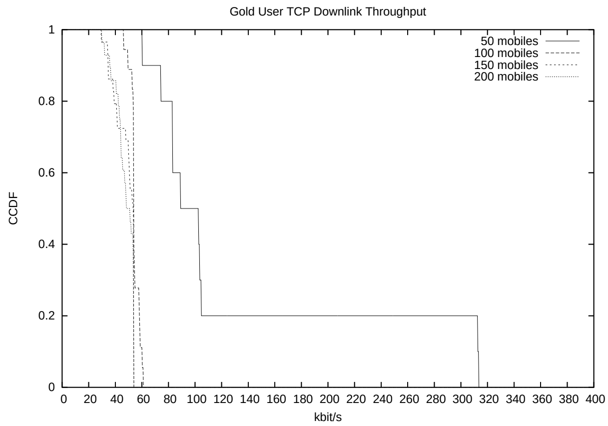

Fig. 11. Gold User TCP throughput for scenario A

VI. Conclusion

One aim of UMTS is the efficient use of the scarce bandwidth resources. To enhance the overall performance several solutions are possible.

We can see that the CAC implementation in URIS is working properly. This implementation is able to protect the network from instability due to overload. The minimum bit rate and BER of the user data connections can be assured as well.

Installing user classes with different maximum bit rates as a QoS parameter can be a used as a network feature because different user classes are able to gain different bit rates according to their maximum spreading factor. The maximum bit rate cannot be guaranteed because it is dependent on the user class composition of the active mobiles.

References

[1] R. Prasad, W. Mohr, and W. Konhäuser, *Third Generation Mobile Communications Systems*, Artech House Publishers, 2000.

[2] 3GPP TS 25.301, "Universal Mobile Telecommunications System (UMTS); Radio Interface Protocol Architecture," Technical Specification V5.3.0, 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, September 2004.

[3] 3GPP TS 125.331, "Radio Resource Control (RRC) protocol specification," June 2003, Version 5.5.0.

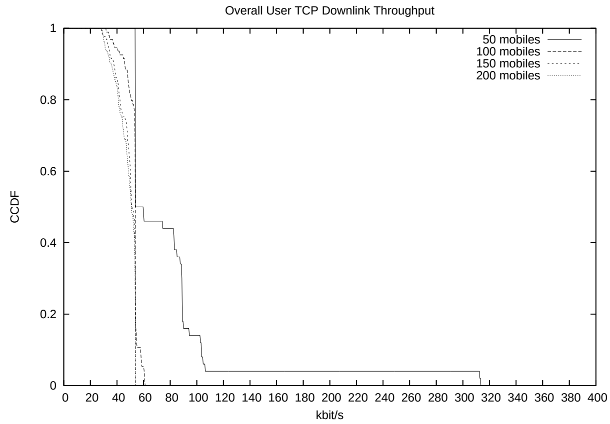

Fig. 12. Overall TCP throughput for scenario A

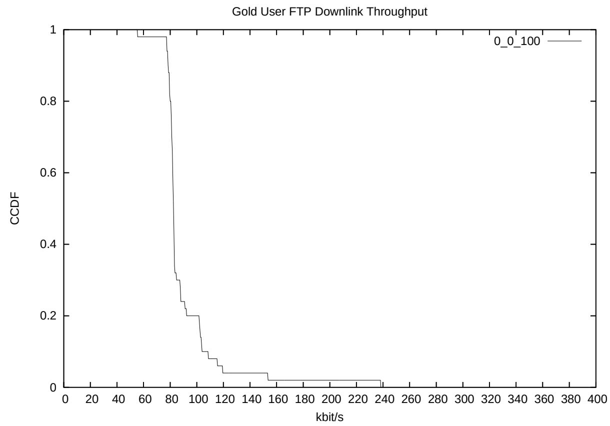

Fig. 13. Overall / Gold User TCP throughput for scenario B

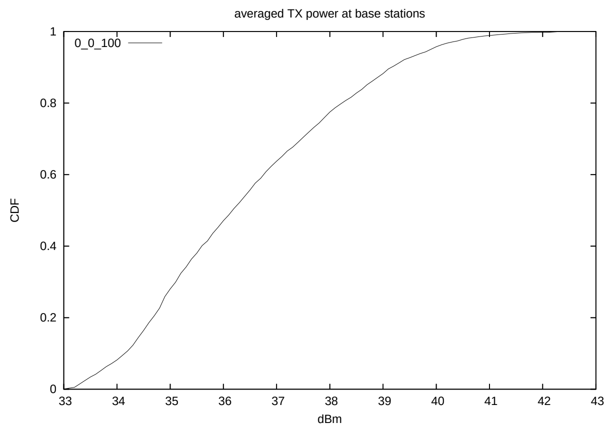

Fig. 14. Tx power at base stations for scenario B

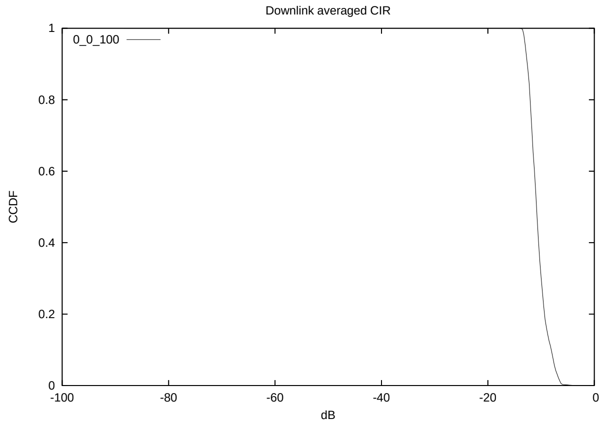

Fig. 15. Downlink CIR for scenario B

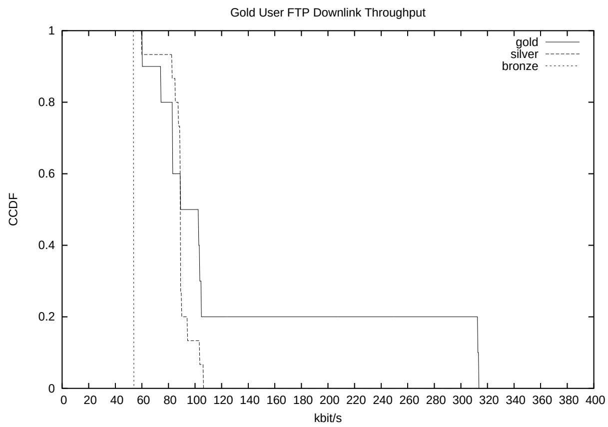

Fig. 16. User Class TCP throughput for scenario C (50_30_20)

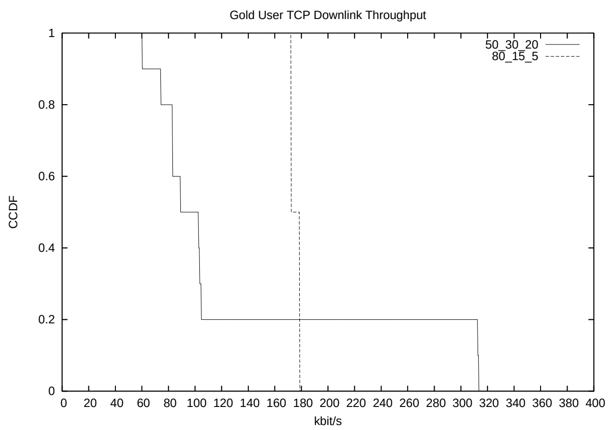

Fig. 17. Gold User TCP throughput for scenario C

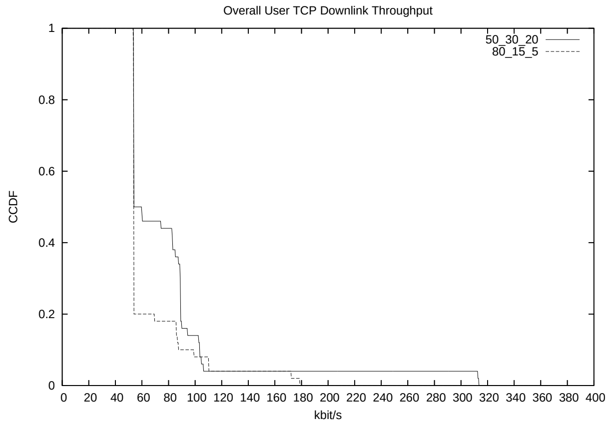

Fig. 18. Overall TCP throughput for scenario C

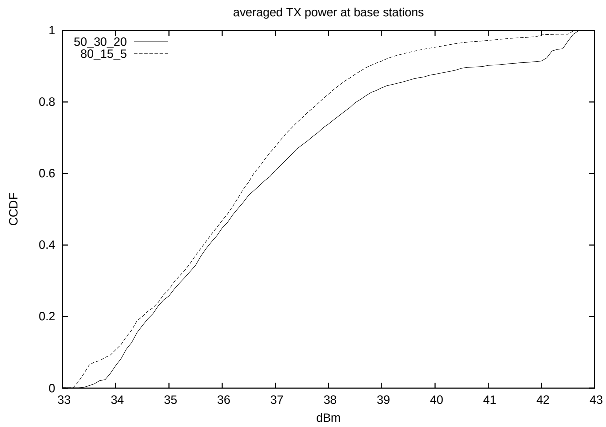

Fig. 19. Tx power at base stations for scenario C

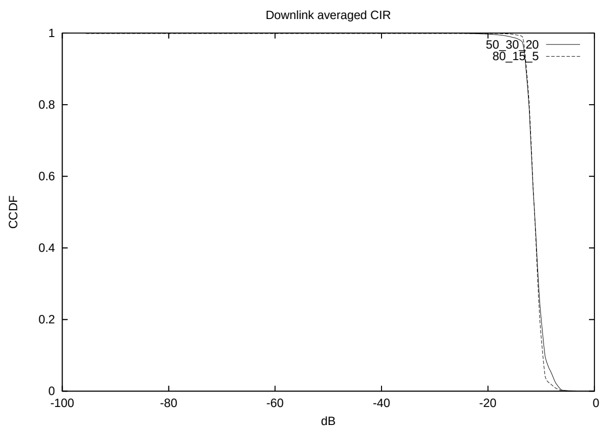

Fig. 20. Downlink CIR for scenario C