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Re:	Harmonized contribution from 802.16m Evaluation Methodology Drafting Group formed at TGm Session #48, March 2007, Orlando, FL
Abstract	This document contains proposed evaluation methodology for IEEE 802.16m technical proposals.
Purpose	For discussion and approval by TGm
Notice	
Release	
Patent Policy and Procedures	

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Abbreviations and Acronyms

3GPP	3G Partnership Project
3GPP2	3G Partnership Project 2
AAS	Adaptive Antenna System also Advanced Antenna System
ACK	Acknowledge
AES	Advanced Encryption Standard
AG	Absolute Grant
AMC	Adaptive Modulation and Coding
A-MIMO	Adaptive Multiple Input Multiple Output (Antenna)
ASM	Adaptive MIMO Switching
ARQ	Automatic Repeat reQuest
ASN	Access Service Network
ASP	Application Service Provider
BE	Best Effort
CC	Chase Combining (also Convolutional Code)
CCI	Co-Channel Interference
CCM	Counter with Cipher-block chaining Message authentication code
CDF	Cumulative Distribution Function
CINR	Carrier to Interference + Noise Ratio
CMAC	block Cipher-based Message Authentication Code
CP	Cyclic Prefix
CQI	Channel Quality Indicator
CSN	Connectivity Service Network
CSTD	Cyclic Shift Transmit Diversity
CTC	Convolutional Turbo Code
DL	Downlink
DOCSIS	Data Over Cable Service Interface Specification
DSL	Digital Subscriber Line
DVB	Digital Video Broadcast
EAP	Extensible Authentication Protocol
EESM	Exponential Effective SIR Mapping
EIRP	Effective Isotropic Radiated Power
ErtVR	Extended Real-Time Variable Rate
FBSS	Fast Base Station Switch
FCH	Frame Control Header
FDD	Frequency Division Duplex
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
FUSC	Fully Used Sub-Channel
HARQ	Hybrid Automatic Repeat reQuest
HHO	Hard Hand-Off

HMAC	keyed Hash Message Authentication Code
HO	Hand-Off
HTTP	Hyper Text Transfer Protocol
IE	Information Element
IEFT	Internet Engineering Task Force
IFFT	Inverse Fast Fourier Transform
IR	Incremental Redundancy
ISI	Inter-Symbol Interference
LDPC	Low-Density-Parity-Check
LOS	Line of Sight
MAC	Media Access Control
MAI	Multiple Access Interference
MAN	Metropolitan Area Network
MAP	Media Access Protocol
MBS	Multicast and Broadcast Service
MDHO	Macro Diversity Hand Over
MIMO	Multiple Input Multiple Output (Antenna)
MMS	Multimedia Message Service
MPLS	Multi-Protocol Label Switching
MS	Mobile Station
MSO	Multi-Services Operator
NACK	Not Acknowledge
NAP	Network Access Provider
NLOS	Non Line-of-Sight
NRM	Network Reference Model
nrtPS	Non-Real-Time Packet Service
NSP	Network Service Provider
OFDM	Orthogonal Frequency Division Multiplex
OFDMA	Orthogonal Frequency Division Multiple Access
PER	Packet Error Rate
PF	Proportional Fair (Scheduler)
PKM	Public Key Management
PUSC	Partially Used Sub-Channel
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RG	Relative Grant
RR	Round Robin (Scheduler)
RRI	Reverse Rate Indicator
RTG	Receive/transmit Transition Gap
rtPS	Real-Time Packet Service
RUIM	Removable User Identify Module
SDMA	Space (or Spatial) Division (or Diversity) Multiple Access

SF	Spreading Factor
SFN	Single Frequency Network
SGSN	Serving GPRS Support Node
SHO	Soft Hand-Off
SIM	Subscriber Identify Module
SINR	Signal to Interference + Noise Ratio
SISO	Single Input Single Output (Antenna)
SLA	Service Level Agreement
SM	Spatial Multiplexing
SMS	Short Message Service
SNIR	Signal to Noise + Interference Ratio
SNR	Signal to Noise Ratio
S-OFDMA	Scalable Orthogonal Frequency Division Multiple Access
SS	Subscriber Station
STC	Space Time Coding
TDD	Time Division Duplex
TEK	Traffic Encryption Key
TTG	Transmit/receive Transition Gap
TTI	Transmission Time Interval
TU	Typical Urban (as in channel model)
UE	User Equipment
UGS	Unsolicited Grant Service
UL	Uplink
UMTS	Universal Mobile Telephone System
USIM	Universal Subscriber Identify Module
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
VSF	Variable Spreading Factor
WiFi	Wireless Fidelity
WAP	Wireless Application Protocol
WiBro	Wireless Broadband (Service)
WiMAX	Worldwide Interoperability for Microwave Access

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Editor's Notes

This document is a harmonized contribution that has been developed from the following contributions submitted in the TGm meeting during Session #48:

Source	Document Reference
Draft IEEE 802.16m Evaluation Methodology Document (Sassan Ahmadi, Roshni M. Srinivasan, Hokyu Choi, Jeongho Park, Jaeweon Cho, DS Park, Louay Jalloul) 2007-03-05	IEEE C802.16m-07/069
Proposed text for P802.16m Evaluation Methodology and Key Criteria Document (Dan Gal et al) 2007-03-05	IEEE C802.16m-07/063
Proposed Text for Evaluation Methodology and Key Criteria for P802.16m (Robert Novak, Mo-Han Fong, Peiying Zhu, Gamini Senarath, Dean Kitchener, Wen Tong, Hang Zhang, David Steer, Derek Yu, Mark Naden, Jianglei Ma, Sang-Youb Kim) 2007-03-10	IEEE C802.16m-07/074r1
Draft Evaluation Methodology and Key Criteria for 802.16m - Advanced Air Interface (Mark Cudak, Ken Stewart, Amitava Ghosh, Fan Wang, Sassan Ahmadi, Roshni M. Srinivasan, Hokyu Choi, Jeongho Park, Jaeweon Cho, Shkumbin Hamiti, Carl Eklund, Louay Jalloul) 2007-03-05	IEEE C802.16m-07/061
Draft IEEE 802.16m Evaluation Methodology and Key Criteria for 802.16m – Advanced Air Interface (Wookbong Lee, Jinsoo Choi, Kiseon Ryu, Ronny (Yong-Ho Kim) 2007-03-15	IEEE C802.16m-07/075r1
Proposal of Simulation Evaluation Methodology for P802.16m (Liu Ying, Dong Xiaolu, Dang Meime) 2007-03-15	IEEE C802.16m-07/064r1

Major sections of this document have been edited by members of the drafting group. Contributions that have been harmonized and section editors are listed at the top of each section.

This document has been both color-coded and encoded using bracketed text. The color coding is used to identify input from the various contributions. At the top of each major section you will find a table assigning a color to particular contribution. Colors have been reused from one major section to another. We have attempted to give like authors the same color throughout the document; however, this was not possible in all cases. All colored-coded text is sourced from the contributions with only minor edits. Black text represents editor's proposed text.

In addition to the color coding, the drafting group has marked some text with brackets and left other text unbracketed. Square brackets [ ] identify text that requires further harmonization. This may include situations where the specified text is proposed for removal by one or more contributors or there are contradictory contributions related to that text.

1. Introduction

A great deal can be learned about an air interface by analyzing its fundamental performance in a link-level setting which consists of one base station and one mobile terminal. This link-level 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 through 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.

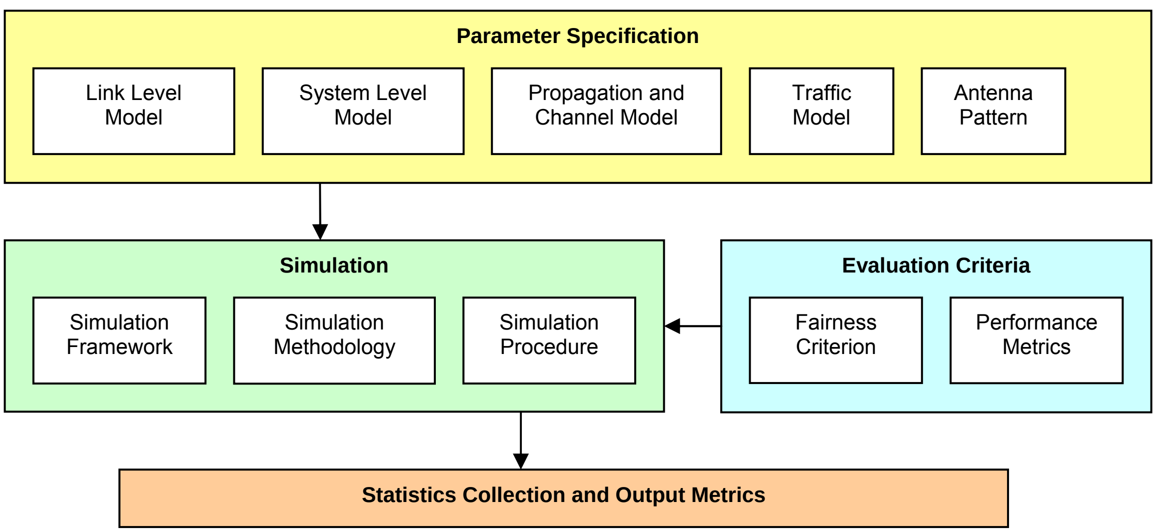

Figure 1-1: Simulation components

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.

The objective of this evaluation methodology is to define link-level and system-level simulation models and associated parameters that shall be used in the evaluation and comparison of technology proposals for IEEE 802.16m. Proponents of any technology proposal using this methodology shall follow the evaluation methods defined in this document and report the results using the metrics defined in this document.

Evaluation of system performance of a mobile broadband wireless access technology requires system simulation that accurately captures the dynamics of a multipath fading environment and the architecture of the air-interface. The main simulation components are illustrated in Figure 1-1.
System-Level Setup

2. System Level Set-up

Editor: Roshni Srinivasan, roshni.m.srinivasan@intel.com

Color	Sources	Document Reference
Green	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc

2.1. Antenna Pattern

The antenna pattern used for each sector is specified as:

$$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3\text{dB}}}\right)^2, A_m\right] \quad (2.1\text{-}1)$$

,
Where $A(\theta)$ is the antenna gain in dBi in the direction θ, $-180 \le \theta \le 180$, and min [.] denotes the minimum function, θ_{3dB} is the 3dB beam width (corresponding to θ_{3dB} = 70 degrees), and A_m= 20 dB is the maximum attenuation.

2.2. Antenna Orientation

The antenna bearing is defined as the angle between the main antenna lobe center and a line directed due east given in degrees. The bearing angle increases in a clockwise direction. Figure 2.2-1 shows an example of the 3-sector 120-degree center cell site, with Sector 1 bearing angle of 330 degrees. Figure 2.2-2 shows the orientation of the center cell (target cell) hexagon and its three sectors corresponding to the antenna bearing orientation proposed for the simulations. The main antenna lobe center directions each point to the sides of the hexagon. The main antenna lobe center directions of the 18 surrounding cells shall be parallel to those of the center cell. Figure

2.2-2 also shows the orientation of the cells and sectors in the two tiers of cells surrounding the central cell.

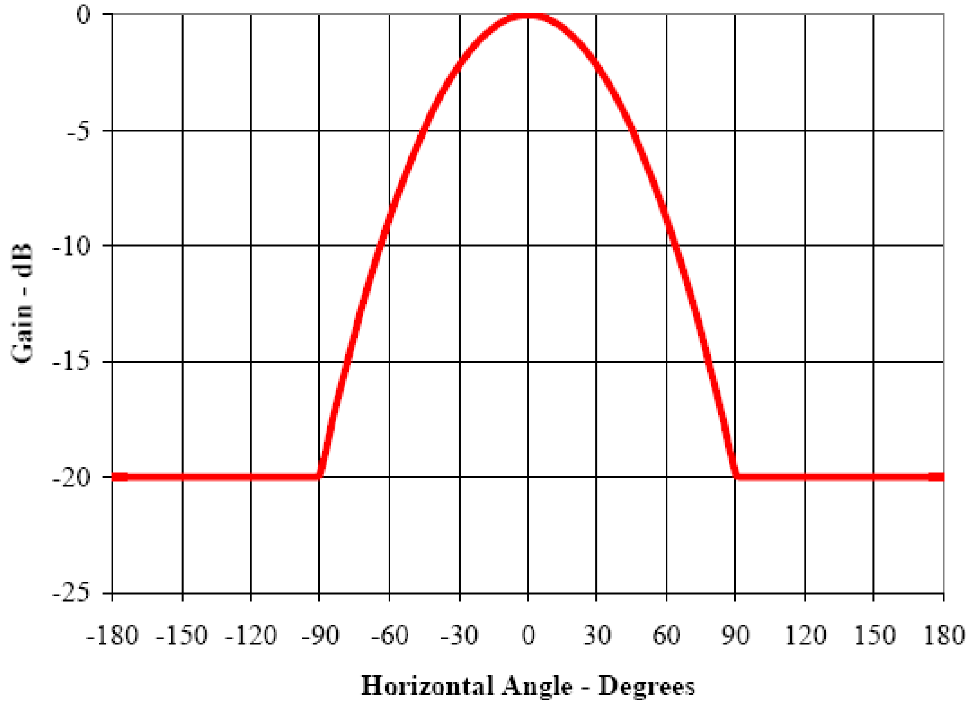

Figure 2.2-1: Antenna Pattern for 3-Sector Cells

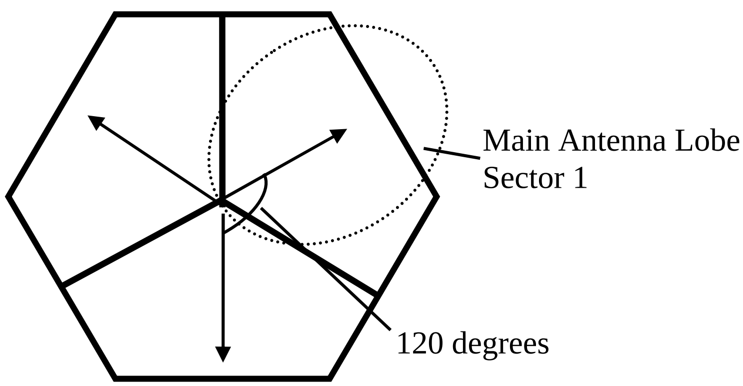

Figure 2.2-2: Centre cell antenna bearing orientation diagram

2.3. Simulation Assumptions

The purpose of this section is to outline simulation assumptions that proponents will need to provide in order to facilitate independent assessment of their proposals. The current table is a template that needs to be developed for a complete description of simulation assumptions. Wherever applicable, default parameters for comparison with the reference system as defined by the 802.16m requirements must also be included in these assumptions.

[

Topic	Assumptions
Basic modulation	TBD [QPSK, 16QAM, 64QAM]
Duplexing scheme	TBD [TDD and FDD]
Resource block definition	TBD
Downlink pilot structure	TBD
Receiver Structure	TBD [MMSE]
Data Channel coding	TBD [Convolutional turbo coding]
Multiple antenna configuration	TBD [2x2 antenna configuration (reference) Overhead calculations for uplink feedback should be shown Downlink overhead should be described and taken into account Channel estimation – should ideally be modelled or at least calculations showing the impact presented Control channels – transmission scheme provide impact on power overhead]
Scheduling	TBD - As required by mandatory traffic mixes
Link adaptation	TBD [Time-domain adaptation only Fixed power allocation]
PHY Abstraction for Link to System Mapping	TBD [MIESM, EESM]
H-ARQ	TBD [Asynchronous, non-adaptive HARQ overhead (associated control) should be accounted for in the system simulations]
Power Control	TBD [Fixed power for data Power control for control signalling - not needed to be considered explicitly in system simulations]
Interference Model	TBD [TBD]
Frequency Reuse	[1]
Control signalling	TBD [Overhead must be described and accounted for assuming at least 95% area coverage reliability.]

Table 2.3-1: System-level simulation assumptions for the downlink

Topic	Baseline for simulation, modelling assumptions
Basic modulation	TBD [QPSK, 16QAM, 64 QAM]
Duplexing scheme	TBD [TDD and FDD]
Resource block definition	TBD [TBD]

Data multiplexing	TBD [Data transmissions: Control transmissions: Control (CQI, ACK/NACK etc) sent in separate resource blocks]
Pilot structure	TBD [TBD]
Receiver Structure	TBD [MMSE]
Data channel coding	TBD [Convolutional turbo coding (CTC)]
Multiple antenna configuration	TBD [Antenna configuration: 1x2 (reference) If MIMO is included in uplink then collaborative MIMO is the preferred scheme Impact on downlink signalling overhead should be described Channel estimation - should be modelled or at least calculations showing the impact presented]
Random Access	TBD [Random access should be separately evaluated. Estimated overhead must be included in final performance estimates.]
Scheduling	TBD - As required by mandatory traffic mixes
Link adaptation	TBD [Base-station (scheduler) controls resources, adaptive modulation and coding] [Slow closed loop power control included, updated at most every 10ms]
H-ARQ	TBD [Asynchronous, non-adaptive HARQ overhead (associated control) should be accounted for in the system simulations]
Power Control	TBD [See link adaptation]
Interference Model	TBD
Frequency Reuse	TBD [1]
Control signalling	TBD [Overhead must be described and accounted for assuming at least 95% area coverage reliability.]

Table 2.3-2: System-level simulation assumptions for the uplink

2.4. Reference System Calibration

This purpose of this section is to provide guidelines for simulation parameters that proponents will need to use in order to evaluate performance gains of their proposals relative to the reference system as defined in the 802.16m requirements document.

2.4.1. Base Station Model

[

Parameter	Description	Value Range
$P1_{BS}$	BS power amplifier 1dB compression point	TBD [39-60 dBm]
PAR_{BS}	Peak-to-average backoff at BS	TBD [9-11 dB]
P_{BS}	MAX transmit power per sector/carrier	TBD [30-51 dBm TBD 43 dBm @ 5MHz bandwidth TBD **46 dBm @ 10MHz bandwidth** 49 dBm @ 20MHz bandwidth and similarly scalable for other bandwidths]
H_{BS}	Base station height	TBD [10-50m (**32 m**)]
G_{BS}	Gain (boresight)	TBD **[17 dBi]**
θ_{BS}	3-dB beamwidth as defined by 3GPP-3GPP2 [10]	TBD [$S = 3 : \theta_{BS} = 70^0$ $S = 6$:] $\theta_{BS} = 35^0$
G_{FB}	Front-to-back power ratio	TBD **[20 dB]**
M_{TX}	Number of transmit antennas	TBD [1,**2**,3,4]
M_{RX}	Number of receive antennas	TBD [1,**2**,3,4]
d_{BS}	BS antenna spacing (ref: ULA)	TBD [$\lambda / 2, 4\lambda, 10\lambda$]
ρ_{MS}	BS Antenna correlation	TBD [0.5]
NF_{BS}	Noise figure (transmit & receive)	TBD [4-6 dB (**5 dB**)]
HW_{BS}	Hardware loss (cable, implementation, etc.)	TBD **[2 dB]**

Table 2.4.1-1: BS equipment model

]

2.4.2. Mobile Station Model

[

Parameter	Description	Value Range

$P1_{SS}$	MS power amplifier 1dB compression point	TBD [29-54 dBm]
PAR_{SS}	Peak-to-average backoff at SS	TBD [9-11 dB]
P_{SS}	RMS transmit power/per SS	TBD [20-45 dBm (**23 dBm**)]
H_{SS}	Subscriber station height	TBD [1.5-7 m (**1.5 m**)]
G_{SS}	Gain (boresight)	TBD **[0 dBi]**
$\{\theta_{SS}\}, G(\{\theta_{SS}\})$	Table of Gains as a function of Angle-of-arrival	TBD [Omni]
N_{TX}	Number of transmit antennas	TBD [**1**,2]
N_{RX}	Number of receive antennas	TBD [1,**2**,3,4]
d_{SS}	SS antenna correlation	TBD [0-0.7 (**0.5**)]
	SS antenna gain mismatch	TBD [0-5 dB (**3 dB**)]
NF_{SS}	Noise figure (transmit & receive)	TBD [6-7 dB (**7 dB**)]
HW_{SS}	Hardware loss (cable, implementation, etc.)	TBD **[2 dB]**

Table 2.4.2-1: MS Equipment Model

]

2.4.3. OFDMA Numerology

[

Parameter	Description	Value Range
OFDMA symbol parameters		
BW	Total bandwidth	TBD [5, **10**, 20 MHz]
N_{FFT}	Number of points in full FFT	TBD [512, **1024**, 2048]
Δ_f	Subcarrier spacing	TBD [10.9375 kHz]
$T_S = 1/\Delta_f$	OFDMA symbol duration	TBD [91.43 us]
CP	Cyclic prefix length (fraction of T_S)	TBD [1/2, 1/4, **1/8**, 1/16]
T_O	OFDMA symbol duration w/ CP	TBD [102.86 us for CP=1/8]

Frame parameters		
T_F	Frame length	TBD [2, **5**, 10, 20 ms]
N_F	Number of OFDMA symbol in frame	TBD [18,**47**,95,193]
R_{DL-UL}	Ratio of DL to UL (TDD mode)	TBD [1:1, **2:1]**
T_{duplex}	Duplex time between UL and DL	TBD [0.67 to 20 ms]
T_{class}	Classification of traffic	TBD Control or Data
Permutation parameters		
DL_{Perm}	DL permutation type	TBD **[PUSC**, AMC, FUSC]
UL_{Perm}	UL permutation type	TBD **[PUSC**, AMC,]
BS_{Nused}	DL: number of sub-carriers for BS TX	TBD [For 10MHz, PUSC: $SS_{Nused} = 841$ FUSC: $SS_{Nused} = 851$ AMC: $SS_{Nused} = 865$]
SS_{Nused}	UL: number of sub-carriers for SS TX	TBD [PUSC/FUSC: $SS_{Nused} = 24$ AMC: $SS_{Nused} = 18$]
F_{sim}	Data sub-carriers explicitly simulated	
$SubCh_{MAX,DL}$	Maximum number of subchannels in DL permutation	TBD [AMC (48), **PUSC (30)**, FUSC (16)]
$SubCh_{MAX,UL}$	Maximum number of subchannels in UL permutation	TBD [AMC (48), **PUSC (35)]**

]

Table 2.4.3-1: OFDMA Air Interface Parameters

[The details of the OFDMA numerology are summarized in Table 2.4.3-2. These parameters are consistent with the IEEE 802.16e .numerology.

Transmission Bandwidth BW (MHz)	5	10	20
Over-sampling Factor n	28/25	28/25	28/25

Sampling Frequency Fs (MHz)	5.6	11.2	22.4
Sample time ($1/F_s$, nsec)	178	89	44.6
FFT Size	512	1024	2048
Number of Guard Sub-Carriers	Permutation Scheme Dependent	Permutation Scheme Dependent	Permutation Scheme Dependent
Sub-Carrier Spacing Δf (kHz)	10.9375	10.9375	10.9375
Useful OFDM Symbol Duration Tu = $1/\Delta f$ (us)	91.4	91.4	91.4
Cyclic Prefix Tg	OFDM Symbol Duration Ts Ts=Tu+Tg (us)	Number of OFDM Symbols per Frame N	Idle Time (us) Frame _Size – N*Ts
Tg=1/8 Tu	91.4 + 11.42=102.82	48	64.64

Table 2.4.3-2: OFDMA numerology for IEEE 802.16m

To compare the performance IEEE 802.16m candidate proposals to the IEEE 802.16e reference system, simulation parameters consistent with the reference system shall be utilized. These parameters are identified in the above table. Link-level and system-level simulation results shall be conducted and reported based on 10 and 20 MHz channel bandwidth using a center frequency of 2.5 GHz.]

3. Duplex Schemes

This document supports the evaluation of both TDD and FDD duplex schemes. The configuration of TDD and FDD duplexing schemes for link-level and system-level simulations is identical except when specified in the evaluation criteria e.g. DL/UL partition, TX/RX switching gaps, channel modeling considerations etc.

4. Channel Models

Editor: Jeff Zhuang, Jeff.Zhuang@motorola.com

Color	Sources	Document Reference
Green	Mark Cudak et al. Jeff Zhuang et al.	C80216m-07_061.pdf, C80216m-07_062.pdf,
Blue	Sassan Ahmadi et al. Sassan Ahmadi et al.	C80216m-07_069.doc, C80216m-07_070.doc
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al. Dean Kitchener et al.	C80216m-07_074r1.pdf C80216m-07_073
Violet	Liu Ting et al.	C80216m-07_064r1.doc
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc

4.1. Introduction

Channel models suitable for evaluation of 802.16m system proposals needs to be developed, taking into account parameters specific to 802.16m including bandwidths, operating frequencies, cell scenario(environment, cell radius, etc), and multi-antenna configurations.

The section gives an overview of channel modeling and high-level considerations to be made to adopt/modify/develop proper channel models for the performance evaluation of 802.16m systems. The section then suggests both link level and system level models with a focus on fulfilling the needs to conduct effective link- and system-level simulations that can generate trustworthy and verifiable results to assess performance related to the 802.16m system requirements.

4.1.1. General Considerations and Overview

The channel models defined in this document are to provide methodologies with sufficient details for the purpose of evaluating the system proposals to 802.16m. Since 802.16m is also targeting IMT-advanced, its system requirements, deployment scenario, and operational bandwidth and frequency should also be considered.

In the ITU-R recommendation ITU-R M.1645 the framework for systems beyond IMT-2000 (IMT-Advanced) shows data rates of up to 1Gbps for nomadic/local area wireless access, and up to 100Mbps for mobile access. As a reference, the European IST WINNER project has devised a method for determining spectrum requirements for IMT-Advanced, and their conclusions are given in [1]. In that report it is stated that in order to achieve the performance targets of IMT-Advanced, sufficiently wide frequency channels need to be provided (i.e. 100MHz channels per transmission direction). In addition, in order to enable several operators to deploy IMT-Advanced compliant systems (to enable competition), several RF channels of such bandwidths are required. Candidate bands for IMT-Advanced are to be considered in 2007 at the WRC-07 conference. When considering candidate bands, the WINNER report suggests that the bands below 3GHz do not have sufficient width to achieve the highest performance goals set for future systems. That reported also concludes that bands above 3GHz present significant challenges in particular for wide area mobile access, due to the increase in path loss with frequency, but is likely to be less of a problem for the shorter range nomadic local area wireless access.

It is noted that the terrain (outdoor/indoor, macro/micro/pico, etc.) dictates the channel modeling, affecting not only parameters but also possibly the methodologies.

4.1.2. Link Level Channel Modelling Considerations

A link level channel model is used mainly for calibrating point-to-point MIMO link performance at various SNR points of interest. Note that any particular link level channel modeling does not contain the information of large-scale fading or how often this kind of link condition occurs in a wireless system.

A link level channel can be developed for a typical propagation environment in a certain terrain and under a particular system setting (e.g., a macro-cell outdoor system with a representative BS and MS antenna configuration). A link-level channel modeling methodology that is consistent with the system level modeling methodology is desirable.

A model based on a conventional tapped-delay-line (TDL) can be considered. For example, three taps are used for the IEEE 802.16d SUI tapped delay line models [2], and six taps are used for the ITU models for IMT-2000 [3], and six taps including 3 or 4 mid-taps are used for the SCME (Spatial Channel Model Extensions) tapped delayed line models for beyond-3G systems [4]. The link level MIMO channels also need to be able to capture the relationship among all the channels between multiple transmit and receive antennas in a point-to-point link. A pre-defined set of antenna correlation matrices could be considered for link level (e.g., [11][12]).

A few important observations should be considered when adopting or modifying a TDL-based model with a defined antenna correlation:

1. The six-tap ITU models were developed for 5MHz bandwidth channels, and as the bandwidth increases, the resolution in the delay domain increases so that more taps are required for higher bandwidth channel models. Each resolvable tap consists of a number of multipath components so that the tap fades as the mobile moves. As bandwidth increases there will be less multipath components per resolvable tap so that the fading characteristics of the taps are likely to change. The tap fading is likely to become more Rician in nature (increasing K-factor with bandwidth) and the Doppler characteristics will not have the classic shape. This also means that the coherence times/distances for the tap fading will most likely be longer for higher bandwidths.

   The above observation suggests that measurement data under bandwidths up to 100MHz needs to be collected and analyzed to obtain the appropriate channel statistics which may vary according to transmission bandwidth. One of alternative solutions is SCME tapped delayed line model which adopts the concept of 'intra-cluster delay spread' to keep the tap resolution. (midpath power-delay profile based on the ITU model is FSS). Also, new simulation procedure and fader that are more flexible than existing Rayleigh fading generators may be required (e.g.[5][6][7][8]), which allow Rician fading or other user-defined Doppler characteristics. Therefore, new fading generators need to be developed and these need to be designed such that their complexity is manageable with a link level simulation (i.e. the run time must not be too high).

2. Multi-antenna correlation should include the potential effect of antenna imbalance with an appropriate branch power ratios (i.e. the ratio of received powers on different branches/antennas), and antenna coupling/correlations under different antenna configurations. The correlations can be calculated using spatial channel

models, such as the SCM from 3GPP [9], or the SCME from the WINNER project. SCME model considers wide bandwidth channels. Ideally the model would include azimuth and elevation (i.e., antenna tilt) angle for the multipath components, and also include polarization conversion in the channel. [Note that the correlation also depends on the location as it determines the mean AoA, and AoD.]

4.1.3. System Level Channel Modelling Considerations

Unlike link level modeling where short-term or small-scale fading behavior of a point-to-point link is modeled, for system level modeling, the large-scale location-dependent propagation parameters such as path loss and shadowing also need to be modeled, including possible relationship among multiple point-to-point links in a multi-user setting. There are in general two types of methodologies for consideration. The first is a physical model in which a propagation environment is modeled using multiple waves/rays parameterized according to the geometrical information. The physical modeling is independent of the antenna configuration, which means that the actual mathematical channel perceived by a receiver will have to incorporate the antenna configuration, traveling speed, trajectory, and velocity, and so on. The other modeling methodology is mathematical or analytical modeling in which the space-time channel as seen by the receiver is abstracted analytically, after factoring in the system and antenna parameters.

As an example of physical model, the 3GPP SCM model [9] has been used in system simulation. It models the physical propagation environment using paths and sub-paths with randomly specified angles, delays, phases, and mean powers. The mathematical MIMO channel for simulation is then derived after defining the antenna configuration and array orientation at both MS and BS. Time-variation is realized after defining MS travel direction and speed. Other ray-based channel models include SCME and WINNER channel model [10]. The ray-based physical models are powerful since they can reproduce truthfully the randomness as observed in measurement campaigns.

For simulation purpose, there are a few characteristics to be considered carefully:

- Simulation run-time. It may be computationally more expensive to generate the actual channel from the ray-based physical models in system simulation, as compared to the tap-delayed-line (TDL) model with fixed power delay profile and a Jakes Doppler profile (note that conventional fixed TDL models are for 5MHz bandwidth and Jakes Doppler is unlikely to be appropriate in wider bandwidth cases). This is due to the needs of summing all the rays with different delay, amplitude, and phase to generate each channel tap.
- Linkage to link-level simulations. In a system level simulation, people may want to use link-level performance results that are obtained under a non-spatial link model. For example, simple spatial extension of the GSM TU channels and ITU Ped-A/B and Veh-A/B channels are still widely used or in legacy results.
- Results convergence. If a model defines some second order statistics as random variable (e.g., angular spread, delay spread, etc.), the simulation may require more realizations and thus longer time to get convergence

4.2. Link Model Definition

A baseline link level channel model should be a tapped-delayed-line (TDL) with a multi-antenna correlation properly defined. An example of the baseline TDL model, in a macro-cell outdoor terrain, is the ITU channel models for the 5MHz bandwidth case. The antenna correlation could be derived from a certain antenna configuration assumption and a certain "typical" spatial parameters. The following sections describe the suggested TDL model and the derivation of antenna correlations.

4.2.1. TDL Models

A link-level channel model defines a specific number of paths, path delay and power profile, and Doppler frequencies for the paths. The tapped-delay line model defined by the ITU is suggested as the baseline TDL model for the 5MHz case. It defines the number of taps (or the number of paths), time delay relative to first tap, average power relative to the strongest tap, and Doppler spectrum of each tap.

The tapped delay line model can be represented in the time-domain as

$$h(t) = \sum_{l=1}^{n} p_l h_l(t)\delta(t - \tau_l) \qquad (4.2.1\text{-}1)$$

Where p_l and τ_l are amplitude and delay of path l, and $h_l(t)$ represents the time varying channel coefficient. All simulations assume that $h_l(t)$ is a temporally correlated random variable with classical (Jakes) Doppler spectrum

$$S(f) = \begin{cases} \dfrac{1}{\pi f_d} \dfrac{1}{\sqrt{1-(f/f_d)^2}} & |f| < f_d \\ 0 & \text{otherwise} \end{cases} \qquad (4.2.1\text{-}2)$$

Where f_d is the appropriate Doppler rate for the subscriber speed and the carrier frequency.

The tapped-delay line parameters of suggested ITU channel models are further summarized in

Table 4.2.1-1 and Table 4.2.1-2. Note that the power values in the tables need to be normalized so that they sum to unit power (0 dB).

Tap	Channel Ped-A $\tau_{rms} = 45ns$	Channel Ped-B $\tau_{rms} = 750ns$	Doppler $v = 3km/h$ and 10 km/h

	Delay (ns)	Power (dB)	Delay (ns)	Power (dB)	Jakes
1	0	0	0	0	Jakes
2	110	-9.7	200	-0.9	Jakes
3	190	-19.2	800	-4.9	Jakes
4	410	-22.8	1200	-8.0	Jakes
5	-	-	2300	-7.8	Jakes
6	-	-	3700	-23.9	Jakes

Table 4.2.1-1: Outdoor to indoor and pedestrian test environment channel impulse response

Tap	**Channel Veh-A** ($\tau_{rms} = 370ns$)		**Channel Veh-B** ($\tau_{rms} = 4000ns$)		**Doppler** $\left(v = 30, 120 \; km/h \right)$
	Delay (ns)	Power (dB)	Delay (ns)	Power (dB)	Jakes
1	0	0	0	0	Jakes
2	310	-1.0	300	-2.5	Jakes
3	710	-9.0	8900	-12.8	Jakes
4	1090	-10.0	12900	-10.0	Jakes
5	1730	-15.0	17100	-25.2	Jakes
6	2510	-20.0	20000	-16.0	Jakes

Table 4.2.1-2: Vehicular test environment channel impulse response

One example for SISO link-level channel model is given below based on ITU models at different velocities. The table is to be finalized to include stationary channel model. Note also that the stationary AWGN channel included could be used mainly for modeling a wired test condition only, not necessary reflecting the realistic stationary channel condition.

ITU Model	**Velocity (km/h)**
AWGN	0
Ped-A	3
Ped-B	{3,30}
Veh-A	{30,120,250}
Veh-B	{30,120,250}

Table 4.2.1-3: ITU Profiles for Link Level Simulations

4.2.2. TDL Models with Antenna Correlation

The MIMO link-level channel model is defined based on the SISO power delay profile, but a pre-defined antenna correlation should be specified on a per-tap basis.

The MIMO channel model is a stochastic channel model for MIMO systems that extends the SISO channel model to the MIMO case by utilizing transmit and receive spatial correlation matrices. Let M and N be the number of TX and RX antennas, respectively, and let $\mathbf{R}_{TX}$, of dimensions MxM, and $\mathbf{R}_{RX}$, of dimensions NxN, be the correlation matrices at the transmit and receive side, respectively. If **H** denotes the NXM discrete-time MIMO channel impulse response matrix between the transmitter and the receiver with entries $H_{n,m}$ expressing the channel impulse response between transmit antenna m, m=1..M, and receive antenna n, n=1..N, then the elements $[R_{\mathrm{Tx}}]_{i,j}$, i,j=1..M, of the MxM spatial correlation matrix $\mathbf{R}_{TX}$ are defined according to

$$[R_{\mathrm{Tx}}]_{i,j} = \langle H_{l,i}, H_{l,j} \rangle \tag{4.2.2-1}$$

where $\langle H_{l,i}, H_{l,j} \rangle$ calculates the correlation coefficient between $H_{l,i}$ and $H_{l,j}$ and is independent of l, l=1..N, i.e., of the receive antennas at the MS. The elements of the NxN correlation matrix $\mathbf{R}_{Rx}$ are defined similarly.

Since the correlation coefficients between any two channel impulse responses connecting two different sets of antennas can be expressed as the product of the correlation coefficients at the transmit and the receive antennas, the spatial correlation matrix of the MIMO channel matrix **H** can be expressed as the Kronecker product of the spatial correlation matrices at the transmit and receive side:

$$\mathbf{R}_{\mathrm{MIMO}} = \mathbf{R}_{\mathrm{Tx}} \otimes \mathbf{R}_{\mathbf{Rx}} = \mathbf{C}_{\mathrm{Tx}}^{*\mathrm{T}} \mathbf{C}_{\mathrm{Tx}} \otimes \mathbf{C}_{\mathrm{Rx}}^{*\mathrm{T}} \mathbf{C}_{\mathrm{Rx}} \tag{4.2.2-2}$$

where $\mathbf{C}_{Tx}$ and $\mathbf{C}_{Rx}$ represent the Cholesky decomposition of $\mathbf{R}_{Tx}$ and $\mathbf{R}_{Rx}$, respectively.

This property of the MIMO channel matrix **H** means that the effects of multipath propagation and mobility can be modeled by generating MxN uncorrelated channel impulse responses, each according to the SISO power delay profile (PDP) and the desired model for including the impact of mobility, e.g., use of the Doppler spectrum, and then for each multipath component w, w=1..W, determine the MIMO channel matrix according to

$$\mathbf{H}_w = \mathbf{C}_{\mathrm{Rx}}^{*\mathrm{T}} \mathbf{H}_{\mathrm{un},w} \mathbf{C}_{\mathrm{Tx}}^{*}, \quad w = 1..W, \tag{4.2.2-3}$$

where $\mathbf{H}_{un,w}$, w=1..W, denotes the NxM MIMO channel matrix created by the N·M uncorrelated channel impulse responses at delay w, w=1..W.

A per-tap antenna correlation matrix should be defined in the MIMO link level model. Such a correlation may be derived based on the per-tap mean angle of arrival (AOA), mean angle of departure (AOD), and a per-tap angular spread (AS) at both BS and MS. For example, using the per-tap AS parameters defined in SCM for urban macro-cell, one can realize a set of per-tap mean AOA and AOD so that the overall AS at BS and AS at MS achieve the values defined in SCM under different power delay profiles. Such an example is given below. The table defines a particular realization of the AOA and AOD for the sake of generating the antenna correlation for link level simulation. For system level simulation, the AOA and AOD certainly vary with locations. For link level simulation, other sets of values could be defined corresponding to different cell environments such as suburban macro, urban macro, and so on.

The antenna correlation can be further computed after assuming a certain antenna configuration. Other modifications are needed for polarized antenna, antenna imbalance, or high K-factor conditions.

Channel Scenario	Urban Macro-Cellular
AS at BS	$\sigma_{AS} = 15^0$
Per-path AS at BS (Fixed)	2 deg
AS at MS	$\sigma_{AS,\,MS} = 68^0$
Per-path AS at MS (fixed)	35^0
AoDs	As specified in Table 4.2.2-2
AoAs	As specified in Table 4.2.2-2

Table 4.2.2-1 ITU Profiles Spatial Extension Parameters

	Path Power	Path AOD (rad)	Path AoA (rad)
Ped-A	0.889345301	0.346314033	1.737577272
	0.095295066	-0.05257642	-1.55645
	0.010692282	-1.817837659	-1.049078459
	0.00466735	-0.836999548	0.345571431
Ped-B	0.405688403	-0.13638548	1.319340881
	0.329755914	0.302249557	-0.119072067
	0.131278194	0.496051618	0.901442565
	0.064297279	0.544719913	-1.424448314
	0.067327516	0.212670549	-3.062670939
	0.001652695	-0.604134536	-1.202289294
Veh-A	0.48500285	-0.46084874	-0.780118399
	0.385251458	-0.897480352	-1.729577654
	0.061058241	-0.525726742	1.792547973
	0.048500285	0.00282531	1.776985779
	0.015337137	-1.016095677	1.386034573
	0.004850029	0.245512493	3.50389557
Veh-B			

Table 4.2.2-2 Path Power, AoD, AoA

4.3. System Model Definition

For system level simulation, certain deployment related parameters need to be defined first that could relates to the channel modeling. For example in macro-cell environment, the cellular system contains three sectors. Antenna pattern and orientation are part of the deployment related parameters that will not be included here. The following sections describe the procedures to simulate MIMO channels in a wireless system and define propagation-related parameters, in sufficient details so that system level simulation can be performed.

A few environments should be considered for IEEE 802.16m system-level simulations:

1. **Suburban macro-cellular**: This scenario is characterized by large cell radius (approximately 1-6 km BS to BS distance), high BS antenna positions (above rooftop heights, between 10-80 m, typically 32 m), moderate to high delay spreads and low angle spreads and high range of mobility (0 – 350 km/h).
2. **Urban macro-cellular**: This scenario is characterized by large cell radius (approximately 1-6 km BS to BS distance), high BS antenna positions (above rooftop heights, between 10-80 m, typically 32 m), moderate to high delay and angle spread and high range of mobility (0 – 350 km/h).
3. **Urban micro-cellular**: This scenario is characterized by small cell radius (approximately 0.3 – 0.5 km BS to BS distance) BS antenna positions at rooftop heights or lower (typically 12.5m), high angle spread and moderate delay spread, and medium range of mobility (0 – 120 km/h). This model is sensitive to antenna height and scattering environment (such as street layout, LOS)]

4.3.1. Channel Mix

In system level simulation, users may be associated with a set of different channel types and velocities should such a case of mixed user speed is evaluated. A few examples are given below.

[If a TDL-based (e.g., ITU) channel with pre-defined antenna correlation is used as the baseline for also system level simulation, the channel models are randomly assigned to the various users according to the probabilities of Table 4.3.1-1 at the beginning of each drop and are not changed for the duration of that drop. The assignment probabilities given in Table 4.3.1-1 are interpreted as the percentage of users with that channel model in each sector. Note the table could be further optimized, especially the model-E (LOS or quasi-LOS) channel definition.

Channel Model	Multi-path Model	# of Paths	Speed (km/h)	Fading	Assignment Probability
Model A	Pedestrian A	4	3	Jakes	0.30
Model B	Pedestrian B	6	10	Jakes	0.30
Model C	Vehicular A	6	30	Jakes	0.20
Model D	Vehicular B	6	120	Jakes	0.10
Model E	Single Path	1	0, f_D=1.5 Hz	Rician Factor K = 10 dB	0.10

Table 4.3.1-1 Channel Models

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[If the SCM urban macro-cell channel model is used, the velocity profile is shown in Table 4.3.1-2 Because of the choice of urban macrocell, velocities are biased towards pedestrian speeds.

Percentage	Velocity (km/h)
35%	3
30%	30
20%	60
15%	120

Table 4.3.1-2: Quantized Velocity Profile

The RF carrier frequency for all link-level and system-level simulations shall be 2.5 GHz.

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[
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 4.3.1-3
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.

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

4.3.1-3: Assumptions on distribution of mobile user speed

]

4.3.2. Interference Channel Modelling

When the interference channels are needed in the system level simulation, the associated MIMO channel may or may not be computed depending on the context of system proposals. For example, in order to investigate the effect of beamforming performed in the neighboring cells to the desired user, it is necessary to simulate the interference MIMO channel. But in other cases, the total interference power is more important than the frequency or spatial selectivity of the interference channel. In this case, the channel between any interfering sector and the subscriber could be modeled as a one-path Rayleigh fading channel, where the Doppler of the fading process is randomly chosen based on the velocities.

4.3.3. Path Loss Model

The proposed path loss model is based on the well known COST-231 modified Hata model for systems supporting carrier frequencies less than or equal to 2.5GHz. The Erceg-Greenstein model [13] may also be used for carrier frequencies up to 3.5GHz. Path loss models applicable to carrier frequencies above 3.5GHz are for further study. Other potential path loss models that could be used are available in the WINNER report [10].

The site-to-site distance used in the simulation should be based on a path loss model and a link budget which closes for both uplink and downlink. As an example, the COST-231 modified Hata model (suburban mode) at 2.0GHz (L=126.2+36log10(R)) may be used when simulating the reference 802.16m system with carrier frequency of 2.5GHz. Other path loss models are need for different environments.

4.3.4. Spatial Channel Model

One of the key elements in channel modeling for 16m system proposal evaluation is how to extend the time-domain (TDL) channels to space and time. Note that the spatial and temporal characteristics of the propagation can often be related.

A physical model such as SCM may be considered for simulation purpose, for example, the parameters for urban macro-cell environment parameters is given in Table 4.3.4-1.

An analytical model may also be considered that can directly generate the mathematical MIMO channel matrix.

Channel Scenario	Urban Macro
Number of paths (N)	6
Number of sub-paths (M) per-path	20
Mean AS at BS	E(σ_{AS})=15 deg
AS at BS as a lognormal RV $\sigma_{AS} = 10 \wedge (\varepsilon_{AS} x + \mu_{AS}),\ x \sim \eta(0,1)$	15deg μ_{AS} = 1.18, ε_{AS} = 0.210
$r_{AS} = \sigma_{AoD} / \sigma_{AS}$	1.3
Per-path AS at BS (Fixed)	2 deg
BS per-path AoD Distribution standard distribution	$\eta(0,\sigma^2_{AoD})$ where $\sigma_{AoD} = r_{AS}\sigma_{AS}$
Mean AS at MS	E(σAS, MS)=68 deg
Per-path AS at MS (fixed)	35 deg
MS Per-path AoA Distribution	$\eta(0,\sigma^2_{AoA}(\mathrm{Pr}))$
Delay spread as a lognormal RV $\sigma_{DS} = 10 \wedge (\varepsilon_{DS} x + \mu_{DS}),\ x \sim \eta(0,1)$	μDS = -6.18 εDS = 0.18

Mean total RMS Delay Spread	E(σ_{DS})=0.65 μs
$r_{DS} = \sigma_{delays} / \sigma_{DS}$	1.7
Lognormal shadowing standard deviation, σ_{SF}	8.9dB
Path loss model (dB), d is in meters	28.6 + 35log10(d)

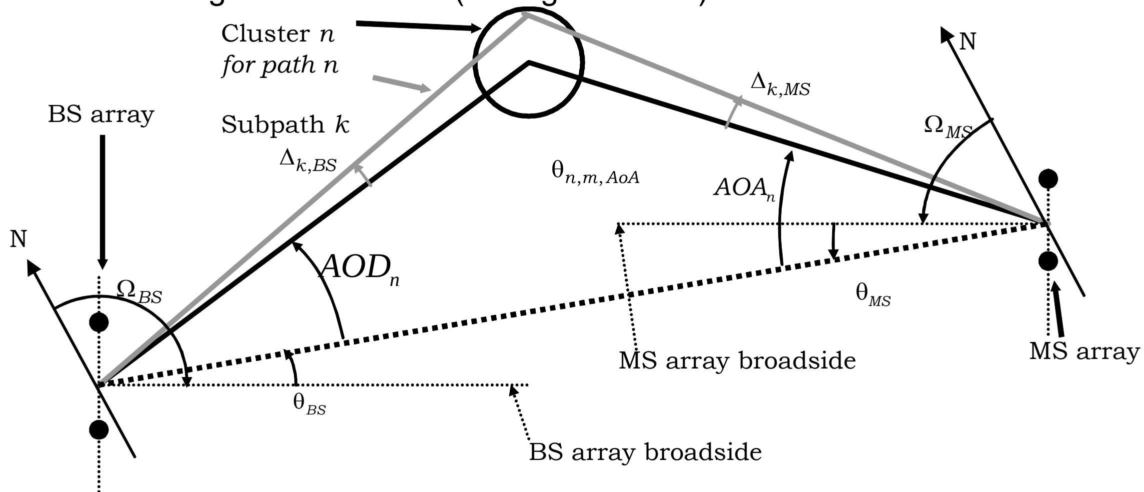
Table 4.3.4-1 Macro-cellular Environment Parameters

The following procedure describes an analytical channel modeling example of how to extend any TDL model to the spatial domain and how to derive the per-tap correlation, in a system level simulation setting. In addition, procedures on how to modify the channel to model correlated shadowing and angular spread, polarized antenna, antenna gain imbalance and coupling, high-K factor propagation, etc.

The procedure is built upon any pre-selected conventional SISO TDL model. A per-tap spatial correlation is then derived in closed-form based on predefined geometrical parameter (e.g., a fixed angular spread, antenna orientation, etc.) and the antenna configuration (spacing, cross-polarization, etc.). The sub-path concepts defined in SCM may be used, but just as one of the options for deriving the closed-form spatial correlation matrix, rather than for deriving the MIMO channel coefficients. The methodology used to extend a conventional TDL model to MIMO link-level model is the same as used to derive the MIMO link model for WiMAX Forum's RCT MIMO test [14][15].

Step 1: *Determine various distance and orientation parameters.*

The placement of the MS with respect to each BS is to be determined according to the cell layout. From this placement, the distance between the MS and the BS (*d*) and the LOS directions with respect to the BS and MS (θ_{BS} and θ_{MS} , respectively) can be determined. Note that θ_{BS} and θ_{MS} are defined relative to the broadside directions. The MS antenna array orientations (Ω_{MS}), are i.i.d., drawn from a uniform 0 to 360 degree distribution. (see figure below)



Step 2: *Calculate the bulk path loss associated with the BS to MS distance.*

A few pathloss models may be considered depending on operating frequency, such as Hata suburban, Erceg, etc. For example in SCM, the macrocell pathloss is based on the modified COST231 Hata urban propagation model:

$$PL[dB] = \left(44.9 - 6.55\log_{10} h_{bs}\right)\log_{10}\left(\frac{d}{1000}\right) + 45.5 + \left(35.46 - 1.1h_{ms}\right)\log_{10}(f_c) - 13.82\log_{10}(h_{bs}) + 0.7h_{ms} + C$$

where h_{bs} is the BS antenna height in meters, h_{ms} the MS antenna height in meters, f_c the carrier frequency in MHz, *d* is the distance between the BS and MS in meters, and *C* is a constant factor (*C* = 0dB for suburban macro and *C* = 3dB for urban macro). Setting these parameters to h_{bs} = 32m, h_{ms} = 1.5m, and f_c =1900MHz, the pathlosses for suburban and urban macro environments become, respectively, $PL = 31.5 + 35\log_{10}(d)$ and $PL = 34.5 + 35\log_{10}(d)$. The distance *d* is required to be at least 35m.

Step 3: *Determine the Shadowing Factor (SF).*

It is randomly generated from a log-normal distribution with a pre-specified standard deviation value. If desired, *SF* can be generated together with the BS angular spread (AS_{BS}) as described in app a, after enforcing a certain intra-site correlation between SF and AS, and an inter-site correlation of SFs.

Step 4: *Choose a well-defined power-delay-profile (PDP) (e.g., GSM TU/HT, ITU Ped-A/B, Veh-A/B) with a specified number of taps (N), their average powers (Pn,, n=1…N) and (τ_n ,n=1…N).*

It is also possible to use a user-defined PDP. For system with a bandwidth of more than 5MHz, a PDP with a larger number of taps may be considered (e.g., SCME is a simple extension of SCM to handle this case).
Average powers are normalized so that the total average power for all paths is equal to one:

$$P_n^{'} = \frac{P_n}{\sum_{n=1}^{N} P_n}.$$

Step 5: *Determine, for each path, the mean "AOD$_n$" (Angle-of-Departure at BS) and the mean "AOA$_n$" (Angle-of Arrival at MS).*

For the *N* <u>departure</u> paths, their mean AOD_n are determined as

$$AOD_1 = \theta_{BS}, \quad AOD_n = \theta_{BS} + \delta_n \quad n = 2 \ldots N$$

where first path (path with τ_1 =0) is assigned with a mean $AOD_1 = \theta_{BS}$ (i.e., LOS direction) and the rest of paths deviate from the LOS direction by some random values δ_n, where δ_n are ordered in increasing absolute value so that $|\delta_2| < |\delta_3| < \ldots < |\delta_N|$ and they are re-ordered from the random realizations of

$$\delta_n^{'} \sim \eta(0, \sigma_{AoD}^2) \quad , \quad n = 2,\ldots, N,$$

where $\sigma_{AoD} = r_{AS} AS_{BS}$ with AS_{BS} being the angular spread of the composite signal with all N paths (as opposed to the per-path $AS_{BS, Path}$) and the value r_{AS} is given in Table 1 (The table could be expanded to accommodate various PDP. Other values "r_{AS}" can be defined for different PDPs).

Table 4.3.4-2: Value of r_{AS} (TBD)

PDP	TU	PB	VA
r_{AS}	1.51	1.78	1.94

The quantity r_{AS} describes the distribution of powers in angle, i.e. the spread of angles to the power weighted angle spread. A value of r_{AS} greater than "1" indicates that there is more power being concentrated in paths that have a smaller AoD. r_{AS} is pre-calculated based on the per-path angular spread $AS_{BS, Path}$ and the PDP pre-defined in Step 4. When r_{AS} and $AS_{BS, Path}$ are chosen properly, the resulting AS of the composite signal should satisfy the mean value of AS_{BS} for the pre-specified PDP. The per-path AS can be chosen as (the choice of "2/5" (TBD) for per-path AS is to simply try to make the overall AS as close to the desired value as possible)

$$AS_{BS,Path} = \frac{2}{5} AS_{BS}$$

where in the simplest model we can define a fixed value for AS_{BS} (e.g., 2-degree, 5-degree, 15-degree), but it can also be generated randomly together with the Shadowing Factor (SF) as described in Appendix-A.

For the N arrival paths, their mean AOA_n are determined as

$$AOA_1 = \theta_{MS}, \quad AOA_n = \theta_{MS} + \Phi_n \quad n = 2 \ldots N$$

where first path (path with τ_1 =0) is assigned with a mean $AOA_1 = \theta_{MS}$ (i.e., LOS direction) and the rest of paths deviate from the LOS direction by some random values Φ_n. Φ_n (in degrees) are random realizations of the following Gaussian variables

$$\Phi_n \sim \eta(0, \sigma_{n,AoA}^2), \quad n = 2,\ldots, N,$$

where $\sigma_{n,AoA} = 104.12\left(1\text{-}\exp\left(\text{-}0.2175\left|10\log_{10}(P_n')\right|\right)\right)$ and $P_n^{'}$ is the mean power of the nth path specified in Step 4. Note that a uniform power angular profile over [0,360] degree will result in an angular spread of 104.12 degree (i.e., the $\sigma_{n,AoA}$ value at $P_n^{'}$=1). The per-path AOAs are taken from Gaussian distributions with different variances that are specified according to the average path power of each path, and there is no need to order the values. The per-path angular spread $AS_{MS, Path}$ is set as 35 degrees in SCM.

To compute the final AS_{BS} and AS_{MS} of the composite signal, refer to Appendix-B. If the value is not desirable, for example, for link level simulation purpose, a value as close to the specified value AS_{BS} (or AS_{MS}) as possible may be desired, this step can be repeated until a satisfactory value is obtained.

Step 6: *Calculate the per-path spatial correlation matrix based on per-path $AS_{BS,\,Path}$, $AS_{MS,\,Path}$, AOD_n, AOA_n, and BS/MS antenna configurations (broadside direction, number and spacing of antennas, polarization., etc.)*

Once the per-path AS, mean AOA, and mean AOD are defined, the theoretical spatial correlation at both BS and MS can be derived, assuming Laplacian power angular distribution. In particular, the antenna spatial correlations between the p-th and q-th antenna at the BS and MS, respectively, are

$$r_{n,BS}(p,q) = \int_{-\infty}^{\infty} p(\alpha)\exp\left\{ j\frac{2\pi d_{BS}}{\lambda}(p-q)\sin(AOD_n+\alpha)\right\} d\alpha$$
$$r_{n,MS}(p,q) = \int_{-\infty}^{\infty} p(\beta)\exp\left\{ j\frac{2\pi d_{MS}}{\lambda}(p-q)\sin(AOA_n+\beta)\right\} d\beta$$

where d_{BS} (d_{MS}) is the antenna spacing at BS (MS) and λ is the wavelength. α is the angular offset around the mean AOD at BS, and β is the angular offset around the mean AOA at MS. The pdf of angular offsets is

$$p(\alpha) = \frac{1}{\sqrt{2}AS_{BS,Path}}\exp\left\{-\frac{\sqrt{2}|\alpha|}{AS_{BS,Path}}\right\}$$
$$p(\beta) = \frac{1}{\sqrt{2}AS_{MS,Path}}\exp\left\{-\frac{\sqrt{2}|\beta|}{AS_{MS,Path}}\right\}$$

The above integration can be computed with two approaches (other alternatives may also exist). See Appendix-C for details. In summary, the first approach is to approximate the Laplacian PDF with 20 subpaths, after which the integration is reduced to a summation. The second approach is to pre-compute the integration using a numerical method. Since the integration depends on mean AOA and AOD, it is possible to quantize them and then pre-compute the integration for each quantized AOA and AOD values.

Denoting the spatial correlation matrix at BS and MS as $\mathbf{R}_{BS,n}$ and $\mathbf{R}_{MS,n}$, the per-path spatial correlation is determined as

$$\mathbf{R}_n = \mathbf{R}_{BS,n} \otimes \mathbf{R}_{MS,n} \text{(Kronecker product)}$$

In the case that the antenna elements are cross-polarization antennas, the per-path channel correlation is determined as

$$\mathbf{R}_n = \mathbf{R}_{BS,n} \otimes \mathbf{\Gamma} \otimes \mathbf{R}_{MS,n}$$

where $\mathbf{\Gamma}$ is a cross-polarization matrix defined in Appendix-D.

Step 8: *Determine the antenna gains of the BS and MS paths as a function of their respective AoDs and AoAs. Calculate the per-path average power with BS/MS antenna gain as*

$$P_n'' = P_n'' * G_{BS}(AOD_n) * G_{MS}(AOA_n)$$

Step 9: *Generate time-variant MIMO channels with above-defined per-tap spatial correlations*.
For each tap, generate M_{BS}xM_{MS} *i.i.d.* channels first assuming Jakes Doppler spectrum $\mathbf{H}_{iid}$ (each tap is a M_{BS}xM_{MS} matrix).

Compute the correlated channel at each tap as

$$\mathbf{H}_n = unvec\left\{R_n^{1/2} vec(H_{iid})\right\}$$

where *vec(H)* denotes the column-wise stacking of matrix H and *unvec* is the reverse operation. $Rn^{1/2}$ denotes the square-root of matrix R.

Step 10: *If a non-zero K-factor wants to be enforced (i.e., K≠0), adjust the LOS path power*
This step is not needed if a K-factor wants to be enforced (see Appendix -E)

Step 11: *Introduce receive antenna gain imbalance or coupling, if need.*
See Appendix-F.

5. Link-to-System Mapping

Editor: Louay Jalloul, jalloul@beceem.com

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Alcatel-Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc
Green	Shiang-Jiun Lin et al	C802.16m-07/059r1

5.1. PHY Abstraction

The objective of a PHY abstraction is to accurately predict link layer performance in a computationally simple way. The requirement for an abstraction stems from the fact that simulating the physical layer links between multiples BSs and MSs in a network/system simulator can be computationally prohibitive. The abstraction should be accurate, computationally simple, relatively independent of channel models, and extensible to interference models and multi-antenna processing.

5.1.1. Background

In the past, system level simulations characterized the average system performance, which was useful in providing guidelines for system layout, frequency planning etc. For such simulations, the average performance of a system was quantified by using the topology and macro channel characteristics to compute a geometric (or average) SINR distribution across the cell. Each subscriber's geometric SINR was then mapped to the highest modulation and coding scheme (MCS), which could be supported based on link level SNR tables that capture fast fading statistics. The link level SNR-PER look-up tables here served as the PHY abstraction for predicting average link layer performance. Examples of this static methodology may be found in [16]-[17].

Current cellular system design is based on exploiting instantaneous channel conditions for performance enhancement. Channel dependent scheduling and adaptive coding and modulation are examples of channel-adaptive schemes employed to improve system performance. Therefore, current system level evaluation methodologies are based on explicitly modeling the dynamic system behavior by including fast fading models within the SLS. Here the SLS must support a PHY abstraction capability to accurately predict the instantaneous performance of the link layer.

5.1.2. Dynamic PHY Abstraction Methodology

In system level simulations, an encoder packet may be transmitted over a selective channel. For example, OFDM systems may experience frequency selective fading, and hence the channel gains of each sub-carrier may not be equal. In OFDM, the coded block is transmitted over several sub-carriers and the post-processing SINR values of the pre-decoded streams are thus non-uniform. Additionally, the channel gains of sub-carriers can change in time due to fading process and possible delay involved in HARQ re-transmissions. The result on a transmission of a large encoder packet is encoded symbols of unequal signal-to-noise ratios at the input of the decoder due to the selective channel response over the encoder packet transmission. PHY abstraction methodology for predicting instantaneous link performance for OFDM systems has received considerable attention in the standards literature [18]-[27]. The role of a PHY abstraction is to predict coded BER/PER given a received channel realization across the OFDM sub-carriers used to transmit the coded FEC block. To predict coded performance, the post-processing SINR values at the input to the FEC decoder are considered as input to the PHY abstraction mapping. As the link level curves are generated assuming a flat channel response at given SNR, an effective SNR, SNR_{eff} is required to accurately map the system level SNR onto the link level curves to determine the resulting probability of packet error. This mapping is termed *effective SIR mapping (ESM).* The ESM PHY abstraction is thus defined as compressing the vector of SINR values to a single effective SINR value, which can then be further mapped to a BLER/PER number.

Several ESM PHY abstractions to predict the instantaneous link performance have been proposed in the literature. Examples include mean instantaneous capacity ([18]-[20]), exponential-effective SINR (EESM, [21],[23]-[25]) and Mutual information effective SINR (MIESM, [26],[27]). Each of these PHY abstractions uses a different function to map the vector of SINR values to a single number. Given the instantaneous EESM SINR, mean capacity or mutual information SINR, the PER (block error rate) for each MCS is calculated using a suitable mapping function for each MCS scheme.

In general, the ESM can be described as follows,

$$SINR_{eff} = \Phi^{-1}\left\{\frac{1}{N}\sum_{n=1}^{N}\Phi\left(SINR_n\right)\right\} \qquad (1)$$

where *N* is the number of symbols in a coded block, and $\Phi(\bullet)$ is a invertible function.

In the case of the mutual information and capacity based ESM the function $\Phi(\bullet)$ is derived from the constrained capacity. While in the case of EESM, the function $\Phi(\bullet)$ is derived based on the Chernoff bound for the probability of error. In the next three sections, we describe these ESM methods.

5.1.2.1. Mutual Information/Capacity ESM

Given a set of *N* received encoder symbol SINRs from the system level simulations, denoted as $SINR_1$, $SINR_2$, $SINR_3$, …, $SINR_N$, the mutual information per symbol is computed as a function of the *M*-ary modulation scheme, with $m = \log_2 M$ bits/symbol, as follows:

$$SI\left(SINR_i, m(i)\right) = E_{XY}\left(\log_2\left(\frac{P\left(Y \mid X, SINR_i\right)}{\sum_X P\left(X\right)P\left(Y \mid X, SINR_i\right)}\right)\right) \qquad (2)$$

In the above, $Y = \sqrt{\rho}X + U$ is the received symbol, $P(Y \mid X, SINR)$ is the channel transition probability density conditioned on the input symbol *X* for a given $SINR = \rho$. It is assumed that the transmitted symbols are equally likely, thus $P\left(X\right) = 1/M$, where *M* is the size of the modulation alphabet and that *U* is modeled as zero mean complex Gaussian with variance $1/2$ per component. The mutual information curves $SI\left(SINR, m\right)$ are generated, and stored once in the system simulator, for each of the modulation and coding formats.

Assuming *N* sub-carriers are used to transmit a coded block, the received mutual information over a coded block is computed as:

$$RBI(i) = \sum_{i=1}^{N} SI\left(SINR_i, m(i)\right) \qquad (3)$$

We note that the even though we refer to the coded block being carried over a set of sub-carriers, in general, the code block may be carried over multiple dimensions, including the spatial dimensions available with MIMO. Also note that in the above, the mutual information may be computed even with non-uniform modulation across the coded block. Next we compute the normalized mutual information per received bit which is given by

$$RBIR(i) = \frac{\sum_{i=1}^{N} SI\left(SINR_i, m(i)\right)}{\sum_{i=1}^{N} m(i)} \tag{4}$$

The advantage of computing the *RBIR* is that the relationship between the *RBIR* and the PER is dependent only on the AWGN curves for the code rate and is independent of the modulation scheme. This feature is a very useful in computing the PHY abstraction for cases where the coded block comprises of mixed modulation symbols.

We further note that an adjustment parameter, $SINR_{adjust}$, may be specified with the RBIR metric that can accounts for deviations of the *RBIR* mapping with respect to the AWGN curves. This adjustment parameter is illustrated by the following equation.

$$RBIR(i) = \frac{\sum_{i=1}^{N} SI\left(SINR_i / SINR_{adjust}, m(i)\right)}{\sum_{i=1}^{N} m(i)} \tag{5}$$

The exact specification of this adjustment parameter is determined through simulations and shall be specified along with the PHY abstraction tables, once the 802.16m numerology is specified.

The pervious computation of the mutual information per coded bit, RBIR, was derived from the symbol-level mutual information. Next we present an alternate method that directly arrives at the bit-level MI. This method will be demonstrated via an example.

[The mutual information of the coded bit is dependent on the actual constellation mapping. The MI of each bit-channel is obtained and averaged across the bits in a QAM symbol. After encoding (e.g. Turbo or CTC), a binary coded bit stream c_k is generated before QAM mapping. The QAM modulation can be represented as a labeling map $\mu : \mathrm{A} \to X$, where A is the set of m-tuples, $m \in \{2,4,6\}$ to represent QPSK, 16 and 64-QAM, of binary bits and X is the constellation. Given the observation y_n corresponding to the n^{th} QAM symbol in a codeword, the demodulator computes the log-likelihood ratio (LLR) $LLR(b_{i,n})$ of the i^{th} bit comprising the symbol via the following expression (where the symbol index n is dropped for convenience)

$$LLR(b_i) = \ln\left(\frac{P(y \mid b_i = 1)}{P(y \mid b_i = 0)}\right) \tag{6}$$

When the coded block sizes are very large in a bit-interleaved coded system (BICM), the bit interleaver effectively breaks up the memory of the modulator, and the system can be represented as a set of parallel independent bit-channels [29]. Conceptually, the entire encoding process can be represented as shown in Figure 5.1.2.1-1.

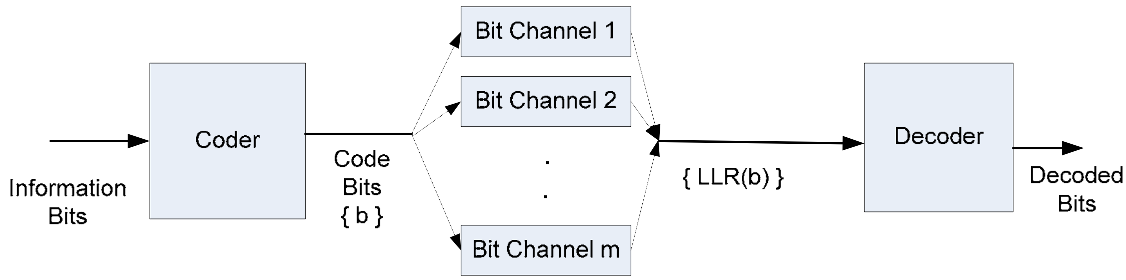


Figure 5.1.2.1-1: Bit-Interleaved Coded Modulation system.

Due to the asymmetry of the modulation map, each bit location in the modulated symbol experiences a different 'equivalent' bit-channel. In the above model, each coded bit is randomly mapped (with probability $1/m$) to one of the m bit-channels. The mutual information of the equivalent channel can be expressed as:

$$I(b, LLR) = \frac{1}{m}\sum_{i=1}^{m} I(b_i, LLR(b_i)) \qquad (7)$$

where $I(b_i, LLR(b_i))$ is the mutual information between input bit and output LLR for i^{th} bit in the modulation map. As can be seen, the bit LLR reflects the demodulation process to compute LLR, which was not reflected in the symbol-level MI and the RBIR defined above. This is the main difference between the bit- and symbol-level MI definitions. More generally, however, the mean mutual information – computed by considering the observations over N symbols (or channel uses) – over the codeword may be computed as

$$M = \frac{1}{mN}\sum_{n=1}^{N}\sum_{i=1}^{m} I(b_i, LLR(b_i)) \qquad (8)$$

The mutual information function $I(b_i, LLR(b_i))$ is, of course, a function of the QAM symbol SINR, and so the mean mutual information M (MMIB) may be alternatively written as

$$M = \frac{1}{mN}\sum_{n=1}^{N}\sum_{i=1}^{m} I_{m,b_i}(SINR_n) = \frac{1}{N}\sum_{n=1}^{N} I_m\left(SINR_n\right) \qquad (9)$$

The mean mutual information is dependent on the SINR on each modulation symbol (index n) and the code bit index i (or i-th bit channel), and varies with the constellation order m. Accordingly, the relationship $I_{m,b_i}(SINR)$ is required for each modulation type and component bit index in order to construct $I_m(SINR)$.[1]

---

[1] *Note that in the 802.16e specification, bit indexing typically proceeds from 0.*

For BPSK/QPSK, a closed from expression is given in [29]-[30], which is a non-linear function that can be approximated in polynomial form. For the particular case of BPSK/QPSK, the function would be the same as that obtained by defining mutual information of a symbol channel (symbol channel is just a bit channel for BPSK). For higher order QAM, the corresponding non-linear function can also be approximated numerically (left for future study).]

5.1.2.2. Exponential ESM (EESM)

The EESM is simply given by

$$SINR_{eff} = -\beta \ln\left(\frac{1}{N}\sum_{n=1}^{N}\exp\left(-\frac{SINR_n}{\beta}\right)\right) \tag{10}$$

where $SINR_n$ is the per-tone SINR, *N* is the number of transmitted sub-carriers and β is a value for optimization/adjustment.

5.1.3. Remarks on PHY Abstraction

The decision on which of the ESM PHY abstraction methods used in the final evaluation report shall be decided based on a rigorous set of comparisons using simulations and analyses.

5.2. Per-tone SINR Computation

[All PHY abstraction metrics are computed as a function of post-processing per-tone SINR values across the coded block at the input to the decoder. The post-processing per-tone SINR is therefore dependent on the transmitter/receiver algorithm used to modulate/demodulate the symbols. As an illustration of how the post-processing per-tone SINR values can be computed, we consider the simple case of a single-input-single output (SISO) system with a matched filter receiver. The received signal at the *i*-th sub-carrier for the *m-th* target user is calculated as:

$$Y^m(i) = \sqrt{P_d^m P_{loss}^m} H^m(i) S(i) + \sum_{j=1}^{N_I} \sqrt{P_I^j P_{loss}^j} G^j(i) X^j + n^m(i) \tag{11}$$

where

P_d^m is the total transmit power from BS (per sector) or MS

P_{loss}^m is the path loss including shadowing and antenna gains

$H^m(i)$ is the channel gain for the desired MS for the *i*-th sub-carrier.

P_I^j, P_{loss}^j are the transmit power, path loss and the channel gain between the desired MS and the *j*-th interferer. $S(i)$, X^j are the transmitted symbols of the desired MS and the *j*-th interferer, respectively. $n^m(i)$ is the receiver AWGN noise with zero mean and variance σ^2.

Using a matched filter (or maximum ratio) receiver, given by $H^m(i)^* Y^m(i)$, the post-processing SINR may be expressed as

$$SINR^m(i) = \frac{P_d^m P_{loss}^m \left|H^m(i)\right|^2}{\sigma^2 + \sum_{j=1}^{N_I} P_I^j P_{loss}^j \left|G^j(i)\right|^2} \qquad (12)$$

In the above we assume that ideal knowledge of interference statistics per sub-carrier is available for post-processing SINR computation. For MIMO systems, post-processing SINR may be similarly calculated, especially if linear receivers are used for MIMO processing. We assume that linear minimum mean square (MMSE) receivers will be used as a baseline receiver for SLS methodology. Advanced receivers, such as the maximum likelihood receiver, and the associated PHY abstractions may be considered on an as needed basis and are for further study (FFS). The following section describes the post-processing SINR calculations for MIMO reception assuming the baseline linear MMSE receiver.

5.2.1. Per-Tone Post Processing SINR Calculation for MIMO

To illustrate the per-tone post processing SINR calculation for a MIMO system based on an linear MMSE receiver, we assume an *K* transmit and *L* receive antennas system for downlink transmission. Since these calculations are illustrative, for the sake of simplicity, we assume that *K* spatial streams are transmitted and $L \geq K$. We also assume that interferers and the desired signal use the same MIMO scheme for transmission. The simplified signal model is described as follows:

$$\underline{Y}^m(i) = \underline{H}^m(i)\underline{S}(i) + \sum_{j=1}^{N_I} \underline{G^j(i)}\underline{X}^j(i) + \underline{n} \qquad (13)$$

where

$\underline{Y}^m(i)$ is the *L* dimensional received signal at the MS for the *i*-th sub-carrier,

$\underline{H}^m(i)$ is the $L \times K$ channel gain matrix between the desired BS and the MS for the *i*-th sub-carrier,

$\underline{G}^m(i)$ is the $L \times K$ channel gain matrix between the interfering BS and the MS for the *i*-th sub-carrier,

$\underline{S}(i)$ and $\underline{X}^j(i)$ are the *M*-dimensional transmit symbol vectors of the desired MS and the *j*-th interferer, with covariance $\sigma_d^2 \underline{I}$ *and* $\sigma_{I,j}^2 \underline{I}$, respectively. $\underline{n}$ is modeled as zero mean AWGN noise vector with covariance $\sigma^2 \underline{I}$, $\underline{I}$ is the $K \times K$ identity matrix.

At the MS, a linear MMSE receiver is used to demodulate the transmitted signal vector, and

$$\hat{\underline{S}}(i) = W(i)\underline{Y}^m(i) \qquad (14)$$

Here, the non-interference-aware MMSE weights $W(i)$ are specified as

$$W(i)=\left(\underline{H}^{m^*}\underline{H}^{m}+\frac{\sigma^2}{\sigma_d^2}I\right)^{-1}\underline{H}^{m^*} \qquad (15)$$

where (.)* is the Hermitian operator.

The post-processing SINR can be computed by defining the following two expressions: $E(i)=W(i)\underline{H}^{m}(i)$, and $D(i)=diag(E(i))$ which denotes the desired signal component. Also, define $I_{self}(i)=E(i)-D(i)$ which is the self interference between MIMO streams.

The post-processing SINR for i^{th} sub-carrier and the k^{th} MIMO stream is thus given as:

$$SINR^k(i)=\frac{diag\left[\sigma_d^2 D(i)D(i)^*\right]_{kk}}{diag\left[\sigma^2 W(i)W(i)^*+\sigma_d^2 I_{self}I_{self}^*+\sum_{j=1}^{N_I}\sigma_{I,j}^2 W(i)G^j(i)G^j(i)^*W(i)^*\right]_{kk}}$$

Here, ideal knowledge of the interfering statistics is assumed for the computation of the post-processing SINR, even though the MMSE receiver does not assume knowledge of the interference statistics.]

5.2.2. Interference Aware PHY Abstraction

As noted earlier, accurate modeling of post-processing SINR in the presence of interference requires perfect knowledge of per-sub carrier interference statistics. The baseline methodology shall assume that perfect knowledge of interference statistics per sub-carrier is available at the receiver. We realize that this is an ideal assumption not likely to be met with practical receivers, and a suitable interference-aware PHY abstraction will become necessary with the use of practical receivers operating in the presence of interference. Proponents should provide justification of assumptions related to knowledge of interference statistics used in system level simulations. The specification of a standard practical interference-aware receiver is for further study (FFS).

5.2.3. Practical Receiver Impairments

The evaluation should account for practical receiver implementation losses resulting from channel estimation errors, synchronization errors, coherence loss due to Doppler and inter-tone interference, etc. Inclusion of channel estimation losses and other receiver impairments is FFS; however the minimum level required is of adding a fixed implementation loss backed by analysis/simulation. The channel estimation algorithm that should be assumed as baseline is a linear channel estimation from known symbols (linear means each channel estimate is a linear function of the received pilots), with reasonable delay considering the link latency requirements.

5.3. Effect of Different Block Sizes

[The PHY abstraction predicts the link performance, in terms of PER, for a coded FEC block. Often it is required to predict the PER of a burst of data comprising several code blocks. Here we need to extrapolate the overall PER of the burst as a function of the predicted PER across the code blocks. We note that if the channel is varying slowly, packet errors across code blocks will be correlated for the blocks that are contiguously allocated along time. However, due to sub-carrier permutation, packet errors across code blocks allocated along the frequency axis will appear independent. The overall PER of the code blocks that are correlated may be computed as the maximum of the individual PER. Whereas, the PER for code blocks that are independent, may be computed via the independence assumption. Therefore the overall predicted PER across the burst shall be modeled as

$$P_{burst} \approx 1 - \prod_{\substack{i \subset Independnet \\ Blocks}} \left(1 - \max_{j \subset correlated\ blcoks} PER_{i,j}\right) \qquad (16)$$

The exact procedure for determining the number of correlated blocks in a burst is dependent on the Doppler and delay spread of the channel, and the exact specification is FFS.]

5.4. PHY Abstraction for H-ARQ

PHY abstraction of H-ARQ depends on the H-ARQ method and should be justified by link level simulations showing the PER forecast versus actual PER in each retransmission.

5.4.1. Baseline Modeling

The following abstraction is proposed as baseline:

- For Chase combining (CC): the SINR values of the sub-carriers are summed across retransmissions, and these combined SINR values will be fed into the PHY abstraction.
- For Incremental redundancy (IR): the transmission and retransmissions are regarded as a single codeword, and all the SINR values are fed into the PHY abstraction, which is calibrated according to link level PER curves of the entire transmission (including the retransmission).

For methods combining CC and IR the second approach is preferred but should be justified by link level simulations.

5.4.2. The Case of Repeated Bits/Symbols

[This section details a method to handle symbol/coded bit repetitions accurately. Depending on the rate matching algorithm used, every H-ARQ transmission could have a set of new parity bits and other bits that are repeated. Accumulating the mutual information is appropriate as long as new parity bits are transmitted in every symbol.

Otherwise, the receiver combines the demodulation symbols or, more typically, the LLRs. In this section, we consider a rate-matching approach that does pure IR transmissions and involves coded bit repetitions once all the coded bits from a base code rate are exhausted.

To handle this case, we consider a code-block transmission of N_{NR} symbols that are not repeated, and a set of N_R symbols (or coded bits) that constitute repeated coded bits. We assume that the block of N_R symbols/coded bits have SINR $\{SINR_{R,i}\}$ $i=1,\ldots, N_R$, each with modulation $\{m_{R,i}\}$ $i=1,\ldots, N_R$. Note that if the $SINR_{R,i}$ corresponds to the SINR of the repeated coded bits, then we can take $m_{R,i} = 1$.

We can then compute an effective SINR using the weight sum of the repeated and non-repeated effective SINR as follows:

$$SINR_{eff} = \frac{\left[\sum_{i=1}^{N_{NR}} m_i\right] f_1^{-1}(I_b) + \sum_{i=1}^{N_R} SINR_{R,i}}{\sum_{i=1}^{N_{NR}} m_i + \sum_{i=1}^{N_R} m_{R,i}} \qquad (17)$$

Here, I_b corresponds to the coded-bit level mutual information using all the non-repeated bits, defined by either Equation (5) or (7), and $\sum_{i=1}^{N_{NR}} m_i$ is the total number of coded bits of the non-repeated portion, each symbol transmitted with modulation order $\{m_i\}$ $i=1,\ldots, N_{NR}$. The function $f_1(\cdot)$ is the mapping from bit SINR to mutual information per bit that is used by a given link-system mapping method. This effective bit-level SINR can be used to compute the PER by looking up the PER curve corresponding to an effective code rate obtained from only the non-repeated portion of the coded bits.

This method involves computing the mutual information, or the effective SINR, at the coded-bit level, which can be easily computed in a recursive fashion using the mutual information of a previous block of bits and the current vector of symbol SINRs. The modification to include repetition is of low-complexity, and can work with any bit-level or symbol-level link to system mapping methodology.]

6. Link Adaptation

Editor: Robert Novak, rnovak@nortel.com

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc
Orange	Mark Cudak et al.	C80216m-07_061.pdf
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc

Violet	Liu Ying et al.	C80216m-07_064r1.doc

Link adaptation can enhance system performance by optimizing resources allocation in varying channel conditions. System level simulations should include adaptation of the modulation and coding schemes, according to link conditions.

The purpose of this section is to provide guidelines for of link adaptation in system evaluations. The use of link adaptation is left to the proponent as it may not pertain to all system configurations. The link adaptation algorithms implemented in system level simulations is left to Individual proponents for each proposal. Proponents should specify link adaptation algorithms including power, MIMO rank, and modulation adaptation per resource block.

6.1. Adaptive Modulation and Coding

The evaluation methodology assumes adaptive modulation and coding with various modulation schemes and channel coding rates is applied to packet data transmissions. In the case of MIMO, different modulation schemes and coding rates may be applied to different streams

6.1.1. Link Adaptation with H-ARQ

The link adaptation [algorithm] should be optimized to maximize the performance at the end of the H-ARQ process (e.g. maximize the average throughput under constraint on the delay and PER, or maximize number of users per service).

6.2. Channel Quality Feedback

A Channel Quality Indicator (CQI) channel is utilized to provide channel-state information from the user terminals to the base station scheduler. Relevant channel-state information can be fed back by the CQICH including: Physical CINR, effective CINR, MIMO mode selection and frequency selective sub-channel selection. With TDD implementations, link adaptation can also take advantage of channel reciprocity to provide a more accurate measure of the channel condition (such as sounding).

[Channel quality measurements sent by the MS are proposal specific and may be different from the metrics used in the PHY abstraction, (for example physical average SNR can be used as metric). PHY abstraction typically yields some metric for predicting actual link performance under the transmission format and channel condition, however as channel quality measurements are affected by practical estimation and implementation considerations, this metric may not be used.]

6.2.1. Channel Quality Feedback Delay

Channel quality feedback processing delay accounts for the latency associated with the measurement of channel at the receiver, the decoding of the feedback channel, and the lead-time between the scheduling decision and actual transmission. The delay in reception of the channel quality feedback delay must be modeled to accurately predict system performance.

Proponents will justify selection of CQI feedback delay for each system proposal.

6.2.2. Channel Quality Feedback Error

System simulation performance should include C/I feedback error by modeling appropriate consequences, such as misinterpretation of feedback or erasure.

[The estimation errors of the CQI metric (or other reports) should be taken into account.] [A proposed way is to measure the estimation errors in link level simulation and feed them artificially into the SLS by adding noise to the true CQI value.]

7. HARQ

Editor: Robert Novak, rnovak@nortel.com

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc
Orange	Mark Cudak et al.	C80216m-07_061.pdf
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc
Violet	Liu Ying et al.	C80216m-07_064r1.doc

The stop-and-wait Hybrid ARQ (HARQ) protocol can be implemented in system simulations. Multiple parallel HARQ streams may be present in each frame each associated with a different packet transmission. Different MIMO configurations may also have an impact on the HARQ implementation.

The each HARQ can results in one of successful decoding of packet, unsuccessful decoding of packet transmission requiring further re-transmission, or unsuccessful decoding of packet transmission after maximum number of re-transmissions resulting in packet error. The effective SNR for packet transmissions after one or more HARQ transmissions used in system simulations is determined according to the PHY abstraction section of the evaluation methodology.

When HARQ is enabled, retransmissions are modeled based on the HARQ option chosen. HARQ can be configured as synchronous/asynchronous with adaptive/non-adaptive modulation and coding schemes for Chase combining or incremental redundancy operation. Adaptive H-ARQ in which the parameters of the retransmission are changed according to channel conditions reported by the MS may be considered. Modeling of adaptive H-ARQ is FFS. [HARQ can also be configured to allow adaptive power boosting for each transmission, or retransmission]

The HARQ model and type shall be specified with chosen parameters, such as maximum number of retransmissions, incremental redundancy, chase combining, etc. HARQ overhead (associated control) should be accounted for in the system simulations on both the uplink and downlink

7.1. ACK/NACK Channel

The positive acknowledgement (ACK) and negative acknowledgement (NACK) channel is used to indicate whether or not a packet transmission was successfully received. An ACK/NACK indication is sent after each HARQ packet transmission.

Modeling of HARQ requires waiting for ACK/NACK feedback after each transmission prior to proceeding to the next HARQ transmission. The feedback delay should include the time required for decoding of the transmission at the receiver.

Misinterpretation, missed detection, or false detection of the ACK/NACK message results in transmission (frame or encoder packet) error or duplicate transmission. Proponents for system each system proposal should justify the system performance in the presence of feedback error of the ACK/NACK channel.

8. Scheduling

Editor: Robert Novak, rnovak@nortel.com

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc
Orange	Mark Cudak et al.	C80216m-07_061.pdf
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc
Violet	Liu Ying et al.	C80216m-07_064r1.doc

The scheduler allocates system resources for different packet transmissions according to a set of scheduling metrics, which can be different for different traffic types. The same scheduling algorithm shall be used for all simulation runs. Various scheduling approaches will have performance and overhead impacts and will need to be aligned. System performance evaluation and comparison require that fairness be preserved or at least known in order to promote comparisons. The owner(s) of any proposal are also to specify the scheduling algorithm, along with assumptions on feedback. The scheduling will be done based on the [reported metric] [CQI (or equivalent feedback) from proposal] rather than ideal abstraction.

Frequency selective scheduling can be applied by channel band selection for MSs. For this mode of scheduling with a contiguous permutation of subcarriers (as the Band Adaptive Modulation and Coding (AMC) mode in IEEE 802.16 systems), the sub-channels may experience different attenuation. A Channel Quality Indicator (CQI) channel is utilized to provide channel-state information from the user terminals to the base station scheduler.

8.1. DL scheduler

A DL scheduler at the BS partitions the two dimensional DL frequency subchannel-OFDMA symbol resources between active DL flows. [In its most generic form, a MAC scheduler may calculate a metric per flow at time t $M_i(t)$ that is a function of many attributes specific to the flow, and serve the flows in descending order of the metric values, where $M_i(t) = f(QoS_i(t), CQI_i(t), Delay_i(t), Throughput_i(t)...)$ [1]. There are several options for constructing the metric function, and each may serve a useful purpose depending on the design objectives.]

For the baseline simulation, a generic proportional fair scheduler with scheduling in time and frequency domain, and shall be used for full-buffer traffic model.

In the general deployment case, the MAC scheduler should be capable of handling traffic mix on different QoS service classes that are enabled by the air interface. The proponent may present additional results with a more sophisticated scheduler other than proportional fair scheduler and shall specify the scheduler algorithm in detail.

8.2. UL scheduler

A UL scheduler at the BS partitions the two dimensional UL frequency subchannel-OFDMA symbol resources between active UL flows. UL scheduler is very similar to Downlink Scheduler. UL scheduler maintains the request-grant status of various uplink service flows. Bandwidth requests arriving from various uplink service flows at the BS will be granted in a similar fashion as the downlink traffic.

8.3. Scheduling Metric for Generic Proportional Fair Scheduler

A generic proportionally fair scheduler may be considered in system level simulations. For informative purposes, the metric for a simple proportionally fair scheduler, in which a single user is scheduled in a given scheduling interval, is described below.
At any scheduling instant t, the scheduling metric $M_i(t)$ for subscriber i used by the proportional fair scheduler is given by

$$M_i(t) = \frac{T\_inst_i(t)}{T\_smoothed_i(t)} \qquad (8.3\text{-}1)$$

where $T\_inst_i(t)$ is the data rate that can be supported at scheduling instant t for subscriber i $T\_inst_i(t)$ is a function of the instantaneous SINR, and consequently of the modulation and coding scheme that can meet the PER requirement. $T\_smoothed_i(t)$ is throughput smoothed by a low-pass filter at the scheduling instant t for user i. For the scheduled subscriber, $T\_smoothed_i(t)$ is computed as

$$T\_smoothed_i(t) = \frac{1}{T_{PF}} * T\_inst_i(t-1) + \left(1 - \frac{1}{T_{PF}}\right) * T\_smoothed_i(t-1) \qquad (2)$$

and for unscheduled subscriber,

$$T\_smoothed_i(t) = (1 - \frac{1}{T_{PF}}) * T\_smoothed_i(t-1) \quad (3)$$

where $T_{PF} > 1$ represents the latency time scale of the PF scheduler.

[The trade-off between spectral efficiency and fairness is implicit in the choice of the latency time scale. Larger values of T_{PF} make it possible for the scheduler to wait longer to schedule subscribers when channel conditions are most favorable..

Shorter time scales reduce the opportunities the scheduler has in scheduling a particular subscriber since the smoothed throughput, which diminishes each time a subscriber is not scheduled, becomes the dominant factor in the value of the metric. Although the smoothed throughput for an unscheduled subscriber drops with every scheduling opportunity that is missed, the metric for the same subscriber increases, ultimately forcing the subscriber to be scheduled within the latency time scale of operation.]

[The scheduler explicitly considers the status of the HARQ streams when computing the scheduling metric and deciding priority for scheduling streams.]

9. Handoff

Editor: Robert Novak, rnovak@nortel.com

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc
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Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc
Violet	Liu Ying et al.	C80216m-07_064r1.doc

The system simulation defined elsewhere in the document deals with throughput, spectral efficiency, and latency. User experience in a mobile broadband wireless system is also influenced by the performance of handoff. This section focuses on the methods to study the performance of handover which affects the end-users experience. Proponents of system proposals specifically relating to handover shall provide performance evaluations according to this section.

For parameters such as cell size, DL&UL transmit powers, number of users in a cell, traffic models, and channel models; the simulation follows the simulation methodology defined elsewhere in the document. Only handoff within the system is considered; inter-system and inter-technology handoffs are not considered.

The handover procedure consists of cell reselection via scanning, handover decision and initiation, and network entry including synchronization and ranging with a target BS.

Latency is a key metric to evaluate and compare various handover schemes as it has direct impact on application performance perceived by a user. Total handover latency is decomposed into several latency elements. Further, data loss rate and unsuccessful handover rate are important metrics.

9.1. System simulation with Mobility

9.1.1. Single Mobile MS Model

For simplicity, one mobile MS and multiple fixed MSs can be modeled as a baseline for the mobility simulations. The mobility related performance metrics shall be computed only for this mobile terminal. [The mobile speed is fixed in every simulation drop and aligned with mobility mix associated with the specified channel models].

The trajectory of the mobile MSs can be chosen from the trajectories shown below.

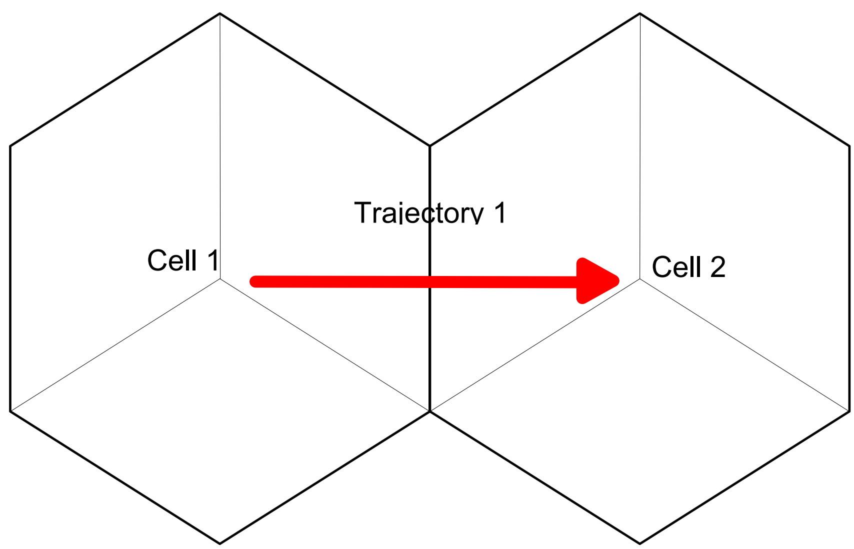
9.1.2. Trajectories

The movement of the single mobile terminal is constrained to one of the following paths (Figure 1 and 2). More detailed and realistic mobility models may be considered.

Trajectory 1: Move from A to B along line joining the cells
Trajectory 2: Move along cell edge. This path is symmetric (the mid-point of Trajectory 2 is on Trajectory 1). The total distance covered by mobile in this case is equal to the cell radius (i.e. distance from center of cell to a vertex).

9.1.2.1. Trajectory 1

In this trajectory, mobile users move from cell 1 to cell 2 along the arrow shown in Figure 9.1.2.1-1



9.1.2.1-1:Trajectory 1 -> Straight path between BSs

9.1.2.2. Trajectory 2

In this trajectory, mobile users move along the cell edge as shown in Figure 9.1.2.2-2

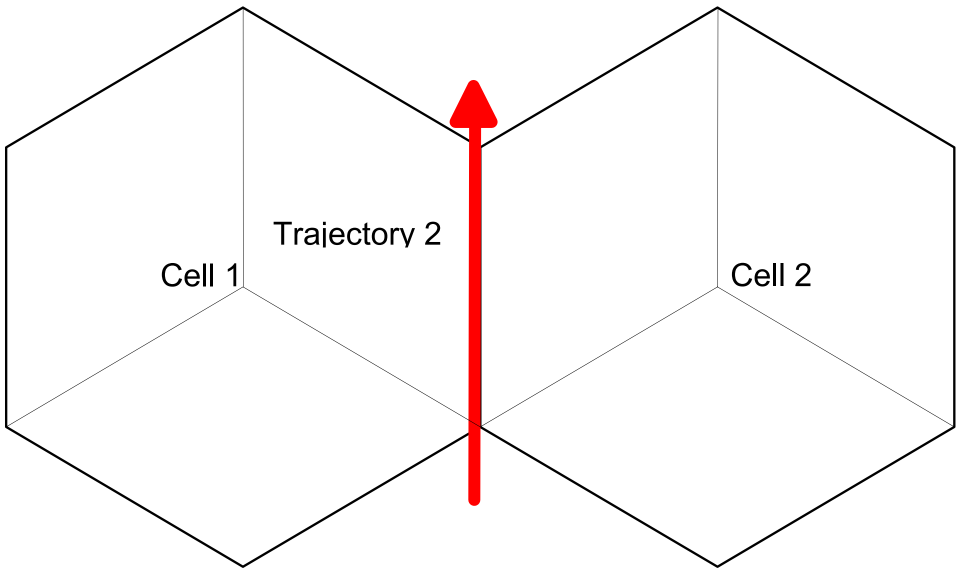
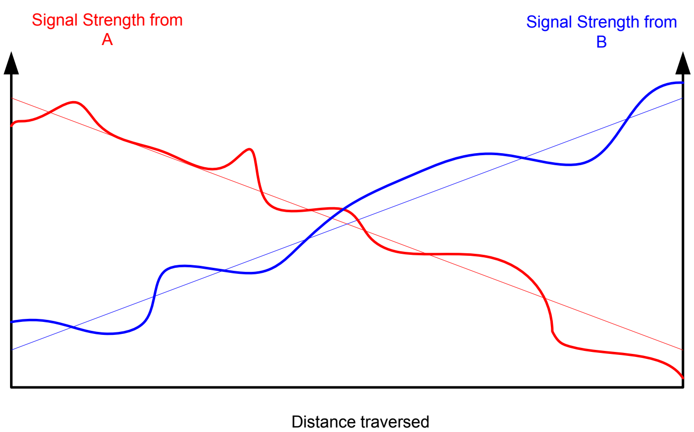
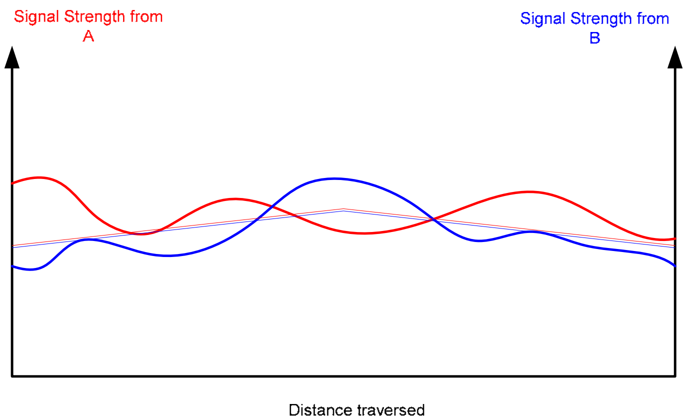


Figure 9.1.2.2: Trajectory 2 -> Straight path along the cell edge

The propagation seen in each of the models is shown in Figures 9.1.2.2-1 and 9.1.2.2-2. The curved lines in the figures include shadow fading, while the straight lines include only path loss. Mobility models 1 and 2 are computed using the path loss and shadowing parameters defined in other parts of the document.



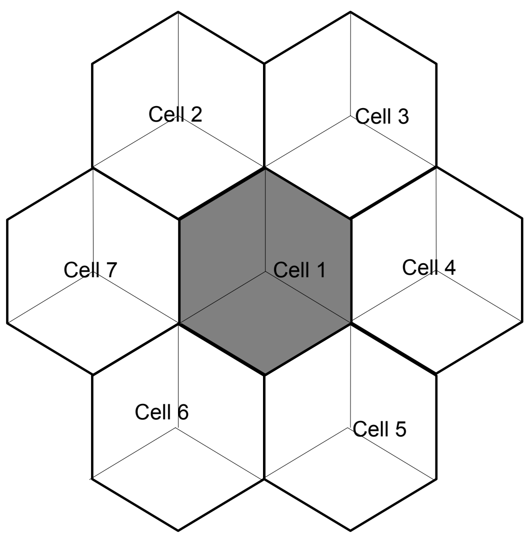
9.1.2.2-1: Propagation for Mobility Path 1



9.1.2.2-2: Propagation for Mobility Trajectory 2

9.1.3. Cell Topology

The simulation scenarios consider system level simulation parameters, channel models, interference modeling, and cellular layouts defined elsewhere in the evaluation methodology. [The time between handoffs shall not be allowed to be less than a specified value. The minimum time between consecutive handoffs is TBD.]



9.1.3-1: 7 Cell Topology

As a reduced complexity option for evaluating handoff, a 7 cell topology may be evaluated. The 7 cell (first tier) network topology is shown in Figure 9.1.3-1.

7 cell topology can be used to evaluate the scenarios in which more than one potential target BSs are involved in the handover procedure. For the 7 cell topology, statistics will be collected only when mobile MSs had handover from or to the center cell (i.e. Cell 1 in Figure 5).

9.2. Handover Performance Metrics

The following parameters should be collected in order to evaluate the performance of different handover schemes. These statistics defined in this section should be collected in relation to the occurrence of handovers. A CDF of each metric may be generated to evaluate a probability that the corresponding metric exceeds a certain value. [For the purpose of evaluating and comparing proposals specially related to handoff performance, proponents shall provide the metrics (TBD).

For a simulation run, we assume:

- Total number of successful handovers occurred during the simulation time = N_{HO}
- Total number of failed handover during the simulation time = N_{HO_fail}
- Total number of handover attempts during the simulation time = $N_{attempt}$

9.2.1. Radio Layer Latency

This value measures the delay between the time instance T1,i that an MS transmits a serving BS its commitment to HO (for a HHO, this is the time that the MS disconnects from the serving BS) and the time instance T2,i that the MS achieves the success of the PHY layer synchronization (i.e., frequency and DL timing synchronization) due to handover occurrence i. The exact thresholds for successful PHY synchronization are for further study. For this metric, the following will be measured.

- Average Radio Layer Latency = $\dfrac{\sum_{i=1}^{N_{HO}} (T_{2,i} - T_{1,i})}{N_{HO}}$
- Maximum Radio Layer Latency =
  $\underset{1 \le i \le N_{HO}}{Max}\, [Radio\ Layer\ Latency\ of\ handover\ occurrance\ i\ ]$

]

9.2.2. Network Entry Time

This value represents the delay between an MS's radio layer synchronization at $T_{2,i}$, and its completion of a Layer 2 network entry procedure at $T_{3,i}$ due to handover occurrence i. This consists of ranging, UL resource request processes (contention or non-contention based), negotiation of capabilities, and registration. All HO MAC messages success/failure rates must be consistent with the packet error rate used for data.

- Average Network Entry Time = $\dfrac{\sum_{i=1}^{N_{HO}}(T_{3,i}-T_{2,i})}{N_{HO}}$
- Maximum Radio Layer Latency = $\max\limits_{1 \le i \le N_{HO}} [\mathit{Network\ Entry\ Time\ of\ handover\ occurance\ i}\ ]$

9.2.3. Connection Setup Time

This value represents the delay between the completion of Layer 2 network entry procedure at $T_{3,i}$ and the reception of first data packet from new BS (target BS) at $T_{4,i}$ due to handover occurrence i.. This consists of DL-UL packet coordination and a path switching time. A path switching time, as a simulation input parameter, may vary depending on network architecture.

- Average Connection Setup Time = $\dfrac{\sum_{i=1}^{N_{HO}}(T_{4,i}-T_{3,i})}{N_{HO}}$
- Maximum Radio Layer Latency = $\max\limits_{1 \le i \le N_{HO}} [\mathit{Connection\ Setup\ Time\ of\ handover\ occurance\ i}\ ]$

9.2.4. Service Disruption Time

This value represents time duration that a user can not receive any service from any BS. It is defined as the sum of Radio Layer Latency, Network Entry Time and Connection Setup Time due to handover occurrence i. [This metric is expressed by the CDF of the service disruption times experienced by the mobile user.]

9.2.5. Data Loss

This value represents the number of lost bits during the handover processes. $D_{RX,i}$ and $D_{TX,i}$ denotes the number of received bits by the MS and the number of total bits transmitted by the serving and the target BSs during the MS performs handover occurrence *i,* respectively. Traffic profiles used for the simulation experiments to compare different handover schemes need to be identical.

Data Loss = $\dfrac{\sum_{i=1}^{N_{HO}} D_{TX,i}-D_{RX,i}}{N_{HO}}$

9.2.6. Handover Failure Rate

This value represents the ratio of failed handover to total handover attempts. A failed handover happens if a handover is executed while the reception conditions are inadequate.

$$\text{Handover Failure Rate} = \frac{N_{HO\_fail}}{N_{attempt}}$$

9.3. 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.]

10. Power Management (informative)

Color	Sources	Document Reference
Blue	Sassan Ahmadi et al.	C80216m-07_069.doc

[The objective of power management is to minimize the power consumption by the subscriber stations (MSs) and thus, to maximize the battery life of the mobile devices. This is achieved through the use of idle mode and sleep mode operations. While idle mode is designed to use the periods of inactivity between calls to save battery power, the sleep mode is designed to use the periods of inactivity in between packets bursts of a call to reduce power consumption by the mobile devices. Thus, the simulation methodology should model sleep and idle mode operations. In addition, it should provide means to investigate the performance of idle and sleep mode operations. For example, some of the critical performance parameters to evaluate the power management in such networks are as follows: the duty cycle of idle and sleep mode operations; time it takes for an MS to return to active from idle and sleep mode. The details of these parameters are FFS.]

11. Traffic Models

Editor: Jeongho Park, jeongho.jh.park@samsung.com

Color	Sources	Document Reference
Red	Alcatel-Dan Gal et al.	IEEE C802.16m-07/063
Green	Sassan Ahmadi et al.	IEEE C802.16m-07/072
Brown	Robert Novak et al.	IEEE C802.16m-07/074r1
Pink	Wookbong Lee et al.	IEEE C802.16m-07/075

This section describes the traffic models in detail. A major objective of system simulations is to provide the operator a view of the maximum number of active users that can be supported for a given service under a specified configuration at a given coverage level. The traffic generated by a service should be accurately modeled in order to find out the performance of a system. This may be a time consuming exercise. Traffic modeling can be simplified, as explained below, by not modeling the user arrival process and assuming full queue traffic which is considered as the baseline. These two assumptions are further discussed proceeding paragraphs. Modeling non-full-queue traffic is also discussed in the next subsections.

Modeling of user arrival process: Typically all the users are not active at a given time and even the active users might not register for the same service. In order to avoid different user registration and demand models, the objective of the proposed simulation is restricted to evaluate the performance with the users who are 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.

Full Queue model: In the full queue 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.

In the following sections, we will concentrate on traffic generation only for the non-full queue case. In addition, the interaction of the generated traffic with the higher layer protocol stack such as TCP is not included here. However, we will provide references to document which provide the detailed TCP transport layer implementation and its interaction with the various traffic models.

11.1. Web Browsing (HTTP) Traffic Model

HTTP traffic characteristics are governed by the structure of the web pages on the World Wide Web (WWW), and the nature of human interaction. The nature of human interaction with the WWW causes the HTTP traffic to have a bursty profile, where the HTTP traffic is characterized by ON/OFF periods as shown in Figure 1-1

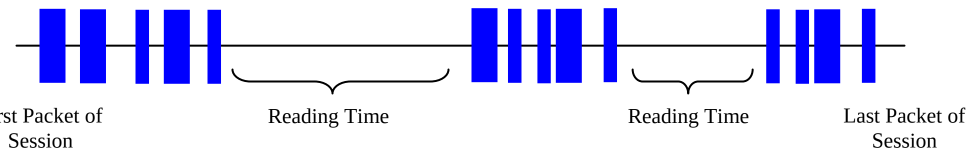


Figure 11.1-1: HTTP Traffic Pattern

The ON periods represent the sequence of packets in which the web page is being transferred from source to destination; while the OFF periods represent the time the user spends reading the webpage before transitioning to another page. This time is also known as Reading Time [34][35].

The amount of information passed from the source to destination during the ON period is governed by the web page structure. A webpage is usually composed of a main object and several embedded objects. The size of the main object, in addition to the number and size of the embedded objects define the amount of traffic passed from source to destination.

In summary, the HTTP traffic model is defined by the following parameters:

SM: Size of main object in page

Nd: Number of embedded objects in a page

SE: Size of an embedded object in page

Dpc: Reading time

Tp: Parsing time for the main page

In addition to the model parameters, HTTP traffic behavior is also dependent on the HTTP version used. Currently HTTP 1.0 and HTTP 1.1 are widely used by servers and browsers [36]-[39]. In HTTP 1.0, also known as burst mode transfer, a distinct TCP connection is used for each object in the page, thereby facilitating simultaneous transfer of objects. The maximum number of simultaneous TCP connections is configurable, with most browsers using a maximum of 4 simultaneous TCP connections. In HTTP/1.1, also known as persistent mode transfer, all objects are transferred serially over a single persistent TCP connection. Table 11.1-1 provides the model parameters for HTTP traffic for downlink and uplink connections [39]-[40].

Component	Distribution	Parameters		PDF
		Downlink	Uplink	
Main object size (SM)	Truncated Lognormal	Mean = 10710 bytes SD= 25032 bytes	Mean = 9055 bytes SD = 13265	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$

		Min = 100 bytes Max = 2 Mbytes $\sigma = 1.37, \mu = 8.35$	bytes Min = 100 bytes Max = 100 Kbytes $\sigma = 1.37, \mu = 8.35$	if x>max or x<min, discard and generate a new value for x
Embedded object size (SE)	Truncated Lognormal	Mean = 7758 bytes SD = 126168 bytes Min = 50 bytes Max = 2 Mbytes $\sigma = 2.36, \mu = 6.17$	Mean = 5958 bytes SD = 11376 bytes Min = 50 bytes Max = 100 Kbytes $\sigma = 1.69, \mu = 7.53$	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \geq 0$ if x>max or x<min, discard and generate a new value for x
Number of embedded objects per page (Nd)	Truncated Pareto	Mean = 5.64 Max. = 53 $\alpha = 1.1, k = 2, m = 55$	Mean = 4.229 Max. = 53 $\alpha = 1.1, k = 2, m = 55$	$f_x = \frac{\alpha k^{\alpha}}{x^{\alpha+1}}, k \leq x < m$ $f_x = \left(\frac{k}{m}\right)^{\alpha}, x = m$ Subtract k from the generated random value to obtain Nd if x>max, discard and regenerate a new value for x
Reading time (Dpc)	Exponential	Mean = 30 sec	Mean = 30 sec $\lambda = 0.033$	$f_x = \lambda e^{-\lambda x}, x \geq 0$
Parsing time (Tp)	Exponential	Mean = 0.13 sec	Mean = 0.13 sec $\lambda = 7.69$	$f_x = \lambda e^{-\lambda x}, x \geq 0$

Table 11.1-1: HTTP Traffic Parameters

To request an HTTP session, the client sends an HTTP request packet, which has a constant size of 350 bytes.

From the statistics presented in the literature, a 50%-50% distribution of HTTP versions between HTTP 1.0 and HTTP 1.1 has been found to closely approximate web browsing traffic in the internet [40].

Further studies also showed that the maximum transmit unit (MTU) sizes most common to in the internet are 576 bytes and 1500 bytes (including the TCP header) with a distribution of 24% and 76% respectively. Thus, the web traffic generation process can be described as in Figure 11.1-2

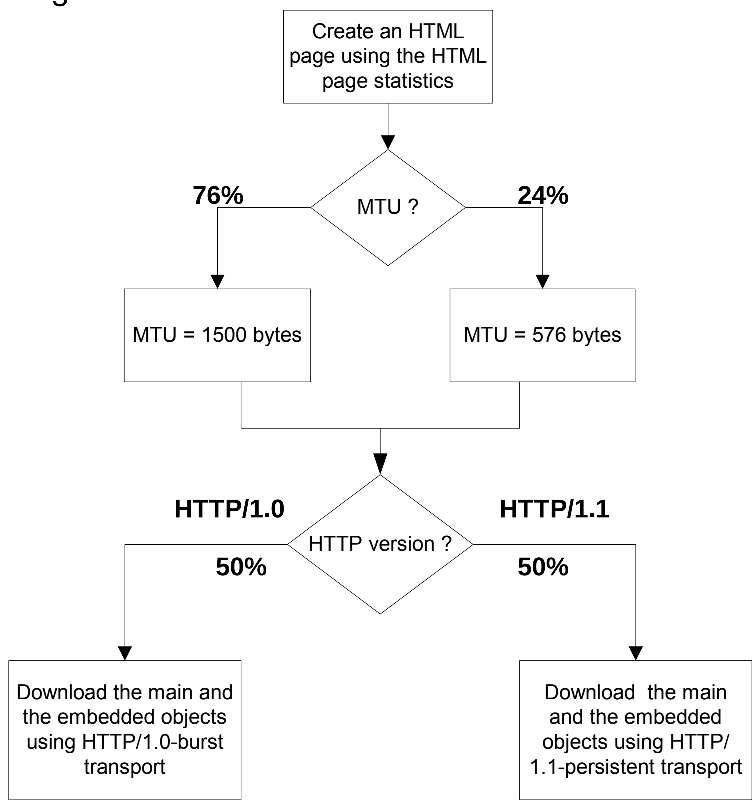


Figure 11.1-2: HTTP Traffic Profiles

11.1.1. HTTP and TCP interactions for DL HTTP traffic

Two versions of the HTTP protocol, HTTP/1.0 and HTTP/1.1, are widely used by servers and browsers. Users shall specify 30% HTTP/1.0 and 70% HTTP/1.1 for HTTP traffic. For people who have to model the actual interaction between HTTP traffic and the underling TCP connection, refer to 4.1.3.2, 4.2.4.3 of [41] for details.

11.1.2. HTTP and TCP interactions for UL HTTP traffic

HTTP/1.1 is used for UL HTTP traffic. For details regarding the modeling of the interaction between HTTP traffic and the underling TCP connection, refer to 4.2.4.1, 4.2.4.2 of [41].

11.2. File Transfer Protocol Model

File transfer traffic is characterized by a session consisting of a sequence of file transfers, separated reading times. Reading time is defined as the time between end of transfer of the first file and the transfer request for the next file. The packet call size is therefore equivalent to the file size and the packet call inter-arrival time is the reading time. A typical FTP session is shown in Figure 11.2-1

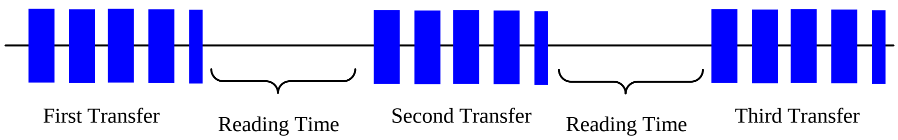


Figure 11.2-1: FTP Traffic Patterns

Table 11.2-1 provides the model parameters for FTP traffic that includes file downloads as well as uploads [42]-[43].In the case of file uploads, the arrival of new users is Poisson distributed and each user transfers a single file before leaving the network.

The FTP traffic generation process is described in Figure 11.2-2. Based on the results on packet size distribution, 76% of the files are transferred using and MTU of 1500 bytes and 24% of the files are transferred using an MTU of 576 bytes. Note that these two packet sizes also include a 40 byte IP packet header and this header overhead for the appropriate number of packets must be added to the file sizes calculated from the probabilistic distributions in Table 11.2-1. For each file transfer a new TCP connection is used whose initial congestion window size is 1 segment.

Component	Distribution	Parameters		PDF
		DL	UL	
File size (S)	Truncated Lognormal	Mean = 2Mbytes SD = 0.722 Mbytes Max = 5 Mbytes $\sigma = 0.35$ $\mu = 14.45$	Min = 0.5 Kbytes Max = 500 Kbytes Mean = 19.5 Kbytes SD = 46.7 Kbytes $\sigma = 2.0899$ $\mu = 0.9385$	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ if x>max or x<min, discard and generate a new value for x
Reading time (D_{pc})	Exponential	Mean = 180 sec. $\lambda = 0.006$	N/A	Download: $f_x = \lambda e^{-\lambda x}, x \ge 0$ Upload: N/A

Table 11.2-1: FTP Traffic Parameters

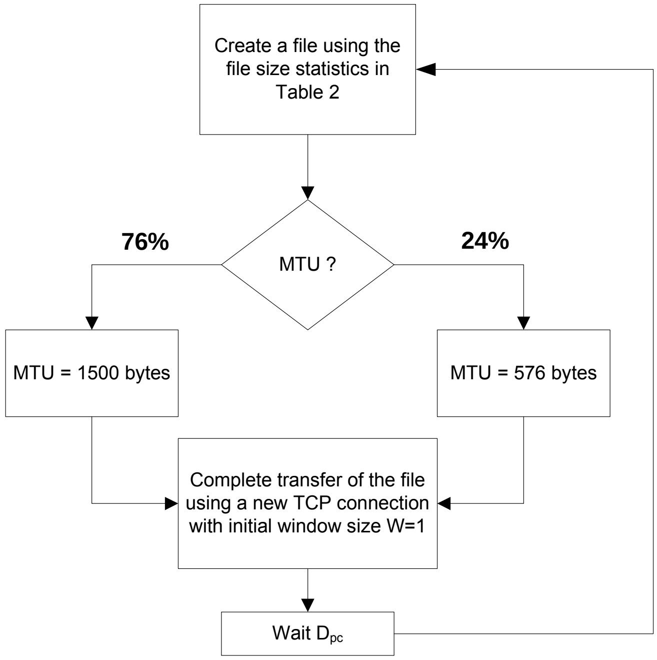


Figure 11.2-2: FTP Traffic Profiles

11.2.1. FTP and TCP interactions

To model the FTP and TCP interactions, please refer to 4.1.4.2 of [41] for details

11.3. Speech Source Model (VoIP)

Color	Source Document Authors	Document Reference
Red	Dan Gal, et al.	IEEE C802.16m-07/063
Green	Sassan Ahmadi, et al.	IEEE C802.16m-07/072
Brown	Robert Novak, et al.	IEEE C802.16m-07/074r1
Violet	Wookbong Lee, et al.	IEEE C802.16m-07/075
Grey	Xin Chang, et al.	IEEE C802.16m-07/067

VoIP refers to real-time delivery of packet voice across networks using the Internet protocols. A VoIP session is defined as the entire user call time and VoIP session occurs during the whole simulation period.

There is a variety of encoding schemes for voice (i.e., G.711, G.722, G.722.1, G.723.1, G.728, G.729, and AMR) that result in different bandwidth requirements. Including the protocol overhead, it is very common for a VoIP call to require between 5 Kbps and 64 Kbps of bi-directional bandwidth.

11.3.1. Basic Voice Model

A typical phone conversation is marked by periods of active talking(or talk spurt(ON period)) interleaved by silence/listening period(or OFF period) as shown in Figure 11.3.1-1.

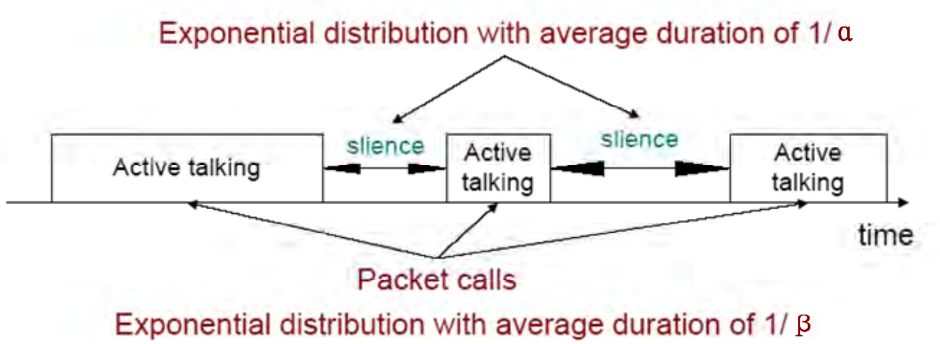


Figure 11.3.1-1: Typical phone conversation profile

Consider the simple 2-state voice activity Markov model shown in Figure 11.3.1-2 [45]

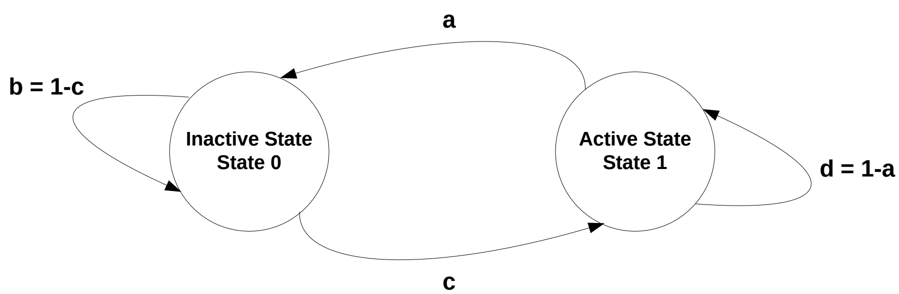


Figure 11.3.1-2: 2-state voice activity Markov model

In the model, the conditional probability of transitioning from state 1 (the active speech state) to state 0 (the inactive or silent state) while in state 1 is equal to a, while the conditional probability of transitioning from state 0 to state 1 while in state 0 is c. The model is assumed updated at the speech encoder frame rate R=1/T, where T is the encoder frame duration (typically, 20ms).

During the active state, packets of fixed sizes are generated at a regular interval. During the inactive state, we have chosen to generate comfort noise with smaller packet sizes at a regular interval instead of no packet transmission. The size of packet and the rate at which the packets are sent depends on the corresponding voice codecs and compression schemes. Table 11.3.1-1 provides information on some common vocoders.

Vocoder	EVRC	AMR	GSM 6.10	G.711	G.723.1		G.729A
Source Bit rate [Kb/s]	0.8/2/4/8.55	4.75-12.2	13	64	5.3	6.3	8
Frame duration [ms]	20	20	20	10	30	30	10
Information bits per frame	16/40/80/171	95-244	260	640	159	189	80

Table 11.3.1-1: Information on various vocoders

Among the various vocoders in Table 11.3.1-1, a simplified AMR (adaptive multi-rate) audio data compression can be used to simplify the VoIP modeling process. AMR is optimized for speech coding and was adopted as the standard speech codec by 3GPP and widely used in GSM. The original AMR uses link adaptation to select from one of eight different bit rates based on link conditions. If the radio condition is bad, source coding is reduced (less bits to represent speech) and channel coding (stronger FEC) is increased. This improves the quality and robustness of the network condition while sacrificing some voice clarity. In our simplified version, we have chosen to disable the link adaptation and use the full rate of 12.2kbps in the active state. This will give us the worst case scenario.

Description	AMR without Header Compression IPv4/IPv6	AMR with Header Compression IPv4/IPv6	G.729 without Header Compression IPv4/IPv6	G.729 with Header Compression IPv4/IPv6
Voice Payload	7bytes for inactive 33 bytes for active	7bytes for inactive 33 bytes for active	0 bytes for inactive 20 bytes for active	0 bytes for inactive 20 bytes for active
Protocol Headers	40 bytes / 60bytes	2 bytes / 4 bytes	40 bytes / 60 bytes	2 bytes / 4 bytes
RTP	12 bytes		12 bytes	
UDP	8 bytes		8 bytes	
IPv4 / IPv6	20 bytes / 40 bytes		20 bytes / 40 bytes	
802.16 Generic MAC Header (default)	TBD (6 bytes)	TBD (6 bytes)	TBD (6 bytes)	TBD (6 bytes)
CRC (default)	TBD (4 bytes)	TBD (4 bytes)	TBD (4 bytes)	TBD (4 bytes)
Total VoIP packet size (default)	TBD (57bytes/77bytes for inactive 83bytes /103bytes active)	TBD (19 bytes/21 bytes for inactive 45 bytes/47 bytes for active)	TBD (0 bytes for inactive 70 bytes/90 bytes for active)	TBD (0 bytes for inactive 32 bytes/ 34 bytes for active)

Table 11.3.1-2: VoIP packet size calculation for simplified AMR and G. 729

Table 11.3.1-2 shows the VoIP packet size calculation for simplified AMR with or without header compression when using IPv4 or IPv6.

To calculate the total packet size, MAC headers and CRC need to be accounted for (example: there are 6 bytes of MAC header and CRC in IEEE 802.16e reference system). For example, without header compression, AMR payload of 33 bytes are generated in the active state for every 20ms and AMR payload of 7 bytes are generated in the inactive state for every 160ms, resulting in a packet size of 83 (57) bytes for the active (inactive) mode, respectively, assuming IPv4 and uncompressed headers.

11.3.2. VoIP Traffic Model Parameters

During each call (each session), a VoIP user will be in the Active or Inactive state. The duration of each state is exponentially distributed. Within the Active/Inactive state, packets of fixed sizes will be generated at a fix interval. Hence, both the datagram size and datagram arrival intervals are fixed within a packet call. Table 11.3.2-1 shows parameters associated with the VoIP traffic model.

Component	Distribution	Parameters	PDF
Active state duration	Exponential	Mean = [1][1.33] second	$f_x = \lambda e^{-\lambda x}$, $x \geq 0$ $\lambda = 1/$ *Mean*
Inactive state duration	Exponential	Mean = [1.5][2] second.	$f_x = \lambda e^{-\lambda x}$, $x \geq 0$ $\lambda = 1/$ *Mean*
Probability of transition from active to inactive state	N/A	[0.02][0.015]	N/A
Probability of transition from inactive to active state	N/A	[0.0133][0.01]	N/A

Table 11.3.2-1: VoIP traffic model parameters specification

Link adaptation of AMR codec is disabled in order to evaluate performance under worst case. The Voice traffic model specifies only one rate during the ON state (talk spurt) of the AMR codec (12.2 kbps) and another rate for the comfort noise (AMR_SID) during the OFF state of the AMR codec. [The 1/8th frame rate blanking is modeled by only transmitting the first 1/8th rate frame of each silence interval.]

Table 11.3.2-2 provides the relevant parameters of the VoIP traffic that shall be assumed in the simulations. The details of the corresponding traffic model are described below:

Parameter	Characterization
Codec	RTP AMR 12.2, Source rate 12.2 kbps
Encoder frame length	20 ms
Voice activity factor (VAF)	40%
SID payload	Modeled TBD according to Table 1-3-2] (5 Bytes + header) SID packet every 160 ms during silence
[Protocol Overhead with compressed header]	[TBD according to Table 1-3-2]
[Total voice payload on air interface]	[TBD according to Table 1-3-2]

Table 11.3.2-2: Detailed description of the VoIP traffic model

11.4. Near Real Time Video Streaming

This section describes a model for streaming video traffic for DL direction. Figure 11.4-1 describes a steady state of video streaming traffic from the network as observed by the base station. Call setup latency and overhead is not considered in this model.

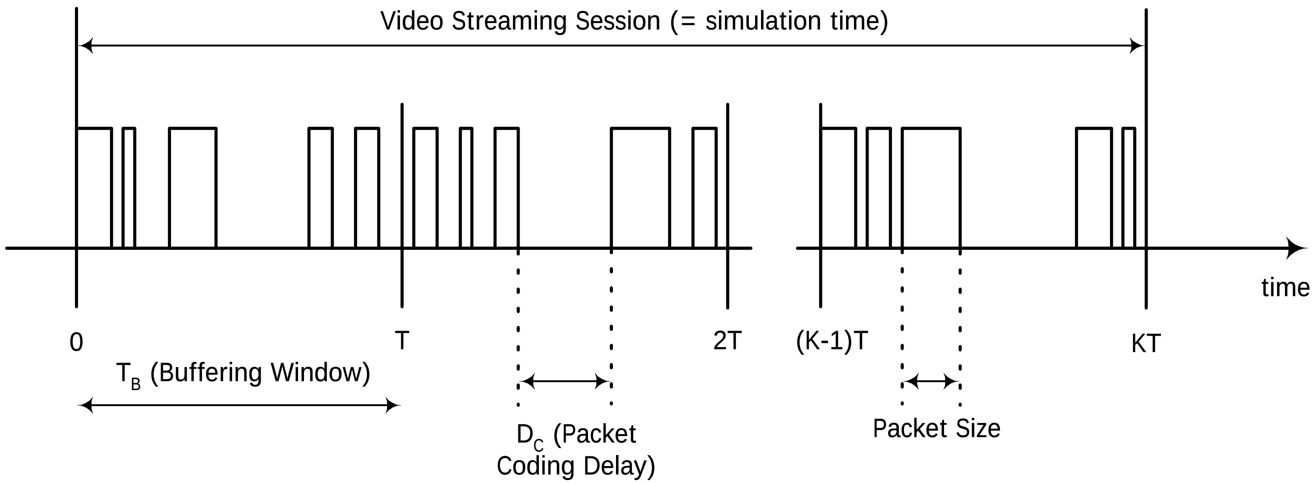


Figure 11.4-1: Video Streaming Traffic Model

11.5. Gaming traffic model

Gaming is a rapidly growing application embedded into communication devices, and thus wireless gaming needs to be considered.

Packet size in gaming traffic is modeled by the Largest Extreme Value distribution. The starting time of a network gaming mobile is uniformly distributed between 0 and 40 m to simulate the random timing relationship between client traffic packet arrival and reverse link frame boundary.

On the uplink, a packet is dropped by the subscriber station if any part of the packet (including HARQ operation) has not started within 160msec of the time the packet entered the subscriber station's buffer. Packet delay of a dropped packet is counted as 180 ms. Currently understanding is that 50 ms lag is considered excellent quality while 100 ms lag is considered good quality. Ping times above 150 ms are often reported to be intolerable [46] . Outage in wireless gaming is defined as average packet delay greater than 60msec, where average delay is the average of the delay of all packets, including the delay of packets delivered and the delay of packets dropped. Table 11.5-1 provides the model parameters for wireless gaming [46].

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	

Initial packet arrival	Uniform	Uniform	a=0, b=40 ms	a=0, b=40 ms	$f(x)=\frac{1}{b-a} \quad a \le x \le b$
Packet arrival time	Extreme	Deterministic	a=55 ms, b=6 ms	40 ms	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $[X=\lfloor a-b\ln(-\ln Y)\rfloor]$ $Y\in U(0,1)$
Packet size	Extreme	Extreme	a=120, b=36 bytes	a=45 bytes, b = 5.7	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=\lfloor a-b\ln(-\ln Y)\rfloor+2,$ $Y\in U(0,1)$

Table 11.5-1: Gaming Traffic Model

Note:

1. To account for UDP header, 2 was added to the size of the packet size

2. 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.

11.6. Traffic Mixes

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.

Table 11.6-1 contains traffic mixes that should be used in system evaluations
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.

	VoIP	FTP	HTTP	NRTV	Gaming	
Voice only	100% [#users = Nv*]	0%	0%	0%	0%	
FTP only	0%	100%	0%	0%	0%	
[HTTP only]	[0%]	[0%]	[100%]	[0%]	[0%]	
[NRTV only]	[0%]	[0%]	[0%]	[100%]	[0%]	
[Gaming only]	[0%]	[0%]	[0%]	[0%]	[100%]	
Traffic Mix	30%	10%	20%	20%	20%	
[Traffic Mix]	[0.5 Nv]	[Remaining capacity for Data Users]				
		[100%]	[0%]	[0%]	[0%]	
[Traffic Mix]	[0.5 Nv]	[Remaining capacity for Data Users]				
		[30%]	[30%]	[30%]	[10%]	
[Traffic Mix]	[0.75 Nv]	[Remaining capacity for Data Users]				
		[30%]	[30%]	[30%]	[10%]	

Table 11.6-1: Traffic Mixes

[* Nv is the system voice capacity that satisfy outage criteria at system and user level]

12. Simulation Procedure and Flow

Editor: Roshni Srinivasan, roshni.m.srinivasan@intel.com

Color	Sources	Document Reference
Green	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc

The nineteen-cell network topology with wrap-around (as shown in the Appendix A) shall be used as the baseline network topology for all system-level simulations. The system simulation flow required in this evaluation methodology is illustrated in Figure 12-1.

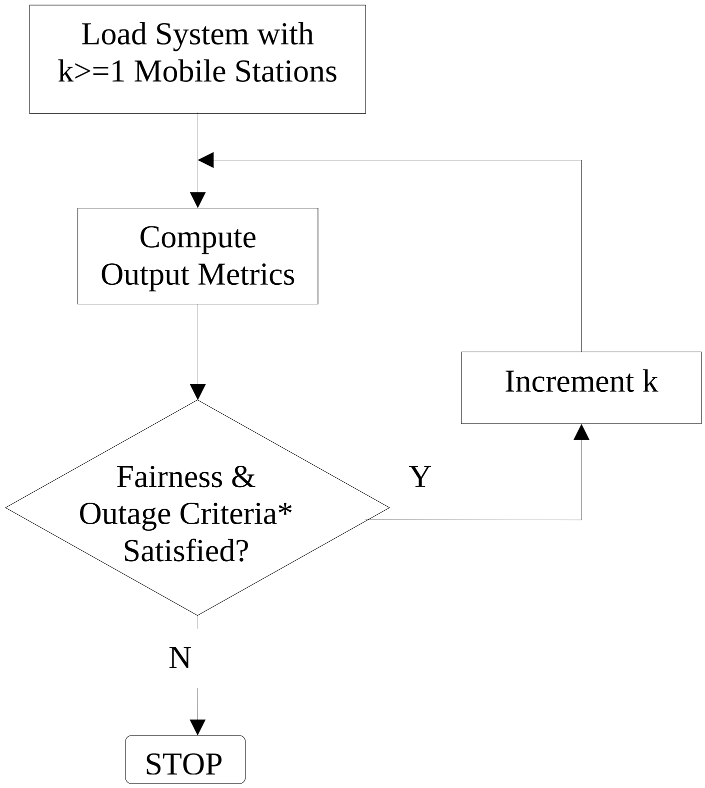


Figure 12-1: Simulation Procedure

** As defined in Sections 11.6 and 13*

1. The system is modeled as a network of 7 clusters. Each cluster has 19 hexagonal cells with six cells of the first tier and twelve cells of the second tier surround the central cell of each cluster. Each cell has three sectors. Frequency reuse is modeled by planning frequency allocations in different sectors in the network
2. MSs are dropped uniformly throughout the system. Each mobile corresponds to an active user session that runs for the duration of the drop.
3. 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.
4. MSs are dropped according to the specified traffic mix. The simulation runs are done with an increment of MSs per sector until a termination condition is met as shown in Figure 12-1.
5. For sectors belonging to the center cluster, sector assignment to an MS is based on the received power at an MS from all potential serving sectors. The sector with best path to MS, taking into account slow fading characteristics (path loss, shadowing, and antenna gains) is chosen as the serving sector.
6. Mobile stations are randomly dropped over the 57 sectors such that each sector has the required numbers of users. Although users may be in regions supporting handoff 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. MS locations for six wrapping clusters are the same as the center cluster.

7. 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.
8. 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 traffic models defined in this document. Start times for each traffic type for each user should be randomized as specified in the traffic model being simulated.
9. Packets are scheduled with a packet scheduler using the required fairness metric. Channel quality feedback delay, PDU errors and ARQ are modeled and packets are retransmitted as necessary. The ARQ process is modeled by explicitly rescheduling a packet as part of the current packet call after a specified ARQ feedback delay period.
10. Simulation time is chosen to ensure convergence in desired output metrics.
11. Performance statistics are collected for MSs in all cells according to the output matrix requirements.
12. All 57 sectors in the system shall be dynamically simulated

13. Simulation Outputs and Performance Metrics

Section Editor: Roshni Srinivasan, roshni.m.srinivasan@intel.com

Color	Sources	Document Reference
Green	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Dan Gal et al.	C80216m-07_063.doc
Brown	Robert Novak et al.	C80216m-07_074r1.pdf
Pink	Wookbong Lee et al.	C80216m-07_075r1.doc
Violet	Liu Ying et al.	C80216m-07_064r1.doc
Orange	Mark Cudak et al.	C80216m-07_061.pdf

13.1. Performance Metrics

[Performance metrics are divided into two categories:

- Single-user performance metrics
- Multi-user performance metrics

Examples of single-user performance metrics are the link budget margins, C/I area coverage and data rate area coverage. These metrics are evaluated assuming that a single user is in a particular cell area utilizing all the resources in that cell while external interference may be evaluated assuming that at least a single active user is available in the external cell (for both downlink and uplink). These metrics are not end-to-end performance metrics and therefore, could be evaluated without modeling higher layer protocols and is independent of applications.

However, when multiple users are in the system the system resources have to be shared and a user's average data rate will be smaller than the single-user rate. Therefore, multi-user metrics are proposed which show how a system behaves under a multi-user environment.
Note that the area coverage mentioned below is equivalent to the percentage of users meeting a given requirement when the users are uniformly distributed in the interested geographical area.]

13.1.1. Single User Performance Metrics

13.1.1.1. Link Budget and Coverage Range (Noise Limited) – single-cell consideration

Link budget evaluation is a well known method for initial system planning that needs to be carried out for BS to MS links. Although a link budget can be calculated separately for each link, it is the combination of the links that determines the performance of the system as a whole. The parameters to be used needs to be agreed upon after obtaining consensus. Using the margins in the link budget, the expected signal to noise ratio can be evaluated at given distances. Using these results, the noise limited range can be evaluated for the system. The link budget template is TBD.

Coverage range is defined as the maximum radial distance to meet a certain percentage of area coverage (x%) with a signal to noise ratio above a certain threshold (target_snr) over y% of time, assuming no interference signals are present. It is proposed that x be 99 and y be 95.

13.1.1.2. C/I Coverage – interference limited multi-cell consideration

The C/I coverage is defined as the percentage area of a cell where the average C/I experienced by a stationary user is larger than a certain threshold (target_ci).

13.1.1.3. Data Rate Coverage – interference limited multi-cell consideration

The percentage area for which a user is able to transmit/receive successfully at a specified mean data rate using single-user analysis mentioned above. No delay requirement is considered here.

13.1.2. Multi-User Performance Metrics

Although a user may be covered for a certain percentage area (e.g. 99%) for a given service, when multiple users are in a sector/BS, the resources (time, frequency, power) are to be shared among the users. It can be expected that a user's average data rate may be reduced by a factor of N when there are N active users (assuming resources are equally shared and no multi-user diversity gain), compared to a single user rate.

For example, assume that there is a system, where a shared channel with a peak rate of 2 Mbps can serve 99% of the area. If a user wants to obtain a video streaming service at 2 Mbps, that particular user will be able to obtain the service, but no other user will be able to get any service during the whole video session (which may extend for more than an hour). Therefore, in this example although 99% area is covered for the video service, this service is not a viable service for the operator and performance of coverage need to be coupled with the capacity in order to reflect viable service solutions. Coverage performance assessment must be coupled with capacity (# of MSs), to obtain a viable metric.

The users having poor channel quality may be provided more resources so that they would get equal service from the cellular operator. This could adversely impact the total cell throughput. Thus, there is a trade-off between coverage and capacity. Any measure of capacity should be provided with the associated coverage. .

Since an operator should be able to provide the service to multiple users at the same time, an increase in the area coverage itself does not give an operator the ability to offer a given service

Therefore, the number of users that can be supported under a given coverage captures actual coverage performance for a given service from a viability point of view.

The suggested performance metric is the number of admissible users (capacity), parameterized by the service (R_{min}), and the coverage (allowable outage probability).

13.2. Definitions of Performance Metrics

It is assumed that simulation statistics are collected from sectors belonging to the test cell(s) of the 19-cell deployment scenario. Collected statistics will be traffic-type (thus traffic mix) dependent.

In this section, we provide a definition for various metrics collected in simulation runs. For a simulation run, we assume:

1] Simulation time per drop = T_{sim}
2] Number of simulation drops = D
3] Total number of users in sector(s) of interest= N_{sub}
4] Number of packet calls for user u = p_u
5] Number of packets in i^{th} packet call = $q_{i,u}$

13.2.1. Throughput Performance Metrics

For evaluating downlink (uplink) throughput, only packets on the downlink(uplink) are considered in the calculations. Downlink and uplink throughputs are denoted by upper case DL and UL respectively (example: R_u^{DL}, R_u^{UL}). The current metrics are given per a single simulation drop.

The throughput metrics below shall be measured at the following layers:

- PHY Layer
- MAC Layer
- TCP Layer

The throughput for those layers is measured at the points identified in Figure 13.2.1-1 where the throughput refers to the payload throughput without the overhead.

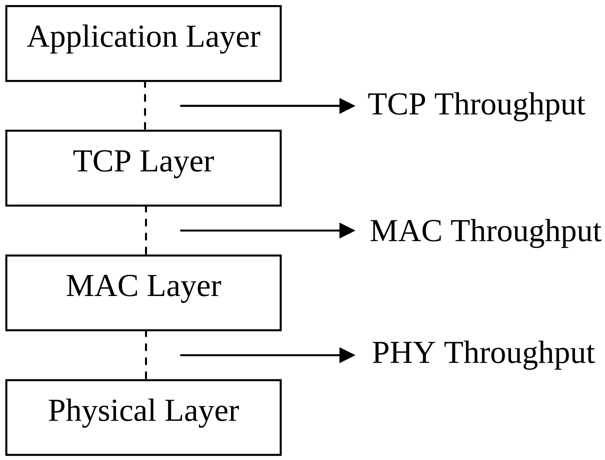


Figure 13.2.1-1: Throughput Metrics Measurement Points

13.2.1.1. Average Data Throughput for User u

The data throughput of a user is defined as the ratio of the number of information bits that the user successfully received divided by the amount of the total simulation time. If user *u* has $p_u^{DL(UL)}$ downlink (uplink) packet calls, with $q_{i,u}^{DL(UL)}$ packets for the i^th downlink (uplink) packet call, and $b_{j,i,u}$ bits for the j^{th} packet; then the average user throughput for user *u is*

$$R_u^{DL(UL)} = \frac{\sum_{i=1}^{p_u^{DL(UL)}} \sum_{j=1}^{q_{i,u}^{DL(UL)}} b_{j,i,u}}{T_{Sim}}$$

13.2.1.2. Sector Data Throughput

Assuming N_{sub} users in sector(s) of interest, and u^{th} user where $u \in N_{sub}$ has throughput $R_u^{DL(UL)}$, then sector(s) data throughput is :

$$R_{sec}^{DL(UL)} = \sum_{u=1}^{N_{sub}} R_u^{DL(UL)}$$

13.2.1.3. Average packet call throughput

Packet call throughput is the total bits per packet call divided by total packet call duration. If user *u* has $p_u^{DL(UL)}$ downlink (uplink) packet calls, with $q_{i,u}^{DL(UL)}$ packets for the i^{th} downlink (uplink) packet call, and $b_{j,i,u}$ bits for the j^{th} packet; then the average packet call throughput is

$$R_u^{pc,DL(UL)} = \frac{1}{p_u^{DL(UL)}} \left(\sum_{i=1}^{p_u^{DL(UL)}} \frac{\sum_{j=1}^{q_{i,u}^{DL(UL)}} b_{j,i,u}}{(T_{i,u}^{end,DL(UL)} - T_{i,u}^{start,DL(UL)})} \right)$$

13.2.1.4. The histogram of users' average packet call throughput

The histogram will display the distribution of the downlink (uplink) average packet call throughput observed at the MS (BS) for the subscribed users.

13.2.1.5. Throughput Outage

[Throughput outage ($O_{thpt}(R_{min})$) is defined as the percentage of users with data rate R_u^{DL}, less than a predefined minimum rate R_{min} (TBD).
]

13.2.1.6. Cell edge user throughput

[The cell edge user throughput can be observed by 5th percentile point of CDF of user average packet call throughput.]

13.2.1.7. Geographical Distribution of Average Packet Call Throughput per User (optional)

The plot will show the geographical distribution of the average packet call throughput observed on the downlink (uplink) by the MS (BS). It provides insight into the throughput variation as a function of distance from the BS. This allows for easy comparison between different reuse scenarios, network loading conditions, smart antenna algorithms, etc.

13.2.2. Performance Metrics for Delay Sensitive Applications

For evaluating downlink (uplink) delay, only packets on the downlink (uplink) are considered in the calculations. Downlink and uplink delays are denoted by upper case DL and UL respectively (example: D_u^{DL}, D_u^{UL}) .

13.2.2.1. Packet Delay

Assuming the *j*th packet of the *i*th packet call destined for user u arrives at the BS (SS) and queued at time $T_{j,i,u}^{arr,DL(UL)}$ and successfully delivered to the MS (BS) MAC-SAP at time $T_{j,i,u}^{dep,DL(UL)}$, then the packet delay is defined as

$$Delay_{j,i,u}^{DL(UL)} = T_{j,i,u}^{arr,DL(UL)} - T_{j,i,u}^{dep,DL(UL)}$$

13.2.2.2. The CDF of packet delay per user

[CDF of the packet delay per user provides a basis in which maximum latency, x%-tile, average latency as well as jitter can be derived. Packets dropped or erased are assigned infinite delay.]

13.2.2.3. X%-tile Packet delay per user

[The x%-tile packet delay is simply the packet delay value for which x% of packets have delay below this value.]

13.2.2.4. The CDF of X%-tile Packet Delays

The CDF of x%-tiles of packet latencies is used in determining the y%-tile latency of the x%-tile per user packet delays.

13.2.2.5. The Y%-tile of X%-tile Packet Delays

The y%-tile is the latency number in which y% of per user x%-tile packet latencies are below this number. This latency number can be used as a measure of latency performance for delay sensitive traffic. A possible criteria for VoIP, for example, is that the 98th %-tile of the 98%-tile of packet latencies per user is 30ms.

13.2.2.6. User Average Packet Delay

The average packet delay is defined as the average interval between packets originated at the source station (either MS or BS) and received at the destination station (either BS or MS) in a system for a given packet call duration. The average packet delay for user u, $D_u^{avg,DL(UL)}$ is given by:

$$D_u^{avg,DL(UL)} = \frac{\sum_{i=1}^{p_u}\sum_{j=1}^{q_{i,u}}\left(T_{j,i,u}^{arr,DL(UL)} - T_{j,i,u}^{dep,DL(UL)}\right)}{\sum_{i=1}^{p_u} q_{i,u}}$$

13.2.2.7. CDF of Users' Average Packet Delay

The CDF will reflect the cumulative distribution of the average packet delay observed by the users.

13.2.2.8. Sector Average Packet Delay

For the application of interest, assume N_{sub} users in sector(s) under consideration. If the application for the u^{th} user, where $u \in N_{sub}$ has average packet delay $D_u^{avg,DL(UL)}$, then:

$$D_{\sec}^{avg,DL(UL)} = \frac{\sum_{u=1}^{N_{sub}}\sum_{i=1}^{p_u}\sum_{j=1}^{q_{i,u}}\left(T_{j,i,u}^{arr,DL(UL)} - T_{j,i,u}^{dep,DL(UL)}\right)}{\sum_{u=1}^{N_{sub}}\sum_{i=1}^{p_u} q_{i,u}}$$

13.2.2.9. Sector Packet Delay Variance

[For the application of interest, assume N_{sub} users in sector(s) under consideration. If the application for the u^{th} user where $u \in N_{sub}$ has average packet delay $D_u^{avg,DL(UL)}$, then the delay variance, $D_{\sec}^{\text{var},DL(UL)}$ observed in the sector for the application is given by:

$$D_{\sec}^{\text{var},DL(UL)} = \frac{\sum_{u=1}^{N_{sub}} \sum_{i=1}^{p_u} \sum_{j=1}^{q_{i,u}} \left(T_{j,i,u}^{arr,DL(UL)} - T_{j,i,u}^{dep,DL(UL)} - D_{\sec}^{avg,DL(UL)} \right)^2}{\sum_{u=1}^{N_{sub}} \sum_{i=1}^{p_u} q_{i,u}}$$

]

13.2.2.10. Maximum Packet Delay

The maximum packet delay is defined as the maximum interval between packets originated at the source station (either MS or BS) and received at the destination station (either BS or MS) in an system for a given packet call duration.

13.2.2.11. User Average Packet Delay Jitter

[For the application of interest, we define delay jitter $J_u^{avg,DL(UL)}$ as the variation in the packet delay times for consecutive packets observed at the receiver by user *u*.

$$J_u^{avg,DL(UL)} = \frac{\sum_{i=1}^{p_u} \sum_{j=2}^{q_{i,u}} \left| \left(T_{j,i,u}^{arr,DL(UL)} - T_{j,i,u}^{dep,DL(UL)} \right) - \left(T_{j-1,i,u}^{arr,DL(UL)} - T_{j-1,i,u}^{dep,DL(UL)} \right) \right|}{\sum_{i=1}^{p_u} q_{i,u}}$$

(13.2.2.11-1)

]

13.2.2.12. CDF of Average Packet Delay Jitter

[The cumulative distribution function (CDF) for the average packet jitter experienced by the users for the application of interest. The CDFs allow for an easy comparison between different reuse scenarios, network loading conditions, and evaluating the percentage of users experiencing excessive jitter.]

13.2.2.13. Packet Loss Ratio

The packet loss ratio per user is defined as

$$Packet\ Loss\ Ratio = 1 - \frac{Total\ Number\ of\ Successfully\ \mathrm{Re}ceived\ Packets}{Total\ Number\ of\ Successfully\ Transmitted\ Packets}$$

13.2.3. System Level Metrics

13.2.3.1. System data throughput

The data throughput of a BS is defined as the number of information bits per second that a site can successfully deliver or receive using the scheduling algorithms.

13.2.3.2. Spectral Efficiency

[Effective system spectral efficiency should be normalized by the downlink/uplink ratio of TDD system. For the DL case:

$$\text{DL Site Spectral Efficiency} = \frac{\text{DL System Data Throughput}}{\text{Total Site BW allocated to DL}}$$

]

Both physical layer spectral efficiency and MAC layer spectral efficiency should be evaluated. Physical layer spectral efficiency should represent the system throughput measured at the interface from the physical layer to the MAC layer, thus including physical layer overhead but excluding MAC and upper layer protocols overhead. MAC layer spectral efficiency should represent the system throughput measured at the interface from the MAC layer to the upper layers, thus including both physical layer and MAC protocols overhead.

The MAC efficiency of the system should be evaluated by dividing the MAC layer spectral efficiency by the physical layer spectral efficiency.

The average cell/sector spectral efficiency is defined as

$$r = \frac{R}{BW_{eff}}$$

Where R is the aggregate cell/sector throughput, BW_{eff} is the effective channel bandwidth i.e., the total spectrum utilized by the system (corresponding to the reuse pattern being simulated). The effective channel bandwidth is defined as

$$BW_{eff} = BW \times TR$$

where BW is the used channel bandwidth, and TR is time ratio of the link. For example, for FDD system TR is 1, and for TDD system with DL:UL=2:1, TR is 2/3 for DL and 1/3 for UL, respectively

13.2.3.3. CDF of SINR

[The cumulative distribution function (CDF) for the signal to interference and noise ratio (SINR) observed by the BS for each user on the uplink channel. The CDF's allow for an easy comparison between different reuse scenarios, network loading conditions, smart antenna algorithms, etc.]

13.2.3.4. Histogram of MCS

[The histogram will display the distribution of MCS for the subscribed users.]

13.2.3.5. Application Capacity

Application capacity *(C_{app})* is defined as the maximum number of application users that the system can support without exceeding the maximum allowed outage probability.

13.2.3.6. System Outage

System outage is defined as when the number of users experiencing outage exceeds k% of the total number of users. User outage is defined in (1.1-1) and **Error! Reference source not found.** ; where k is dependent on the application of interest.

13.2.3.7. Combined Coverage and Capacity Index (cc)

[The number *N* of simultaneous users per cell that can be supported achieving a target information throughput $R_{\min}$ with specified coverage reliability.

This performance metric can be approximated using either a simplified approximate evaluation methodology or a more detailed simulation as described below. Both methods are useful since the approximation methodology can be used to quickly compare two coverage enhancement techniques during the initial system concept development stage. The detailed simulations are useful to evaluate more carefully the most promising concepts. When results are presented the evaluation method used should be reported.]

13.2.3.7.1. Method 1: Simplified Combined Coverage and Capacity Index Evaluation

[This is a Simplified Methodology to evaluate Combined Coverage and Capacity Index (cc) using only the rate capability of each user. This can be evaluated without modeling higher layer protocols.

Assume that in a simulation N users are dropped uniformly in the service area. Let the required coverage for a given service be x% and the required information rate for that service be R_{min}. The first step in evaluating *cc* is to sort the MSs in descending order of achievable rate, assuming each utilizes the entire resources. Then, only the top x% of the MSs are considered.. Assume the number of users in the remaining group is *k*, and the data rate capability of user i is r_i (i = 1 to N) by using a scheduler that provides equal throughput to all the serviced users.

Then,

if the min(r_i) < R_{min}, $cc = 0$ (i.e. indicating that the service cannot be provided with the required coverage, regardless of the number of users).

Else, $$cc = \frac{k}{\sum_{i=1}^{k} \frac{R_{\min}}{r_i}},$$

Letting N become large, *cc* approaches the expected value of the number of users that can be supported by the system for that service with the given coverage (i.e. x%).

If a user communicates directly with BS, r is its effective rate to BS.]

13.2.3.7.2. Method 2: Detailed Combined Coverage and Capacity Index Evaluation

[The following is a more detailed methodology to evaluate the combined coverage and capacity metric.

Coverage reliability for a particular system (cell radius, shadow fading environment, relay station placement if present, and so on) with a particular number of users n each requiring information throughput $R_{\min}$ is calculated using a static system simulator. The static simulator shall model all other-user interference affects using appropriate path loss models and power control models (if any). The static simulator shall model a scheduler and resource manager that allocates resources to as many users as possible such that the target information throughput $R_{\min}$ is achieved. The static system simulator is run repeatedly with each run modeling a different instance of random drops of n MSs. Each simulator run results in $n_{s,i}$ MSs being served with the required information throughput and $n_{b,i}$ MSs being blocked due to insufficient carrier to interference plus noise ratio and/or insufficient time-frequency (or power) resources. $n = n_{b,i} + n_{s,i}$. In this equation, i is an index identifying a particular simulation run. Coverage reliability is a function of n and is:

$$\frac{1}{M \times n} \sum_{i=1}^{M} n_{s,i}$$

where M is the total number of simulation runs. The Combined Coverage and Capacity Index *cc* is the largest n for which

$$\frac{1}{M \times n} \sum_{i=1}^{M} n_{s,i} > x$$

]

13.3. Fairness Criteria

It may be an objective to have uniform service coverage resulting in a fair service offering for best effort traffic. A measure of fairness under the best effort assumption is important in assessing how well the system solutions perform.

The fairness is evaluated by determining the normalized cumulative distribution function (CDF) of the per user throughput. The CDF is to be tested against a predetermined fairness criterion under several specified traffic conditions. The same scheduling algorithm shall be used for all simulation runs. That is, the scheduling algorithm is not to be optimized for runs with different traffic mixes. The owner(s) of any proposal are also to specify the scheduling algorithm.

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

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

13.3.1. Equal Throughput or Full Fair Criterion

[To satisfy equal throughput requirement, all the users who are admitted to the system should get equal per user throughout if they have same amount of traffic to send/receive. In a full queue scenario, where traffic is assumed to be always available for transmission, the equal throughput requirement can be achieved by allocating time slots to users, such that the time allocated during a certain period for that user is inversely proportional to the data rate capability of the user.

If the data rate capability of the ith user is *r(i)*, under the equal throughput criterion, time allocated to each user should be proportional to 1 / *r(i)* (assuming equal input traffic).

The resulting equal aggregate throughput is, $C = \dfrac{1}{\sum_{i=1}^{n} 1/(r(i)}$

For some systems, such as those involving relays, one of the primary objectives of is to provide uniform service offering across users, so the total aggregate throughput under equal throughput criterion is a good metric to compare systems.]

13.3.2. Moderately Fair Solution

The CDF of the normalized throughputs with respect to the average user throughput for all users is determined. This CDF shall lie to the right of the curve given by the three points in Table 13.3.2-1.

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

Table 13.3.2-1: Moderately Fair Criterion CDF

13.3.3. Minimum Average Throughput Fairness Criterion

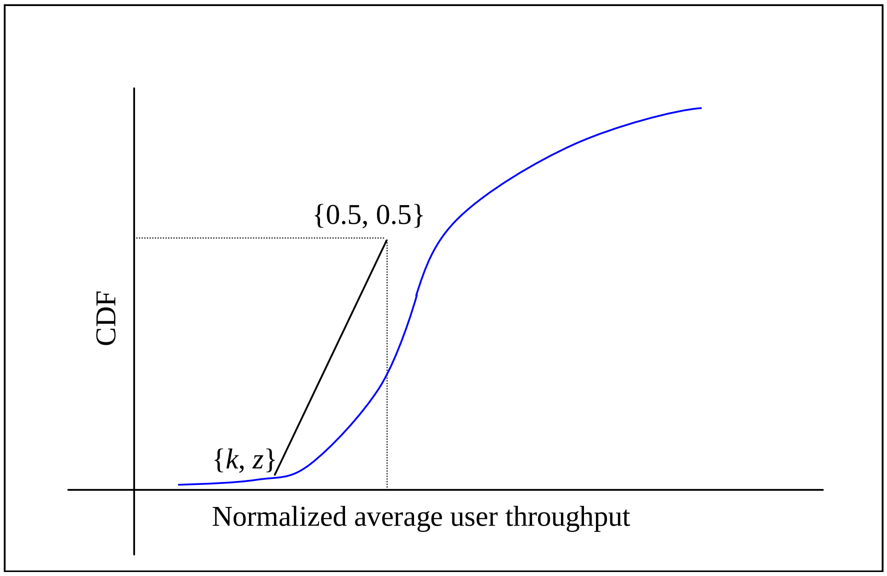
[This fairness criterion ensures a level of fairness of average user throughputs, with a requirement of a specific minimum average user throughput, R_{min}. The minimum average user throughput that all systems must satisfy is not a normalized value, but rather a minimum performance for a given service. The CDF of the normalized throughputs and the minimum normalized average user throughput, *k*, with respect to the average user throughput for all users, are determined This CDF shall lie to the right of the curve given by the three points in Table 13.3.3-1, where the line passes through the point {0.5, 0.5}, as in the Moderately Fair Solution.

Normalized Throughput w.r.t average user throughput (*x*)	CDF
k	*Z*
$k \le x \le 0.5$	$\frac{(1-2z)x+z-k}{1-2k}$
0.5	0.5

Table 13.3.3-1: Minimum Average Throughput Fairness Criterion

As some systems may tolerate some number of users in outage for the entire simulation window, up to a fraction of *z* users do not have to satisfy the minimum system requirements. The minimum average user throughput R_{min} is set to the service minimum. The allowed fraction of users in outage is *z*, and is selected based on the service.

The criterion addresses both coverage for a minimum average user throughput through point {*k, z*}, and average user throughput fairness through the requirement of being to the right of curve defined by entries of Table 13.3.3-1. Note that this criteria defaults to the moderately fair solution if *k* = 0.1 and *z* = 0.1. The equal throughput criteria can achieved by setting *k* = 1.0 and *z* = 0.0. Note that for *k* >0.5 the criteria is effectively described by a single point at {*k*, *z*}, which a minimum normalized average user throughput *k*, given some allowable outage *z*.



] **Figure 13.3.3-1: Minimum average throughput fairness criterion**

13.4. Additional Metrics (optional)

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

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

Figure 13.4.2-1shows 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.

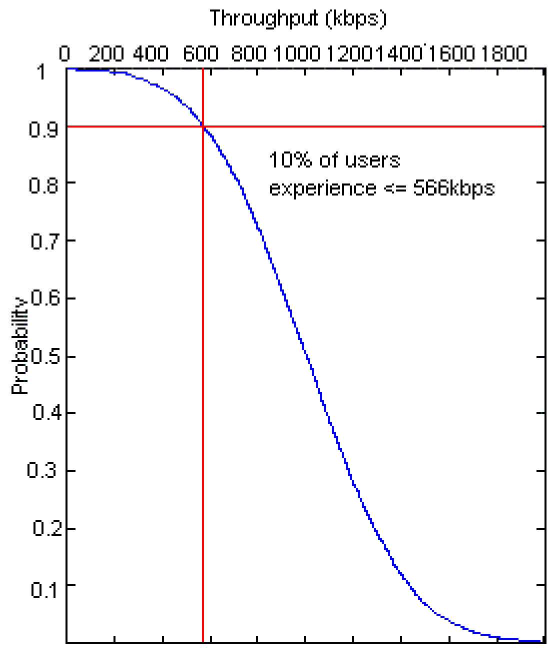


Figure 13.4.1-1: Service Distribution for a fixed load/coverage operating point

13.4.2. 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 **Error! Reference source not found.**, 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 13.4.2-1. 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.

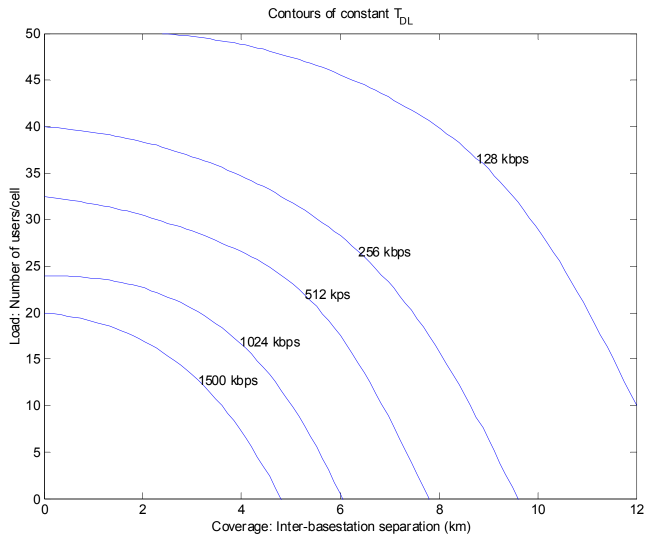


Figure 13.4.2-1: Contours of constant minimum service level

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 13.4.2-1 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.

]

14. Template for Reporting Results

Color	Sources	Document Reference
Green	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Dan Gal et al.	C80216m-07_063.doc

[The link-level and system-level simulation results shall be reported in the format exemplified in the following tables and graphs. Where applicable, one table and figure per simulation case should be provided. Models and assumptions should be aligned with those listed in this document. Any deviations should be highlighted.

Metric	Peak Rate (Mbps)	C-plane latency (ms)	U-plane delay (ms)	Intra- FA HO delay (ms)	Inter-FA HO delay (ms)
IEEE 802.16e Reference System					
IEEE 802.16m DL					
IEEE 802.16m UL					

Table 14-1: System Analysis Results

Metric	Average cell throughput and spectrum efficiency (x IEEE 802.16e Reference System)	Average user throughput and spectrum efficiency	Cell-edge user throughput and spectrum efficiency
IEEE 802.16e Reference System			
IEEE 802.16m baseline SU-MIMO			
IEEE 802.16m baseline MU-MIMO			
IEEE 802.16m			
IEEE 802.16m			

Table 14-2: Downlink full buffer system evaluation

Metric	Average cell throughput and spectral efficiency (x IEEE 802.16e Reference System)	Average user throughput and spectral efficiency	Cell-edge user throughput and spectral efficiency
IEEE 802.16e Reference System			
IEEE 802.16m baseline			
IEEE 802.16m MU-MIMO			
IEEE 802.16m SU-MIMO			
IEEE 802.16m …			

Table 14-3: Uplink full queue system evaluation results.

Metric	VoIP Capacity
IEEE 802.16m	

DL	
IEEE 802.16m UL	

Table 14-4: VoIP Results

]

14.1. 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 14.1-1. 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.

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[2], 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	-

Table 14.1-1: Evaluation Reports

]

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

Appendix-A: Correlation of Angular Spread and Shadowing Factor

Assuming correlation between AS and SF (both are log-normal distributed), randomly determine the AS and SF for the *d*th base station (*d* = 1 … *D*) as in the following with respect to a given mobile user:

$$\sigma_{AS,d} = 10 \wedge \left(\varepsilon_{AS}\beta_d + \mu_{AS}\right)$$
$$\sigma_{SF,d} = 10 \wedge \left(\sigma_{SH}\gamma_d / 10\right)$$

where $\mu_{AS} = E\left(\log_{10}(\sigma_{AS})\right)$ is the logarithmic mean of the distribution of AS (pre-defined), and $\varepsilon_{AS} = \sqrt{E\left[\log_{10}(\sigma^2_{AS,n})\right] - \mu^2_{AS}}$ is the logarithmic standard deviation of the distribution of AS (pre-defined). σ_{SH} is the shadow fading standard deviation given in dB (pre-defined). β_n and γ_n are generated together by:

$$\begin{bmatrix} \beta_n \\ \gamma_n \end{bmatrix} = \begin{bmatrix} c_{11} & c_{12} \\ c_{21} & c_{22} \end{bmatrix} \begin{bmatrix} w_{n1} \\ w_{n2} \end{bmatrix} + \begin{bmatrix} 0 & 0 \\ 0 & \sqrt{\zeta} \end{bmatrix} \begin{bmatrix} \xi_1 \\ \xi_2 \end{bmatrix}$$

where the two correlated Gaussian random variables are in turn respectively generated from independent Gaussian random variables w_{n1} and w_{n2}, as well as two global (applicable to all bases) independent Gaussian random variables ξ_1, ξ_2. The matrix **C** with elements c_{ij} multiplying the *w*'s is given by

$$\mathbf{C} = \left(\mathbf{A} - \mathbf{B}\right)^{1/2} = \left(\begin{bmatrix} 1 & \rho_{\gamma\beta} \\ \rho_{\gamma\beta} & 1 \end{bmatrix} - \begin{bmatrix} 0 & 0 \\ 0 & \zeta \end{bmatrix} \right)^{1/2}$$

where the superscript "½" denotes the matrix square root. The intra-site correlation between SF and AS is □□□= -0.6 and inter-site SF correlation is ζ=0.5 in SCM.

Appendix-B: Calculation of Circular Angular Spread

The following derivation of the angular spread assumes that each multipath further contains *M* sub-paths. The creation of subpath is to approximate the power angular distribution (Laplacian for example) with a finite number of points. With the introduction of subpath, the AS of the overall signal can be more easily computed as, for a signal with *N* multi-paths,

$$\sigma_{AS} = \sqrt{\frac{\sum_{n=1}^{N}\sum_{m=1}^{M}\left(\theta_{n,m,\mu}\right)^2 \cdot P_{n,m}}{\sum_{n=1}^{N}\sum_{m=1}^{M} P_{n,m}}}$$

where $P_{n,m}$ is the power for the *m*th subpath of the *n*th path, $\theta_{n,m,\mu}$ is defined as

$$\theta_{n,m,\mu} = \begin{cases} 2\pi + \left(\theta_{n,m} - \mu_\theta\right) & \text{if } \left(\theta_{n,m} - \mu_\theta\right) < -\pi \\ \left(\theta_{n,m} - \mu_\theta\right) & \text{if } \left|\theta_{n,m} - \mu_\theta\right| \le \pi \\ 2\pi - \left(\theta_{n,m} - \mu_\theta\right) & \text{if } \left(\theta_{n,m} - \mu_\theta\right) > \pi \end{cases},$$

μ_θ is defined as

$$\mu_\theta = \frac{\sum_{n=1}^{N}\sum_{m=1}^{M}\theta_{n,m} \cdot P_{n,m}}{\sum_{n=1}^{N}\sum_{m=1}^{M} P_{n,m}}$$

and $\theta_{n,m}$ is the AoA (or AoD) of the *m*th subpath of the *n*th path. Note that we have dropped the AoA (AoD) subscript for convenience.

We note that the angle spread should be independent of a linear shift in the AoAs. In other words, by replacing $\theta_{n,m}$ with $\theta_{n,m} + \Delta$, the angle spread $\sigma_{AS}(\Delta)$ which is now a function of Δ should actually be constant no matter what Δ is. However, due to the ambiguity of the modulo 2π operation, this may not be the case. Therefore the angle spread should be the minimum of $\sigma_{AS}(\Delta)$ over all Δ:

$$\sigma_{AS} = \min_{\Delta} \sigma_{AS}(\Delta) = \sqrt{\frac{\sum_{n=1}^{N}\sum_{m=1}^{M}\left(\theta_{n,m,\mu}(\Delta)\right)^2 \cdot P_{n,m}}{\sum_{n=1}^{N}\sum_{m=1}^{M} P_{n,m}}}$$

where $\theta_{n,m,\mu}(\Delta)$ is defined as

$$\theta_{n,m,\mu}(\Delta) = \begin{cases} 2\pi + \left(\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right) & \text{if } \left(\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right) < -\pi \\ \left(\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right) & \text{if } \left|\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right| \le \pi \\ 2\pi - \left(\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right) & \text{if } \left(\theta_{n,m}(\Delta) - \mu_\theta(\Delta)\right) > \pi \end{cases},$$

$\mu_\theta(\Delta)$ is defined as

$$\mu_\theta(\Delta) = \frac{\sum_{n=1}^{N}\sum_{m=1}^{M}\theta_{n,m}(\Delta)\cdot P_{n,m}}{\sum_{n=1}^{N}\sum_{m=1}^{M}P_{n,m}}$$

and $\theta_{n,m}(\Delta) = \theta_{n,m} + \Delta$.

Note that when the angular spread is small, the angular spread computed conventionally and the circular angular spread often give the same value.

Appendix-C: Spatial Correlation Calculation

In order to compute the integration defined in step 6, two methods can be considered here:

Method-1: Using 20 subpaths to approximate the Laplacian PDF

For each path, generate 20 subpaths with some angular offsets from the per-path AOD_n and AOA_n. The angular offsets of the k-th (k=1..20) subpath are determined by (the offsets are the same for all paths)

$$\psi_{k,BS} = \Delta_k AS_{BS,Path}$$
$$\psi_{k,MS} = \Delta_k AS_{MS,Path}$$

where the values of Δ_k are given below.

Table C-1: Value of Δ_k

Sub-path number *k*	Δ_k
1,2	± 0.0447°
3,4	± 0.1413°
5,6	± 0.2492°
7,8	± 0.3715°
9,10	± 0.5129°
11,12	± 0.6797°
13,14	± 0.8844°
15,16	± 1.1481°
17,18	± 1.5195°
19,20	± 2.1551°

Derive the antenna spatial correlation at the BS and MS between the p-th and q-th antenna as:

$$r_{n,BS}(p,q) = \frac{1}{20}\sum_{k=1}^{20} \exp\left\{ j\frac{2\pi d_{BS}}{\lambda}(p-q)\sin(AOD_n + \psi_{k,BS}) \right\}$$
$$r_{n,MS}(p,q) = \frac{1}{20}\sum_{k=1}^{20} \exp\left\{ j\frac{2\pi d_{MS}}{\lambda}(p-q)\sin(AOA_n + \psi_{k,MS}) \right\}$$

where d_{BS} (d_{MS}) is the antenna spacing at BS (MS) and λ is the wavelength.

Method-2:Pre-compute the correlation values with quantized AOA, AOD

Pre-calculate the BS spatial correlation matrices for a set of $AOD \in \{-90^o, -80^o, \ldots, 0^o, \ldots, 80^o, 90^o\}$ and the MS spatial correlation matrices for a set of $AOA \in \{-90^o, -80^o, \ldots, 0^o, \ldots, 80^o, 90^o\}$

$$R_{BS}(m,p,q) = \int_{-\infty}^{\infty} p(\alpha)\exp\left\{ j\frac{2\pi d_{BS}}{\lambda}(p-q)\sin(AOD[m]+\alpha) \right\} d\alpha$$
$$R_{MS}(m,p,q) = \int_{-\infty}^{\infty} p(\beta)\exp\left\{ j\frac{2\pi d_{MS}}{\lambda}(p-q)\sin(AOA[m]+\beta) \right\} d\beta$$

where m is the quantization step index, α , β are the angular offset at BS and MS, respectively with Laplacian PDF as defined in step-6.

For each path, determine the index m_{BS} corresponding to AOD$_{n,}$

$$m_{BS} = \left\lfloor \frac{AOD_n}{10} \right\rfloor$$

and the index m_{MS} corresponding to AOA$_n$

$$m_{MS} = \left\lfloor \frac{AOA_n}{10} \right\rfloor$$

The spatial correlation matrix for this path is then

$$\begin{aligned} r_{n,BS}(p,q) &= R_{BS}(m_{BS}, p, q) \\ r_{n,MS}(p,q) &= R_{MS}(m_{MS}, p, q) \end{aligned}$$

Appendix-D: Polarized Antenna

Correlation between polarized antennas results from the cross polarization power ratio (XPR). The polarization matrix is given by:

$$\mathbf{S} = \begin{bmatrix} s_{vv} & s_{vh} \\ s_{hv} & s_{hh} \end{bmatrix},$$

where *v* denotes vertical and *h* horizontal polarization, the first index denoting the polarization at BS and the second the polarization at MS. In the ITU scenarios we assume -8 dB per-tap power ratio between vertical-to-horizontal and vertical-to-vertical polarisations (also P_{hv}/P_{hh} = -8dB). The -8dB value was adopted from reference [11][12]. The following derivation of antenna correlation due to polarization with -8dB XPD can also be found in [15]. This results in the following mean power per polarization components

$$\begin{aligned} p_{vv} &= E\left\{ |s_{vv}|^2 \right\} = 0\,\mathrm{dB} = 1 \\ p_{vh} &= E\left\{ |s_{vh}|^2 \right\} = -8\,\mathrm{dB} = 0.1585 \\ p_{hv} &= E\left\{ |s_{hv}|^2 \right\} = -8\,\mathrm{dB} = 0.1585 \\ p_{hh} &= E\left\{ |s_{hh}|^2 \right\} = 0\,\mathrm{dB} = 1 \end{aligned}$$

If the MS polarizations are assumed to be vertical and horizontal, but the BS polarizations are slant +45° and -45°. The MS and BS polarization matrices $\mathbf{P}_{MS}$ and $\mathbf{P}_{BS}$ respectively are rotation matrices, which map vertical and horizontal polarizations to MS and BS antenna polarizations.

$$\mathbf{P}_{MS} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix}$$

$$\mathbf{P}_{BS} = \frac{1}{\sqrt{2}} \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix}$$

The total channel is the matrix product of the BS polarization, the channel polarization, and the MS polarization:

$$\mathbf{Q} = \mathbf{P}_{BS}\,\mathbf{S}\,\mathbf{P}_{MS} = \frac{1}{\sqrt{2}} \begin{bmatrix} s_{vv} + s_{hv} & s_{vh} + s_{hh} \\ s_{vv} - s_{hv} & s_{vh} - s_{hh} \end{bmatrix}$$

The covariance matrix of the channel is

$$
\begin{aligned}
\mathbf{\Gamma} &= E\left\{vec(\mathbf{Q})\cdot vec(\mathbf{Q})^H\right\} \\
&= E\left\{\frac{1}{2}\begin{bmatrix}
(s_{vv}+s_{hv})(s_{vv}+s_{hv})^* & (s_{vv}+s_{hv})(s_{vv}-s_{hv})^* & (s_{vv}+s_{hv})(s_{vh}+s_{hh})^* & (s_{vv}+s_{hv})(s_{vh}-s_{hh})^* \\
(s_{vv}-s_{hv})(s_{vv}+s_{hv})^* & (s_{vv}-s_{hv})(s_{vv}-s_{hv})^* & (s_{vv}-s_{hv})(s_{vh}+s_{hh})^* & (s_{vv}-s_{hv})(s_{vh}-s_{hh})^* \\
(s_{vh}+s_{hh})(s_{vv}+s_{hv})^* & (s_{vh}+s_{hh})(s_{vv}-s_{hv})^* & (s_{vh}+s_{hh})(s_{vh}+s_{hh})^* & (s_{vh}+s_{hh})(s_{vh}-s_{hh})^* \\
(s_{vh}-s_{hh})(s_{vv}+s_{hv})^* & (s_{vh}-s_{hh})(s_{vv}-s_{hv})^* & (s_{vh}-s_{hh})(s_{vh}+s_{hh})^* & (s_{vh}-s_{hh})(s_{vh}-s_{hh})^*
\end{bmatrix}\right\} \\
&= \frac{1}{2}\begin{bmatrix}
p_{vv}+p_{hv} & p_{vv}-p_{hv} & 0 & 0 \\
p_{vv}-p_{hv} & p_{vv}+p_{hv} & 0 & 0 \\
0 & 0 & p_{vh}+p_{hh} & p_{vh}-p_{hh} \\
0 & 0 & p_{vh}-p_{hh} & p_{vh}+p_{hh}
\end{bmatrix}
\end{aligned}
$$

Here the property of uncorrelated fading between different elements in **S** (i.e. $E\left\{s_{ij}s_{kl}^*\right\}=0, \quad i\neq k, j\neq l$) has been used to simplify the expressions. Plugging the numericacl example of -8dB XPD, we have

$$
\mathbf{\Gamma}=\frac{1}{2}\begin{bmatrix}
1+0.1585 & 1-0.1585 & 0 & 0 \\
1-0.1585 & 1+0.1585 & 0 & 0 \\
0 & 0 & 0.1585+1 & 0.1585-1 \\
0 & 0 & 0.1585-1 & 0.1585+1
\end{bmatrix}=\begin{bmatrix}
0.5793 & 0.4208 & 0 & 0 \\
0.4208 & 0.5793 & 0 & 0 \\
0 & 0 & 0.5793 & -0.4208 \\
0 & 0 & -0.4208 & 0.5793
\end{bmatrix}
$$

When all of the diagonal elements are equal, the covariance matrix can be further normalised to correlation matrix:

$$
\Gamma=\begin{bmatrix}
1 & \gamma & 0 & 0 \\
\gamma & 1 & 0 & 0 \\
0 & 0 & 1 & -\gamma \\
0 & 0 & -\gamma & 1
\end{bmatrix}
$$

Value of □ depends only on XPR and it is obtained from the previous matrix after the normalization of the diagonal values to “1”. With different orientations of MS and BS antenna polarizations, also the covariance matrix structure will be different.

Appendix-E: LOS Option with a K-factor

A single-tap MIMO channel can be added to the TDL channels in this case and then modify the time-domain channels as:

$$\mathbf{H}_n = \begin{cases} \sqrt{\frac{1}{K+1}}\mathbf{H}_n + \sqrt{\frac{K}{K+1}}\mathbf{H}^{LOS} & n = 1(\text{first tap}) \\ \sqrt{\frac{1}{K+1}}\mathbf{H}_n & n \neq 1 \end{cases}$$

where the K-factor is in decimal and the LOS component is defined as, between p-th BS antenna and q-th MS antenna

$$\mathbf{H}^{LOS}(p,q) = \exp\left(j\frac{2\pi d_{BS}(p-1)}{\lambda}\sin\left(\theta_{BS}\right)\right) \times \exp\left(j\frac{2\pi d_{MS}(q-1)}{\lambda}\sin\left(\theta_{MS}\right)\right)$$

where d_{BS} and d_{MS} are antenna spacing at the BS and MS, respectively, assuming uniform linear array in this case.

Appendix-F: Antenna Gain Imbalance and Coupling

Overall receive correlation matrix is

$$\mathbf{H}_n^{'} = \begin{bmatrix} \sqrt{\dfrac{1}{c+1}} & \sqrt{\dfrac{c}{c+1}} \\ \sqrt{\dfrac{c}{c+1}} & \sqrt{\dfrac{1}{c+1}} \end{bmatrix} \begin{bmatrix} 1 & 0 \\ 0 & \sqrt{a} \end{bmatrix} \mathbf{H}_n$$

Where antenna-1 to antenna-2 coupling coefficient (leakage of ant-1 signal to ant-2) is "c" (linear) and the antenna-1 and antenna gain ratio is "a" (linear).]

Appendix-G: 19-Cell Wrap-Around Implementation

G-1. Multi-Cell Layout

In Figure G.1-1, a network of cells is formed with 7 clusters and each cluster consists of 19 cells. Depending on the configuration being simulated and required output, the impact of the outer 6 clusters may be neglected. In such cases, only 19 cells of the center cluster may be modeled.

For the cases where modeling outer-cells are necessary for accuracy of the results, the wrap around structure with the 7 cluster network can be used. In the wrap around implementation, the network is extended to a cluster of networks 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 G.1-1. 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 G-3-3-1.

An example of the antenna orientations in case of a sectorized system is defined in . 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 $(a+3R, b+8\sqrt{3}R/2)$;
c. Distance between (x,y) and $(a-3R, b-8\sqrt{3}R/2)$;
d. Distance between (x,y) and $(a+4.5R, b-7\sqrt{3}R/2)$;
e. Distance between (x,y) and $(a-4.5R, b+7\sqrt{3}R/2)$;
f. Distance between (x,y) and $(a+7.5R, b+\sqrt{3}R/2)$;
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.

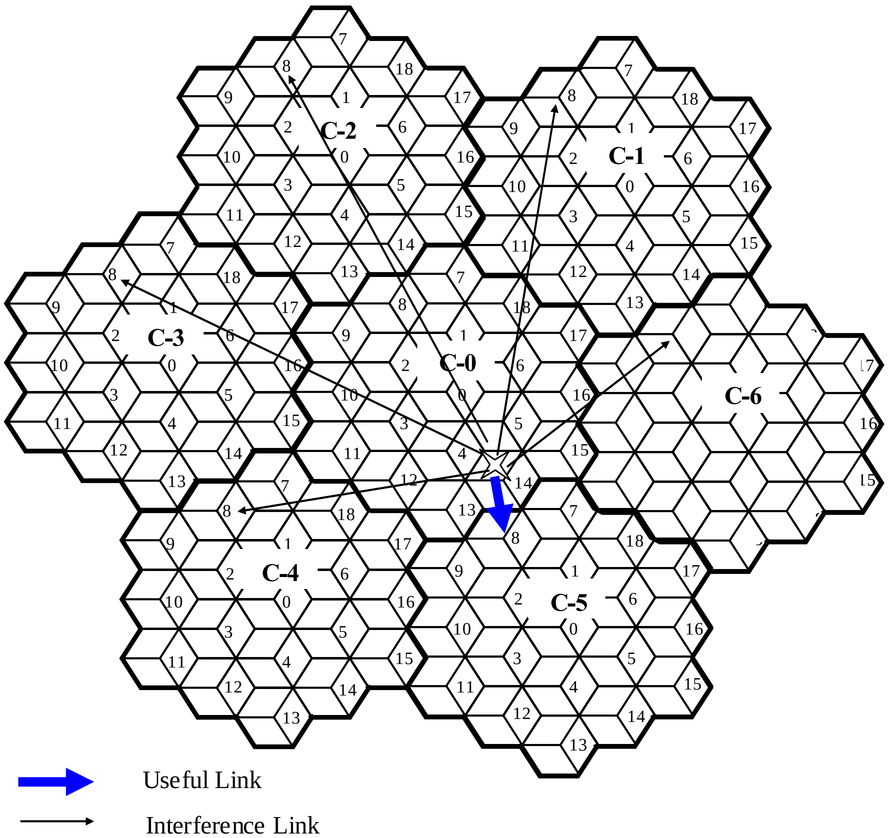


Figure G.1-1: Multi-cell Layout and Wrap Around Example

G-2. Obtaining virtual MS locations

The number of MSs is predetermined for each sector, where each MS location is uniformly distributed. The MS assignment is only done for the cluster-0 from where the decided MSs are replicated in the other six clusters. The purpose to employ this wrap-around technique, as will be discussed in later section, is to easily model the interferences from other cells.

G-3. Determination of serving cell/sector for each MS in a wrap-around multi-cell network

The determination of serving cell for each MS is carried out by two steps due to the wrap-around cell layout. The first step is to determine the 19 shortest distance cells for each MS from all seven logical cells clusters, and the second step is to determine the serving cell/sector among the nearest 19 cells for each MS based on the strongest link according to the path-loss and shadowing.

To determine the shortest distance cell for each MS, the distances between the target MS and all logical cell clusters should be evaluated and the 19 cells with a shortest distance in all 7 cell clusters should be selected. Figure G.1-1 illustrates an example for determination of the shortest distance cell for the link between MS and cell-8. It can be seen that the cell-8 located in cluster-5 generates the shortest distance link between MS and cell-8.

To determine the serving cell for each MS, we need to determine 19 links, whereby we may additionally determine the path-loss, shadowing and transmit/receive antenna gain in consideration of antenna pattern corresponding to the nearest 19 cells/sectors. The serving cell for each MS should offer a strongest link with a strongest received long-term power. [If the serving cell/sector doesn't belong to the center 19 cell cluster, this MS shall be redropped.] It should be noted that the shadowing experienced on the link between MS and cells located in different clusters is the same.

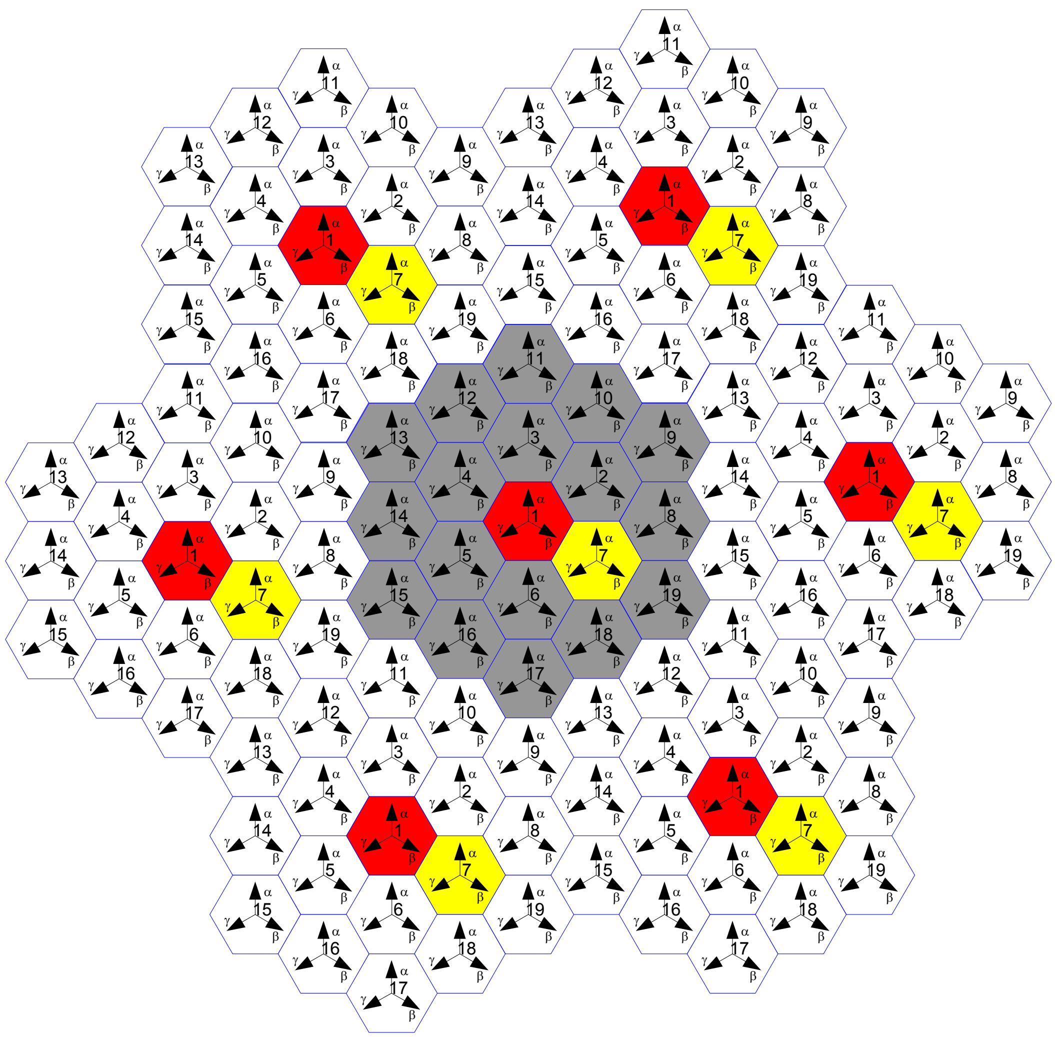


Figure G-3-3-1 : An example of the antenna orientations for a sectorized system to be used in the wrap-around simulation. The arrows in the Figure show the directions that the antennas are pointing

Appendix-H: Calculation of PAPR and Cubic Metric

Color	Sources	Document Reference
Green Orange	Sassan Ahmadi et al. Jeff Zhuang et al.	C80216m-07_069.doc

When power amplifier and radiated spectrum performance related subjects are discussed, for example to evaluate the amount of backoff needed to achieve a certain radiated spectrum requirement (e.g., Adjacent Channel Loss Ratio or ACLR), PAPR and CM are two options that are commonly used.

H-1. Peak to Average Power Ratio (PAPR)

PAPR could be used to calculate the TX power amplifier back-off value. Since it is not desirable to change the back-off value at every symbol, the rms transmit power should be evaluated over a long period of time. For the calculation of average power, the worst case assumption, which causes maximum average power, should be made; e.g., maximum loading of data. In addition it is desirable for the P_{av1} defined below to have a similar value from symbol-to-symbol.

P_{av1} denotes the average power over one OFDM symbol duration defined by

$$P_{av1} = \frac{1}{T}\int_0^T |s(t)|^2 dt \qquad \text{(G-1-1)}$$

where s(t) is the time domain OFDM signal. Note that s(t) may include a weight matrix to take into account the transmit beamforming, and closed-loop MIMO if any. Note that the weight matrix is a function of channel and can increase the peak power.

P_{av} is an ensemble average of OFDM symbols over all possible random data, and channel

$$P_{av} = E\left\{|s(t)|^2\right\} \qquad \text{(G-1-2)}$$

In transmit antenna selection, the average power should be evaluated only when the antenna is selected in order to get the maximum (or worst case) average power.

P_{max} is the peak power measured over one OFDM symbol duration and it is defined as

$$P_{\max} = \max_{0 \le t < T} |s(t)|^2 \qquad \text{(G-1-3)}$$

Then, the PAPR1 and PAPR are defined as follows

$$PAPR_1 = \frac{P_{\max}}{P_{av1}}$$

$$PAPR = \frac{P_{\max}}{P_{av}}, \qquad \text{(G-1-4)}$$

The CCDF of PAPR1 and PAPR should be evaluated over all possible channel conditions, and random OFDM signals. In the PAPR simulations, the signal shall be over-sampled at least 8 times. The guard interval can be included in the OFDM symbol duration (T). From P_{av} and CCDF of PAPR, the back-off value and operating point of PA can be obtained.

H-2. Instantaneous PAPR

The instantaneous power can be written as

$$P_i = |s(t)|^2 .$$

We can define two different instantaneous PAPR by normalizing using either P_{av1} or P_{av}

$$PAPR_{i1} = \frac{P_i}{P_{av1}}, \text{ and}$$

$$PAPR_i = \frac{P_i}{P_{av}} .$$

Note that we should look at lower probability in CCDF of $PAPR_i$ than that in PAPR case depending on the number of samples during one OFDM symbol.

H-3. Cubic Metric

The cubic metric is a well established approach ([47], Section 6.2.2) to assessing the impact of the peak excursions of a power amplifier transmitted waveform. Originally specified for direct sequence spread CDMA waveforms (such as 3GPP WCDMA), the cubic metric method has been further studied in the specification of variable bandwidth OFDM systems [48], where it has again been found to be an accurate predictor of critical power amplifier and radiated spectrum performance criteria such as Adjacent Channel Loss Ratio (ACLR).

A larger CM denotes a more difficult waveform type for power amplification, leading to greater spectrum expansion or adjacent channel power insertion. Alternatively, CM can be used to compute the amount of "power back-off" required to achieve the same ACLR. A waveform CM value which is greater by 1dB than the CM for a second waveform implies the first waveform, for the same PA and current drain, can be radiated at 1dB higher mean power level.

The cubic metric (CM) of a waveform $v(t)$ is formally defined as

$$CM\,(dB) = \frac{20\log_{10}\{rms\left[v_{norm}^{3}(t)\right]\} - 20\log_{10}\{rms\left[v_{ref}^{3}(t)\right]\}}{K} \quad (0.18)$$

where $rms(x) = \sqrt{\frac{x^T x}{N}}$ (where x is sampled as a length-N vector), and $v_{norm}(t) = \frac{|v(t)|}{rms[v(t)]}$.

In (0.18) the reference CM of $20\log_{10}\{rms\left[v_{ref}^{3}(t)\right]\}$ and the coefficient K are determined empirically by studying the behaviour of contemporary mobile station power amplifiers [48].

For the present purpose, the following expression is applicable

$$CM = \frac{20\log_{10}\left\{rms\left[v_{norm}^{3}\left(t\right)\right]\right\} - 1.52}{1.56} + 0.77 \quad dB$$

Appendix-I: Overhead Calculations

Color	Sources	Document Reference
Green	Sassan Ahmadi et al.	C80216m-07_069.doc
Red	Dan Gal et al.	C80216m-07_063.doc

I-1. Overhead Channels

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I-1.1. Dynamical Simulation of the Downlink 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 e.g., 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 demodulation performance (i.e., frame error rate) of the downlink control channel could be assessed using the link abstraction method used to model traffic channels, with proper modifications, if necessary, to reflect any difference in the transmission or coding format of the control channel.

[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)] [SNR] and predict the demodulation performance.

For dynamic overhead channels with open-loop control (if used), the simulations should take into account the estimate of the required downlink 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 [SNR] [Eb/No (or similar metric)].

Once the received [SNR] [Eb/No (or similar metric)] is obtained and the frame error rate is predicted, 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 downlink and uplink, 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.

I-1.2. Uplink Modelling in Downlink 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.

I-1.3. Signalling 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.

I-2. Calculation of Overhead

[

I-2.1. Calculation of L1 overhead

The L1 overhead shall be accounted for the purpose of calculation of system performance metrics such as spectral efficiency, user throughput, etc. The following are the list of L1 overhead that shall be accounted for in the overhead calculation:

1. Number of sub-carriers that carry preamble ($N_{preamble}$)
2. Number of sub-carriers that are used as the guard carriers and DC sub-carrier (N_{guard}).
3. Number of sub-carriers that are assigned to the TX/RX switching points (N_{gap}) (for TDD duplex scheme only).
4. Number of sub-carriers that are used to carry pilots (N_{pilot}) in downlink or uplink sub-frame.
5. The cyclic prefix (CP) or the guard interval (GI) is accounted for through the actual number of OFDM symbols within a frame.

Given transmission bandwidth (BW) and sub-carrier spacing (1/T when there is no cyclic prefix), the ideal number of sub-carriers per OFDM symbol becomes $BW \cdot T$. The number of OFDM symbols over frame duration (T_f) excluding L1 overhead can be expressed as $\frac{T_f}{T}$. Hence, the total number of available sub-carriers over BW and T_f becomes $M = BW \cdot T_f$.

However, due to L1 overhead, the actual number of available sub-carriers (N) over a frame and the transmission bandwidth will be smaller than M.

$$L1_{overhead} = \frac{M-N}{M} \times 100 \text{ in \%} \qquad \text{(I-2.1-1)}$$

As an example in IEEE 802.16e reference system, M=10 MHz * 5 ms = 50000. Using DL PUSC, 47 OFDM symbols are available (including the preamble and cyclic prefix overhead). The number of usable sub-carriers for both data and pilot is 840*47 including the effect of guard sub-carriers. In PUSC, the number of pilots is 840*47/56*8=5640. Thus, N is equal to 840*47*48/56=33840. Hence, the L1 overhead becomes 32.3%.

I-2.2. Calculation of L1+L2 overhead

The L2 overhead includes the following:

1. The number of OFDM symbols (or sub-carriers) that are used for MAP
2. The average number of ODFM symbols (or sub-carriers) that are used for system configuration information (FCH, DCD/UCD).
3. The number of OFDM symbols (or sub-carriers) that are used for ACK/NACK,
4. The number of OFDM symbols (or sub-carriers) that are used for CQICH,
5. The number of OFDM symbols (or sub-carriers) that are used for Ranging

Let L denote the number of sub-carriers over a frame and the bandwidth excluding L2 overhead such as MAP, control channel, etc. Then, the L1+L2 overhead is defined

$$L1 + L2_{overhead} = \frac{M-L}{M} \times 100 \text{ in \%.}$$

The L2 overhead can be written as

$$L2_{overhead} = \frac{N-L}{N} \times 100 \text{ in \%.}$$

Note that $L1 + L2_{overhead} \neq L1_{overhead} + L2_{overhead}$.

Appendix-J: 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.

J-1. Cell/Sector Locations

The inter-cell distance is 2.5 km and the cells are located as shown in the Figure J 1.

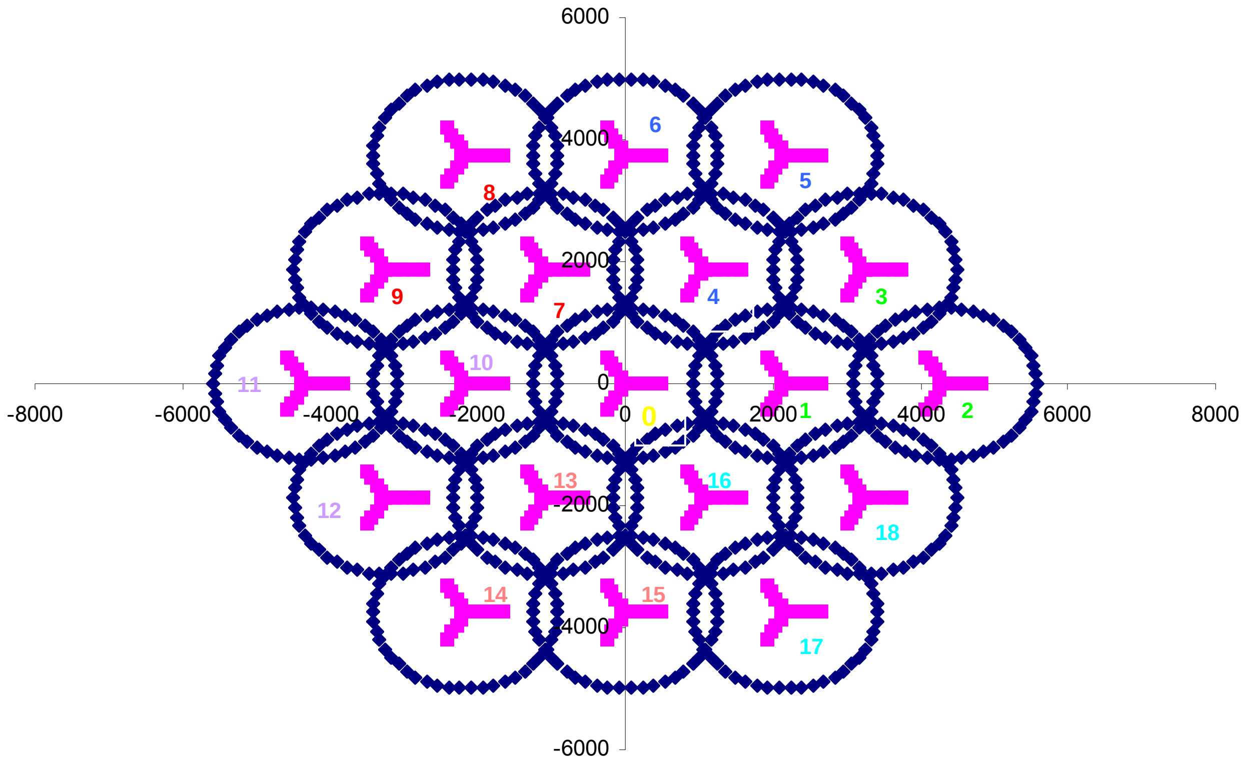

Figure J 1:Cell definition in the Cartesian Coordination System and the Numbering of Cells

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.

J-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:

10users..xls

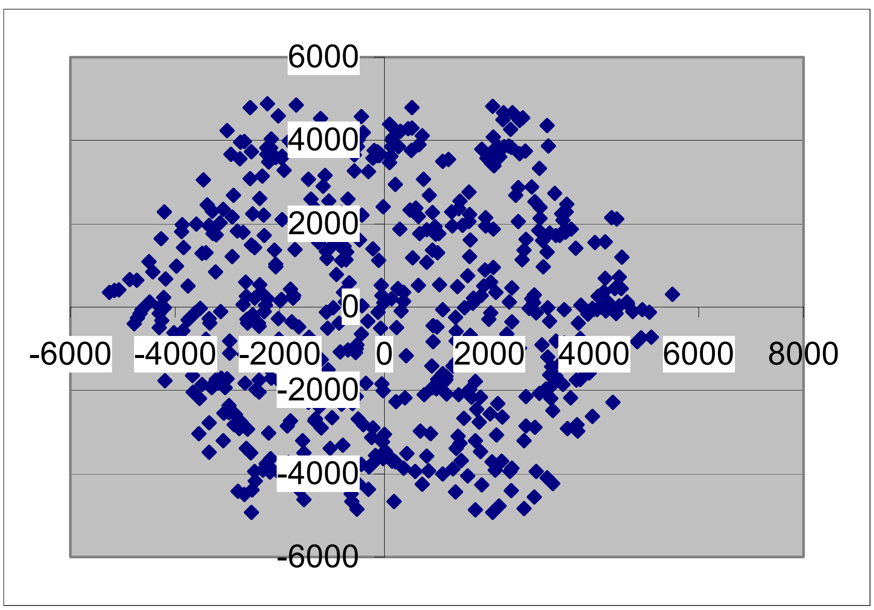

Figure J- 2: MS Locations

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