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Proposed Text for
P802.16m Evaluation Methodology and Key Criteria Document

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Table of Contents

1. SCOPE **1**

2. MODELING **1**

3. SIMULATION REQUIREMENTS **28**

4. EVALUATION REPORT **39**

REFERENCES **40**

APPENDIX A: DEFINITION OF TERMS **41**

APPENDIX B: 19-CELL WRAP-AROUND IMPLEMENTATION **41**

APPENDIX C: FIXED USER LOCATIONS FOR SYSTEM LEVEL CALIBRATION **43**

.... **44**

.... **44**

.... **44**

.... **44**

.... **44**

.... **44**

P802.16m - Evaluation Methodology and Key Criteria

Acknowledgement
The referenced Evaluation Criteria document [2], which was created by IEEE 802.20 is, to a large extent, the basis for the present document, albeit with significant changes and adaptations tailored to project 802.16m.

1. Scope

This document describes the evaluation criteria that IEEE 802.16 Task Group m ("TGm") shall use to assess and compare technical proposals which address the 802.16m Requirements [1].

1.1 Document structure

The document consists of two major parts: (a) ***modeling*** requirements and (b) ***simulation*** requirements. Figure 1 describes the composition of the evaluation methodology system.

2. Modeling

2.1 Link level and System Level analyses

A great deal can be learned about an air interface by analyzing its fundamental performance in a, so-called, link-level settings which consists of one base station and one mobile terminal. The analysis can provide information on the system's fundamental performance metrics such as: noise-limited range, peak data rate, maximum throughput, etc. The actual performance, in real-world settings, where multiple base stations are deployed in a service area and operating in the presence of a large number of active mobile users, can only be evaluated in a System-level analysis. The extension of the link-level analysis methods to a system-level analysis may start with adding multiple users in a single-cell setting. This technique is generally straightforward and provides a mechanism for initial understanding of the multiple-access characteristics of the system. Ultimately, however, quantifying the system level performance, although difficult, carries with it the reward of producing results that are more indicative of the system performance.

Since system level results vary considerably with different propagation and interference environments, as well as with the number and distribution of users within the cells, it is important that the assumptions and parameters, used in the analysis, be reported carefully lest the quoted network-level performance be misleading.

This document specifies detailed requirements for both the link-level and the system-level analyses.

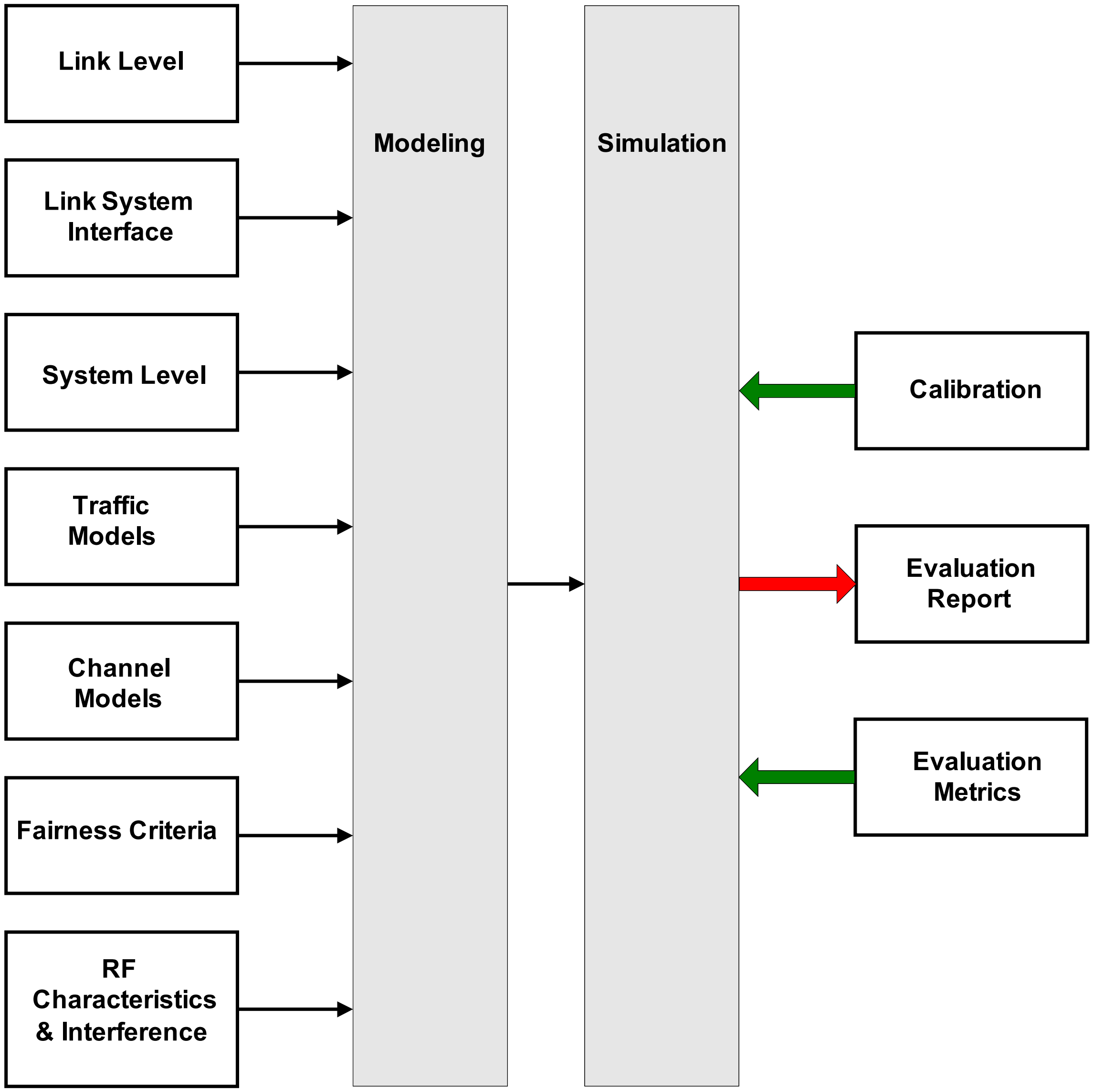

Figure 1: Composition of the Evaluation Methodology

2.2 Link level modeling

Single user link-level analysis focuses on the performance of a single user terminal operating in a specified propagation and interference environments. This is an important performance assessment tool for understanding the air interface as it yields important information about the system, including:
- the effectiveness of the link-adaptation and power control,
- the noise-limited base-station to terminal range,
- the SNR required to support various classes of service,
- the impact on performance of multipath and fading.

It should be emphasized that due to the variability and complexity of the propagation environment and the inter-cell interference, a single-user link-level analysis cannot be directly extrapolated to the system level to determine the actual system performance.

2.3 Modeling assumptions

The performance of modulation and coding schemes is to be evaluated using all channel environments associated with the channel models specified for the evaluation.

2.4 Link level performance metrics

FER vs. SINR curves are generated by the link-level simulations. Systems with adaptive modulation and coding should produce a set of curves (one curve per modulation and coding class). A second set of curves is the link-level *throughput vs. SINR*. The link-level *throughput* is derived from the FER values using the following formula:

$$Throughput \quad n \quad 1 \quad FER \ / T$$

where, T is the frame duration in seconds, and n is the number of information bits/frame supported by a modulation-coding class.

2.5 Traffic models

This section describes the traffic models in detail. Sections and clarify some aspects of the modeling approach and the remaining sections provide detailed models for traffic type listed in Table 1.

Table 1: Characteristics of traffic types used for evaluating

#	Application	Traffic Category	Mandatory/ Optional
1	FTP	Best-effort	M
2	Web Browsing	Interactive	M
3	Video Streaming	Streaming	M
4	VoIP	Real-time	M
5	Gaming	Interactive Real-time	O

2.5.1 User/Traffic Modeling Approach

One of the objectives of a modeling and simulation exercise is to determine the number of users a MOBILE BROADBAND WIRELESS system can support. The proposed approach here is to have traffic models for a user who is maintaining a session with transmission activity. These can be used to determine the number of such registered users that can be supported. This document does not address the arrival process of such registered users, i.e. it does not address the statistics of subscribers that register and become active.

Modeling of an aggregated load from a number of user nodes for background loading purposes may not be feasible for a wireless network. Such an abstraction is particularly difficult with adaptive antenna technologies and systems with complex channel dependencies. So, our traffic models apply to one user terminal.

2.5.2 Packet Generation

In some of the traffic models, there is a statistical description of the workload or the content of the application rather than the actual packet stream. This is consistent with the state of the art in evaluation of multi-service data systems. For example, the Web browsing model describes the Web pages and the timing between the Web pages. Depending on the details of the underlying TCP model (e.g. MTU size, max receive window) and the HTTP protocol version (HTTP v1.0 versus HTTPv1.1 versus HTTPv2.0), the actual stream of packets will vary. In some cases, as in the Voice models, the model may describe the packet stream more directly.

2.5.3 Web Browsing

Web browsing is the dominant application for broadband data systems, and has been studied extensively. See references [4], [5]

The parameters for the web browsing traffic are as follows:

S_M: Size of the main object in a page
S_E: Size of an embedded object in a page
N_d: Number of embedded objects in a page
D_{pc}: Reading time
T_p: Parsing time for the main page

Table 2 *(continued on the next page)* summarizes the HTTP traffic model parameters.

Table 2: HTTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes Std. dev. = 25032 bytes Minimum = 100 bytes Maximum = 2 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 1.37, \mu = 8.35$
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758 bytes Std. dev. = 126168 bytes Minimum = 50 bytes Maximum = 2 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 2.36, \mu = 6.17$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64 Max. = 53	$f_x = \frac{\alpha k^\alpha}{x^{\alpha+1}}, k \le x < m$ $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$ $\alpha = 1.1, k = 2, m = 55$ Note: Subtract k from the generated random value to obtain N_d
Reading time (D_{pc})	Exponential	Mean = 30 sec	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.033$
Parsing time (T_p)	Exponential	Mean = 0.13 sec	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 7.69$

Note: When generating a random sample from a truncated distribution, discard the random sample when it is outside the valid interval and regenerate another random sample.

2.5.4 FTP

In FTP applications, a session consists of a sequence of file transfers, separated by *reading times*. The two main parameters of an FTP session are:

S : the size of a file to be transferred

D_{pc}: reading time, i.e., the time interval between the end of the download of the previous file and the user request for the next file.

The underlying transport protocol for FTP is TCP. The parameters for the FTP application session are summarized in Table 3.

Table 3 FTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
File size (S)	Truncated Lognormal	Mean = 2Mbytes Std. Dev. = 0.722 Mbytes Maximum = 5 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[-\frac{(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 0.35, \mu = 14.45$
Reading time (D_{pc})	Exponential	Mean = 180 sec.	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.006$

2.5.5 Voice over IP (VoIP)

A VoIP call shall be assumed to be between one user (mobile station) and one wired user. In order to get an evaluation of the air interface the wireline and core network impairments are set to 0.

VoIP Traffic Source

The ITU G.729A voice encoder should be simulated with an assumed 4 byte IP header. Each packet produced by the encoder shall be appended with a 4 byte header that accounts for UDP/IP overhead, after header compression.

VoIP Performance Metric

VoIP performance shall be compared on the basis of the CDF of the R values generated as a result of simulating voice traffic. R values with the corresponding impairment factors shall be obtained for the forward and reverse links.

The following metrics shall be evaluated for each VoIP user:

(a) Mean Delay (T_a): On the forward link, the delay is measured from the point the voice packet is generated at the wired origin point to the time it is delivered at the mobile station. On the reverse link, the delay is measured from the point the voice packet is generated at the mobile station to the point it is delivered to the wired termination point.

(b) Packet Loss Probability (Ppl, measured in percents): The packet loss probability is measured separately on the reverse and forward links.

The following set of formulas (defined in ITU-T G.107) shall be used to compute the *R-factor* as a function of the delay and packet loss probabilities. Proposals with better R-factors shall be considered to have better performance.

$$R_{802.16M} = 93.2 - I_d - I_{e\text{-}eff}$$

The quantity I_d is defined as given below

$$Id = Idd$$

For $Ta \le 100$ ms:

$$Idd = 0$$

For $Ta > 100$ ms:

$$Idd = 25\left\{\left(1+X^6\right)^{\frac{1}{6}} - 3\left(1+\left[\frac{X}{3}\right]^6\right)^{\frac{1}{6}} + 2\right\}$$

with:

$$X = \frac{\lg\left(\frac{Ta}{100}\right)}{\lg 2}$$

Further, $I_{e\text{-}eff}$ is defined as shown below, with Ie=11 and Bpl = 19% (note that Bpl is measured in percents based on random packet loss).

$$Ie\text{-}eff = Ie + (95 - Ie)\cdot\frac{Ppl}{Ppl + Bpl}$$

The results shall include a histogram of R values for VoIP users in the system. Additionally, a histogram of packet delays may be included.

Explanation

ITU-T G.107 defines an objective model known as E-Model based on Network, Speech, Terminal/Device parameters to estimate/predict the perceived quality of VoIP session. The primary output of the E-Model is transmission rating factor R (Total Value Index) that can be mapped one-to-one to an estimated MOS.

The E-model defines 20 different parameters each with a default value and their ranges of values are defined. If all parameters are set to the default values, the calculation results in a very high quality with a rating factor of $R_{default} = 93.2$, which is also defined as the intrinsic quality of a voice call with a mouth-to-ear delay of 0 ms. The intrinsic quality of a packetized voice call transported without packet loss in the G.711 format corresponds to this $R_{default} = 93.2$.

However, for system specific impairments such as Packet Loss, Delay etc considered, the effective R factor for such system needs to be estimated by incorporating equipment impairment factor, delay factor. The effective R factor is

$R_{802.16M} = 93.2 - I_d - I_{e\text{-}eff}$

Here I_d, the impairment factor representing all impairments due to delay of voice signals Talker Echo, Listener Echo and I_{dd}, a loss of interactivity, represents the impairment caused by too-long absolute delay Ta, which occurs even with perfect echo canceling. Here we assume perfect Jitter operation resulting no packet loss and additional delay introduced by jitter.

The packet-loss dependent Effective Equipment Impairment Factor Ie-eff is derived using the codec specific value for the Equipment Impairment Factor at zero packet-loss Ie and the codec specific Packet-loss Robustness Factor *Bpl*

I_e represents the effect of degradation introduced by CODECs, Packet Loss. ITU-T G.113 (Appendix-

2002) provides parameters for use in calculating Ie for CODEC type and Packet Loss rate. For a G.729 Codec Ie = 11, and for random packet loss Bpl = 19.

2.5.6 Video Streaming

The following describes a model for streaming video traffic. Figure 2 shows the video stream in a steady state of flow from the network to the base station. Latency of starting up the call is not considered in this steady state model.

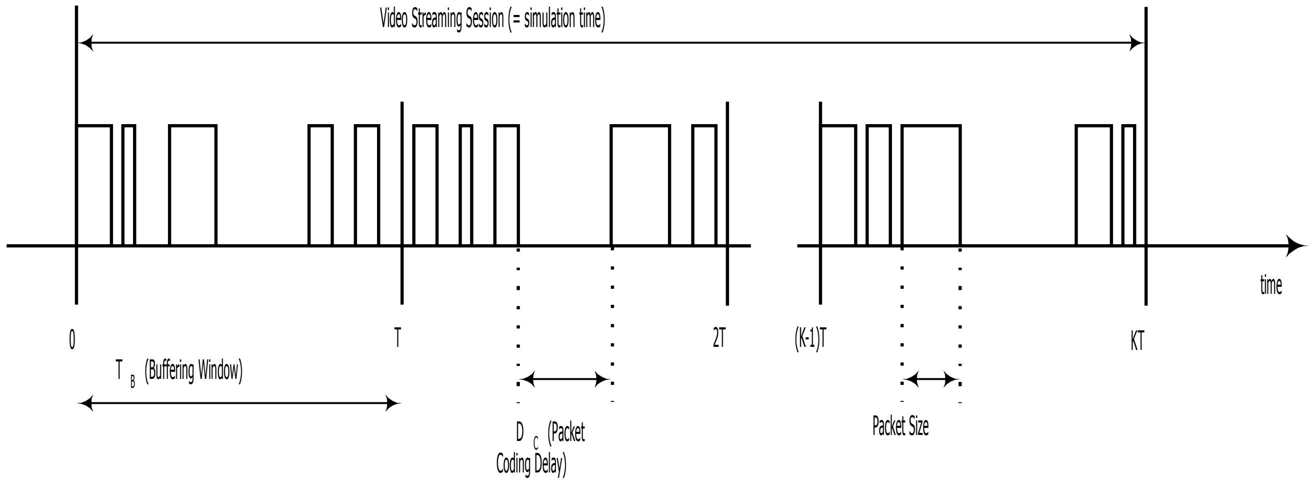

Figure 2: Near Real-Time Video Traffic Model

A video streaming session is defined as the entire video call time, including the associated audio streaming, which is equal to the simulation time for this model.

Each frame of video data arrives at a regular interval T determined by the number of frames per second (fps). Each frame is decomposed into a fixed number of slices, each transmitted as a single packet. The size of these packets/slices is distributed as a truncated Pareto. Encoding delay, Dc, at the video encoder introduces delay intervals between the packets of a frame. These intervals are modeled by a truncated Pareto distribution. The parameter T_B is the length (in seconds) of the de-jitter buffer window in the mobile station used to guarantee a continuous display of video streaming data. This parameter is not relevant for generating the traffic distribution but is useful for identifying periods when the real-time constraint of this service is not met. At the beginning of the simulation, it is assumed that the mobile station de-jitter buffer is full with (T_B x source video data rate) bits of data. Over the simulation time, data is “leaked” out of this buffer at the source video data rate and “filled” as forward link traffic reaches the mobile station. As a performance criterion, the simulation shall record the length of time, if any, during which the de-jitter buffer runs dry.
The de-jitter buffer window for the video streaming service is a maximum of 5 seconds.
Using a source rate of 64 kbps, the video traffic model parameters are defined Table 4.

Table 4: Near Real-Time Video Traffic Model Parameters

Information types	Inter-arrival time between the beginning of each frame	Number of packets (slices) in a frame	Packet (slice) size	Inter-arrival time between packets (slices) in a frame
Distribution	Deterministic (Based on 10fps)	Deterministic	Truncated Pareto (Mean= 50bytes, Max= 125bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution Parameters	100ms	8	K = 20bytes = 1.2	K = 2.5ms = 1.2

2.5.7 Wireless Multi-Party Gaming Traffic

Wireless gaming is an important application that should be considered in 802.16m system evaluation. This section describes a model for mobile network gaming traffic on the forward link and reverse link. This model is a combination of a standardized reverse link model (see cdma2000 Evaluation Methodology, C.P1002, Version 0.3, July 2004) and a forward link model developed from the research literature.

Reverse Link

Table 5 describes the parameters for the mobile network gaming traffic on the reverse link.

Table 5 Mobile Reverse Link network gaming traffic model parameters

Component	Distribution	PDF and generation method
Initial packet arrival	Uniform (a=0, b=40ms)	$f(x) \quad \frac{1}{b \quad a} \quad a \quad x \quad b$
Packet arrival	Deterministic (40ms)	
Packet size	Extreme (a=45 bytes, b = 5.7)	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln \quad \ln Y$, $Y \quad U(0,1)$ Because packet size has to be integer number of bytes, the largest integer less than or equal to X is used as the actual packet size.
UDP header	Deterministic (2 bytes)	

This model uses Largest Extreme Value distribution for the packet size. For cellular system simulation, 2-byte UDP header (after header compression) should be added to the packet size X. Because the packet size has to be an integer number of bytes, the largest integer less than or equal to X is used as the actual packet size. To simulate the random timing relationship between client traffic packet arrival and reverse link frame boundary, the starting time of a network gaming mobile is uniformly distributed within [0, 40ms].

A maximum delay of 160ms is applied to all reverse link packets, *i.e.*, a packet is dropped by the mobile station if any part of the packet have not started physical layer transmission, including HARQ operation, 160ms after entering the mobile station buffer.. A packet can start physical layer transmission at the 160ms time instant. Packet dropping should be the last operation of mobile station buffer management, if any, at any time instant. The packet delay of a dropped packet is counted as 180ms.

A mobile network gaming user is in outage if the average packet delay is greater than 60ms. The average delay is the average of the delay of all packets, including the delay of packets delivered and the delay of packets dropped.

Forward Link Model

Table 6 describes the parameters for the mobile network gaming traffic on the forward link.

Table 6 Forward Link network gaming traffic model parameters

Component	Distribution	PDF and generation method
Initial packet arrival	Uniform (a=0, b=40ms)	$f(x) = \frac{1}{b-a} \quad a \le x \le b$
Packet arrival	Extreme (a=55, b=6)	
Packet size	Extreme (a=120 bytes, b = 36)	$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0$ $X = a - b \ln(-\ln Y)$, $Y \sim U(0,1)$ Because packet size has to be integer number of bytes, the largest integer less than or equal to X is used as the actual packet size.
UDP header	Deterministic (2bytes)	

This model uses Largest Extreme Value distribution for the packet size. For cellular system simulation, a 2-byte UDP header (after header compression) should be added to the packet size X. Because the packet size has to be an integer number of bytes, the largest integer less than or equal to X is used as the actual packet size. To simulate the random timing relationship between client traffic packet arrival and reverse link frame boundary, the starting time of a network gaming mobile is uniformly distributed within [0, 40ms].

A maximum delay of 160ms is applied to all reverse link packets, *i.e.*, a packet is dropped by the mobile station if any part of the packet have not started physical layer transmission, including HARQ operation, 160ms after entering the mobile station buffer.. A packet can start physical layer transmission at the 160ms time instant. Packet dropping should be the last operation of base station buffer management, if any, at any time instant. The packet delay of a dropped packet is counted as 180ms.

A mobile network gaming user is in outage if the average packet delay is greater than 60ms. The average delay is the average of the delay of all packets, including the delay of packets delivered and the delay of packets dropped.

2.5.8 Full buffers (Infinite backlog) model

In the full buffers (Infinite backlog) user traffic model, all the users in the system always have data to send or receive. In other words, there is always a constant amount of data that needs to be transferred, in contrast to bursts of data that follow an arrival process. This model allows the assessment of the spectral efficiency of the system independent of actual user traffic distribution type.

2.5.9 Traffic Mix

A MOBILE BROADBAND WIRELESS system is expected to support a mix of simultaneous traffic types. There can be different types of usage scenarios (multi-service v. single-type), different types of devices (notebook PCs v. PDAs or smart phones), different usage levels (intense v. light) and different delay/latency requirements (real-time v. best-effort).

The previous sections are primarily concerned with the traffic models for each of the potential traffic types. As discussed in the previous section, these models are based on statistical analysis of measured traffic that yielded some invariant patterns that are not very dependant on the specific system. It is more difficult to describe a similar invariant mix of traffic types since these tend to depend more heavily on the type of system and the actual deployment mix of user device types.

In the context of system performance evaluation, using traffic models, the specific traffic-mix should emphasize different aspects of the system performance, e.g. sustained throughput for file downloads v. faster response times for interactive applications.
A short list of representative applications and their corresponding percentage in a simulated system-wide traffic mix is shown in Table 7.

For system level simulation purposes, “traffic mix” refers to the percentage of users in the system generating a particular type of traffic. In this context, each user is assumed to be generating only one type of traffic, recognizing that in an actual network a single user’s terminal could support multiple applications and generate several types of traffic simultaneously.

Table 7 Traffic mix: percentage of different Traffic Types

Traffic Category	Application	Percentage (%)	Percentage with Gaming
Best Effort	FTP	30	30
Interactive	Web browsing	30	25
Streaming	Video streaming	30	30
Real-time	VoIP	10	10
Interactive	Gaming – (Optional)	-	5

2.6 Fairness Criteria

In the evaluation of spectral efficiency and in order to make a fair comparison of different proposals, it is important that all mobile users be provided with a minimal level of throughput. The fairness for best effort traffic (HTTP, FTP and full buffers) is evaluated by determining the normalized cumulative distribution function (CDF) of the user throughput, which meets a predetermined function given in Table 8. For applications other than best effort, *application specific outage criteria are applicable*.

Let $T_{put}[k]$ be the throughput for user k. The normalized throughput with respect to the average user throughput for user k, $\tilde{T}_{put}[k]$ is given by:

$$\tilde{T}_{put}[k] \quad \frac{T_{put}[k]}{\underset{i}{\mathrm{avg}}\, T_{put}[i]}.$$

The CDF of the normalized throughput with respect to the average user throughput is determined. The CDF shall lie to the right of the curve given by the points in Table .

Table 8: Fairness Criteria CDF

Normalized Throughput w.r.t average user throughput	CDF
0.1	0.1
0.2	0.2
0.5	0.5

2.7 Channel Modeling

The channel models, associated parameters and parameter values used to describe the propagation channel environment are included in a separate P802.16m document.

2.7.1 Channel Mix

At the link level, the channel models shall include the following non-spatial-varying parameters:
Case-I: Pedestrian A: NLOS, speed: 3, 30, 120 km/h; 4 paths
Case-II: Vehicular A: Speed: 30, 120, 250 km/h; 6 paths
Case-III: Pedestrian B: Speed: 3, km/h; 6 paths
Case-IV: Vehicular B: Speed: 30, 120, 250 km/h; 6 paths

A channel mix, based on the link-level channel models with fixed path delays, should also be used.

The following channel mix, based on the link-level channel models with fixed path delays, is to be applied.
Two scenarios are to be analyzed: suburban macro cell and urban micro cell which represent the two typical extremes of deployment environment. The assumptions on user speed distribution, quantized to 3, 30, 120 and 250 km/h, for each scenario are shown in Table 9.
For each channel power delay profile corresponding to each of the above speeds, the probability of users is equally distributed.

Note that in the tables below, the percentage of users at 250 km/hr for the suburban macro cells and the percentage of users at 250 km/hr and 120 km/hr for the urban micro cell are set to zero. In a realistic scenario, this would be a very small percentage, but, in order to achieve statistically meaningful simulation results, they are set to zero in this table. However, in order to understand the system performance under these speeds, a separate set of link curves for the suburban macro cells at 250 km/hr and urban macro cells at 120 km/h should be provided.

Table 9: Assumptions on distribution of mobile user speed

User distribution percentage per speed

User speed (km/h)	3	30	120	250
Suburban macro cell	40%	36%	24%	0
Urban micro cell	58%	42%	0	0

Scenario 1: Suburban Macro cells

Channel PDP Models	I			II			III	IV		
User speed (km/h)	3	30	120	30	120	250	3	30	120	250
Probability	0.20	0.12	0.08	0.12	0.08	0.0	0.20	0.12	0.08	0.0

Scenario 2: Urban Micro cells

Channel PDP Models	I			II			III	IV		
User speed (km/h)	3	30	120	30	120	250	3	30	120	250
Probability	0.29	0.14	0	0.14	0	0	0.29	0.14	0	0

2.8 TCP Model

Many Internet applications including Web browsing and FTP use TCP as the transport protocol. Therefore, a TCP model is introduced to more accurately represent the distribution of TCP packets from these applications. This section specifies separate, independent downlink and uplink TCP simulation models.

2.8.1 TCP Connection Set-up and Release Procedure

The TCP connection set-up and release protocols use a three-way handshake mechanism as described in Figure 3 and Figure 4. The connection set-up process is described below:

The transmitter sends a 40-byte SYNC control segment and waits for ACK from the remote server. The receiver, after receiving the SYNC packet, sends a 40-byte SYNC/ACK control segment. The transmitter, after receiving the SYNC/ACK control segment starts TCP in slow-start mode (the ACK flag is set in the first TCP segment).

The procedure for releasing a TCP connection is as follows:
The transmitter sets the FIN flag in the last TCP segment sent.
The receiver, after receiving the last TCP segment with FIN flag set, sends a 40-byte FIN/ACK control segment.
The transmitter, after receiving the FIN/ACK segment, terminates the TCP session.

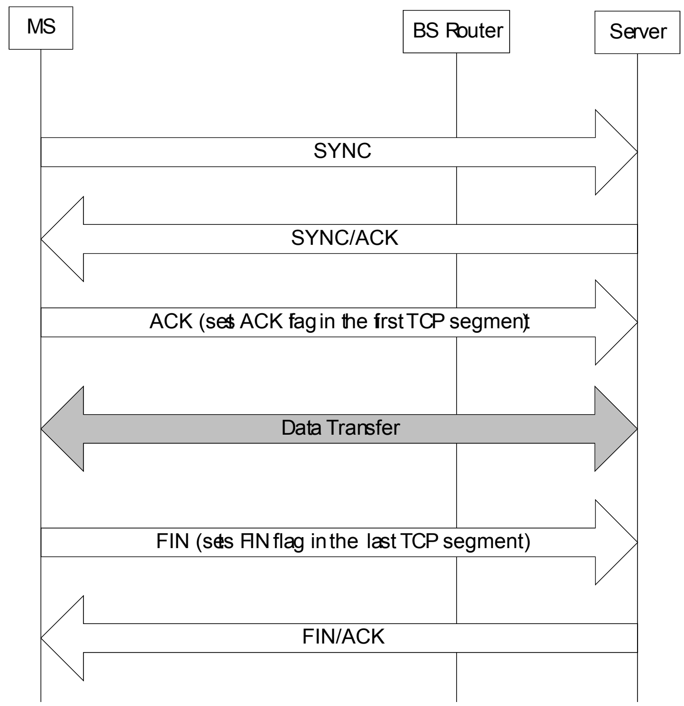

Figure 3: TCP connection establishment and release for Uplink data transfer

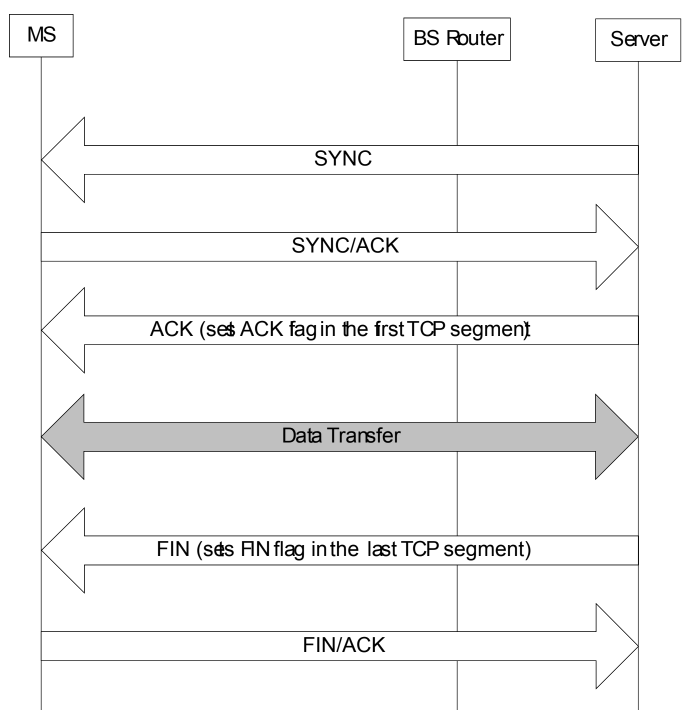

Figure 4: TCP connection establishment and release for Downlink data transfer

2.8.2 TCP slow start Model

The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size of the transmitter and the receiver window size. After the connection establishment is completed, the transfer of data starts in slow-start mode with an initial congestion window size of 1 segment. The congestion window increases by one segment for each ACK packet received by the sender regardless of whether the packet is correctly received or not, and regardless of whether the packet is out of order or not. This results in exponential growth of the congestion window i.e. after n RTTs (Round Trip Times), the congestion window size is 2^n. segments

2.8.3 UL (Uplink) slow start model

The UL slow start process is illustrated in Figure 5. The round-trip time $_{rt,}$ consists of two components $_{rt} = {}_u + {}_l$ where: $_u$ = the sum of the time taken by a TCP data segment to travel from the base station router to the server plus the time taken by an ACK packet to travel from the server to the client; $_l$ = the transmission time of a TCP data segment over the access link from the client to the base station router. $_u$ is further divided into two components; $_2$ = the time taken by a TCP data segment to travel from the base station router to the server plus the time taken by an ACK packet to travel from the server back to the base station router and $_3$ = the time taken by the ACK packet to travel from the base station router to the client. Table 10 summarizes this information.

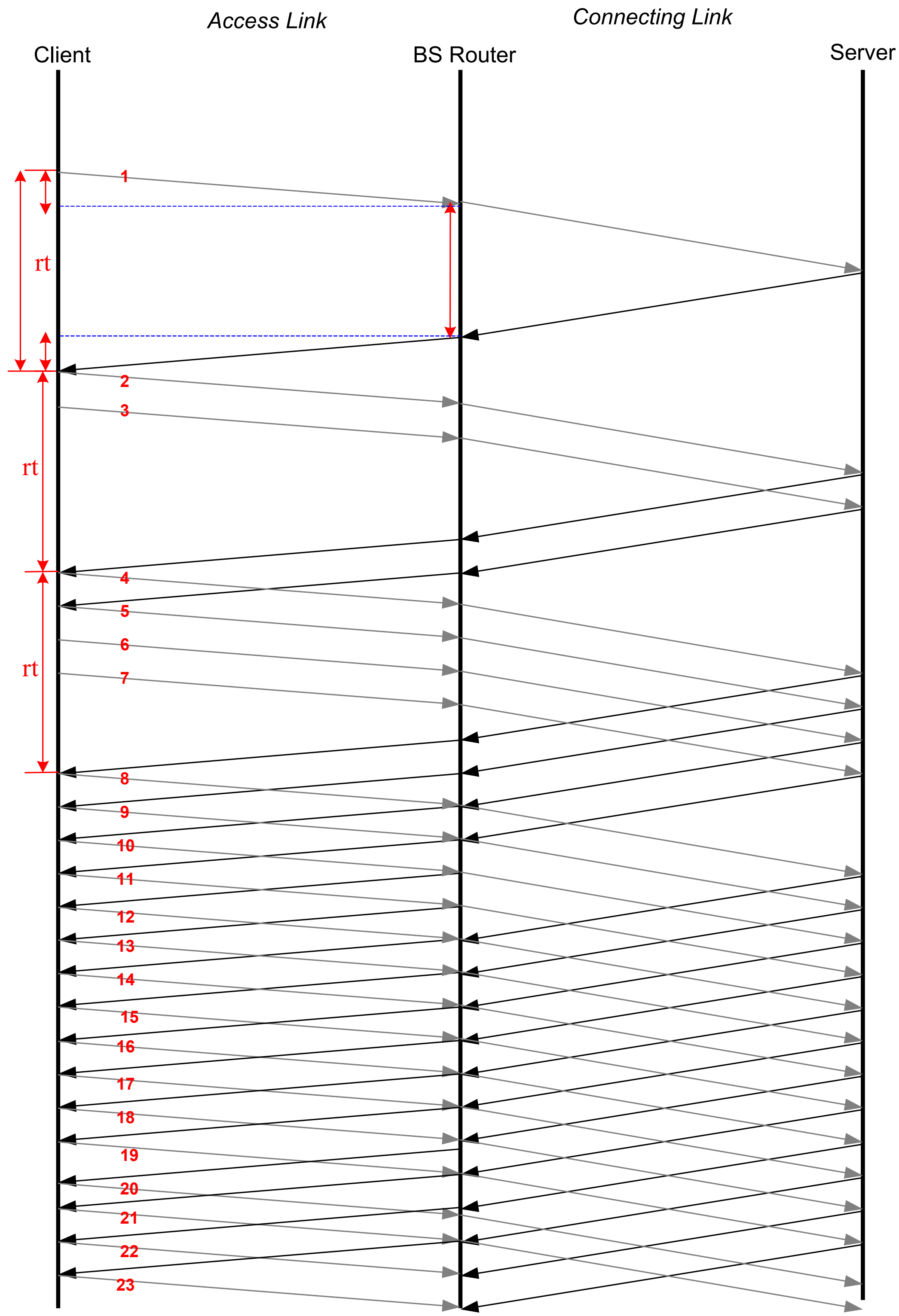

Figure 5: TCP Flow Control During Slow-Start; ${}_{l}$ = Transmission Time over the Access Link (UL); ${}_{rt}$ = Roundtrip Time

Table 10 Delay components in the TCP model for the UL upload traffic

Delay component	Symbol	Value
The transmission time of a TCP data segment over the access link from the client to the base station router.	$_{1}$	Determined by the access link throughput
The sum of the time taken by a TCP data segment to travel from the base station router to the server and the time taken by an ACK packet to travel from the server to the base station router.	$_{2}$	
The time taken by a TCP ACK packet to travel from the base station router to the client.	$_{3}$	

2.8.4 DL (Downlink) slow start model

This DL slow start process is illustrated in Figure 66. The round-trip time in Figure 66, $_{rt,}$ consists of two components: $_{rt} = _{d} + _{4}$ where: $_{d}$ = the sum of the time taken by an ACK packet to travel from the client to the server and the time taken by a TCP data segment to travel from the server to the base station router; $_{4}$ = the transmission time of a TCP data segment over the access link from the base station router to the client. $_{d}$ is further divided into two components; $_{5}$ = the time taken by a TCP ACK to travel from the base station router to the server plus the time taken by a TCP packet to travel from the server back to the base station router and $_{3}$ = the time taken by the TCP packet to travel from the base station router to the client. Table 11 summarizes this information.

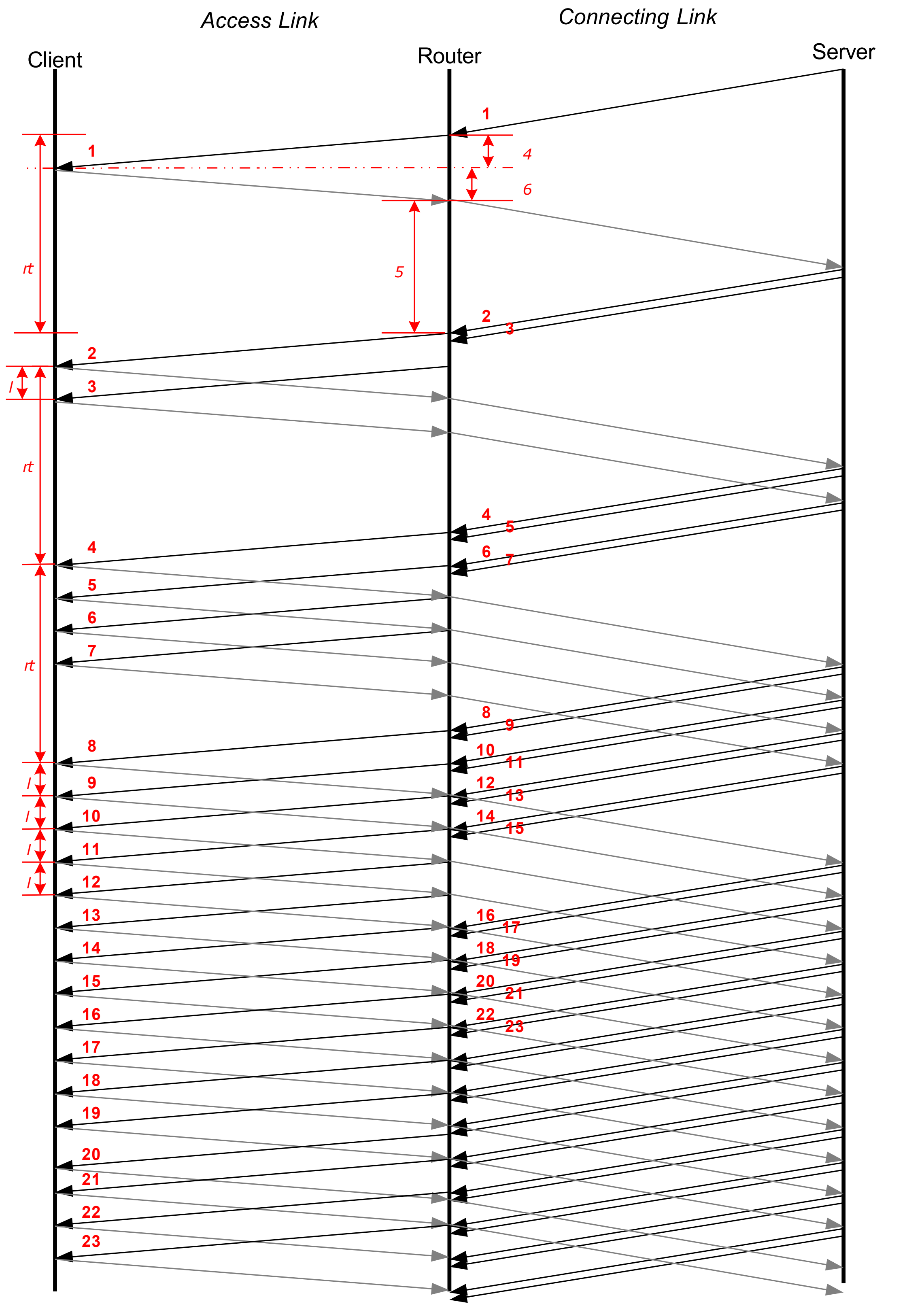

Figure 6: TCP Flow Control During Slow-Start; $_{l}$ = Transmission Time over the DL; $_{rt}$ = Roundtrip Time

Table 11 Delay components in the TCP model for the DL traffic

Delay component	Symbol	Value
The transmission time of a TCP data segment over the access link from the base station router to the client.	4	Determined by the access link throughput
The sum of the time taken by a TCP ACK to travel from the base station router to the server and the time taken by TCP data packet to travel from the server to the base station router.	5	
The time taken by a TCP ACK to travel from the client to the base station router.	6	

From Figure 5 and 6, it can be observed that, during the slow-start process, for every ACK packet received by the sender two data segments are generated and sent back to back. Thus, at the mobile station (base station), after a packet is successfully transmitted, two segments arrive back-to-back after an interval u= 2+ 3 (d= 5+ 6). Based on this observation, the packet arrival process at the mobile station for the upload of a file is shown in Figure 7. It is described as follows:

Let S = size of the file in bytes. Compute the number of packets in the file, N = S/(MTU-40) . Let W = size of the initial congestion window of TCP. The MTU size is fixed at 1500 bytes

If N>W, then W packets are put into the queue for transmission; otherwise, all packets of the file are put into the queue for transmission in FIFO order. Let P=the number of packets remaining to be transmitted beside the W packets in the window. If P=0, go to step 6, wait until a packet of the file in the queue is transmitted over the access link. Schedule arrival of next two packets (or the last packet if P=1) of the file after the packet is successfully ACK'ed. If P=1, then P=0, else P=P-2

If P>0 go to step 3

End.

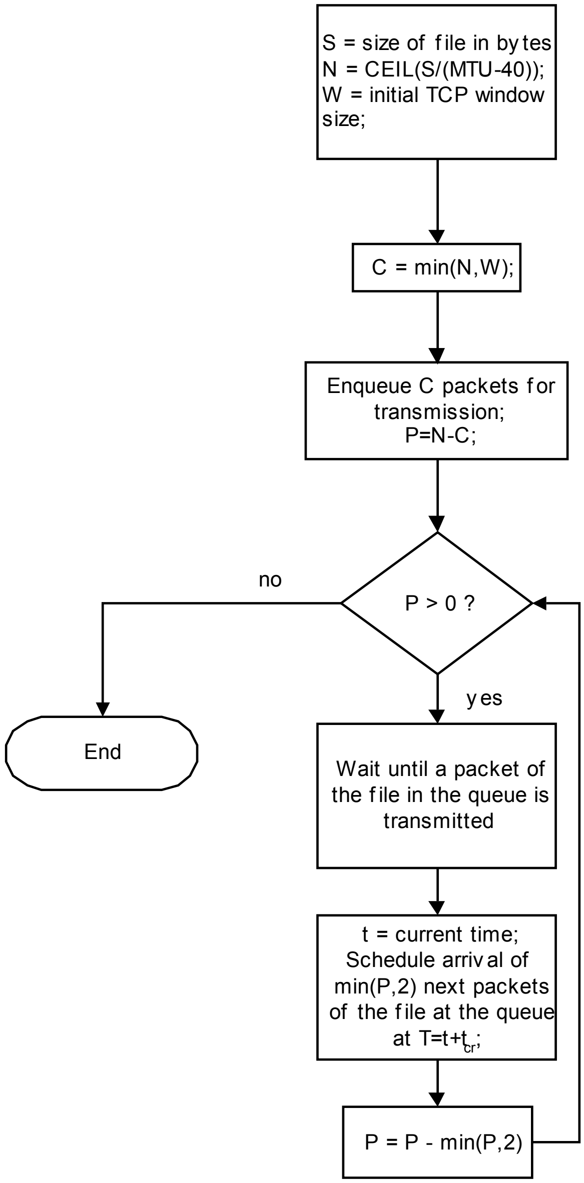

Figure 7: Packet arrival process at the mobile station (base station) for the upload (download) of a fi e using TCP

2.9 Mobility Modeling for Signaling Robustness Evaluation

The system simulation defined elsewhere in the document deals with sector throughput, spectral efficiency, latency and fairness. However, user experience in a **MOBILE BROADBAND WIRELESS** system is also influenced by the performance of handoff and paging. The objective of this section is to propose methods to study the performance of handoff and paging. Only handoff within the system is considered; inter-system and inter-technology handoffs are not considered.

2.10 Higher Layer and Network Modeling

The analysis and modeling of the types of traffic discussed in section 4 of this document is based on the working assumption that the *Network Layer* protocol (layer 3) is IP and the *Transport Layer* (layer 4) protocols are TCP and UDP. Different applications use different higher layer protocols such as HTTP, RTP, DNS, Telnet, FTP, SMTP, etc. The extent of the modeling of such application protocols – as far as traffic packet generation is concerned - is defined in section 4. In the following sections we discuss TCP modeling in detail, primarily due to its major impact on system performance.

2.10.1 Network Delay model

The one-way Internet packet delay is modeled using a shifted Gamma distribution with the parameters shown in Table 12. The packet delay is independent from packet to packet.

Table 12: Parameters for the shifted Gamma Distribution

Scale parameter ( )	1
Shape parameter ( )	2.5
Probability density function (PDF)	$f(x) \quad \frac{(x/\ \ )\ \ ^{1} e^{x/a}}{(\ \ )}$ (.) is the gamma function
Mean	
Variance	
Shift	See Table 3

Two values, 7.5ms and 107.5ms are used for the shift parameter in order to model the domestic routes and the International routes respectively. The users' routes are selected randomly at the time of drop with the distribution shown in Table 13.

Table 13: Shift parameters for domestic and international IP routes

IP Route Type	Percentage of users	Shift parameter	Mean one-way IP packet delay
Domestic	80%	7.5ms	10ms
International	20%	107.5ms	110ms

2.11 Performance Metrics

Wireless systems often divide operation in two states: a connected state and a power save state. The terminology Connected State and Power Save State in this section is meant as an example, and proposals are free to either select alternative terminology, or to select more or fewer operating states.
Connected State: A terminal is said to be in connected state if it has an assigned traffic channel on both the uplink and the downlink.

Power Save State: A terminal is said to be in Power Save State if it has no assigned traffic channel on the uplink and downlink (it may have common/broadcast channels on the downlink). In this state the terminal can exchange data with a base station only by first transitioning to a Connected State

Proposals that have two operating states that are logically equivalent to a connected state and a power save state shall be evaluated based on the following mobility metrics.

Editor Note: Contributor requested to provide more detailed definitions.

Connected State Handoff Metrics

Outage period on uplink and downlink in case of handoff: The downlink outage period is the length of time during handoff when the terminal receives no new data. The uplink outage period is the length of time during handoff when the base station receives no new data.

Probability of connection drop during handoff: A connection drop is defined to occur when the outage period on the uplink or downlink crosses a threshold. This probability can be computed from the CDF of the outage period.

Power Save Mode Metrics

Probability of missed pages due to base station reselection. Base station reselection is defined as the process where a mobile terminal changes the base station (or set of base stations) from which it monitors pages, or the base station to which it directs access attempts).

Delay in transition to connected state upon base station reselection. This delay corresponds to the extra time taken to acquire the signal and parameters of the newly selected base station.

Average power consumption (duty cycle) in power save mode. The duty cycle in power save mode is defined as the fraction of time for which the receiver is on.

All other proposals (proposals with alternate definitions of operating states) shall define metrics that characterize performance under equivalent mobility situations.

The objective of the evaluation criteria in this section is not to obtain precise values for the metrics, but rather to obtain "ballpark" performance numbers that enable proponents to justify that their proposals have efficient support for mobility related performance.

In order to permit evaluation of the mobility metrics, a candidate proposal shall include details about the signaling required to implement the following

Connected state handoff
Power save state base station reselection
Page reception in power save mode
System acquisition for transition from power save state to connected state
General operation in power save state

The signaling details in a candidate proposal may be in the form of call flows or timing diagrams. If signaling messages are used for any handoff or paging operation, the proposal shall specify the format of the message.

The performance of signaling can be evaluated once an appropriate model for the event is available. Each proposal shall provide a model that contains sufficient information to evaluate the performance metrics discussed in this section, as well as those additional metrics deemed necessary by candidate proposals.

2.11.1 Proposal Requirements

In order to evaluate the metrics, a model for the signaling event needs to be developed. The nature of this model will depend on the candidate system. A few examples of event models are given here.

Example 1: Consider the case of *handoff in connected state*. A typical implementation for handoff from sector A to sector B (other implementations are allowed) has the following steps

The user terminal measures a signal-strength, in dBm (or C/I in dB), of sector B [time depends on measurement procedure and structure of pilots]
Terminal sends a Pilot Report to sector A [time calculated based on terminal position]
Sector A sets up resources on sector B [time depends on backbone as per Section 5.3. For simplicity, processing time at the sectors shall be ignored.]
Sector A sends Handoff Direction to terminal [time calculated based on terminal position]
Terminal establishes communication with sector B.

The first relevant performance metric in this case is the *Probability of Connection Drop:* This is the probability that step 4 above will fail (due to failure of one of the earlier events, or a failure in step 4 alone). The second performance metric of interest is the *handoff delay*: delay between the time of degradation of the signal from sector A and the time communication with sector B is established.
Example2: Consider the case of page reception during mobility from sector A to sector B. A typical implementation has the following steps.
Terminal wakes up some time before paging slot
Terminal acquires beacon from sector A
Terminal detects low signal strength on sector A
Terminal acquires pilot from sector B
Terminal attempts to decode the paging channel from sector B
The relevant performance metric in this case is the probability that a page is missed because of delay in acquiring the paging channel from sector B.

2.12 RF Environment

2.12.1 Radio Transceiver Characteristics

The RF environment defined herein for the evaluation model consists of a set of RF parameters that should be used as common constrains as well as by some proposal-specific parameters.

Table 12 defines the transmitter and receiver parameters for the base station (BS) and the mobile station (MS) radios. Note that some parameters vary with channel bandwidth. Base values are proposed for 5MHz, 10 MHz and 20 MHz channels.

2.12.2 Transmitter

For the evaluation purposes, the maximum base station (BS) **transmit power** is specified here as **peak power** per 1 MHz and is **+43 dBm/MHz**. The mobile station (MS) maximum transmit power is fixed as +27 dBm for all channel bandwidths.
For the evaluation purposes, the **out-of-band emission** limit shall be that which is specified for the block edge, by the FCC for the PCS band.

For clarification: for multi-carrier technologies, the transmit power is the total power of all carriers transmitted simultaneously and intended to be received by one receiver simultaneously.

Note that when applying the **FCC emission attenuation formula,** the maximum permissible out-of-band power is -13dBm for all transmit powers and all channel bandwidths. Also note that the resolution-bandwidth for out-of-band power measurements is 1 MHz.

2.12.2 Receiver

Noise Figure

To achieve a fair performance comparison, different technologies should be constrained by similar environmental interference conditions and should have identical receiver noise figure specifications. Thus, for the evaluation purposes, it is required that all proposals assume the same receiver **noise figure (NF)** value of **10 dB** for the **MS** and **5 dB** for the **BS** without regard to channel bandwidth.

Receiver Sensitivity

The theoretical **receiver sensitivity** is expected to vary from one technology to another, but again, for the sake of a fair comparison, it is required that the receiver sensitivity be specified for raw data bit error rate (BER) of 0.1%.

The receiver sensitivity (in dBm) is calculated using the following formula:

$$\text{Sensitivity} = (-174.5\text{ dBm}) + \text{NF (in dB)} + 10\log(\text{channel-BW in Hz}) + C/N_{min} \text{ for } 0.1\%\text{ BER}).$$

Table 14: Evaluation Criteria RF Parameters

#	RF Parameter	Base Value 1 MHz Channel	5 MHz Ch BW	10 MHz Ch BW	20 MHz Ch BW
1	Transmitter Power -- **BS**	43 dBm/MHz	+50 dBm	+53 dBm	+56 dBm
2	Transmitter Power -- **MS**	27 dBm	+27 dBm	+27 dBm	+27 dBm
3	Out of Band emission limits – BS and MS (emission measured in 1 MHz resolution bandwidth)	Attenuation of the transmit power P by: 43 +10 log(P) dB	-13 dBm	-13 dBm	-13 dBm
4	Receiver noise figure -- **BS**	5 dB	5 dB	5 dB	5 dB
5	Receiver noise figure -- **MS**	10 dB	10 dB	10 dB	10 dB
6	Receiver reference sensitivity (to be proposed by each technology)	Specify at BER of 0.1%	value 2 (proposal specific)	value 3 (proposal specific)	value 4 (proposal specific)
7*	Receiver Selectivity -- **BS**	63 dB	63 dB	63 dB	63 dB
8*	Receiver Selectivity -- **MS**	33 dB	33 dB	33 dB	33 dB

* Recommended values. Proposals may choose (and commit to) different values.

2.12.3 Link Budget

The link budget template – shown on the next page, as Table 15 – was adopted from ITU-R M.1225 with slight modifications. Table entries that have explicit numerical values (such as power levels, cable losses, etc) shall be used by in the system simulations. Values for the blank data items in the table (such as diversity gain, soft handoff gain etc) should be provided in the technical proposals.

id/ii	Item	Downlink	Uplink
	Test environment	Suburban/urban macro-cell, micro-cell,	Suburban/urban macro-cell, micro-cell,
	Operating frequency (for simulation purposes)	2.5 GHz	2.5 GHz
	Test service		
	Multipath channel class	Cases I-IV	Cases I-IV
ii/id	(a0) Average transmitter power per traffic channel	dBm	dBm
id	(a1) Maximum transmitter power per traffic channel	dBm	dBm
id	(a2) Maximum total transmitter power	43 dBm/MHz	27dBm
ii	(b) Cable, connector, and combiner losses (enumerate sources)	3 dB	0 dB
	Body Losses	0 dB	3 dB
ii	(c) Transmitter antenna gain	17 dBi	0 dBi
id	(d1) Transmitter EIRP, per traffic channel (a1 – b c)	dBm	dBm
id	(d2) Total transmitter EIRP (a2 – b c)	57 dBm	27 dBm
	Penetration Loss (Ref: 3GPP2)	20 dB (Building) 10 dB (Vehicular)	20 dB (Building) 10 dB (Vehicular)
ii	(e) Receiver antenna gain	0 dBi	17 dBi
ii	(f) Cable and connector losses	0 dB	3 dB
	Body Losses	3 dB	0 dB
ii	(g) Receiver noise figure	10 dB	5 dB
ii	(h) Thermal noise density (H) (linear units)	–174 dBm/Hz $3.98 \quad 10^{-18}$ mW/Hz	–174 dBm/Hz $3.98 \quad 10^{-18}$ mW/Hz
id	(i) Receiver interference density (NOTE 1) (I) (linear units)	dBm/Hz mW/Hz	dBm/Hz mW/Hz
id	(j) Total effective noise plus interference density $10 \log (10^{((g \quad h)/10)} \quad I)$	dBm/Hz	dBm/Hz
ii	(k) Information rate ($10 \log (R_b)$)	dB(Hz)	dB(Hz)
id	(l) Required $E_b/(N_0 \quad I_0)$	dB	dB

id	(m)	Receiver sensitivity = (j k l)		
id	(n)	Hand-off gain	dB	dB
id	(o)	Explicit diversity gain	dB	dB
id	(o)	Other gain	dB	dB
id	(p)	Log-normal fade margin	dB	dB
id	(q)	Maximum path loss {d1 – m (e – f) o n o – p}	dB	dB
id	(r)	Maximum range	Km	Km

Table 15: Link Budget Template

Note: Peak power is equivalent to maximum power according to ITU-R M.1225. For definition of maximum power and average power, please refer to ITU-R M.1225.

NOTES to the Link Budget table:

NOTE 1 – Since the significance and method of calculating this value will vary from RTT to RTT, the proponent must give a detailed explanation of their method for calculating this value and its significance in determining capacity and coverage of the RTT. In particular, the proponent must state explicitly what frequency reuse ratio and traffic loading per sector are assumed in determining this quantity. Interference has to be evaluated for the specified low traffic level given for each test environment.

The following sections provide descriptions of the individual link budget template items. Descriptions apply to both forward and reverse links unless specifically stated otherwise. For the forward link the base station is the transmitter and the mobile station the receiver. For the reverse link the mobile station is the transmitter and the base station the receiver.

id: Implementation dependent

ii: Implementation independent

(a0) Average transmitter power per traffic channel (dBm)

The average transmitter power per traffic channel is defined as the mean of the total transmitted power over an entire transmission cycle with maximum transmitted power when transmitting.

(a1) Maximum transmitter power per traffic channel (dBm)

Maximum transmitter power per traffic channel is defined as the total power at the transmitter output for a single traffic channel. A traffic channel is defined as a communication path between a mobile station and a base station used for user and signalling traffic. The term traffic channel implies a forward traffic channel and reverse traffic channel pair.

(a2) Maximum total transmitter power (dBm)

Maximum total transmit power is the aggregate maximum transmit power of all channels.

(b) Cable, connector, and combiner losses (transmitter) (dB)

These are the combined losses of all transmission system components between the transmitter output and the antenna input (all losses in positive dB values). The value is fixed in the template.

(c) Transmitter antenna gain (dBi)

Transmitter antenna gain is the maximum gain of the transmitter antenna in the horizontal plane (specified as dB relative to an isotropic radiator). The value is fixed in the template.

(d1) Transmitter EIRP. per traffic channel (dBm)

This is the summation of transmitter power output per traffic channel (dBm), transmission system losses (–dB), and the transmitter antenna gain (dBi), in the direction of maximum radiation.

d2) Transmitter EIRP. (dBm)

This is the summation of the total transmitter power (dBm), transmission system losses (–dB), and the transmitter antenna gain (dBi).

(e) Receiver antenna gain (dBi)

Receiver antenna gain is the maximum gain of the receiver antenna in the horizontal plane (specified as dB relative to an isotropic radiator).

(f) Cable, connector, and splitter losses (receiver) (dB)

These are the combined losses of all transmission system components between the receiving antenna output and the receiver input (all losses in positive dB values). The value is fixed in the template.

(g) Receiver noise figure (dB)

Receiver noise figure is the noise figure of the receiving system referenced to the receiver input. The value is fixed in the template.

(h), (H) Thermal noise density, N_0 (dB(m/Hz))

Thermal noise density, N_0, is defined as the noise power per Hertz at the receiver input. Note that (h) is logarithmic units and (H) is linear units. The value is fixed in the template.

(i), (I) Receiver interference density I_0 (dBm/Hz)

Receiver interference density is the interference power per Hertz at the receiver front end. This is the in-band interference power divided by the system bandwidth. The in-band interference power consists of both co-channel interference as well as adjacent channel interference. Thus, the receiver and transmitter spectrum masks must be taken into account. Note that (i) is logarithmic units and (I) is linear units. Receiver interference density I_0 for forward link is the interference power per Hertz at the mobile station receiver located at the edge of coverage, in an interior cell.

(j) Total effective noise plus interference density (dBm/Hz)

Total effective noise plus interference density (dBm/Hz) is the logarithmic sum of the receiver noise density and the receiver noise figure and the arithmetic sum with the receiver interference density, i.e:

$$j \quad 10 \log (10^{((g \quad h)/10)} \quad I)$$

(k) Information rate (10 log R_b) (dB(Hz))

Information rate is the channel bit rate in (dB(Hz)); the choice of R_b must be consistent with the E_b assumptions.

(l) Required $E_b/(N_0 \quad I_0)$ (dB)

The ratio between the received energy per information bit to the total effective noise and interference power density needed to satisfy the quality (BER) objectives specified in section 10.1.2 under condition of channel model cases I-IV. Power control should not exceed the ceiling established by the sum of the log-normal fade margin plus hand-off gain. Diversity gains included in the $E_b/(N_0 \quad I_0)$ requirement should be specified here to avoid double counting. The translation of the threshold error performance to $E_b/(N_0 \quad I_0)$ performance depends on the particular multipath conditions assumed.

(m) Receiver sensitivity (j k l) (dBm)

This is the signal level needed at the receiver input that just satisfies the required $E_b/(N_0 \quad I_0)$.

(n) Hand-off gain/loss (dB)

This is the gain/loss factor (or –) brought by hand-off to maintain specified reliability at the boundary. Assume equal average loss to each of the two cells. The hand-off gain/loss shall be calculated for 50% shadowing correlation. The proponent must state explicitly the other assumptions made about hand-off in determining the hand-off gain.

(o) Explicit diversity gain (dB)

This is the effective gain achieved using diversity techniques. It should be assumed that the correlation coefficient is zero between received paths. Note that the diversity gain should not be double counted. For example, if the diversity gain is included in the $E_b/(N_0 \quad I_0)$ specification, it should not be included here.

(o) Other gain (dB)

An additional gain may be achieved due to future technologies. For instance, space diversity multiple access (SDMA) may provide an excess antenna gain. Assumptions made to derive this gain must be given by the proponent.

(p) Log-normal fade margin (dB)

The log-normal fade margin is defined at the cell boundary for isolated cells. This is the margin required to provide a specified coverage availability over the individual cells.

(q) Maximum path loss (dB)

This is the maximum loss that permits minimum RTT performance at the cell boundary:

Maximum path loss d1 – m (e – f) o o n – p

(r) Maximum range (km)

The maximum range is computed for each deployment scenario. Maximum range, R_{max}, is given by the range associated with the maximum path loss. The equations to determine path loss are given in the 802.16m channel models document.

3. Simulation Requirements

3.1 System Level Modeling

In order to accurately model the traffic, physical and MAC layer dependencies between the uplink (UL) and the downlink (DL), the system simulations should include both UL and the DL.

3.2 Simulation Flow and User Loading

The system simulation flow, required in this evaluation methodology, is illustrated in Figure 8. The simulation shall follow the following rules:

The system consists of 19 hexagonal cells (as shown in the Appendix B). Each cell has three sectors. Each mobile dropped corresponds to an active user session. A session runs for the duration of the drop. Mobiles are randomly assigned channel models. Depending on the simulation, these may be in support of a desired channel model mix, or separate statistical realizations of a single type of channel model. Users may be designated a load type (full buffer and best effort) or a probe type (users with specific QoS requirements).

The runs are done with an increment of two probe users per sector until a termination condition is met. By incrementing the number of probe users, system performance under a variety of traffic conditions is tested. This method provides greater insight into performance than provided by a single traffic mix. The process may be repeated for different kinds of probe users.

Mobile stations are randomly dropped over the 57 sectors such that each sector has the required numbers of probe users (QoS users) and load users. Although users may be in regions supporting handoff (i.e. either soft-handoff or hard handoff depending on the technology), each user is assigned to only one sector for counting purposes. All sectors of the system shall continue accepting users until the desired fixed number of probe and load users per sector is achieved everywhere. Users dropped within 35 meters of a sector antenna shall be redropped.

Fading signal and fading interference are computed from each mobile station into each sector, and from each sector to each mobile for each simulation interval.
The total simulation time per drop will be 5 minutes excluding any time required for initialization.

Packets are not blocked when they arrive into the system (i.e. queue depths are infinite).
Users with a required traffic class shall be modeled according to the appropriate sections in this document. Start times for each traffic type for each user should be randomized as specified in the traffic model being simulated.

The ARQ process (if proposed) is modeled by explicitly rescheduling a packet as part of the current packet call after a specified ARQ feedback delay period.
Results are collected from all cells according to the output matrix requirements.
All 57 sectors in the system shall be dynamically simulated.

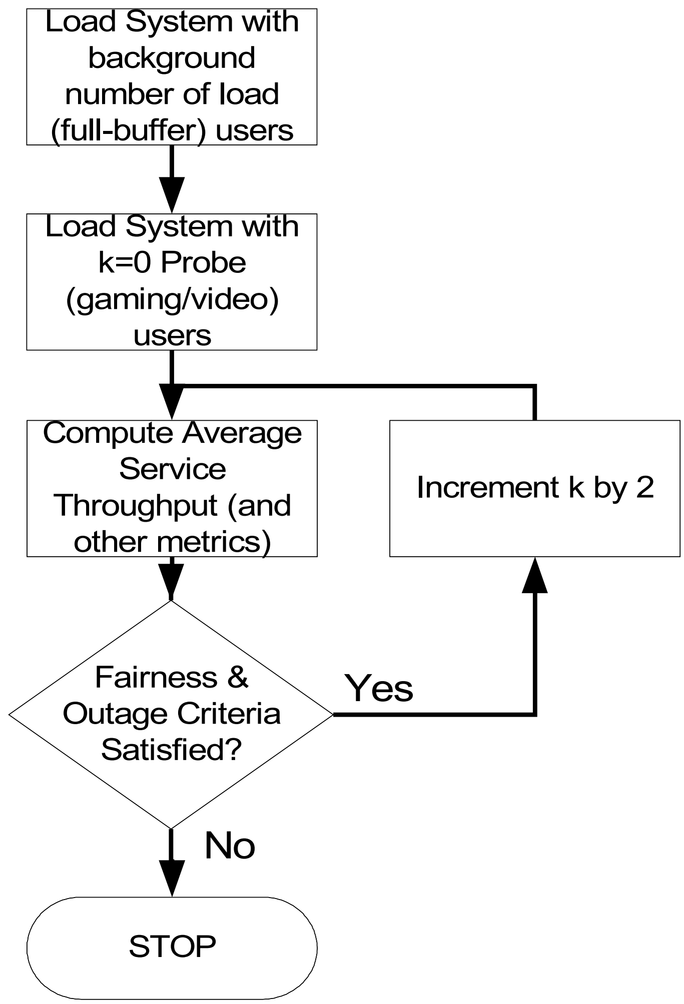

Figure 8: Simulation Flow Chart

3.3 System simulation with one mobile user

All terminals except one shall be fixed. The mobility related performance metrics shall be computed only for this mobile terminal.

Mobility Model: The movement of the single mobile terminal is constrained to one of the following paths (Figure 9). More detailed and realistic mobility models may be considered.

Path 1: Move from A to B along line joining the cells

Path 2: Move from A to B with "around the corner" effect that causes rapid signal loss from A, signal gain to B. (built into the propagation model)

Path 3: Move along cell edge. This path is symmetric (the mid-point of Path 3 is on Path 1)

Cells A and B are two cells in the center of the simulation region (cells 1 and 2 of the cell layout in the appendix). Details about the paths are provided in Table 6

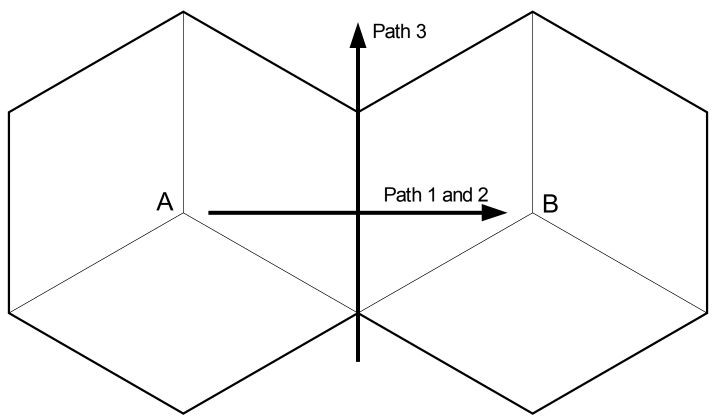

Figure 3: Path of Mobile in models 1, 2 and 3

The propagation seen in each of the models is shown in figures 10, 11 and 12. The curved lines in the figures include shadow fading, while the straight lines include only path loss. Mobility models 1 and 3 are computed using the path loss and shadowing parameters defined in other parts of the document. Mobility model 2 assumes that there is a sudden propagation loss of EdgeLoss dB as the terminal moves across the cell boundary. This stringent model is useful to test the robustness of handoff signaling.

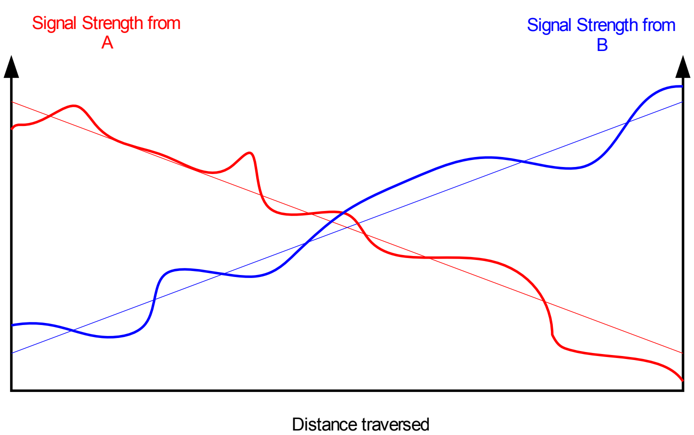

Figure 4: Propagation for Mobility Path 1

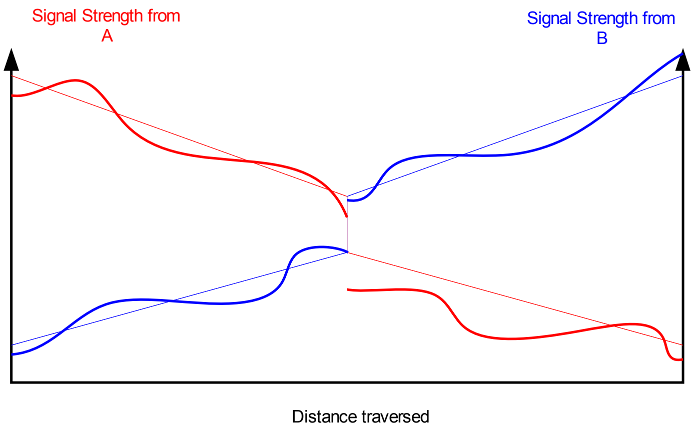

Figure 5: Propagation for Mobility Path 2

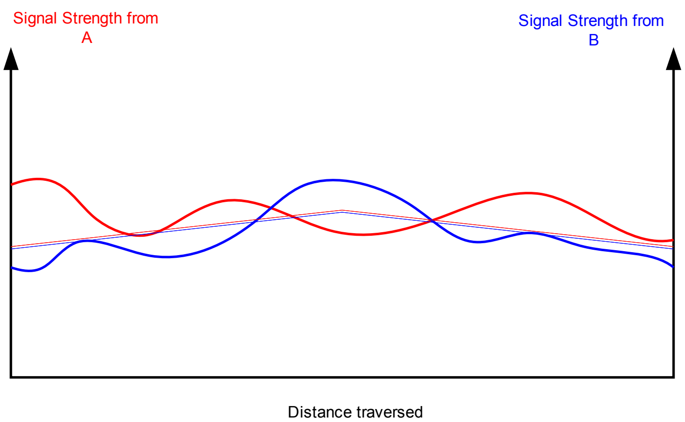

Figure 6: Propagation for Mobility Path 3

Table 16: Parameters for the Mobility Model

Parameter Name	Interpretation	Value
R	Distance between A and B	As in system simulation.
EdgeLoss	Sudden propagation loss at cell edge	3, 6, 9 dB
V	Mobile Speed	3, 30, 120 Km/h
D_{corr}	Shadow Fading Corr. Distance	30 m
D_0	Distance of starting point from A in paths 1 and 2. (same as distance of ending point from B)	30 m
D_3	Total distance covered by terminal in path 3	Same as R

3.4 Simulation Procedure for Mobility

For parameters such as cell size, terminal density and channel models, the simulation follows the simulation methodology defined elsewhere in the document.
For all channels relevant to the mobility scenario, a realistic channel load shall be simulated.

Simulation Realizations and Averaging: The simulation shall construct realizations using the following rules

Each realization shall consist of independent positions and channels for the stationary terminals

Each realization shall consist of independent channel realizations along the mobile terminal's path.

Internal random variables (if any) that govern any signaling event (such as exponential backoff) shall be drawn independently for each realization.

The performance metrics shall be obtained by averaging across each realization.

Mobile Channel Modeling: The channel for the mobile terminal shall be modeled by adding a fast fading model on top of a first order auto-regressive shadow fading model along the mobile terminal's path. The fast fading model shall use the mobile speed being studied.

The correlation distance D_{corr} shall be interpreted as follows. Let Z_a and Z_b be the shadow fading in dB at points 'a' and 'b' along a linear mobile trajectory, such that 'a' is *a* meters and 'b' is *b* meters away from the starting point. Then

$E[Z_a Z_b] = E[Z_a Z_a] \exp(-|a-b|/D_{corr})$

Note that in the equation above Z_a and Z_b are Gaussian random variables.

Results: Results shall be presented in the form of values of the Connected State Handoff Metrics, and the Power Save State metrics, as defined in the beginning of this section. Separate metric values shall be given for the paths 1, 2 and 3. Further, for path 3, results shall be given for all Edge Loss values given in Table 6.

3.5 Overhead Channels

Dynamical Simulation of the Forward Link Overhead Channels

Dynamically simulating the overhead channels is essential to capture the dynamic nature of these channels. The simulations shall be done as follows:

The performance of the overhead channels shall be included in the system level simulation results (unless the overhead channel is taken into account as part of fixed overhead). (For example, if an overhead channel is time division multiplexed, and takes all the bandwidth, the percentage of time used translates into the same percentage decrease of throughput.)

There are two possible types of overhead channels depending on the proposal: static and dynamic. A static overhead channel requires fixed base station power. A dynamic overhead channel requires dynamic base station power.

The link level performance should be evaluated off-line by using separate link-level simulations. The performance is characterized by curves of detection, miss, false alarm, and error probability (as appropriate) versus Eb/No (or some similar metric depending on the interface between the link and system simulations).

The system level simulations need not directly include the coding and decoding of overhead channels. There are two aspects that are important for the system level simulation: the required Ec/Ior (or some similar metric depending on the interface between the link and system simulations) during the simulation interval, and demodulation performance (detection, miss, and error probability — whatever is appropriate).

For static overhead channels, the system simulation should compute the received Eb/No (or similar metric).

For dynamic overhead channels with open-loop control (if used), the simulations should take into account the estimate of the required forward link power that needed to be transmitted to the mobile station for the overhead channels. During the reception of overhead information, the system simulation should compute the received Eb/No (or similar metric).

Once the received Eb/No (or similar metric) is obtained, then the various miss error events should be determined. The impact of these events should then be modeled. The false alarm events are evaluated in link-level simulation, and the simulation results shall be included in the evaluation report. The impact of false alarm, such as delay increases and throughput reductions for both the forward and reverse links, shall be appropriately taken into account in system-level simulation.
All overhead channels should be modeled or accounted for.

If a proposal adds messages to an existing channel (for example sending control on a data channel), the proponent shall justify that this can be done without creating undue loading on this channel. The system level and link level simulation required for this modified overhead channel as a result of the new messages shall be performed according to 3) and 4), respectively.

Reverse Link Modeling in Forward Link System Simulation
The proponents shall model feedback errors (e.g. power control, acknowledgements, rate indication, etc.) and measurements (e.g. C/I measurement). In addition to supplying the feedback error rate average and distribution, the measurement error model and selected parameters, the estimated power level required for the physical reverse link channels shall be supplied.

Signaling Errors
Signaling errors shall be modeled and specified as in the following table.

Table 17 Signaling Errors

Signaling Channel	Errors	Impact
ACK/NACK channel (if proposed)	Misinterpretation, missed detection, or false detection of the ACK/NACK message	Transmission (frame or encoder packet) error or duplicate transmission
Explicit Rate Indication (if proposed)	Misinterpretation of rate	One or more Transmission errors due to decoding at a different rate (modulation and coding scheme)
User identification channel (if proposed)	A user tries to decode a transmission destined for another user; a user misses transmission destined to it.	One or more Transmission errors due to HARQ/IR combining of wrong transmissions
Rate or C/I feedback channel (if proposed)	Misinterpretation of rate or C/I	Potential transmission errors
Transmit sector indication, transfer of H-ARQ states etc. (if proposed)	Misinterpretation of selected sector; misinterpretation of frames to be retransmitted.	Transmission errors

Proponents shall quantify and justify the signaling errors and their impacts in the evaluation report.

3.6 Link-System Interface (LSI)

An interface between link and system simulations is required because the link and system simulations are performed separately (the simulation complexity would be very high if joint link and system simulations are required).

Using the actual link curves is the default methodology for the link-system interface. The link curves can always be used. A technology specific methodology can be used if provided with full verification subject to the satisfaction of the group.

The link level simulation is to produce the statistical profile of the packet error as function of the measured C/I as well as other system design parameters. It should be based on detailed modeling of the relevant components in the transmitter and receiver.

The system level simulation is to capture the macroscopic statistical profile of the PHY, MAC and upper-layers performance for one cell or a cluster of cells. It should be based on the appropriate modeling as required by this document, and make use of the result produced by the link level simulation.

Because the foci of the link level simulation and the system level simulation are different, there is no need to repeat the same details in both simulation stages, so that the computer resource can be used effectively. While the link level simulation has to capture the transceiver specific details, the system level simulator is allowed to use a macroscopic model of the transceiver to determine the measured C/I, in order to reduce unnecessary simulation work. The caveat of such a model for the system level simulation is to consider the impact of certain details in the transceiver as noise sources, so that the computed C/I matches those used in the link level simulation.

3.7 System Simulation Calibration

The purpose of system-level calibration is to ensure that, under a set of common assumptions and models, the simulator platforms that will be used by various proponents can produce results that are similar. The calibration procedures and metrics specified in this section are intended to be as technology-independent as possible.

Simulation assumptions

The link budget defined in this document and the channel models adopted for use in the evaluation criteria should be used in the calibration.

Deterministic Calibration

The purpose of deterministic calibration is to assure that the basic configuration and layout of the simulation environment is coded in accordance with the evaluation criteria. The configuration of the simulation scenario is characterized by the following parameters:

Base Station (BS) to BS distance: 2R=2.5 km, or as otherwise defined, for several scenarios, in the system requirements document.

Path loss model: as specified in the channel model document for suburban macro

For the forward link, maximum C/I = 30 dB , where C/I is defined as the ratio of carrier traffic power to the total interference and noise power at the receiver

Mobiles are put in deterministic locations in each sector. For instance, 3 mobiles in each sector, located at (-60, R/2), (0,R/2) and (60, R), respectively, where (theta, r) refers to the polar coordinate-system of the sector with the reference direction (theta=0) being the antenna main lobe direction and the maximum radius of the cell being (r=R).

A single antenna is used for BS and for MS, respectively.
Results of C/I for each mobile are recorded in a spreadsheet. Cells are numbered according to the following scheme: a=index of the cell (0,1,2…,18) ,b=index of the sector (0, 1, 2, numbered clockwise from the upward position), c=index for location within the sector (0,1,2 in counter clock-wise, in case of 3 mobiles per sector)

Traffic Model Calibration
The traffic models shall be calibrated by plotting and comparing the CDF of packet sizes, inter-packet arrival times and related traffic model parameters.

3.8 Equipment Characteristics

3.8.1 Antenna Characteristics

Each proposal will specify its antenna characteristics, e.g. antenna pattern, number of antennas, antenna array geometry (if applicable), orientation, number of sectors.

3.8.2 Hardware Characteristics

The assumed hardware parameters of both the base station and the user terminals are necessary to interpret the quoted results. For example, differences in specification (both BS and UT) significantly affect performance results:
- maximum output power
- noise figures
- antenna gain, pattern, and height
- cable loss (if applicable).

A proposal shall include detailed information regarding the amplifier/s used in the simulation. The information shall be sufficiently detailed such that the claimed simulation results can be verified by others and that the practicality of the proposed amplifier arrangement is justified.

3.9 Deployment Characteristics

Information such as values of system-level parameters shall be provided to allow evaluation of the proposed technology in a typical deployment scenario. Relevant system-level parameters used for an 802.16m deployment include:
- number of carriers
- total spectral bandwidth
- system frequency allocation
- sectorization

3.10 Output Metrics

In this section, statistics for quantifying the aspects of network-level performance for full buffer user traffic models are described.

3.10.1 User data rate CDF for a fixed specified load and base station separation

Figure 7 shows a qualitative example of a cumulative distribution function (CDF) of the distribution of downlink data rates $D(N_u, S)$ in the interior sectors of a network for a specified load/coverage operating point (N_u, S) where (by definition) the number of full buffer users per sector is (N_u), and the (nearest neighbor) base station separation is (S). This distribution of data rates is taken on the *ensemble* of random placements of N_U full buffer active users in each sector of the network and all other stochastic input parameters.

Proponents shall provide at least one CDF of user data rates for a fixed load N_u and separation S to be specified by the proponent. CDF plots of user data rates shall be provided for both uplink and downlink user data rates.

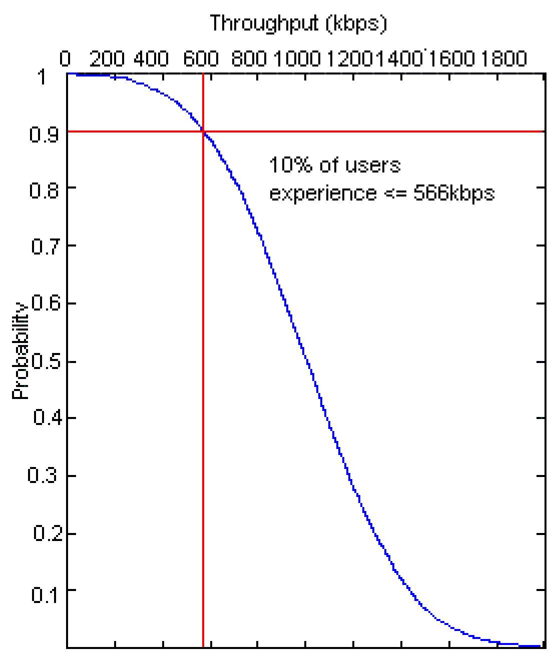

Figure 7: Service Distribution for a fixed load/coverage operating point

3.10.2 Aggregate Throughput vs Base Station Separation at Minimum Service Level

The downlink minimum service level at each load/coverage operating point, denoted $T_{DL}(N_u, S)$, is defined as the downlink per user data rate which is exceeded at least 80% of the time. .For example in Figure 7, 90% of the full buffer users will be served with a *minimum service level* of approximately 600 kbits/sec at the load/coverage operating point (N_u, S).

Proponents shall provide contour plots of constant downlink minimum service levels against full buffer users per sector versus base station separation.

Similarly the uplink minimum service level at each load/coverage operating point is defined as the uplink per user data rate which is exceeded at least 80% of the time.

Proponents shall provide contour plots of constant uplink minimum service levels against full buffer users per sector versus base station separation.

An example is shown in Figure 8. This example (produced for illustrative purposes), reveals the tradeoff between the base station separation (S) and the number of full buffer users per sector (N_u). For example, to guarantee an expected minimum service rate of, say, 1024 kbits/sec across 90% of the sector area, few full buffer users (less than 5) can be supported per sector at the inter-base station separation of 6 km. Conversely, many full buffer users per sector (more than 20) can be supported in the interference-limited case when the base stations are closely spaced.

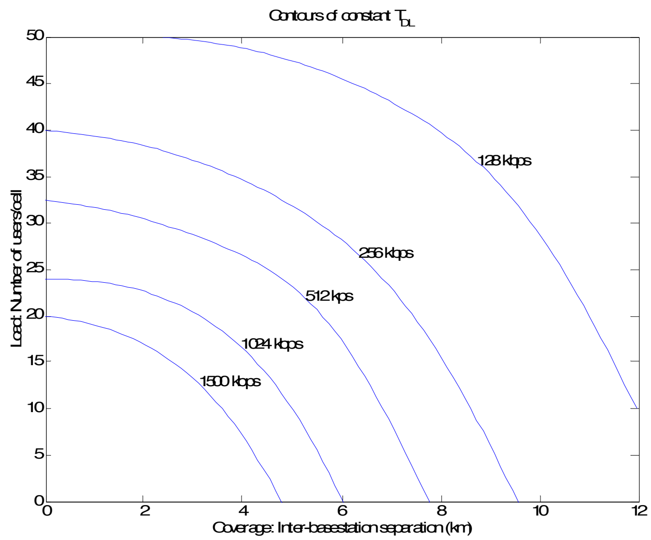

Figure 8: Contours of constant minimum service level

The downlink aggregate throughput per sector (A_{DL}) at each load/coverage operating point is the number of full buffer users per sector N_u multiplied by the mean user data rate of the pdf associated with CDF $D(N_u, S)$.

The proponent shall provide a plot of the downlink aggregate throughput (A_{DL}) versus base station separation for constant minimum service levels. As an example, the plot resembles Figure 8 with the vertical axis being aggregate throughput instead of number of users.

Similarly the proponent shall provide a plot of the uplink aggregate throughput (A_{UL}) versus base station separation for constant minimum service levels.

Note: provision of the complete contours is required to enable comparison of different proposals.

3.10.3 Computing Spectral Efficiency

Spectral efficiency figures shall be computed, separately, for the downlink (DL) and the uplink (UL) as follows:

Use the simulation results to calculate the average throughput per sector, by dividing the aggregate throughput across the 19-cell network, by 57. Then, divide the average throughput per sector by the total spectrum "deployed" in the network.

It is desirable that proposals provide plots of the downlink and uplink spectral efficiency versus base-station separation distance for constant minimum service levels.

4. Evaluation Report

The 802.16m evaluation report may be submitted as two separate reports. A summary of the items to be simulated and submitted in each report is defined in Table 18. The items marked with an 'X' are included in the corresponding report.

The goals of the first report are, first, to achieve confidence that different simulation models are calibrated and, second, to present fundamental performance metrics for the physical and link layer of various proposals.

The system level calibration shall follow the procedures described in this document.

Table 18 Evaluation Reports

Items		Evaluation Report 1	Evaluation Report 2
Link Level Simulation		X	
System simulations with 19 tri-sector cells layout		X	X
System Simulation calibration		X	-
Applications	Full Buffers	X	-
	Traffic Type Mix		X
Channel Models	Suburban macro, 3 Km/h pedestrian B[1], 100% (No channel mix)	X	X
	Suburban macro, 120Km/h Vehicular B, 100% (No channel mix)	X	X
	Link-level 250 Km/h suburban macro model and system level Channel Mix Models		X
Network delay and loss model			X
Mobility (i.e. Handoff) model			X
Overhead Channels model			X
RF characteristics		X	
Link Budget		X	-

[1] Link curves should be provided for the considered channel models

References

1. IEEE 802.20-PD-09 – “802.20 Evaluation Criteria v1.0”
2. IEEE 802.16m-07/002 version x – P802.16m Requirements Document
3. A Corlett, D.I. Pullin and S. Sargood, “Statistics of One-Way Internet Packet Delays,” *53rd IETF*, Minneapolis, March 2002.
4. D. Staehle et al, “Source Traffic Modeling of Wireless Applications,” Research Report, June 2000. available at: http://www3.informatik.uni-wuerzburg.de/TR/tr261.pdf
5. Mena and J. Heidemann, “An empirical study of real audio traffic,” INFOCOM 2000. Proceedings. IEEE , volume: 1 , 26-30 March 2000. pp 101 -110 vol.1
6. R. A. Bangun and E. Dutkiewicz, “Modelling multi-player games traffic,” Information Technology: Coding and Computing, 2000. Proceedings. International Conference on , 27-29 March 2000. pp: 228 –233
7. C. Heyaime-Duverge and V. K. Prabhu, “Modeling action and strategy Internet-games traffic Vehicular Technology Conference, 2002. VTC Spring 2002., Vol. 3 , 6-9 May 2002, pp: 1405 -1409 vol.3.
8. “3GPP TR 25.848 V4.0.0, Physical Layer Aspects of UTRA High Speed Downlink Packet Access”, March 2001.

Appendix A: Definition of terms

Number of Active Users per Cell
For the purposes of this analysis, an *active user* is a terminal that is registered with a cell and is using or seeking to use air link resources to receive and/or transmit data within the simulation interval. Evaluating service quality as a function of the well-defined concept of the number of active users per cell is a natural way of comparing how well disparate systems behave under increasing network load.

Inter-base Station Separation
For the purposes of defining network load, it is natural to treat inter-base station distance as a parameter. Closely spaced deployments will stress the interference-limited performance of the network while widely spaced deployments will stress the range-limited performance. In any case, users of an 802.16m system will likely experience different link quality at locations throughout the cell that depend both on the distance from the base station and the inter-base station separation. Thus, we include inter-base station separation in our definition of the load/coverage operating point.

One-Way Internet packet delay
One-way Internet packet delay is defined as the time it takes for an IP packet to travel from the base station (server) to the server (base station).

System Spectral Efficiency
System Spectral Efficiency is defined in the context of a full block assignment deployment and is calculated as the average aggregate throughput per sector (in bps/sector), divided by the spectrum block assignment size (in Hz) (excluding all PHY/MAC layer overhead).

Appendix B: 19-Cell Wrap-Around Implementation

In order to allow for data collection in all cells within the hexagonal network, it is necessary to extend the network to a cluster of network consisting of 7 copies of the original hexagonal network, with the original hexagonal network in the middle while the other 6 copies are attached to it symmetrically on 6 sides, as shown in Figure 9. The cluster can be thought of as 6 displacements of the original hexagon. There is a one-to-one mapping between cells/sectors of the center hexagon and cells/sectors of each copy, so that every cell in the extended network is identified with one of the cells in the central (original) hexagonal network. Those corresponding cells have thus the same antenna configuration, traffic, fading etc. except the location. The correspondence of those cells/sectors is illustrated in Figure 9.

An example of the antenna orientations in case of a sectorized system is defined in Figure 9. The distance from any MS to any base station can be obtained from the following algorithm: Define a coordinate system such that the center of cell 1 is at (0,0). The path distance and angle used to compute the path loss and antenna gain of a MS at (x,y) to a BS at (a,b) is the minimum of the following:

a. Distance between (x,y) and (a,b);
b. Distance between (x,y) and $(a \quad 3R, b \quad 8\sqrt{3}R/2)$;
c. Distance between (x,y) and $(a \quad 3R, b \quad 8\sqrt{3}R/2)$;
d. Distance between (x,y) and $(a \quad 4.5R, b \quad 7\sqrt{3}R/2)$;

e. Distance between (x,y) and $(a \quad 4.5R, b \quad 7\sqrt{3}R/2)$;

f. Distance between (x,y) and $(a \quad 7.5R, b \quad \sqrt{3}R/2)$;

g. Distance between (x,y) and $(a - 7.5R, b - \sqrt{3}R/2)$,

Where, R is the radius of a circle which connects the six vertices of the hexagon.

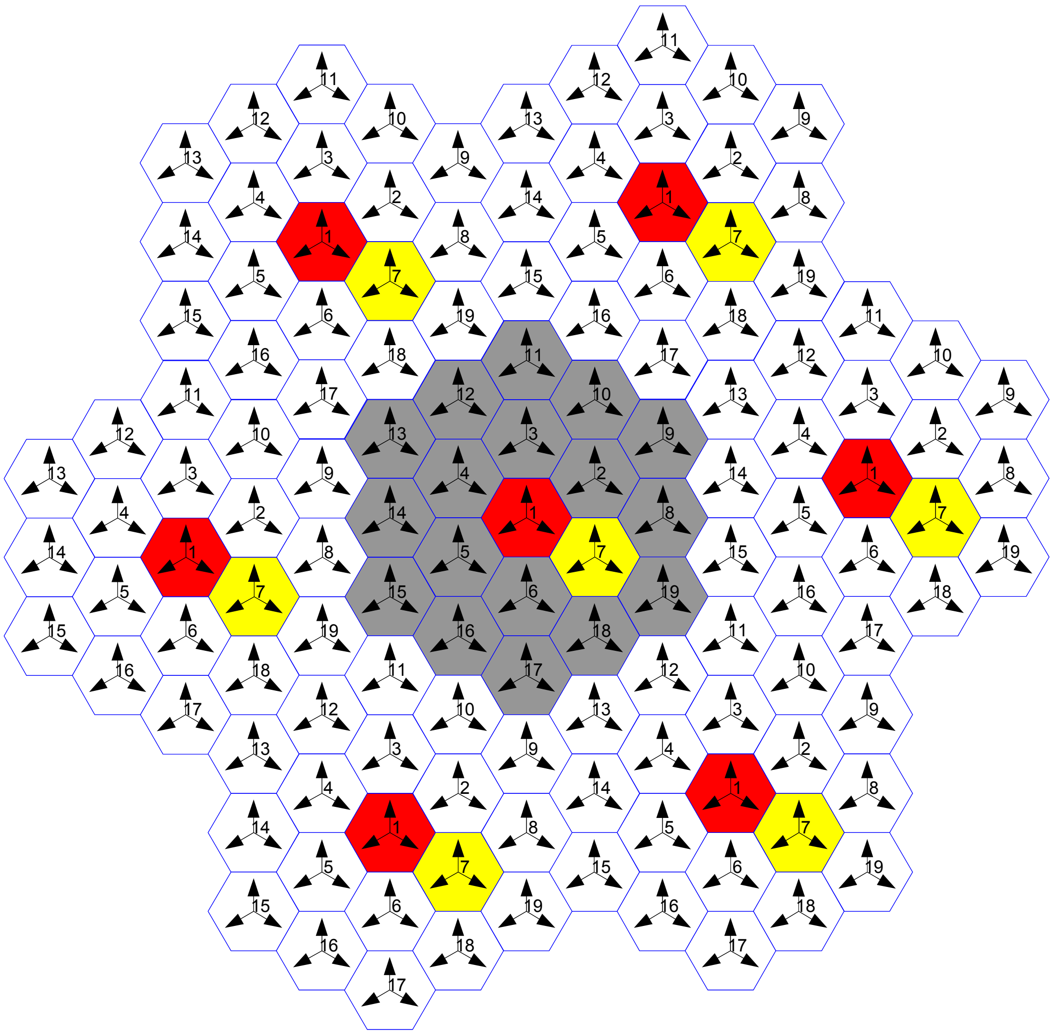

Figure 9: An example of the antenna orientations for a sectorized system to be used in the wrap-around simulation. The arrows in the Figure show the directions that the antennas are pointing.

Appendix C: Fixed user locations for system level calibration

In order to assure a fair and accurate comparison of technical proposals, it was proposed to calibrate the simulation tools, starting from a deterministic configuration []. For the deterministic simulation, it was proposed to use fixed but random dropped mobiles. An assignment to the author is to provide a list of such mobile locations. Locations for 10 mobiles per sector are generated for 19 cell sites, each with 3 sectors, and shown in the attached file below. The followings are some due explanations to the data.

C.1 Cell/Sector Locations

The inter-cell distance is 2.5 km and the cells are located as shown in the Figure 10:

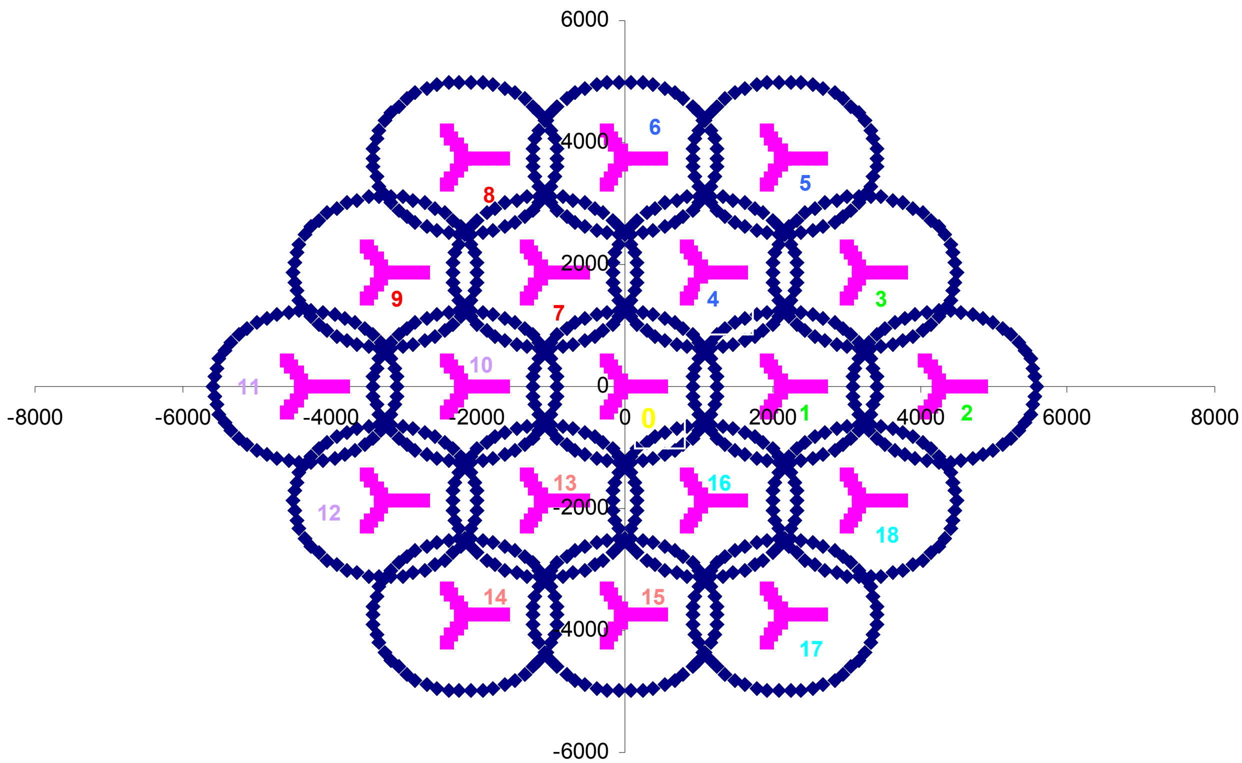

Figure 10: Cell definition in the Cartesian Coordination System and the Numbering of Cells

The rules used for numbering the cells are the following, in the given order:

Sector-wise starting from the sector one, which is the center cell. The sector of cells is numbered counter-clock wise, where a sector of cells is defined as those cells that are confined within an area between two radiation lines from the original with an angle of 60 degree.

Row-wise from inner to outer rows within each sector

Cell-wise from right to left on each row of cells.

Each cell is divided into 3 sectors, characterized by the antenna direction of each sector. The number of sector is counter-clock wise with 0, 1 and 2, respectively, where the respective antenna direction is

0: theta=0 degree,

1: theta=120 degree,

3: theta=240 degree,

where theta is the local polar angle of the cell. By this convention, the first sector of the center cell has the index (0, 0), while the last sector has the index (18,2). Mobiles are uniformly dropped in each sector, where an area around the cell center with radius 35 meters are excluded for mobiles. The unit of distance is meter.

C.2 Location Data

The generated locations are shown in the attached spreadsheet, where the names have the following meaning:

bs.id=index of the base stations as given above

l.sc.id=local sector identifier=sector index as given above

g.sc.id=global sector identifier=3*bs.id+l.sc.id

l.ms.id=local mobile station identifier

bs.loc.x=x-coordinate in Cartesian Coordination System of base station

bs.loc.y=y-coordinate in Cartesian Coordination System of base station

ms.loc.x=x-coordinate in Cartesian Coordination System of mobile station

ms.loc.y=y-coordinate in Cartesian Coordination System of base station

Random locations of 10 mobiles per sector for 57 sectors can be found in the embedded spreadsheet below:

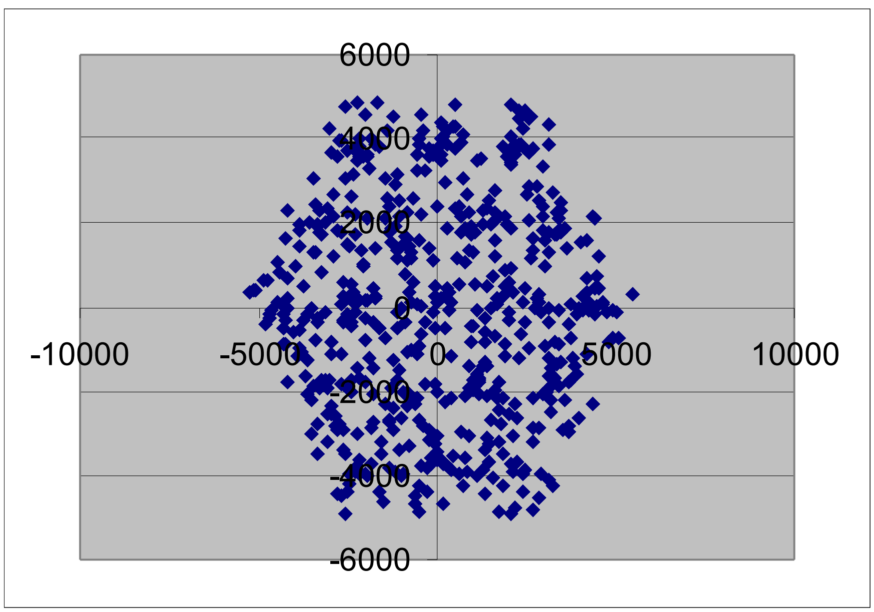

Figure 11: MS Locations