3GPP TSG RAN WG1 **TSGR1#15(00)1094**
Meeting #15
22-25th, August, 2000
Berlin, Germany

Agenda item:

Source: Ericsson, Motorola, Nokia

Title: Common HSDPA system simulation assumptions

Document for: Discussion and approval

Summary:

In TSG-R1 meeting #14, both link and system level simulation assumptions for High Speed Downlink Packet Access (HSDPA) studies were presented. After discussion it was agreed that joint contributions presenting link and system level simulation assumptions applicable to different simulation platforms should be produced based on the earlier contributions.

This contribution presents assumptions for system level simulations. They are proposed to be used to verify the potential performance gains due to the currently proposed HSDPA features, such as adaptive modulation and coding scheme (AMCS), fast Hybrid ARQ, and fast cell site selection (FCSS).

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1. INTRODUCTION

In TSG-R1 meeting #14 both link and system level simulation assumptions for High Speed Downlink Packet Access (HSDPA) studies were presented [1, 2, 3]. Two contributions presenting link and system level simulation assumptions applicable to different simulation platforms were agreed to be produced based on [1], [2], and [3]. This contribution presents assumptions for system level simulations.

2. SCOPE AND OBJECTIVES OF THE SIMULATIONS

The scope of this document is to propose a set of definitions and assumptions on which HSDPA simulations can be based. The initial objective of such system simulations should be to illustrate/verify the potential performance gains due to the currently proposed HSDPA features, such as adaptive modulation and coding scheme (AMCS), fast Hybrid ARQ, and fast cell site selection (FCSS).

3. COMMON SYSTEM LEVEL SIMULATION ASSUMPTIONS

As system level simulation tools and platforms differ between companies very detailed specification of common simulation assumptions is not feasible. Yet, basic simulation assumptions and parameters should be harmonized as proposed in the subsequent chapters.

Various kinds of system performance evaluation methods may be used. In Annex 1, two different methods are outlined. They should be seen as examples and therefore other methods can be used.

3.1 Basic system level parameters

The basic system level simulation parameters are listed in the Table 1.

Table 1. Basic system level simulation assumptions.

Parameter	Explanation/Assumption	Comments
Cellular layout	Hexagonal grid, 3-sector sites	Provide your cell layout picture
Site to Site distance	2800 m	
Antenna pattern	As proposed in [2]	Only horizontal pattern specified
Propagation model	$L = 128.1 + 37.6\ Log_{10}(R)$	R in kilometres
CPICH power	-10 dB	
Other common channels	- 10 dB	
Power allocated to HSDPA transmission, including associated signaling	Max. 80 % of total cell power	
Slow fading	As modeled in UMTS 30.03, B 1.4.1.4	
Std. deviation of slow fading	8 dB	
Correlation between sectors	1.0	
Correlation between sites	0.5	
Correlation distance of slow fading	50 m	
Carrier frequency	2000 MHz	
BS antenna gain	14 dB	
UE antenna gain	0 dBi	
UE noise figure	9 dB	
Max. # of retransmissions	Specify the value used	Retransmissions by fast HARQ
Fast HARQ scheme	Chase combining	For initial evaluation of fast HARQ
BS total Tx power	Up to 44 dBm	
Active set size	3	Maximum size
Specify Fast Fading model	Jakes spectrum	Generated e.g. by Jakes or Filter approach

3.2 Data traffic model

The described data-traffic model simulates bursty web traffic. The parameters of the model are based on [4] but have been tailored to reduce simulation run time by decreasing the number of UEs required to achieve peak system loading. The main modification is to reduce the reading time between packet calls. In addition, TCP/IP rate adaptation mechanisms have been included to pace the packet arrival process of packets within a packet call.

The model assumes that all UEs dropped are in an active packet session. These packet sessions consist of multiple packet calls representing Web downloads or other similar activities. Each packet call size is modeled by a truncated Pareto distributed random variable producing a mean packet call size of 25 Kbytes. Each packet call is separated by a reading time. The reading time is modeled by a Geometrically distributed random variable with a mean of 5 seconds. The reading time begins when the UE has received the entire packet call.

Each packet call is segmented into individual packets. The time interval between two consecutive packets can be modeled in two ways, as an open loop process or as a closed loop process. The open loop process models the timer interval as a geometrically distributed random variable. Specifically, the mean packet inter-arrival time will be set to the ratio of the maximum packet size divided by the peak link speed. The closed loop model will incorporate the “slow-start” TCP/IP rate control mechanism for pacing packet traffic. Slow-start will be implemented as described in [5]. A total round trip network delay of 100 ms will be assumed for TCP ACK feedback.

The fundamentals of the data-traffic model are captured in Table 2.

Table 2. Data-traffic model parameters.

Process	**Random Variable**	**Parameters**
Packet Calls Size	Pareto with cutoff	=1.1, k=4.5 Kbytes, m=2 Mbytes, = 25 Kbytes
Time Between Packet Calls	Geometric	= 5 seconds
Packet Size	Segmented based on MTU size	(e.g. 1500 octets)
Packets per Packet Call	Deterministic	Based on Packet Call Size and Packet MTU
Packet Inter-arrival Time (open- loop)	Geometric	= MTU size /peak link speed (e.g. [1500 octets * 8] /2 Mbps = 6 ms)
Packet Inter-arrival Time (closed-loop)	Deterministic	TCP/IP Slow Start (Fixed Network Delay of 100 ms)

3.3 UE mobility model

A static or dynamic UE mobility model can be used. Both fixed UE speed or a speed distribution may be used. In the latter case the speed distribution given in Figure 1 shall be used, see also Table 3. A speed is assigned to each user at the beginning of the simulation and will not be changed during the simulation. Stationary UEs signal paths will be Rician faded with K factor of 12dB and 2Hz Doppler spread.

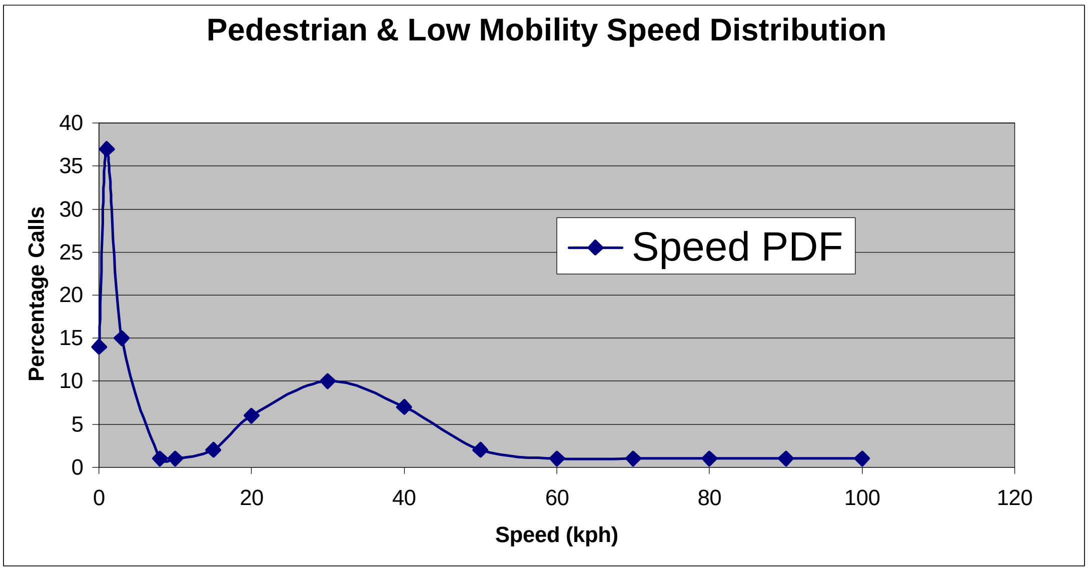

Figure 1. Pedestrian and low mobility speed distribution.

Table 3. Speed distribution

Speed (kph)	0	1	3	8	10	15	20	30	40	50	60	70	80	90	100
Percentage	14	37	15	1	1	2	6	10	7	2	1	1	1	1	1

3.4 Packet scheduler

Multiple types of packet schedulers may be simulated. However, initial results may be provided for the two simple schedulers provided below that bound performance. The first scheduler (C/I based) provides maximum system capacity at the expense of fairness, because all frames can be allocated to a single user with good channel conditions. The Round Robin (RR) scheduler provides a more fair sharing of resources (frames) at the expense of a lower system capacity.

Both scheduling methods obey the following rules:

- An ideal scheduling interval is assumed and scheduling is performed on a frame by frame basis.
- The “frame” is defined by the HSDPA concept, e.g. 0.67ms (1 slot), 3.33ms (5 slots) , or 10 ms (15 slots).
- A queue is 'non-empty' if it contains at least 1 octet of information.
- Packets received in error are explicitly rescheduled after the ARQ feedback delay consistent with the HSDPA definition.
- A high priority queue is maintained to expedite the retransmission of failed packet transmission attempts. Entry into the high priority queue will be delayed by a specified time interval (e.g. 5 frame intervals) to allow for scheduler flexibility[1]. If the packet in the high-priority queue is not rescheduled after a second time interval (e.g. 10 frame intervals) it is dropped.
- Packets from the low priority queue may only be transmitted after the high-priority queue is empty.
- Transmission during a frame cannot be aborted or pre-empted for any reason

The C/I scheduler obeys the following additional rules:

- At the scheduling instant, all non-empty source queues are rank ordered by C/I for transmission during a frame.
- The scheduler may continue to transfer data to the UE with the highest C/I until the queue of that UE is empty, data arrives for another UE with higher C/I, or a retransmission is scheduled taking higher priority.
- Both high and low priority queues are ranked by C/I.

[1] The delayed entry into the high priority queue can be used to reduce compulsory retransmission of a single packet. A fast retransmission mechanism, such as N-channel stop-and-wait ARQ, would provide one packet to the high priority queue if the delayed entry mechanism were not provided. As a result, this single packet would be retried in lieu of all other packets regardless of the channel conditions. Note that the case when retransmitted packets always have priority over new transmissions is included in this description as a special case.

The RR scheduler obeys the following rules:

- At the scheduling instant, non-empty source queues are serviced in a round-robin fashion.
- All non-empty source queues must be serviced before re-servicing a user.
- Therefore, the next frame cannot service the same user as the current frame unless there is only one non-empty source queue.
- The scheduler is allowed to group packets from the selected source queue within the frame.

3.5 Outputs and performance metrics

The following suggested performance metrics for both the entire system and the center site taken over each simulation run may be provided. In all cases, a packet is as defined by the traffic model.

1. Percentage of users as a function of throughput for different loading levels

Throughput is measured on a per packet basis and is equal to the number of information bits divided by the total transmission time. In other words, retransmissions are accounted for and reduce the peak data rate statistic. The total transmission time is defined to include the time to transmit the initial attempt and each subsequent retry.

For example, consider a packet "*m*":

- Packet *m* contains I_m information bits.
- Packet requires three attempts to transmit.
- Packet m takes $T_{m,j}$ seconds to transmit for attempt j

$$R(m) = \frac{I_m}{\sum_{j=1}^{3} T_{m,j}} \tag{1}$$

Figure 2 shows a sample output graph.

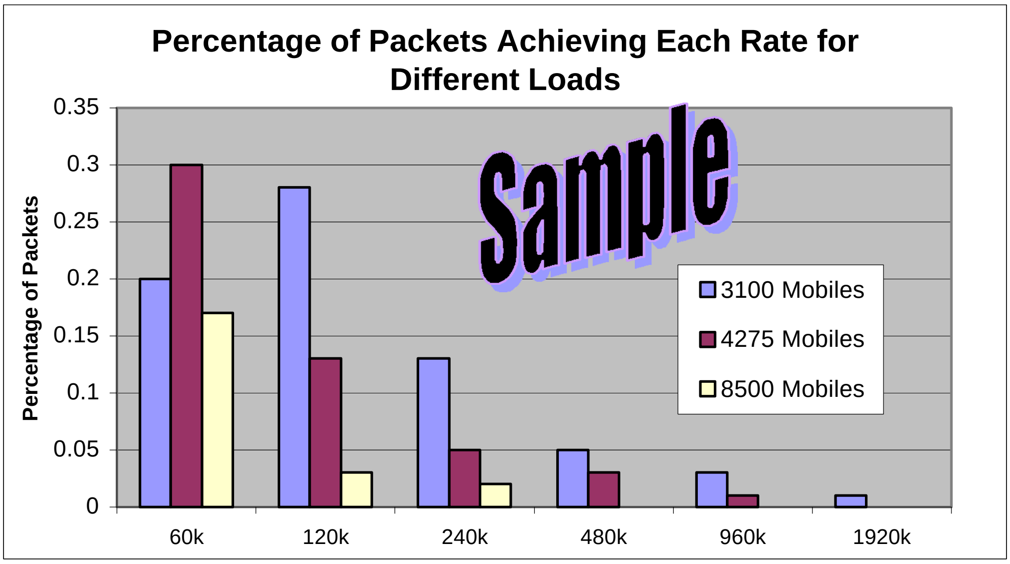

Figure 2. Percentage of Packets as function of throughput for the different loading levels.

2. Mean distance from serving site for each throughput level, measured per packet

The rate of each packet is calculated as in the previous section. A sample output graph is shown in the Figure 3.

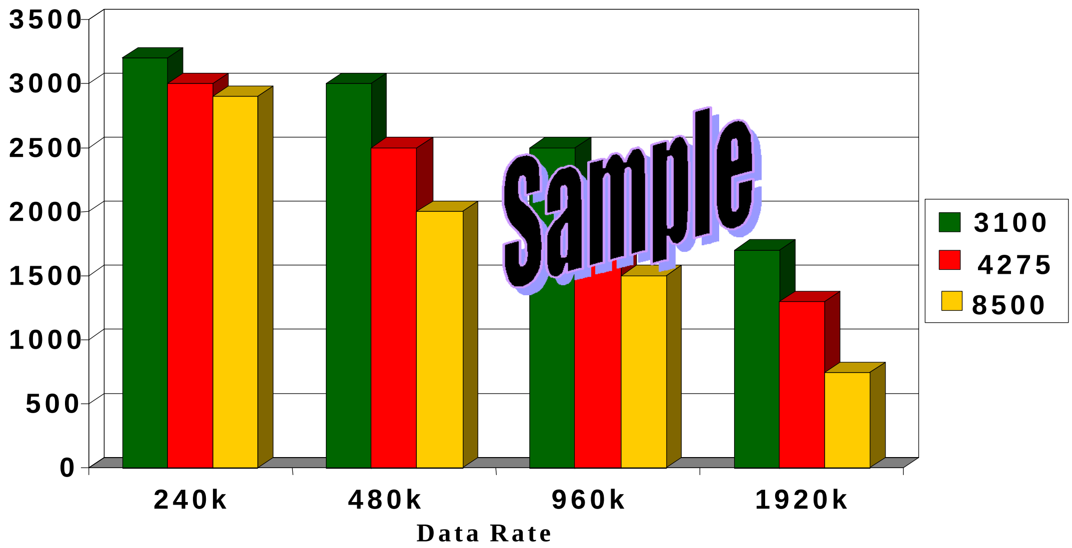

Figure 3. Mean Distance from serving site for each throughput level.

3. The following statistics as a function of offered load may also be provided

   3.1 Throughput per sector: Total number of bits successfully transferred divided by the total number of sectors and simulation duration.

3.2 Average and Variance of Packet Call Completion Time – measured from when the first packet of a packet call arrives at the base station's queue to when the final packet of the packet call is received by the UE station

3.3 Average and Variance Packet Call Transfer Rate - defined as the payload size of a packet call divided by the transfer time where transfer time is measured from when the first packet of a packet call is transmitted by the base station to when the final packet of the packet call is received by the UE station

3.4 Service Rate – the number of completed packet calls per second.

3.6 Simulation cases

In order to evaluate the performance of the basic features proposed for HSDPA (AMCS, fast HARQ and FCSS), at least the simulation cases described below should be conducted. In both cases the performance reference is the Rel.-99 system.

3.6.1 Case 1

In case 1, adaptive modulation and coding (AMCS) and fast HARQ will be modeled.

The following parameters will be used:

- MCS may be selected based on CPICH measurement, e.g. RSCP/ISCP, or power control feedback information
- MSC update rate: once per 3.33 ms (5 slots)
- CPICH measurement transmission delay: 1 frame
- Selected MCS applied with 1 frame delay after receiving measurement report
- Std. dev. of CPICH measurement error: 0, 3dB
- CPICH measurement rate: once per 3.33 ms
- CPICH measurement report error rate: 1 %
- Frame length for fast HARQ: 3.33 ms
- Fast HARQ feedback error rate: 0%, 1% or 4 %.

3.6.2 Case 2

In case 2 all the three techniques (fast HARQ, AMC, and FCSS) will be modeled. The parameters are as for case 1, with the addition of:

- Cell selection rate: once per 3.33 ms
- Cell selection error rate: 1 %
- FCSS request transmission and cell selection delay: 2 frames

4. CONCLUSIONS

A basic set of definitions and assumptions which HSDPA system simulations can be based on have been presented in this document. They are proposed to be used to verify the potential performance gains due to the currently proposed HSDPA features, such as adaptive modulation and coding scheme (AMCS), fast Hybrid ARQ, and fast cell site selection (FCSS).

REFERENCES

[1] Motorola. Link Evaluation Methods for High Speed Downlink Packet Access (HSDPA). TSG-R1 document, TSGR#14(00)0910, 4-7th, July, 2000, Oulu, Finland, 6 pp.

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[4] ETSI SMG2. Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS. ETSI SMG2 Technical Report, TR 101 112 v3.2.0 (UMTS 30.03), 83 pp.

[5] Gary R. Wright, Richard Stevens. TCP/IP Illustrated, Volume 2: The Implementation;, Addison-Wesley Professional Computing.

ANNEX 1: EXAMPLES OF PERFORMANCE EVALUATION METHODS

In the following, two examples of system performance evaluation methods are briefly described. First one is a combination of simulations and analytic evaluation and second one is based on dynamic system level simulations.

A. Analytic Simulation

In this method C/I statistics for all locations in a 19 cell, 3-sectored system is created and the corresponding C/I histogram is obtained. Next, from the link simulations the Throughput vs. C/I results are obtained for various MCS with STTD and Hybrid ARQ. The link and system simulation results are then combined and post-processed to obtain average sector throughput for various classes of scheduler. The flow diagram of this method is shown in Figure 4.

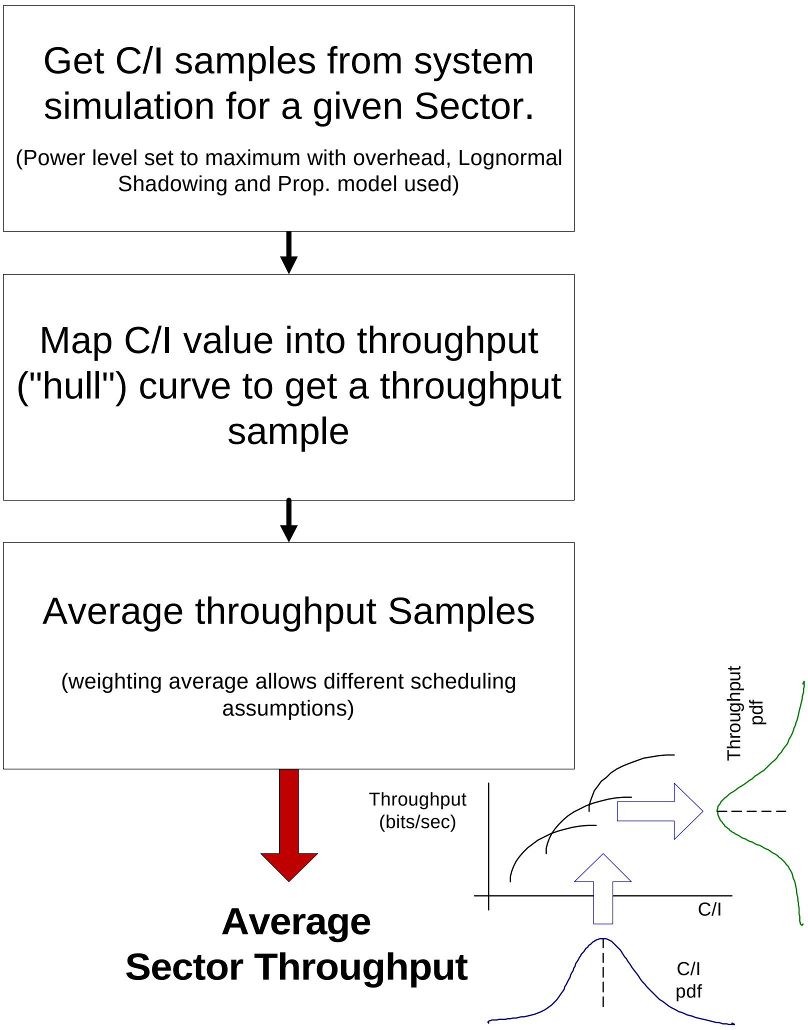

Figure 4. Analytic Simulation Flow Chart

In order to get an estimate of capacity the Max/Min scheduler as shown in Figure 5 is used. In this scheduler users with throughput above sector average gets Max/Min more packets than users below the sector average.

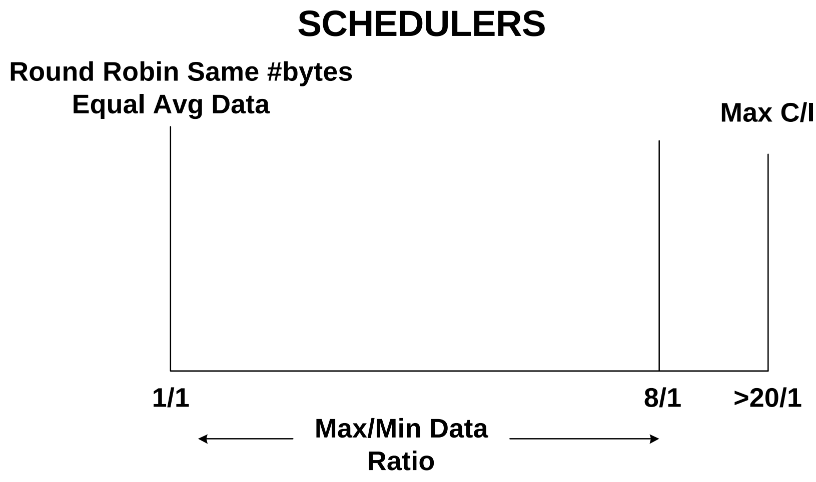

Figure 5. Max/Min schedulers

B. Dynamic system level simulations

Determining high rate packet data system performance requires a dynamic system simulation tool to accurately model feedback loops, signal latency, site selection, protocol execution, random packet arrival, and mobility in a multipath fading environment. The packet system simulation tool will include Rayleigh and Rician fading and evolve in time with discrete steps (e.g. time steps of 0.667ms). The time steps need to be small enough to correctly model feedback loops, latencies, scheduling activities, measurements of required system metrics (e.g. C/I similar to CPICH Ec/No), and fast cell site selection. A Ec/Ior vs. FER curve for a AWGN (static) channel will be created using a link level simulation for each data rate, modulation and coding scheme to determine successful over the air packet delivery. Sampling Eb/Nt points over each frame creates a frame metric. For a given frame the metric is used with the static curve to determine if the frame is erased. Alternatively, one can also use an array of Ec/Ior vs. FER curve for different fading conditions, geometries, speeds and MCS which will then be used in the system simulation to determine whether a frame is erased or not. Lognormal shadowing, delay spread, and fractional recovered power (per ray) will also be modeled. Scheduling and MAC will be included in the simulation to the detail necessary to model resource allocation latencies.

The data traffic model is intended to capture the interaction between radio link algorithms/designs and end-user applications. As such, it is proposed that both best effort and real-time models be simulated to capture air-interface performance. Ideally, best effort services should be modeled by a closed-loop traffic model in the form of a full web browsing model operating over a TCP/IP stack. The close-loop traffic model provides a variable IP packet stream that reacts to the quality of the radio link and the policies of the radio network's packet scheduler. Furthermore, the close-loop traffic model should properly model the bursty nature of data traffic and limit the simulation scheduler to a causal implementation that operates on available information such as the current queue depths and bounds buffering delays to practical levels. The ideal real-time model combines specific frame-erasure rates and delay guarantees to test the capability of the air interface. These real-time models will likely consume greater resources than best effort service. The ability of the air-interface to meet these guarantees may be measured by both the probability of blocking and the residual capacity remaining for best effort services.