On the Capacity of a UTRA-TDD Network with Multiple Services

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***Abstract*— *UMTS Terrestrial Radio Access* (UTRA) in *Time Division Duplex* (TDD) mode is a combination of *Time Division Multiple Access* (TDMA) and *Code Division Multiple Access* (CDMA). Radio resources are disposed in the time as well as in the code domain and allow a variable allocation to either uplink or downlink transmission. With this characteristic, UTRA-TDD allows to flexibly adapt the physical channel and radio frame structure to various kinds of services in different environments and traffic conditions. Depending on the choice of *Radio Resource Management* (RRM) algorithms, this paper focuses on simulative capacity estimation in a typical micro-cellular dense urban environment with a service mix of speech and Internet data traffic.**

I. Introduction

The main advantages from *Universal Mobile Telecommunications System* (UMTS) are expected in the transmission of high bit rate multimedia services to the customer at every location anytime. Hence, an important issue for network design is the development of intelligent *Radio Resource Management* (RRM) algorithms to flexibly and fairly distribute the scarce available bandwidth to a variety of users aiming to provide sufficient *Quality of Service* (QoS) for every connection. The future mobile multimedia services like web browsing, video or audio streaming, games and many more significantly differ from speech in their characteristic. Therefore, the network must be dimensioned concerning both, the expected amount of traffic generated by the customers and the kind of traffic with respect to amount of data, delay constraints, error sensitivity, service asymmetry and jitter.

The UMTS standard now contains three *Code Division Multiple Access* (CDMA) techniques referred as *UMTS Terrestrial Radio Access* (UTRA) modes. The *Frequency Division Duplex* (FDD) mode requires the same amount of bandwidth for both, the forward link or *Downlink* (DL) and reverse link or *Uplink* (UL) direction. Hence, it appears quite inflexible in assigning the required capacity to UL and DL and therewith considering the service's asymmetry. Taking into account, e.g., the amount of data in UL and DL for a typical web browsing session, the DL of a UTRA-FDD network would be the limiting factor and capacity in UL would be rendered useless—a circumstance highly intolerable when considering the enormous cost for spectrum. The *High Speed Downlink Packet Access* (HSDPA) is a means to provide higher capacity in DL by offering higher order modulation schemes. However, this will only be possible for the users in close distance to the base station since this kind of transmission is more susceptible to link quality and therefore requires a higher *Carrier to Interference Ratio* (CIR) [1].

Two *Time Division Duplex* (TDD) modes with different chip rates (*High Chip Rate* (HCR) with 3.84 Mchip/s and *Low Chip Rate* (LCR) with 1.28 Mchip/s per carrier) seem to be more capable to cope with the expected highly asymmetric multimedia traffic [2, 3]. The available bandwidth in UTRA-TDD can be divided in the time domain so that the community of users and their requested service gets the necessary radio resources for a restricted time interval only. Hence, the overall bandwidth can easily be sub-divided depending on the required QoS. In HCR UTRA-TDD each radio frame of fixed length is therefore divided into so-called time slots and a *Time Division Multiple Access* (TDMA) component is added to the physical layer structure. Moreover, with the ability to define a different number of time slots for data transmission in DL and UL, TDD offers also mechanisms to adapt to service asymmetry. In the case of web traffic, longer intervals would be defined for DL transmission and short intervals for UL transmission [4].

This paper introduces and evaluates basic RRM strategies for UTRA-TDD networks operating with HCR transmission. The remainder is organized as follows. In Sec. II. basic RRM mechanisms are discussed and the algorithms applied in the simulation environment are described. Sec. III contains important simulation parameters and models. Results are presented in Sec. IV.

II. Radio Resource Management

This section summarizes the basic RRM algorithms applied for the simulations of UTRA-TDD networks. RRM generally includes the organization and distribution of the system's radio *Resource Units* (RU)—defined as the combination of spreading code, time slot, and frequency—to the requested services. Focusing the HCR mode with a maximum number of 16 codes simultaneously and 15 time slots per radio frame of 10 ms duration on a single frequency with 5 MHz bandwidth, 240 RU are to be considered [5]. Fig. 1 illustrates the organization of the physical layer in case of multiple service transmission and asymmetric radio frame configuration.

Following is a collection of four RRM key issues and the description of how they are realized in the simulations.

1) Channel Allocation: An interference-based dynamic channel allocation scheme is applied in the simulations. Always the *Least Interfered Resource* (LIR) is assigned to a connection. A performance comparison with a static *Fixed Channel Allocation* (FCA) with re-use distance of one and a random assignment of channels to voice connections is presented in [6].

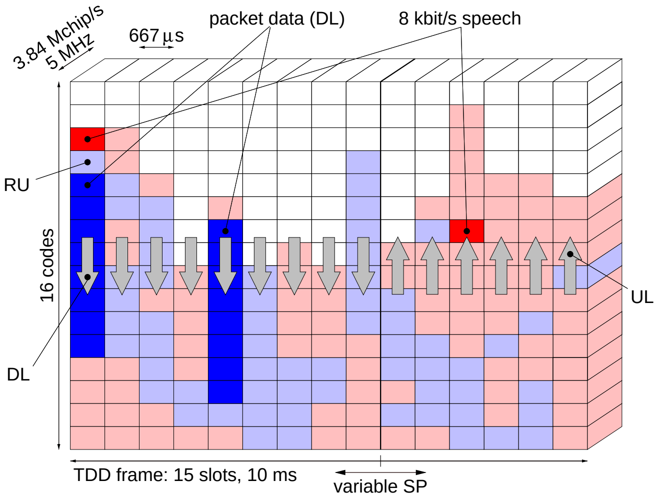

Fig. 1. Radio frame configuration in UTRA-TDD with asymmetric time slot allocation and multiple service transmission

2) Scheduling: Two scheduling strategies are investigated, a *First Come, First Serve* (FCFS) strategy without any packet queueing and a random scheduling algorithm with queueing. Further strategies and their performance in UTRA-TDD are investigated in [7].

The first method really involves no scheduling, but the packet switched services are treated by circuit switched transmission as long as data is present. This includes also connection blocking. Some simulation results for the FCFS strategy are already published in [2]. As a result, the distribution of transmission delays is expected to follow the packet size distribution transformed into the time domain with respect to the physical channel's capacity.

However, the second routine introduces an additional delay due to the waiting times in the scheduler queues. All packet data connections will be admitted to the system and put into waiting queues. As long as there are resources available, user data flows are scheduled by randomly choosing them subsequently out of the list of queued requests. If there are not enough free resources left, all the other users remain waiting until the next scheduling phase will be initiated. The scheduling interval is set to 5 frames, e.g. 50 ms. After this interval all the scheduled connections will be suspended and put back into the list of waiting requests. A new random determination of connections to be serviced in the next phase is initiated afterwards.

3) Call Admission Control: There is no *Call Admission Control* (CAC) scheme included in the simulations. All connections will be admitted to the network with a resource-availability constraint for circuit switched traffic. That means in our case that speech connections require at least one RU in DL and one RU in UL. A speech connection will be blocked otherwise.

In case of FCFS scheduling without any waiting buffers, also packet data connections will be blocked by the network if there are not enough resources available. With the introduction of waiting queues, no packet data service request is rejected. All are admitted to the network.

4) Service Priorities: A speech service priority is applied in the simulations with random scheduling. If there are no free RU in DL, one packet switched service connection will be suspended and the RU of this connection are freed for the usage by circuit switched speech. Therefore, speech service blocking probability is determined only by physical capacity restrictions, e.g. the minimum number of RU available in either link direction. If an asymmetric time frame with 10 slots for DL transmission and the remaining 5 slots for UL transmission is chosen, the system's capacity in terms of speech connections is hard limited by 80 available RU in UL.

III. Simulation Parameters

A small Manhattan-like urban model with 12 base stations based on [8] is generated for simulation purpose. Nine squared blocks with size 400 m×400 m and streets of width 30 m in between give a simulation area of 1.742 km^2. Fig. 2 illustrates the scenario settings.

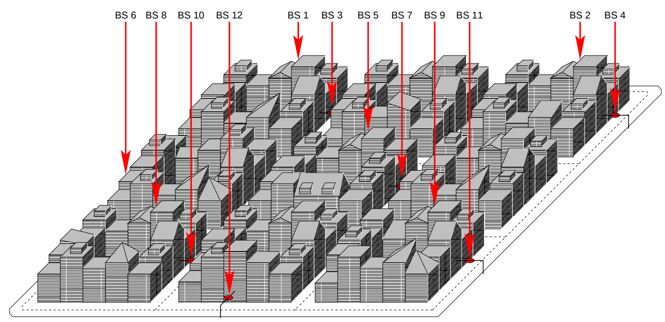

Fig. 2. Manhattan-like urban simulation scenario

Base stations are located in the center of the streets below roof tops of the surrounding buildings. Propagation modeling along the streets is based on Berg's model with few modifications [8, 9]. User's mobility along the defined road map also conforms to [8]. Turning probability at intersections equals 1/4 in either direction and with probability 1/2 the preferred movement is straight forward.

In CDMA systems, *Power Control* (PC) algorithms are mandatory for the operation of radio communications [1]. In addition to joint detection as applied in UTRA-TDD which removes intra-cell interference, PC can be seen as a method to reduce inter-cell interference in the network. Since no clustering and thus channel grouping is necessary in CDMA systems, PC significantly improves the system capacity. As a result, the TD-CDMA network seems to be no longer limited by intra-system interference but by the physical capacity in terms of available RU in either UL or DL direction.

Two different kinds of services are modeled within the simulations. As an example for circuit switched traffic the speech service is chosen, whereas Internet data is transmitted as packet switched traffic via an *Unconstrained Delay Data* (UDD) radio bearer. See [2, 8] for more details on traffic modeling and service parameters.

Circuit Switched Traffic Source: Circuit switched speech connections are characterized by Poisson arrivals of calls with negative exponentially distributed call durations with mean $T_\mathrm{D} = 120\,\mathrm{s}$. To provide the speech service at *Adaptive Multi Rate* (AMR) coder level with 7.95 kbps, two RU with *Spreading Factor* (SF) 16 are needed—one in UL and one in DL.

A speech connection is alternating between active and silent period times (see Fig. 3). Both times are negative exponentially distributed with mean $T_\mathrm{active} = T_\mathrm{silent} = 3\,\mathrm{s}$. This results in a channel activity of 50 %.

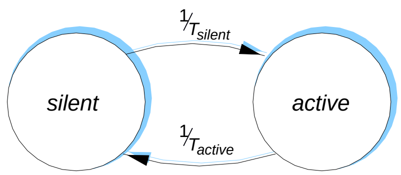

Fig. 3. Traffic model for speech service

Packet Switched Traffic Source: Web browsing packet data traffic is characterized to appear during so-called browsing sessions as illustrated in Fig. 4. Session call arrivals are modeled by a Poisson process similar to the arrivals of speech service calls. During a session, several packet call requests will be initiated by the user. The number of packet calls within a session is a geometrically distributed random variable with an average value of five. This means, an average of five packet calls according to five page requests will appear within one session.

The reading time between consecutive packet calls T_reading is a negative exponentially distributed random variable with mean 12 s. The reading time starts after the packet is completely received and afterwards, a new packet call is initiated.

Each packet call contains single packets representing the main or inline objects of the requested web page. The number of packets per packet call is a geometrically distributed random variable with mean 25. The average amount of data per single packet is 480 byte data which is also negative exponentially distributed. A minimum packet size of 50 byte accounts for header and signaling overhead.

A single RU is able to carry a certain amount of data. Following the UMTS specifications [5], this will be approximately 29 byte when spreading with SF 16 is used and some bits are spent for protocol overhead. Resulting is a bit rate of $\frac{29 \cdot 8\,\mathrm{bit}}{10\,\mathrm{ms}} = 23.2\,\mathrm{kbps}$ per allocated RU. UMTS defines multiple radio bearers with a wide range of user bit rates and certain requirements for transmission delay. For web traffic, the UDD bearer with 384 kbps is chosen for transmission.

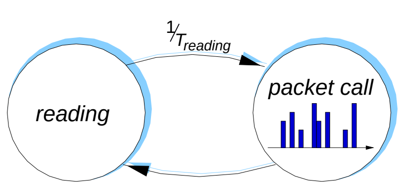

Fig. 4. Traffic model for packet data services

IV. Results

This section contains the results of two different simulation runs. The first is an evaluation of FCFS scheduling without waiting queues. A comparison between two different physical channel capacities is drawn. The latter introduces the described random scheduling mechanism and speech service priority to the network. Results are compared to the FCFS scheduling with data channels containing a pool of 4 RU.

For the following simulations, a service mix with speech and packet data web traffic is chosen. The PC target CIR level for speech is set to $-10\,\mathrm{dB}$ and for packet data at $-7\,\mathrm{dB}$ in order to consider the lower *Bit Error Ratio* (BER) required from data services in comparison to speech. The traffic mix contains 2/3 of speech and 1/3 packet data traffic only in DL direction. This does not correspond to the number of users served but to the amount of RU capacity in use. Overall offered traffic is varied from 40 % to 90 %, i.e. this average amount of the total available 240 RU is occupied by active connections. Having half of the speech traffic in UL and the remaining traffic (speech and web) in DL, a time slot allocation ratio of 2:1 in favor of DL transmission would be optimal. Therefore, the radio frame is configured with 10 time slots for DL and 5 time slots for UL transmission.

Service Mixture with FCFS Scheduling and Hard Blocking

Two kinds of simulations are performed, a fixed assignment of 2 RU and 4 RU per connection, respectively. SF 16 is chosen in both cases. Blocking probability is obviously larger in the case where 4 RU are searched for. It is more likely to find an available channel with only 2 RU. Thus, the blocking rates increase with increasing number of RU required for the channel. Fig. 5 illustrates the according *Satisfied User Criteria* (SUC), i.e. the percentage of satisfied users according to [8] for these two kinds of allocation policies. Due to the higher blocking probability if 4 RU channels are required, the overall SUC is significantly lower in this case.

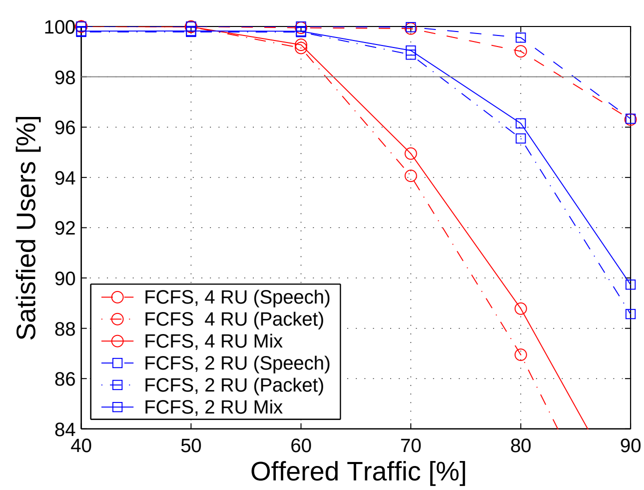

Fig. 5. SUC for speech and data users with FCFS scheduling and hard blocking over differently configured physical channel

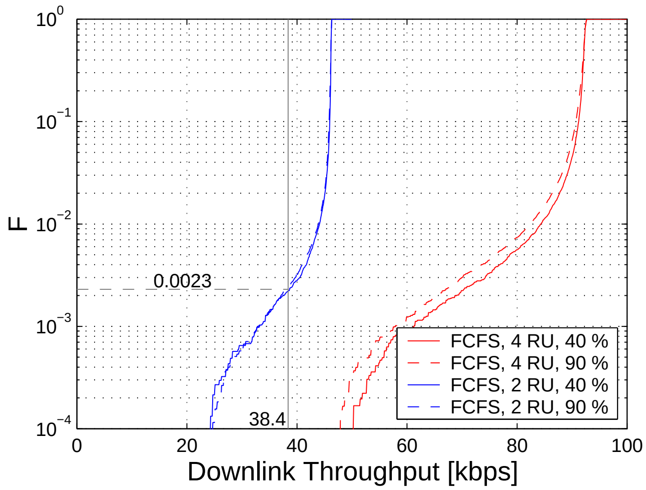

Fig. 6. Distribution of DL throughput with 2 RU and 4 RU physical channels

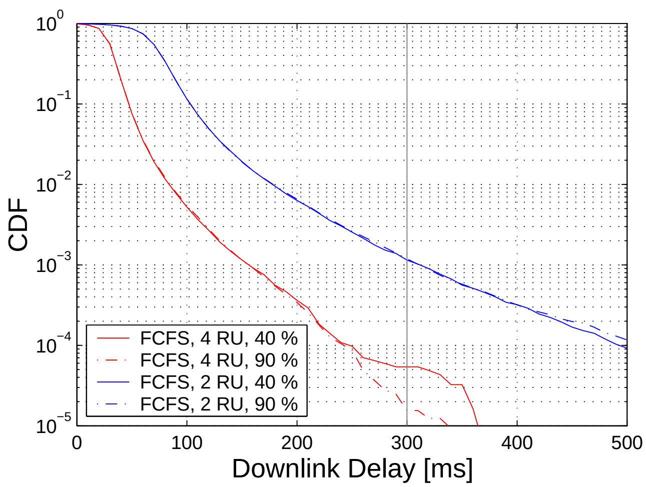

Fig. 7. Complementary distribution function of DL packet delay without waiting queues at different loads

In addition to call blocking, SUC also considers QoS constraints in terms of active session throughput provided by the network. According to the SUC, a packet data user is satisfied if he gets at least 10 % of the required throughput as an active session throughput. For the UDD bearer service with 384 kbps, an active session throughput of 38.4 kbps means satisfactory QoS. The distribution functions of throughput in DL presented in Fig. 6 are obtained with a scheduler parameterization to exclusively allocate a 92.8 kbps (4 RU) channel to a packet data connection, or a 46.4 kbps (2 RU) channel, respectively. Lower throughput values result only from not completely filled RU combinations (remaining packet size is less than 116 or 58 byte depending on the number of RU assigned to a connection). With a minimum packet size of 50 byte, the worst case packet sizes are $4 \times 29 + 1 = 117$ byte for channels with 4 RU (resulting in the minimum throughput of 46.8 kbps) and $2 \times 29 + 1 = 59$ byte for channels with 2 RU (resulting in the minimum throughput of 23.6 kbps). The effect that the higher the channel capacity the higher the throughput is clearly shown. When 4 RU are pooled together to build up one channel every connection will experience a sufficient active session throughput. With only 2 RU allocated to a single connection, throughput drops below the required 10 % threshold of 38.4 kbps for 0.23 % of the sessions.

A comparable behavior can be seen from the delay distributions as presented in Fig. 7. The delay measurements double when the channel capacity is decreased from a 92.8 kbps (4 RU) to a 46.4 kbps (2 RU) channel. As the channel capacity is fixed during the active session times, i.e. during the packet calls, the delay distributions are just a transformation of the packet size distribution into the time domain. Some additional effects due to the UTRA-TDD frame length of 10 ms and some not completely used transmission channels modify the negative exponential appearance.

Traffic Mixture with Queueing and Random Scheduling

The scheduling strategy bases on a random selection of waiting connections to be set active for the duration of the next scheduling interval. At the beginning of each scheduling operation, all connections are suspended and put into the waiting queues. Hence, there's no correlation between the queueing for the next scheduling period and the queued connections during the last period. The active session throughput for the connections is now dependent on transmission time caused by the limited channel capacity and waiting delays in the scheduler queues.

Fig. 8 shows the SUC for speech and packet data users for different traffic conditions. As speech users are prioritized, SUC for speech is only affected by call blocking and therefore corresponds to the Erlang-B blocking. The curve for packet data without queueing corresponds to the curve in Fig. 5 where 4 RU are allocated for each packet data connection. Unsatisfied users are caused by hard blocking of packet connections. If all users are considered for queueing, there are no more blocked connections and SUC improves significantly.

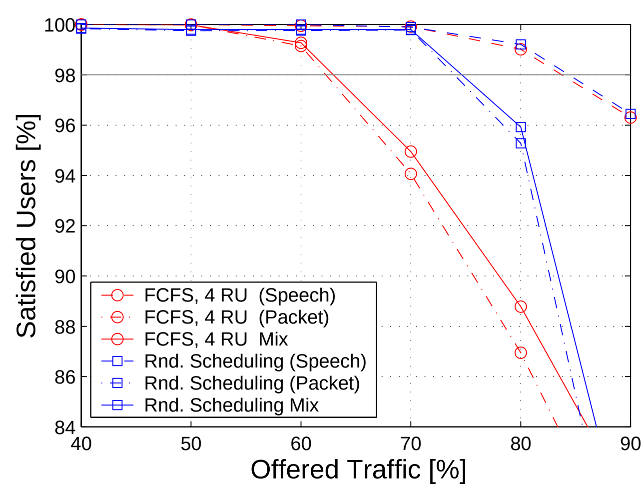

Fig. 8. SUC for speech and data users in a mixed service environment with random scheduling for packet data

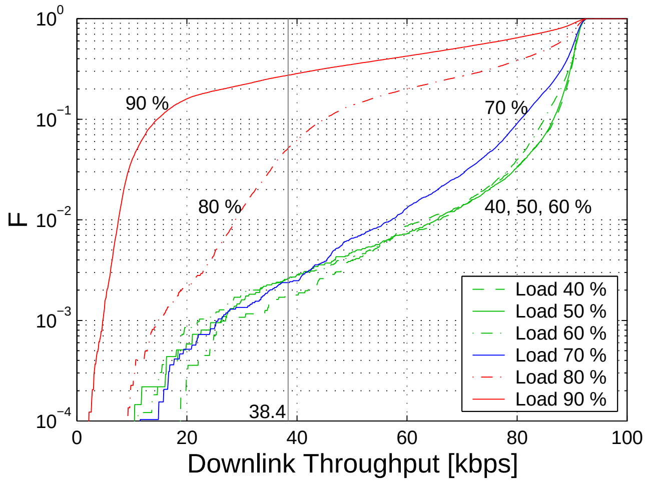

Fig. 9. Distribution of DL throughput with random scheduling algorithm

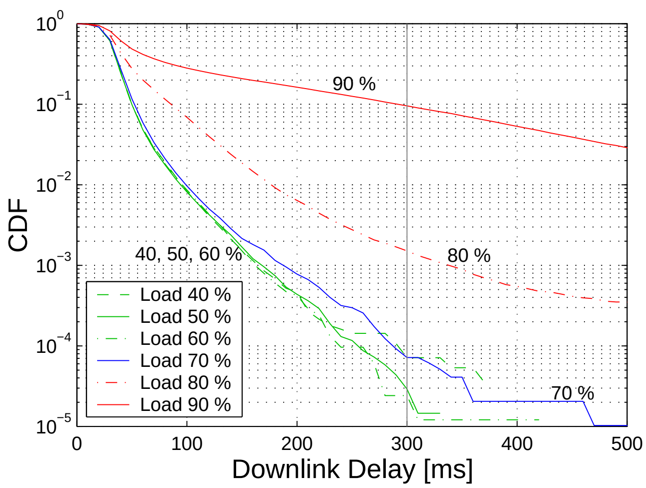

Fig. 10. Complementary distribution function of DL packet delay with random scheduling at different loads

However, when the offered traffic increases, the average throughput per session decreases as illustrated in Fig. 9. At 80 % offered traffic there are about 5 % sessions with unsatisfactory throughput. This corresponds to 5 % unsatisfied packet data users in the statistics in Fig. 8. At 90 % offered traffic, more than 25 % of the users experience an active session throughput below the requirement and remain unsatisfied. The network capacity is defined as the load where any of the services has at least 98 % satisfied users. This is reached at an offered traffic of about 75 %.

A similar tendency can be seen in the packet delay distributions presented in Fig. 10. If the offered traffic increases above 70 %, the packet delay increases according to the decreasing throughput. Whereas the distributions for offered traffics up to 70 % of the system capacity are approximately the same, a considerable fraction of waiting delays appears in addition to the transmission delay for higher traffic. Concerning the 300 ms delay bound as required by *Long Constrained Delay* (LCD) data services, most real-time services might also be transmitted via the UDD bearer. Only 0.01 % of the data packets are too large or experience a longer waiting delay than to be transmitted within the 300 ms interval. However, if scheduling delays come into effect, the overall transmission delays increase rapidly and real-time data transmission will need some sort of priority. Furthermore, CAC strategies are necessary to prevent system instabilities and a quick degradation of the SUC. From the states of the waiting queues a practical threshold for CAC can be derived.

V. Conclusion

This paper contains simulative results on the capacity of a HCR UTRA-TDD network providing speech and Internet data services. Two methods for the improvement of overall system capacity are investigated. The first method is the allocation of low capacity physical channels to packet data connections and with that reducing connection blocking probability. In this case, the active session throughput is decreased as trade-off. Secondly, with the application of simple scheduling mechanisms and the introduction of waiting queues a higher system capacity can be achieved. Due to the lack of CAC strategies, SUC decreases rapidly if the waiting delays in the scheduler queues increase. Nevertheless, in an optimal HCR TDD micro-cellular network 75 % of the system capacity (in terms of available radio resources) can be used. With a traffic mixture of ⅔ speech and ⅓ packet data, 50 % of the capacity go to speech and 25 % go to packet data. This corresponds to approximately 60 voice connections and about 140 web sessions in parallel per cell.

Acknowledgment

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