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Title	Multi-hop System Evaluation Methodology (Channel Model and Performance Metric)	
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Re:	Response to a call for contributions for the Relay TG, see C80216j-06/001.pdf
Abstract	This document captures scope of the Multi-hop System Evaluation Methodology including the Channel Model, Traffic Model and Performance Metrics.
Purpose	System Evaluation Methodology including the Channel Model, Traffic Model and Performance Metrics documented in this contribution is used as a reference for the performance evaluation for the IEEE802.16j Task Group.
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Multi-hop System Evaluation Methodology

1 Introduction

The scope of this Multi-hop System Evaluation Methodology is to develop and specify parameters and methods associated with the channel models, traffic models, performance metrics that would serve as guidelines to aid in the evaluation and comparisons of technology proposals for IEEE 802.16 TGj. It is not the intention of this document to mandate the use of this evaluation methodology in the comparisons of proposals.

1.1 Simulation overview

In this section, an example of the Simulation model is provided. Figure 1 shows the components and methodology that would serve as a baseline for the rest of this document.

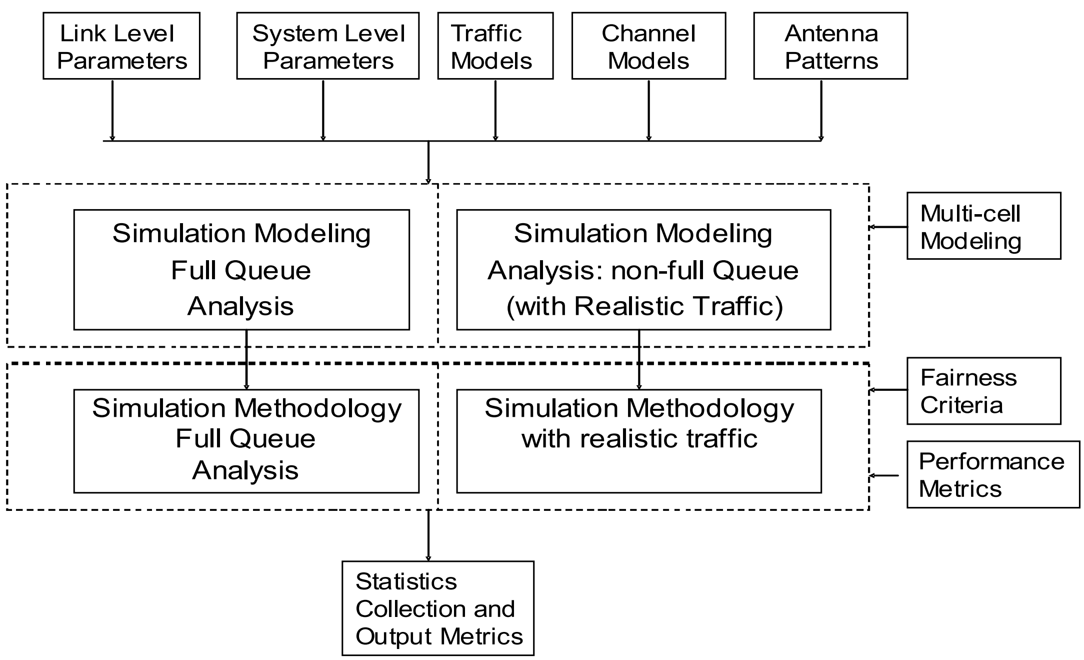

Figure 1 Simulation Components and Overall Methodology

2 Channel Models

[Editor's note: adopt the modified IEEE802.16d SUI channel model as baseline [14], and open for further comparison with other models such as the path-loss models in [6]]

2.1 Path-Loss Model

2.1.1 Path-loss Types

The path loss for the IEEE802.16j system contains the basic models for the IEEE802.16-2004 and additional path-loss associated with RS nodes. The path-loss types are listed in Table 1

Table 1 Summary Table of Path-loss Types for IEEE802.16j Relay System

Category	Links	Description		Reference	Note
Type A	BS–MS	Hilly terrain with moderate-to-heavy tree densities		Section 2.1.2.1	IEEE 802.16 Type A model
Type B		Intermediate path-loss condition			IEEE 802.16 Type B model
Type C		Flat terrain with light tree densities			IEEE 802.16 Type C model
Type D	BS–RS RS–RS	Both node-antennas (BS/RS) above rooftop	LOS	Section 2.1.2.2	Modified IEEE 802.16 model
			NLOS		
Type E	BS–RS RS–RS RS–MS	Only one node-antenna (BS/RS) above rooftop	NLOS	Section 2.1.2.4	Modified IEEE 802.16 model
Type F	RS–RS RS–MS	Both node-antennas (BS/RS) below rooftop	LOS	Section 2.1.2.5	Advanced LOS
			NLOS	Section 2.1.2.6	Berg/WiNNER
Type G	RS–RS RS–MS	Indoor Office	NLOS	Section 2.1.2.7	ITU model

2.1.1.1 The relationship path-loss models with the relay system usage models

[Editor's note: The linkage with the path-loss models defined in Table 1 and the usage models for the IEEE802.16j is FFS]

2.1.2 Detailed Path-loss Models

2.1.2.1 Type-A/B/C: BS ↔ MS, BS ↔ MRS, BS ↔ NRS)

The IEEE 802.16 path-loss and shadow fading model is given by [14]

$$PL = A + 10 \cdot \gamma \cdot \log_{10}(d / d_0) + \Delta PL_f + \Delta PL_h + s\ dB \qquad (1)$$

where d_0=100m and $d>d_0$. $A=20\cdot\log_{10}(4\pi d_0/\lambda)$ and $\gamma=(a - b\cdot h_b + c/h_b)$. λ is the wavelength in meter and h_b is the base station antenna height, which is between 10m and 80m. "***s***" is the log-normal shadow fading component in dB. Three propagation scenarios are categorized as

Terrain Type A: Hilly terrain with moderate-to-heavy tree densities

Terrain Type B: Intermediate path-loss condition

Terrain Type C: Flat terrain with light tree densities

The corresponding parameters for each propagation scenario are

Table 2 Parameters for the Type A/B/C

Model Parameter	Terrain Type A	Terrain Type B	Terrain Type C
a	4.6	4	3.6
b	0.0075	0.0065	0.005
c	12.6	17.1	20

Moreover, the correction factors for carrier frequency (ΔPL_f) and receive antenna height (ΔPL_h) are:

$$\Delta PL_f = 6 \cdot \log_{10}(f / 2000)\ dB \qquad (2)$$

where f is the carrier frequency in MHz.

ΔPL_h = - 10.8 · $\log_{10}(h / 2)\ dB$; for Terrain Type A and B (3)
ΔPL_h = - 20 · $\log_{10}(h / 2)\ dB$; for Terrain Type C
where h is the receive antenna height between 2m and 10m.

2.1.2.2 Type-D: BS ↔ RS, LOS (ART-to-ART)

This scenario is shown in Figure 2, where both the BS and RS antennas are mounted above the rooftops (ART) and they have a LOS between them.

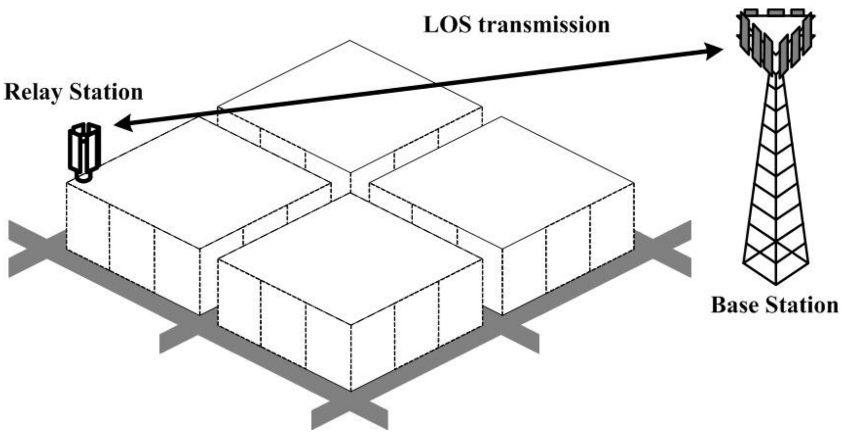

Figure 2 BS-RS link with LOS/NLOS

For this link a modified IEEE 802.16d channel model in section 2.1.2.1. There are three categories for this model, where each category represents a different environment. The most benign category (category C) is chosen for this scenario to allow for the fact that the relays in this case are assumed to have been deployed with a good LOS back to the BS. The model is equal to the free space path loss up to a breakpoint, which is determined by the transmission frequency and the relay antenna height. Beyond the breakpoint, the path loss exponent increases, and this is to account for the fact that LOS probability will decrease with distance from the BS. This factor is also important for multi-cell simulations for interference calculations. The relay will only be deployed to try to give LOS back to the 'wanted' BS. Interfering BSs (at greater distance) will most likely not have a LOS back to the BS, and the path loss model will account for this.

$$PL(dB) = \begin{cases} 20\log\left(\dfrac{4\pi d}{\lambda}\right) & \text{for } d \le d_0 \\ A + 10\gamma\log\left(\dfrac{d}{d_0}\right) + PL_f + PL_{ht} & \text{for } d > d_0 \end{cases}$$

where,

$$A = 20\log\left(\frac{4\pi d_0^{'}}{\lambda}\right)$$

$$d_0 = 100m$$

$$d_0^{'} = d_0 10^{-\frac{PL_f + PL_{ht}}{10\gamma}}$$

$$\gamma = a - bh_b + \frac{c}{h_b}$$

$$PL_f = 6\log\left(\frac{f(MHz)}{2000}\right)$$

$$PL_{ht} = \begin{cases} -10\log\left(\dfrac{h_t}{3}\right) & \text{for } h_t \le 3m \\ -20\log\left(\dfrac{h_t}{3}\right) & \text{for } h_t > 3m \end{cases}$$

d = distance between basestation and terminal

h_b = height of basestation

h_t = height of terminal

$a = 3.6$

$b = 0.005$

$c = 20$

Note that the MS height correction factor is Okumura's correction factor.

2.1.2.3 Type-D: RS ↔ RS, LOS (ART to ART)

For this scenario we assume that both relays are deployed above the rooftops, and they are deployed such that a LOS exists between them. Note that interfering relays at greater distances will not necessarily have a LOS path, and so the model proposed in section 2.1.2.2 can be used.

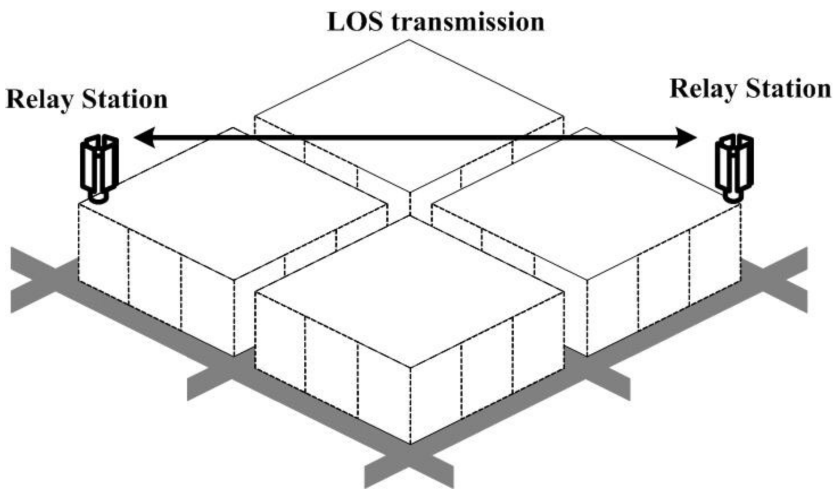

Figure 3 RS-RS LOS link (ART to ART)

2.1.2.4 Type-E: BS ↔ RS, NLOS (ART-to-BRT)

This scenario is shown illustrated in Figure 4, where in this case the BS antenna is mounted above the rooftops and the relay antenna is mounted below the rooftop (BRT).

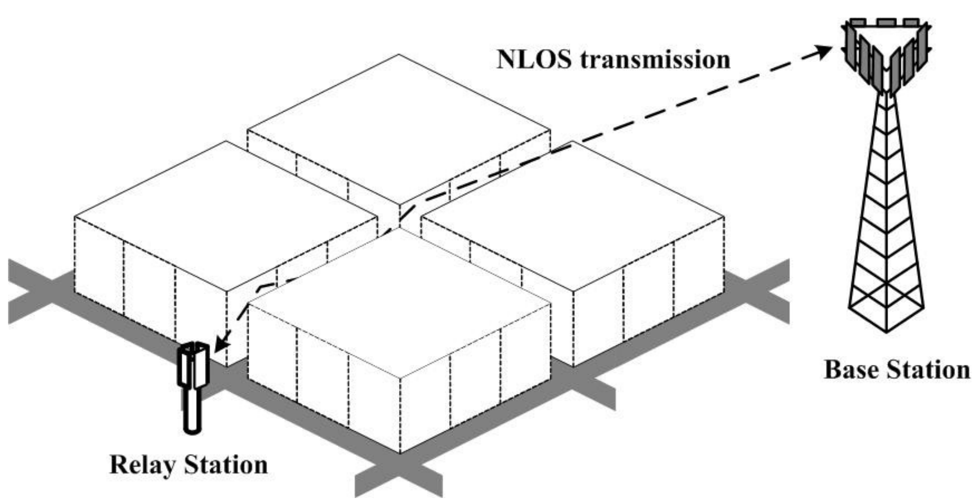

Figure 4 BS-RS NLOS (ART to BRT)

For this case the link is like a standard macro-cellular link, except that the relay antenna height is likely to be higher than the height of a typical MS. Consequently, the section 2.1.2.1 is a good model for this case, where all three categories (A, B, and C) are now applicable to cover different environments. The model includes a MS antenna height correction factor, and it includes a frequency correction factor.

The model is identical to that given in section 2.1.2.1 except for the following changes to allow for three different environment types: see [6]

Category A: Hilly terrain with moderate-to-heavy tree densities
Category B: Mostly flat terrain with moderate-to-heavy tree densities, or hilly terrain with light tree densities
Category C: Flat terrain with light tree densities

$$PL_{ht} \quad \begin{array}{ll} 10.8 \log \dfrac{h_t}{2} & \text{Categories A \& B} \\ 10 \log \dfrac{h_t}{3} & \text{Category C}, h_t \quad 3 \\ 20 \log \dfrac{h_t}{3} & \text{Category C}, h_t \quad 3 \end{array}$$

2.1.2.5 Type-E: RS ↔ RS, NLOS (ART-to-BRT)

This scenario is similar to the BS-MS link, where it is assumed that one relay is mounted above the rooftop and one relay is mounted below the rooftop. Therefore, the model proposed in section 2.1.2.2 can be used.

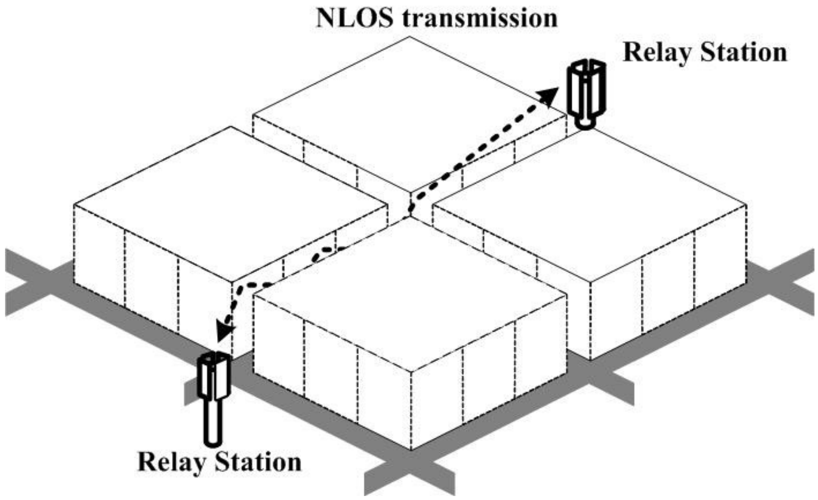

Figure 5 RS-RS NLOS (ART to BRT)

2.1.2.6 Type-F: RS ↔ MS, LOS (BRT-to-BRT)

For this scenario we assume that both the relay antenna and the MS antenna are located below the rooftop, and that they are located on the same street.

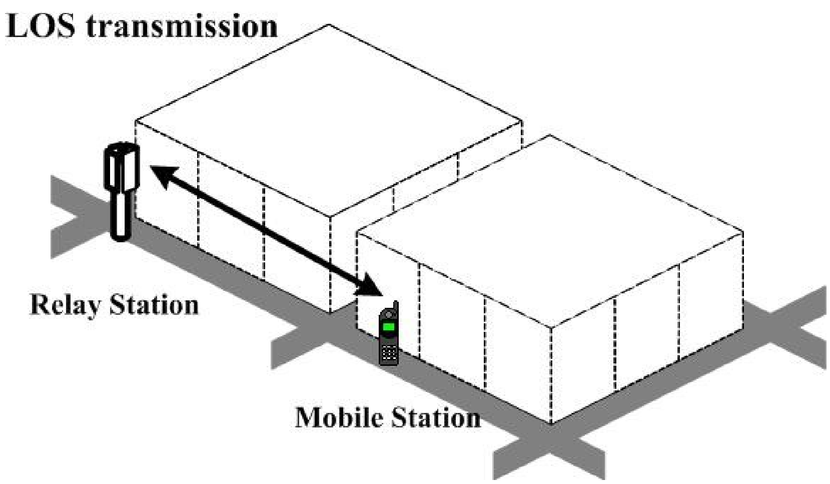

Figure 6– RS-MS LOS Scenario

For this case an advanced LOS model is a two-slope model, where the breakpoint is dependant on the relay and MS antenna heights. However, the effect of traffic is taken into account by defining an effective road height, which reduces the relay and MS heights. In addition, a visibility factor is included which reduces the path loss further as distance increases, and this factor accounts for the fact that LOS decreases with distance along a street. The model is given below:-

$$PL\ dB \quad 20\log \frac{e^{sr}4\ \ rD\ r}{}$$

where,

r distance between Tx and Rx

e^{sr} Visibility factor s 0.002

Wavelength

$$D\ r \quad \begin{matrix} 1 & r & r_{bp} \\ \dfrac{r}{r_{bp}} & r & r_{bp} \end{matrix}$$

$$r_{bp} \quad \frac{4\ h_t \quad h_0\ \ h_r \quad h_0}{}$$

h_t Height of transmitter above ground

h_r Height of receiver above ground

h_0 Effective road height $1.0m$

Note, for the distance between RS-RS or RS-MS less than 10m case, the free-space model is used.

2.1.2.7 Type-F: RS ↔ MS, NLOS (BRT-to-BRT)

For this scenario the RS and MS antenna heights are below rooftop and they are located on different streets.

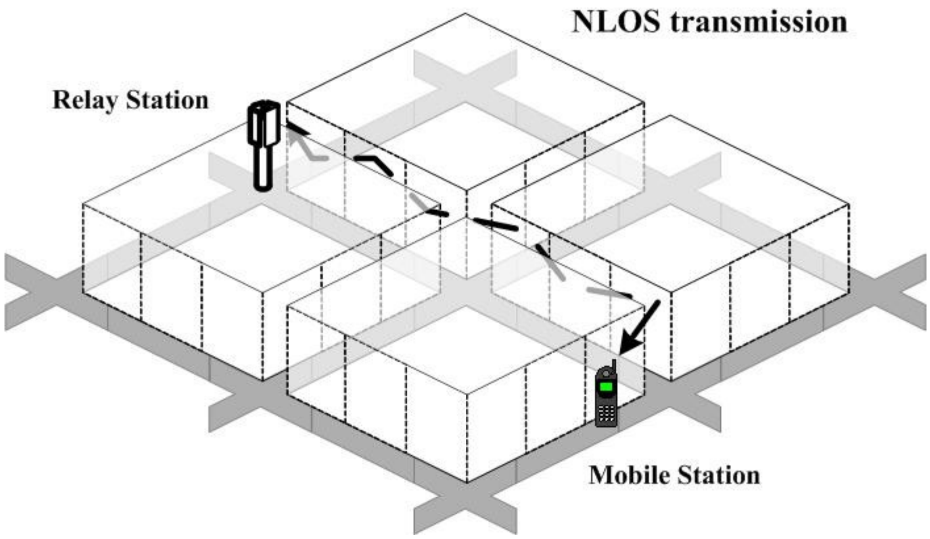

Figure 7 RS-MS NLOS scenario

For this case, the model takes minimum of an over-the-rooftop component and a round-the streets component. The round-the-streets component is based on a model by Berg, although this has been modified to be compatible with the advanced LOS model, such that the visibility is included, and the effective road height to give the correct breakpoint in the first street section. The full model is shown below:

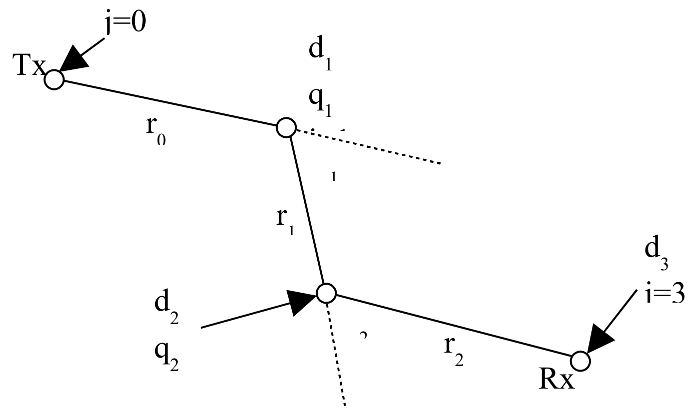

Figure 2-8 – Geometry of street sections used for Berg model

$$PL_{Berg}(dB) = 20\log\left(\frac{4\pi d_n D\left(\sum_{j=1}^{n} r_{j-1}\right)\prod_{j=1}^{n} e^{s r_{j-1}}}{\lambda}\right)$$

$R = \sum_{j=1}^{n} r_{j-1}$ Distance along streets between Tx and Rx

r_j Length of the street between nodes j and $j+1$ (there are $n+1$ nodes in total)

$$r_{bp} = \begin{cases} r_0 \text{ if } r_0 > \dfrac{4(h_t - h_0)(h_r - h_0)}{\lambda} \\ \dfrac{4(h_t - h_0)(h_r - h_0)}{\lambda} \text{ if } r_0 \le \dfrac{4(h_t - h_0)(h_r - h_0)}{\lambda} \end{cases}$$

$$D(R) = \begin{cases} 1 \text{ if } R \le r_{bp} \\ \dfrac{R}{r_{bp}} \text{ if } R > r_{bp} \end{cases}$$

The distance d_n is the illusory distance and is defined by the recursive expression,

$k_j = k_{j-1} + d_{j-1} q_{j-1}$

$d_j = k_j r_{j-1} + d_{j-1}$

with $k_0 = 1$ and $d_0 = 0$

$$q_j(\theta_j) = \left(\theta_j \frac{q_{90}}{90}\right)^{\nu}$$

θ_j Angle between streets at junction j

$q_{90} = 0.5$, and $\nu = 1.5$

$PL_{over\_the\_rooftop}(dB) = 24 + 45\log(r_{Eu})$

r_{Eu} Euclidean distance between Tx and Rx

$PL(dB) = \min\left(PL_{Berg}(dB), PL_{over\_the\_rooftop}(dB)\right)$

Note that the one-street turn corner modeling is recommended for the most of case.

For Type-F NLOS scenario the alternative path-loss model can be:

$$PL = 65 + 0.096 \cdot d_1 + (28\text{-}0.024 \cdot d_1) \cdot \log_{10}(d_2)\ dB$$

where d_1 is the distance along the main street in meter, which is valid from 10m to 550m. d_2 is the distance for perpendicular street, which is valid from w/2m to 450m. w is the street width, and the carrier frequency is 5 GHz

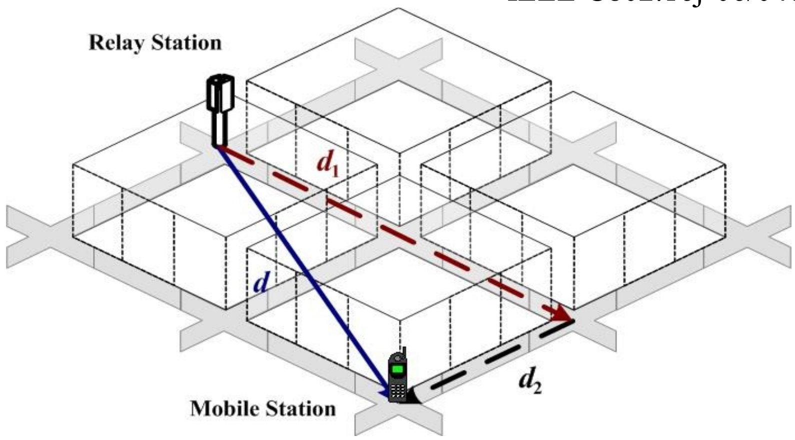

Figure 9 The alternative model for RS-MS NLOS scenario

2.1.2.8 Type-G Indoor Office Environment path-loss Model

[Editor: The indoor model is FFS, the default model is shown in this section]

The path-loss model for indoor environment is

$$PL = 37 + 30 \cdot \log_{10}(d) + 18.3 \cdot n^{((n+2)/(n+1)-0.46)}\ dB \qquad (4)$$

where d is the distance in meters and n is the number of floors in the path.

2.2 Shadowing modeling

The level of shadow fading (in dB) is usually simulated by dropping a normal distributed random variable, this refers to typical log-normal shadow fading model. However, the correlation of the propagation environment for different observation time or different radio links can not be presented if the simulator drops these variables independently. The standard deviation of the shadowing is introduced in Section 2.2.1 and two types of correlation models for shadow fading are introduced in this section 2.2.2

2.2.1 Standard deviation of the shadowing

The typical values of the standard deviation for lognormal shadowing is listed in Table 3,

Table 3 Standard Deviation Values

	Type-A	Type-B	Type-C	Type-D		Type-E		Type-F		Type-G
				LOS	NLOS	LOS	NLOS	LOS	NLOS	
Std (dB)	10.6	9.6	8.2	1.5	[4.5]	[FFS]	[FFS]	[FFS]	[FFS]	[12]

2.2.1.1 Correction factor for standard deviation of the shadowing

[Editor's note: The following informative text captures the advanced standard deviation correction factor for the lognormal shadowing]

A model is proposed where the lognormal standard deviation increases with excess path loss over free space loss. This is to prevent excessively large shadowing components when the path loss is equal to (or close to) the free space path loss, which occurs at shorter ranges typically. In particular, when the shadowing component is from the 'negative' side of the lognormal distribution, this model prevents the path loss from becoming unrealistically low.

$$(r) \quad 1 \quad e^{\frac{|P(r) \quad P_{fs}(r)|}{4}} \quad 1.5$$

Where,

is the maximum standard devation

$P(r)$ is the mean path loss (dB)

$P_{fs}(r)$ is the free space path loss (dB)

For short ranges where the path loss is equal to (or close to) the free space loss the lognormal standard deviation reduces to a lower value of 1.5dB, which accounts for variations due to interference of the direct and ground reflected components. The value of 1.5dB is based on an evaluation of the path loss using a two-ray model.

As the excess path loss increases (with distance) the standard deviation increases to an upper value of ($_u$+*1.5*). This upper value can be specified for the various multi-hop links.

2.2.2 Correlation Model for Shadow Fading

Two types of correlation model for shadowing fading are described in section 2.2.2.1 and 2.2.2.2.

2.2.2.1 Auto-correlation Model for Shadow Fading

The auto-correlation of shadow fading should be used for IEEE802.16j relay system. The_auto-correlation of shadow fading represents the correlation among the shadowing effects among the same radio link in different locations, which is illustrated in Figure 9.

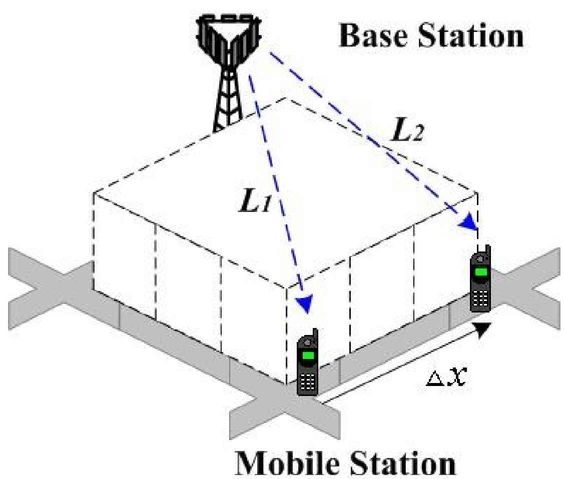

Figure 10 Auto-correlation of shadow fading

A popular model is:

$$\rho(\Delta x) = e^{-\frac{|\Delta x|}{d_{cor}}\ln 2} \tag{13}$$

where ρ is the correlation coefficient and Δx is the distance between adjacent observation locations. d_{cor} is the de-correlation distance, which was suggested as 20m in vehicular test environment.

The way to apply this model in system level simulation is briefly described as follows:

Consider the log-normal shadow fading model with zero mean and variance σ^2 in dB. If L_1 is the log-normal component at position P_1 and L_2 is the one for P_2, which is Δx away from P_1. Then L_2 will be normally distributed in dB with mean $\rho(\Delta x)\cdot L_1$ and variance $(1-[\rho(\Delta x)]^2)\cdot\sigma^2$..

2.2.2.2 Cross-Correlated Shadowing Model

The advanced cross-correlation of shadow fading model may be employed to evaluate the IEEE802.16j relay system. The cross-correlation model represents the correlation among the shadow effect of different radio links at the same time. In general, longer common propagation path will induce higher correlation. For example, the cross-correlation among the shadow fading of the radio links in Figure 10(a) should be lower than the one in Figure 10(b).

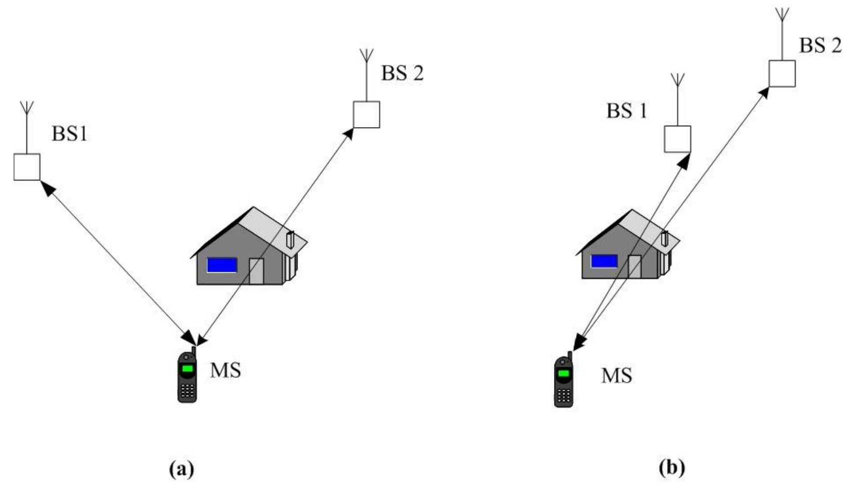

Figure 11 Cross-correlation of shadow fading

The correlation for the shadowing effect is modeled between the BSs and RSs for case of BS/RS deployed above the rooftop and for the case of RS below the rooftop. In additions the spatial de-correlation is also modeled for BS-MS and RS-MS links. For RS below the rooftop, the RS-MS link path-loss-dependent shadowing is modeled.

2.2.2.2.1 BS-MS/RS-MS link (BS/RS above rooftop)

In a network of BSs, the lognormal shadowing from two different base sites at a given MS location will have some level of correlation. In order to correctly model the benefits of relaying this correlation needs to be modelled. In addition, the shadowing from a given base site at two different MS locations will be correlated if they are within the spatial decorrelation distance of the shadowing. Therefore relays need to be beyond the spatial decorrelation distance to have a beneficial effect for a subscriber, and the spatial correlation of the shadowing also needs to be modelled.

2.2.2.2.2 Correlation between MSs

For modelling the shadowing correlation between two BSs at a given MS location a model based is given by Saunders. The geometry for the model is shown in Figure 11.

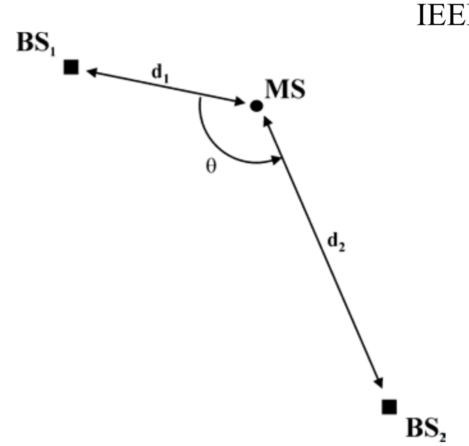

Figure 12– Geometry for correlation between two BSs

The correlation is then calculated using the following equations:

$$\frac{d_1}{d_2} \quad \text{for } 0^\circ \quad {}_T \text{ and } d_1 \quad \frac{d_c}{2}$$

$$\frac{\ {}_T\ }{} \quad \sqrt{\frac{d_1}{d_2}} \quad \text{for } {}_T \quad \text{and } d_1 \quad \frac{d_c}{2}$$

$$\sqrt{\frac{d_c}{2d_2}} \quad \text{for } d_1 \quad \frac{d_c}{2}$$

where,

Correlation

$d_c \quad \frac{1}{e}$ decorrelation distance

${}_T \quad 2\sin^{\ 1} \frac{d_c}{2d_1}$

constant depending on size and height of terrain and clutter, and height of the basestations relative to them. A value of 0.3 is used in [1], based on comparison with measured data.

For a given network of BSs a correlation matrix, $\mathbf{R}_{xx}$, can be calculated using the above model. If a vector of independent lognormal samples, $\mathbf{x}$, are generated at a given MS location, representing the shadowing from each BS, then these samples can be correlated using $\mathbf{R}_{xx}$.

$$\mathbf{y} \quad \mathbf{Tx}$$

where,

$\mathbf{x}$ Vector of independent lognormal samples

$\mathbf{y}$ Vector of correlated lognormal samples

$\mathbf{T}$ Transformation matrix

To calculate T : -

$$\mathbf{R_{xx}} \quad E\ \mathbf{yy^H} \quad E\ \mathbf{Tx}\ \mathbf{Tx}^{\mathbf{H}}$$
$$\mathbf{T}E\ \mathbf{xx^H}\ \mathbf{T^H}$$
$$\mathbf{TIT^H}$$
$$\mathbf{TT^H}$$

where $\mathbf{I}$ is the identity matrix

$$\mathbf{R_{xx}} \quad \mathbf{UDU^H}$$
$$\mathbf{D} \quad \mathbf{D}^{1/2}\mathbf{D}^{1/2}$$
$$\mathbf{R_{xx}} \quad \mathbf{UD}^{1/2} \quad \mathbf{D}^{1/2}\mathbf{U^H}$$
$$\mathbf{UD}^{1/2} \quad \mathbf{UD}^{1/2}\ ^{\mathbf{H}}$$
$$\mathbf{TT^H}$$

where : -

$\mathbf{A^H}$ complex conjugate transpose of matrix $\mathbf{A}$

2.2.2.2.3 Spatial correlation of shadowing

In order to model spatial correlation of the lognormal shadowing along a route a simple sum of sinusoids approach can be used:

$$L\ dB \quad \overset{N}{\underset{n\ 1}{}} \ a \cos k_{n\_1}x \quad {}_n \ \cos k_{n\_2}y \quad {}_n$$

$$a \quad \sqrt{\frac{4\quad ^2}{N}}$$

${}_n$ and ${}_n$ are random phase terms uniformly distributed between 0 - 2

k_{n_1}, k_{n_2} wavenumbers of the n^{th} sinusoids

The maximum values of the wave-numbers determine the de-correlation distance of the shadowing. For the urban environment, if the wave-numbers are randomly distributed between [0, 2 /75] then a 0.5 de-correlation distance of 20m results, and the 1/e de-correlation distance is 23m (value of d_c required for calculating correlation between two BS). A suggested number of sinusoids is 100.

2.2.2.2.4 RS-MS link (RS below rooftop)

The lognormal shadowing from two different below rooftop RSs at a given MS location will have some level of correlation. In order to correctly model the benefits of relaying this correlation needs to be modelled. In addition, the shadowing from a given RS site at two different MS locations will be correlated if they are within the spatial de-correlation distance of the shadowing.

2.2.2.2.4.1 Correlation between RSss

For the below rooftop case, the correlation between RSs is FFS.

2.2.2.2.4.2 Spatial correlation

For the below rooftop case, the same model can be used as for the BS-MS link. The de-correlation distance for this link is FFS.

2.2.3 Tap-Delayed-Line channel modeling

[Editor's note: IEEE802.16d SUI channel model for fixed/nomadic RS as baseline. Simplified channel model for MIMO simulation is FFS]

2.2.3.1 Multipath fading model parameters

A tap delay line is used to emulate the multipath fading channel. The channel parameters are derived from actual channel measurements. Depending on the K-factor, each tap coefficient is generated from either a Ricean or Rayleigh random variables. 802.16 (derived from SUI), ITU and WINNER multipath fading model parameters are summarized in . Details regarding the channel models can be found in [14].

Table 4 802.16 - SUI channel models

Terrain Type A: Hilly terrain with moderate-to-heavy tree densities: SUI 1				
	Tap1	Tap2	Tap3	Unit
Delay	0	0.4	0.9	s
Power	0	-15	-20	dB
K factor	4	0	0	
Doppler	0.4	0.3	0.5	Hz
Terrain Type A: Hilly terrain with moderate-to-heavy tree densities: SUI 2				
	Tap1	Tap2	Tap3	Unit
Delay	0	0.4	1.1	s
Power	0	-12	-15	dB
K factor	2	0	0	
Doppler	0.2	0.15	0.25	Hz
Terrain Type B: Intermediate path-loss condition: SUI 3				
	Tap1	Tap2	Tap3	Unit
Delay	0	0.4	0.9	s
Power	0	-5	-10	dB
K factor	1	0	0	
Doppler	0.4	0.3	0.5	Hz
Terrain Type B: Intermediate path-loss condition: SUI 4				
	Tap1	Tap2	Tap3	Unit
Delay	0	1.5	4.0	s
Power	0	-4	-8	dB
K factor	0	0	0	
Doppler	0.2	0.15	0.25	Hz
Terrain Type C: Flat terrain with light tree densities: SUI 5				
	Tap1	Tap2	Tap3	Unit

Delay	0	4	10	s
Power	0	-5	-10	dB
K factor	0	0	0	
Doppler	2.0	1.5	2.5	Hz
Terrain Type C: Flat terrain with light tree densities: SUI 6				
	Tap1	Tap2	Tap3	Unit
Delay	0	14	20	s
Power	0	-10	-14	dB
K factor	0	0	0	
Doppler	0.4	0.3	0.5	Hz

2.2.4 Antenna pattern

2.2.4.1 BS antenna

For omni-directional antenna, the antenna gain should be $0\ dBi$ for each direction. For 3-sector or 6-sector antenna, the antenna pattern specified as:

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

where

$-180^\circ < \theta \le 180^\circ$;

θ is the angle between the direction of interest and the steering direction of the antenna;

$\theta_{3db} = 70^o$ is the 3 dB beam width for 3 sector antenna, 35^o for 6 sector antenna.

A_m = 20dB maximum attenuation (front-to-back ratio) for 3 sector antenna, 23dB for 6 sector antenna.

2.2.4.2 RS antenna

[Editor's note: FFS]

2.2.4.3 MS antenna

Omni antenna pattern is assume for MS

3 Traffic models

[Editor's note: Full buffer is baseline model, need to choose on real-time traffic model, adopt [4] and use references]

This section describes the traffic models in detail. Section 3.1 addresses DL and Section 3.2 the UL A major objective of multihop simulations is to provide the operator a view of how many users can be supported for a given service under a specified multihop configuration at a given coverage level. The traffic generated by a service should be accurately modeled in order to find out the performance. Traffic modeling can be simplified, as explained below, by not modeling the user arrival process and assuming full queue traffic. These are explained below.

Modeling of user arrival process: All the users are not active and they might not register for the same service. In order to avoid different user registration and demand models, the objective of the proposed simulation is made limited 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.

At the relay station, however, the traffic availability depends on the forwarded traffic from either base station, user or by another relay even in the full queue model.

The traffic models provided in the next sections describe only the non-full queue case.

3.1 Traffic Modeling for IEEE802.16j Services

The traffic model is listed in Table 5.

Table 5: Services to be considered

#	Application	Traffic Category	Priority
1	Full buffer		x
2	FTP	Best-effort	
3	Web Browsing	Interactive	
4	VoIP	Real-time	TBD
5	Video Streaming	Streaming	TBD
6	Live Video	Interactive Real-time	

.

3.2 Traffic Modeling for UL Services

This section discusses the traffic modeling related to reverse link data traffic.

3.2.1 FTP Upload / Email

The file upload and email attachment upload are modeled as Table 6.

Table 6FTP Characteristics

Arrival of new users	Poisson with parameter
Upload file size	Truncated lognormal; lognormal pdf: $f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[-\frac{(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 2.0899, \quad \mu = 0.9385$ Min = 0.5 kbytes Max = 500 kbytes If the value generated according to the lognormal pdf is larger than Max or smaller than Min, then discard it and regenerate a new value. The resulting truncated lognormal distribution has a mean = 19.5 kbytes and standard deviation = 46.7

	kbytes

The FTP traffic is simulated as follows:

- At the beginning of the simulation there are 5 FTP users waiting to transmit.
- Before transmitting, call setup is performed for each user
- Afterwards, FTP upload users arrive according to the Poisson arrival process, as defined in Table 6.
- For each new FTP upload user coming into the system, call setup is performed
- Each FTP upload user stays in the system until it finishes the transmission of its file
- After an FTP upload user finishes the transmission of its file, it immediately leaves the system.

Since the arriving FTP users are dropped uniformly over 19 cells, it is possible the number of users can exceed the sector capacity. In that case, the new arrival should be blocked. The sector capacity is 43 in total. The blocking rate should be recorded.

3.2.2 HTTP Model

The following figure is an example of events occurring during a HTTP session.

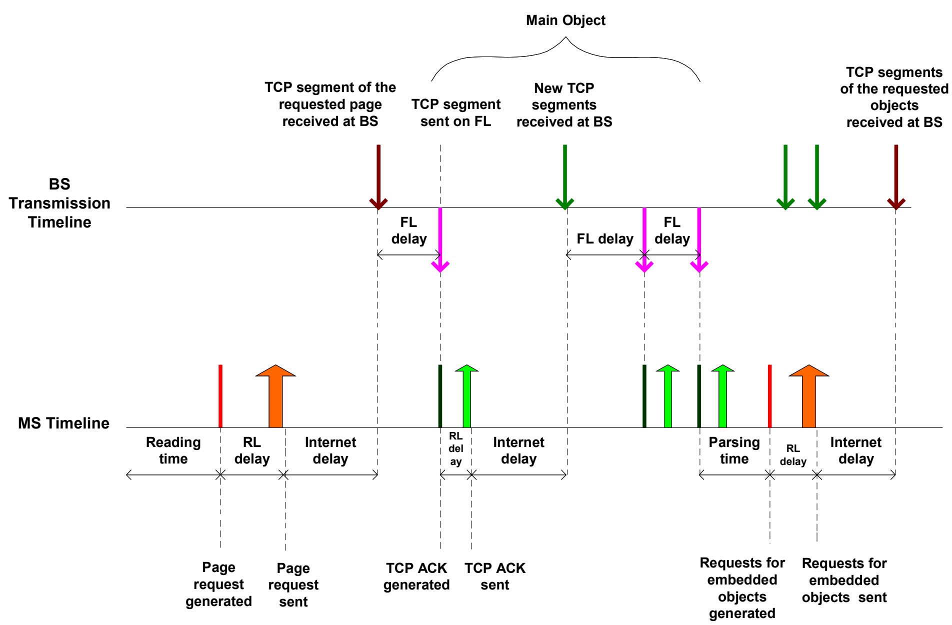

Figure 13 Example of events occurring during web browsing.

HTTP Traffic Model Parameters

Reading time (D_{pc}): modeled as in Table 6

Internet delay (D_I): modeled as an exponentially distributed random variable with a mean of 50ms

Parsing time (T_p): modeled as in Table 6

UL delay: specific for the implemented system. Includes UL packet transmission delay and scheduling delay (if scheduled)

FL delay (D_{FL}): defined as the time a TCP segment is first in the queue for transmission until it finishes transmission on DL. The delay includes transmission delay and DL scheduling delay. If there are multiple packets, each packet has its own additional contribution to the overall D_{FL}.

Number of TCP segments in the main object (N_M). $N_M = S_M /(MTU-40)$. The main object size, S_M, is generated according to Table 6

Number of TCP segments in embedded object (N_E). $N_E = S_E /(MTU-40)$. The embedded object size, S_E, is generated according to Table 6

Number of embedded objects (N_d). Modeled according to Table 6

HTTP1.1 mode

The opening and the closing of the TCP connections is not modeled

HTTP request size = 350 bytes

Requests for embedded objects are pipelined – all requests are buffered together

MTU size = 1500 bytes

ACK size = 12 bytes

Every received TCP segment is acknowledged.

Table 6 HTTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 9055 bytes Std. dev. = 13265 bytes Minimum = 100 bytes Maximum = 100 Kbytes	If x > max or x < min, then discard and re-generate a new value for x. $f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 1.37, \quad \mu = 8.35$
Embedded object size (S_E)	Truncated Lognormal	Mean = 5958 bytes Std. dev. = 11376 bytes Minimum = 50 bytes Maximum = 100 Kbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 1.69, \quad \mu = 7.53$ If x > max or x < min, then discard and re-generate a new value for x.
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 4.229 Max. = 53	$f_x = \frac{\alpha k^\alpha}{x^{\alpha+1}}, k \leq x < m$ $\alpha = 1.1, k = 2, m = 55$ Note: Subtract k from the generated random value to obtain N_d If x > max, then discard and re-generate a new value for x
Reading time (D_{pc})	Exponential	Mean = 30 sec	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.033$
Initial reading time (D_{ipc})	Uniform	Range [0, 10] s	$f_x = \frac{1}{b-a}, \quad a \leq x \leq b$ $a = 0, b = 10$

Parsing time (T_p)	Exponential	Mean = 0.13 sec	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 7.69$

Packet Arrival Model for HTTP

At the beginning of the simulation, call setup is performed for all HTTP users. After that, the simulation flow is described as follows:

Generate an initial reading time D_{ipc}.[1] Wait D_{ipc} seconds.

Initiate the TCP window size W=1

Generate a request for the main page

Wait for the requests to go through the UL and reach the bases station (UL delay):

In case these are requests for embedded objects, wait until all requests reach the base station.

Generate an Internet delay D_I. Wait D_I seconds.

Generate random delays, which define the time instances when each of the TCP segment transmission is completed the DL. The number of these instances is:

For the main page:

At the very beginning of the packet call: 1

Afterwards: min(2n, #of outstanding TCP segments on DL), where n is the number of ACKs received in the last physical layer packet (from the step 3.2.2)

For embedded objects:

At the very beginning of the transmission of embedded objects: min(W, $\sum_{i=1}^{N_d} N_E^i$).

Afterwards: min(2n, #of outstanding TCP segments on FL), where n is the number of ACKs received in the last physical layer packet

Every time instance of the completed TCP segment transmission on FL generates an ACK on UL

Continue UL simulation – when ACK is generated, reduce the number of outstanding TCP packets by 1

Examine if the transmission of the very last TCP segment of the HTTP object is completed:

If no:

Proceed with simulation until next ACK or a group of n ACKs within a single physical layer packet is transmitted

Increase W:=W+n

Go to step 3.2.2

If yes, for main page:

Generate T_p (parsing time)

Generate requests for embedded objects

Continue UL simulation - transmit outstanding ACK(s) for the main page and accordingly increment W:=W+n for each

1

group of n ACKs transmitted, until requests for embedded objects are generated

Go to step 3.2.2

If yes, for embedded objects:

Generate D_{pc} (reading time)

Continue UL simulation - transmit outstanding ACK(s) for the embedded objects

Go to step 3.2.2 when reading time expires or until all ACKs are transmitted, whichever is longer.

3.2.2.1 Modeling of DL Traffic when carrying out a Up Link only simulation

HTTP requests and TCP ACKs come under this category. It is not known, what percentage of traffic would be acks and HTTP requests in a broadband systems. It is clear that the size of the access page increases with time, while ACK messages and HTTP requests remains the same. Therefore, we can expect that in the future systems, the impact of AC K and HTTP requests will be negligible compared to size of the data contents.

3.3 Non Real-Time Traffic

3.3.1 Full buffer model

The transmitter buffer is full.

3.3.2 FTP

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

S : the size of a file to be transferred

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

The underlying transport protocol for FTP is TCP. The packet trace of an FTP session is shown in Figure 13.

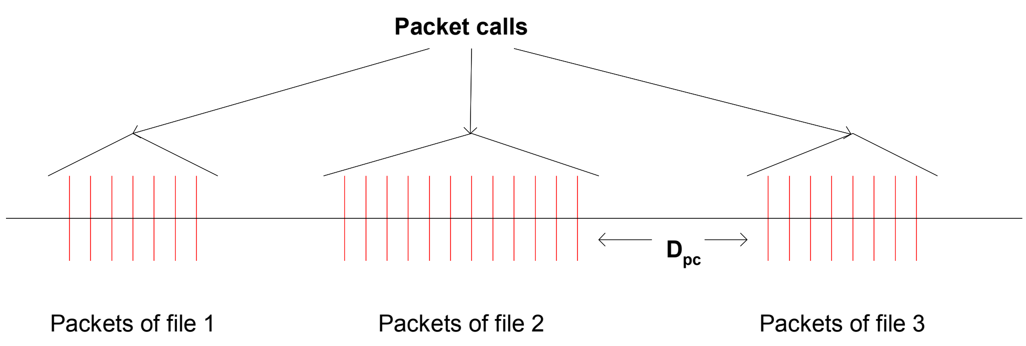

Figure 14 Packet Trace in a Typical FTP Session

The parameters for the FTP application session are described in Table 7

Table 7 FTP Traffic Model Parameters

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

3.3.3 Web Browsing

Web browsing is the dominant application for broadband data systems, and has been studied extensively.

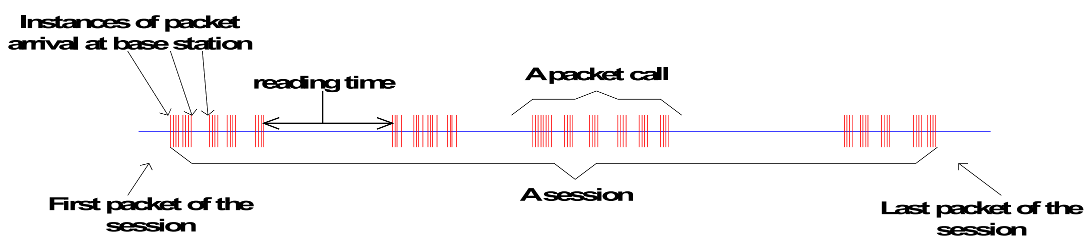

Figure 15 Packet Trace of a Typical Web Browsing Scheme

Figure 14 shows the packet trace of a typical web browsing session. The session is divided into ON/OFF periods representing web-page downloads and the intermediate reading times. In Figure 14, the web-page downloads are referred to as packet calls. These ON and OFF periods are a result of human interaction where the packet call represents a user's request for information and the reading time identifies the time required to digest the web-page.

As is well known, web-browsing traffic is self-similar. In other words, the traffic exhibits similar statistics on different timescales. Therefore, a packet call, like a packet session, is divided into ON/OFF periods as in . Unlike a packet session, the ON/OFF periods within a packet call are attributed to machine interaction rather than human interaction. In general, a web-page is constructed from many individually referenced objects. A web-browser will begin serving a user's request by fetching the initial HTML page using an HTTP GET request. After receiving the page, the web-browser will parse the HTML page for additional references to embedded image files such as the graphics on the tops and sides of the page as well as the stylized buttons. The retrieval of the initial page and each of the constituent *objects* is represented by ON period within the packet call while the parsing time and protocol overhead are represented by the OFF periods within a packet call. For simplicity, the term "page" will be used in this paper to refer to each packet call ON period. As a rule-of-thumb, a page represents an individual HTTP request explicitly initiated by the user. The initial HTML page is referred to as the "main object" and the each of the constituent objects referenced from the main object are referred to as an "embedded object".

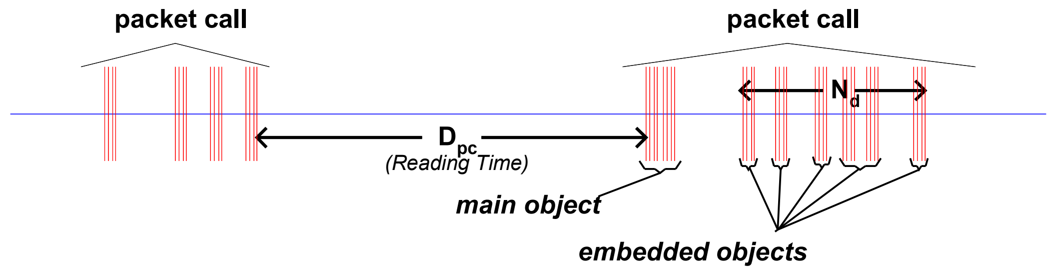

Figure 16 Contents in a Packet Call

The parameters for the web browsing traffic are as follows:

S_M: Size of the main object in a page

S_E: Size of an embedded object in a page

N_d: Number of embedded objects in a page

D_{pc}: Reading time

T_p: Parsing time for the main page

Table 8 HTTP Traffic Model Parameters

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

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

3.4 Real-Time Traffic

3.4.1 Voice over IP (VoIP)

A VoIP call shall be assumed to be between one user and one wired user. In order to get an evaluation of the air interface the wireline and core network impairments are neglected.

VoIP Traffic Source

The G.729A decoder shall be simulated with an assumed 4 byte IP header. Each packet produced by the

G.729A vocoder shall be appended with a 4 byte header that accounts for UDP/IP overhead, after header compression.

3.4.2 Video Streaming

The steady state of video streaming traffic from the network as seen by the base station. Latency of starting up the call is not considered in this steady state model.

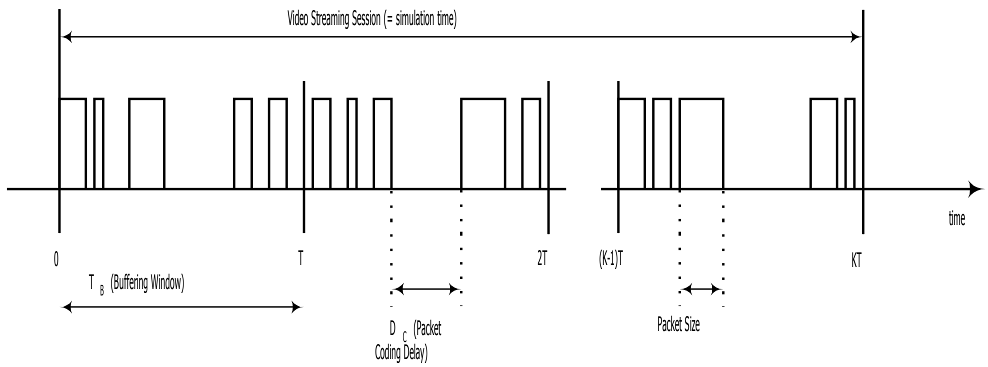

Figure 17Near Real-Time Video Traffic Model

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

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

Using a source rate of 64 kbps, the video traffic model parameters are defined Table 9.

Table 9 Near Real-Time Video Traffic Model Parameters

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

Distribution Parameters	100ms	8	K = 20bytes = 1.2	K = 2.5ms = 1.2

3.4.3 Modeling UL Traffic for the DL only Simulations

HTTP requests and TCP ACKs come under this category. It is not known, what percentage of traffic would be acks and HTTP requests in a broadband systems. It is clear that the size of the access page increases with time, while ACK messages and HTTP requests remains the same. Therefore, we can expect that in the future systems, the impact of AC K and HTTP requests will be negligible compared to size of the data contents.

4 Performance Metrics

The performance metrics are divided into two categories. They are:

Single-user performance; and

Multi-user performance.

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 forward and UL). 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.

In order to evaluate multi-user performance accurately, scheduling and higher layer traffic behaviors and protocols need to be modeled. However, simulation run times can be prohibitively large. Specially, in the case of multihop systems, each sector can have several relay stations and there are a large number of relay stations and relay to user and relay to base links need to be modeled sand simulated. Therefore, such simulations can be very CPU intensive. Therefore, we suggest that initial design validations be done using a simple but representative analysis using a full queue traffic without modeling higher layers. These are described under multi-user performance metrics.

4.1 Single-user performance Metrics

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.

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

Link budget evaluations is a well known method for initial system planning and this needs to be carried out for relay to base, relay to user and base to user links separately. 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 when the relays are deployed. Link budget analysis are provided in detail in Section 5.

Since relays can be used to extend the range covered by a cell under noise limited environment (i.e. no interference from other cells but the limitation coming from the fact that the transmit power is not enough to provide a sufficient signal strength above thermal noise) coverage range is a metric of importance in such cases.

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.

4.1.2 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).

4.1.3 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.

4.1.4 Multi-user Performance Metrics

4.1.4.1 Combined Coverage and Capacity Metric (cc)

There are three important aspects that need to be considered when the multi-user performance is evaluated for a multi-hop system.

4.1.4.1.1 Sharing the shared channel among users:

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) are to be shared with other 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..

The low rate users can be provided more resources so that they would get equal service from the cellular operator but that would impact capacity. Thus, there is a trade-off between coverage and capacity and any measure of capacity should be provided with the associated coverage. .

Since an operator should be able to provide the service to multiple users in 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 of a given service from a viability point of view.

<u>Combined Coverage and Capacity Index (*CC*)</u>: The number N of simultaneous users per cell (e.g. MMR-cell or legacy cell) that can be supported achieving a target information throughput $R\min$ with a 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 at 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.

<u>**Method 1:**</u>

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, in a simulation that number of users are dropped uniformly in the service area. Let the required coverage for a given service is x% and the required information rate for that service is Rmin. The first step in evaluating *cc* is to take out the lowest (100-x)% of users out of the evaluation. Assume the number of users in the remaining group is *k*, and the average effective data rate that can be supported by the ith user is r_i (i = 1 to N).

Then,

if the min(r_i) < Rmin, $cc = 0$ (i.e. the service cannot be provided with the required coverage).

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

this is the maximum # 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.

Method 2:

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

Coverage reliability for a particular system (cell radius, shadow fading environment, relay station placement, 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 and all relays supporting those users such that the target information throughput is $R\min$ 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 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:

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

where M is the total number of simulation runs. The <u>Combined Coverage and Capacity Index *N*</u> is the largest n for which

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

4.2 Definitions of Performance Metric

4.2.1 System data throughput

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

4.2.2 Packet call throughput:

Packet call throughput which is the total bits per packet call divided by total packet call duration.

4.2.3 Effective system spectral efficiency

Effective system spectral efficiency normalized by the downlink/uplink ratio of TDD system, for the DL case:

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

4.2.4 CDF of data throughput per user

The throughput of a user is defined as the ratio of the number of information bits that the user successfully receives during a simulation run and the simulation time.

4.2.5 The CDF of packet delay per user

4.3 Fairness Criteria

Since one of the primary objectives of the introduction of relays is to have uniform service coverage resulting in a fair service offering, a measure of fairness is very important in assessing how well the relaying 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, $\widetilde{T}_{put}[k]$ is given by

$$\widetilde{T}_{put}[k] \quad \frac{T_{put}[k]}{\operatorname*{avg}_{i} T_{put}[i]}.$$

4.3.1 Fairness Index

Since CDF does not provide a quantitative measure of fairness it is important to define a metric to measure fairness. Since fairness of a system can be increased by providing more resources to low rate users which result in a reduction of the system capacity, when performance is measured it is important to specify the associated fairness. Then, the performance of two systems can be compared under same fairness conditions. For this purpose, fairness index of a resulting throughput distribution is defined as,

$$\text{Fairness Index (FI)} = e^{-\sigma}$$

where σ is the standard deviation of the normalized per user throughput distribution.

Note that higher the FI higher is the fairness of a system and FI =1 corresponds to the case where all the users receive same throughput.

Depending on the service type and test case being simulated, different fairness requirements may be specified. Three such fairness criteria are specified in this document for this purpose. The evaluation methodology should specify what fairness criterion has to be met for a given test case.
Equal Throughput Criterion:
To have a reasonably compromise fairness as specified to meet a CDF requirement.

To meet a specified level of fairness

4.3.2 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, $C = \frac{1}{\sum_{i=1}^{n} 1/(r(i))}$

Since one of the primary objectives of relays is to provide uniform service offering across users, the total aggregate throughput under equal throughput criterion, is a good metric to compare two systems.

4.3.3 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 0.

Table 10 Criterion CDF

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

4.3.4 Fairness Criterion to meet a Specified Fairness Index

Under this fairness criterion, the fairness index of the normalized per user throughput should be higher than a target value. This target value is to be specified under each test case. i.e., the fairness requirement is,

Fairness Index of the resulting distribution > target_fairness_index.

5 References

[1] 3GPP2/TSG-C.R1002, "1xEV-DV Evaluation Methodology (V14)", June 2003.
[2] 3GPP TR 25.996 V6.1.0, "Spatial channel model for Multiple Input Multiple Output (MIMO) simulations," September 2003.
[3] IEEE C802.16j-6/023, "Multi-hop System Evaluation Methodology" May, 2006
[4] IEEE C802.16j-6/024, "Multi-hop System Evaluation Methodology: Traffic Models", May, 2006

[5] IEEE C802.16j-6/025, “Multi-hop System Evaluation Methodology–Performance Metrics” May, 2006
[6] IEEE C802.16j-6/020r1,“Channel Models & Performance Metrics for IEEE 802.16j Relay Task Group” May, 2006
[7] IEEE C802.16j-6/009, “Correlated Lognormal Shadowing Model”, May, 2006
[8] IEEE C802.16j-6/010, “Below Rooftop Path Loss Model”, May, 2006
[9] IEEE C802.16j-6/011, “Multi-hop Path Loss Model (Base-to-Relay and Base-to-Mobile)”, May, 2006
[10] IEEE C802.16j-6/012, “Multi-hop Network Simulation with Street Layout “May, 2006
[11] IEEE C802.16j-6/014, “Metrics for Multi-hop Systems” May, 2006
[12] IEEE C802.16j-6/015, “RF compatibility of RS with other MSS” May, 2006
[13] IEEE C802.16j-6/003, “Seaport Path Loss Model for Fixed Wireless Applications”, May, 2006
[14] IEEE 802.16.3c-01/29r4, “Channel Models for Fixed Wireless Applications” July 21, 2001 http://www.wirelessman.org/tg3/contrib/802163c-01_29r4.pdf

Appendices

A.1 Multi-Cell Layout

In Figure 6, 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 7 clusters may be neglected. In those cases, only 19 cells and associated relays may be modeled. These cases are identified in the sections below.
For the cases where modeling outer-cells are necessary for accuracy of the results, the 7 cluster network can be used. However, the six of the seven clusters are just virtual clusters repeating the middle cluster in its surroundings as shown in the figure. Each cell with generic hexagonal grid is separated to 3 sectors, each is formed by a panel directional antennas.

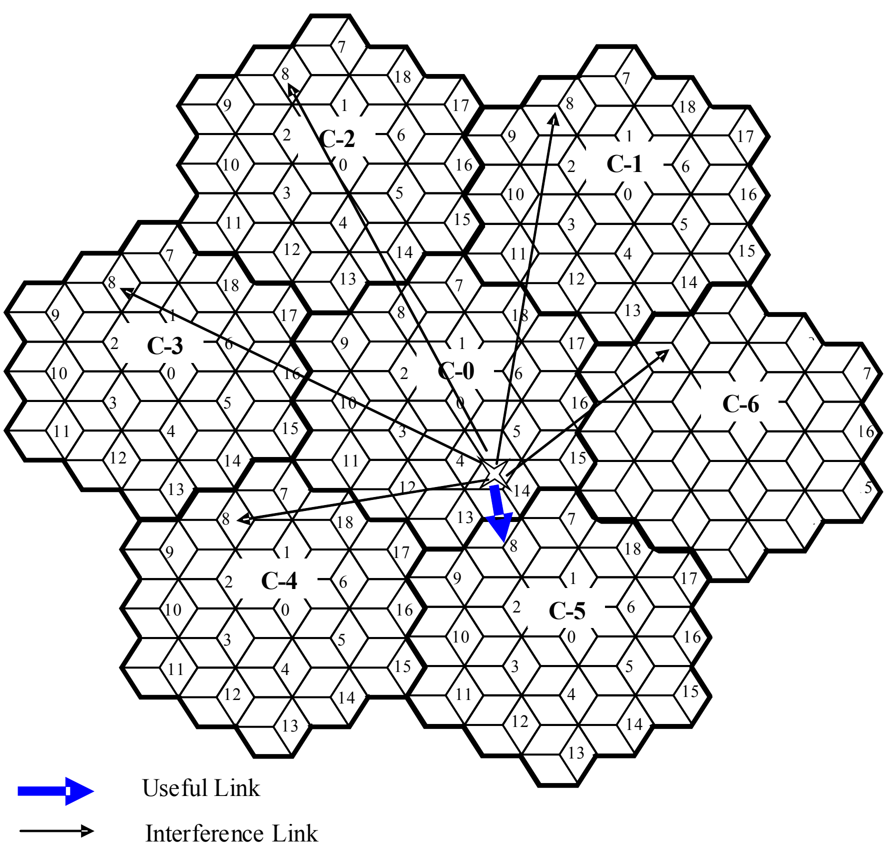

Figure 18 Multi-cell Layout and Wrap-around Example

A1.1 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 in 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.

A1.2 Determination of severing cell 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; one is to determine the shortest distance cell for each MS from all seven logical cells, and the other is to determine the severing cell for each MS based on the strongest link among 19 cells related to the path-loss and shadowing.

To determine the shortest distance cell for each MS, the distances between the target MS and all logical cells should be evaluated and select the cell with a shortest distance in 7 clusters. Figure 2 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 severing cell for each MS, we need to determine 19 links, whereby we may additionally determine the corresponding path-loss, shadowing and transmit/receive antenna gain in consideration of antenna pattern. The serving cell for each MS should offer a strongest link with a strongest received long-term power. It should be noted that the shadowing experienced on the link between MS and cells located in different clusters is the same.

B Link Budget

Link Budget should include all the key items defined below

$Si = Pout + Gt - A_{Backoff} - Pl - Ls + Gr - Ml + Fm$

Pout Output power of transmitter in dBm
Gt Transmitting antenna gain in dBi
$A_{Backoff}$ Amplifier Backoff
Pl Path loss in dB
Ls Shadowing loss in dB
Gr Receiving antenna gain in dBi
Ml Miscellaneous losses (include cable losses, nonlinearity, body loss, polarization mismatch, other losses etc.)
Fm Fade Margin in dB

Si Received power level at receiver input in dBm