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Re:	Response to a call for comments and contributions for the Relay TG, see C80216j-06/006.pdf
Abstract	This document captures scope of the Multi-hop System Evaluation Methodology including the Channel Model, Traffic Model and Performance Metrics. This document is a Task Group document to which the harmonized contribution document C802.16j-06/042r2 has been converted as one of the baseline documents defined during the 2nd Relay TG in session #44.
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 IEEE802.16 Relay TG.

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Multi-hop Relay System Evaluation Methodology

1 Introduction

The scope of this Multi-hop Relay 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 Relay TG. It is not the intention of this document to mandate the use of this evaluation methodology in the comparisons of proposals. Proposors should provide sufficient details of simulation parameters such that it is possible for other proposors to replicate their results.

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.

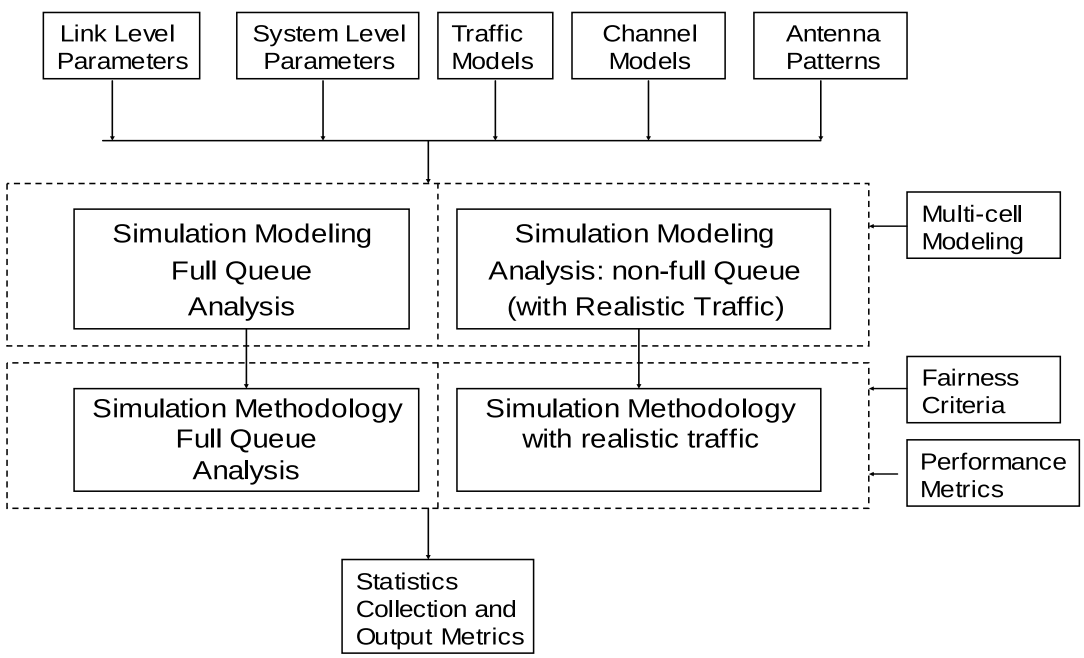


Figure 1. Simulation Components and Overall Methodology

2 Channel Models

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	Description		Reference
Type A	Macro-cell suburban, ART to BRT for hilly terrain with moderate-to-heavy tree densities.	LOS/NLOS	Section 2.1.2.1
Type B	Macro-cell suburban, ART to BRT for intermediate path-loss condition.	LOS/NLOS	
Type C	Macro-cell suburban, ART to BRT for flat terrain with light tree densities.	LOS/NLOS	
Type D	Macro-cell suburban, ART to ART	LOS	Section 2.1.2.2
Type E	Macro-cell, urban, ART to BRT	NLOS	Section 2.1.2.3
Type F	Urban or suburban, BRT to BRT.	LOS	Section 2.1.2.4.1
		NLOS	Section 2.1.2.4.2
Type G	Indoor Office	LOS/NLOS	Section 2.1.2.5
Type H	Macro-cell, urban, ART to ART.	LOS	Section 2.1.2.6
Type J	Outdoor to indoor	NLOS	Section 2.1.2.7

Note: LOS (Line Of Sight), NLOS (Non Line Of Sight), ART (Above Roof Top), BRT (Below Roof Top)

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

Table 2. Relationship between path-loss and usage models.

Links	Path-loss Type	Applicable Usage Model	Note
BS-RS	Type A/B/C	I, III, IV	Suburban, RS antenna is BRT
	Type D	I, III	BS antenna is ART and RS antenna is ART
	Type E	I, III, IV	Urban, BS antenna is ART and RS antenna is BRT
	Type H	I, III	Urban, BS antenna is ART and RS antenna is ART
	Type J	II	BS is outdoor and RS is indoor/tunnel
BS-MS	Type A/B/C	I, III, IV	Suburban, BS antenna is ART
	Type E	I, III, IV	BS antenna is ART
	Type J	II	BS is outdoor and MS is indoor/tunnel
RS-RS	Type A/B/C	I, III, IV	Suburban, one RS antenna is ART

	Type D	I, III	Both RS antennas are BRT
	Type E	I, III, IV	Urban, One RS antenna is ART and another one is BRT
	Type F	I, III, IV	Both RS antennas are BRT
	Type G	II	Both RS antennas are inside building
	Type J	II	One RS is outside and the other inside a building/tunnel
RS-MS	Type A/B/C	I, III	Suburban, RS antenna is ART
	Type E	I, III	RS antenna is ART
	Type F	I, III, IV	RS antenna is BRT
	Type G	II	Both RS and MS antennas are inside building
	Type J	II	RS is outside and MS is inside or RS is inside and MS is outside

The usage models referenced from IEEE 802.16j-06/015 are:

I. Fixed Infrastructure Usage Model

II. In-Building Coverage Usage Model

III. Temporary Coverage Usage Model

IV. Coverage on Mobile Vehicle Usage Model

2.1.2 Detailed Path-loss Models

2.1.2.1 Type-A/B/C (Suburban, ART-to-BRT)

The modified IEEE 802.16 path-loss model is recommended for these links where this is given in [9].

The basic IEEE 802.16 model is provided below for reference.

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

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 BS/RS antenna height, which is between 10m and 80m. 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 given in Table 3.

Table 3. 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
where f is the carrier frequency in MHz.

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

Modified IEEE 802.16 model

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

where,

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

$$d_0 = 100m$$

$$d_0^{'} = d_0 10^{-\left(\frac{\Delta PL_f + \Delta PL_{ht}}{10\gamma}\right)}$$

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

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

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

d = distance between transmitter and receiver

h_b = height of BS/RS ART antenna

h_t = height of RS/MS BRT antenna

The model parameters for Types A, B and C are the same as those specified for the basic model, as provided in Table 3.

2.1.2.2 Type-D: LOS (ART-to-ART)

This scenario is shown in Figure 2 and Figure 3, where both node antennas are mounted above the rooftops (ART) and they have a LOS between them.

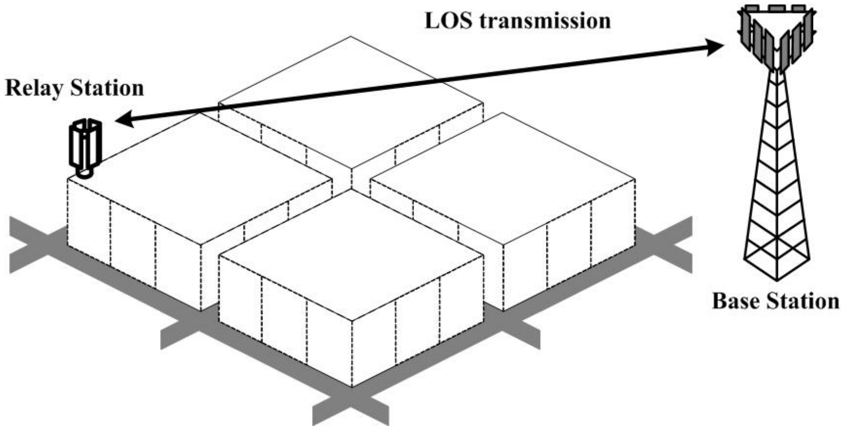


Figure 2. BS-RS link with LOS

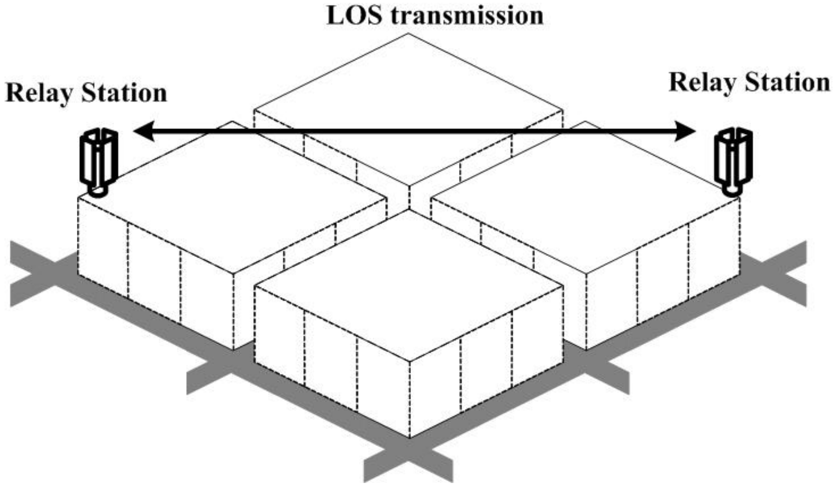


Figure 3. RS-RS LOS link (ART to ART)

For this link the modified IEEE 802.16d channel model is recommended as presented in this section. There are three categories for this model, as shown in the previous section, 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.

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

where,

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

$$d_0 = 100m$$

$$d_0^{'} = d_0 10^{-\left(\frac{\Delta PL_f + \Delta PL_{ht}}{10\gamma}\right)}$$

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

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

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

d = distance between transmitter and receiver

h_b = height of BS/RS antenna

h_t = height of RS antenna

$a = 3.6$

$b = 0.005$

$c = 20$

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

2.1.2.3 Type-E: (Urban ART-to-BRT, NLOS)

For the urban NLOS case which is shown as the examples in Figure 4 and 5, the COST 231 Walfisch-Ikegami model is recommended and given in [14].

(Editor's Note: The text of COST 231 Walfisch-Ikegami model is in the Appendix A of [14])

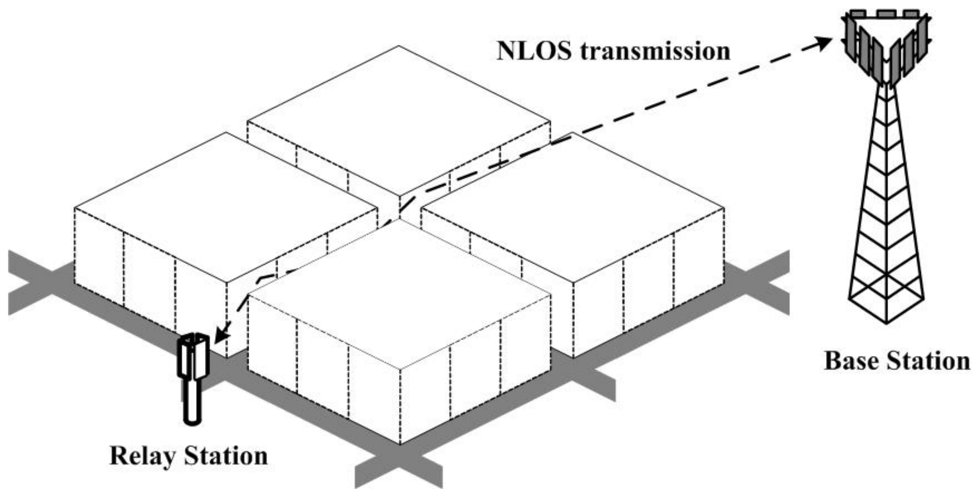


Figure 4, BS-RS NLOS (ART to BRT)

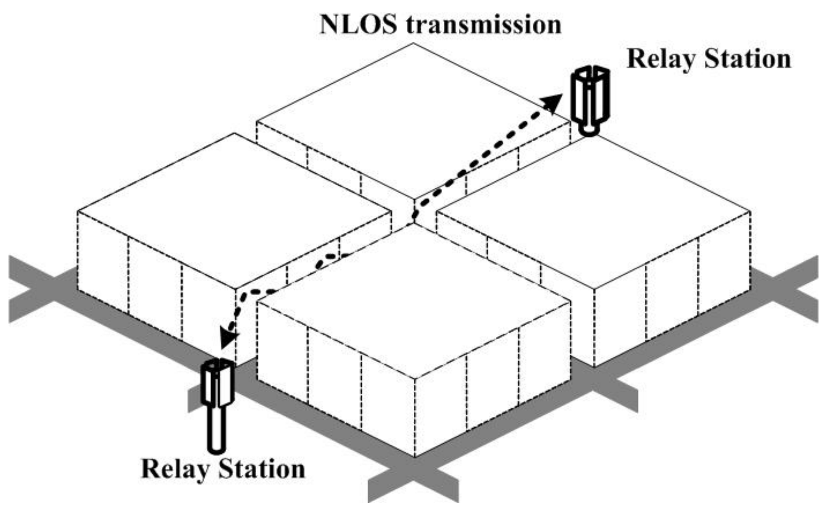


Figure 5. RS-RS NLOS (ART to BRT)

Parameter values to be used for this model are provided below. The use of these values is not mandatory:

Building spacing, b = 60m *(this is the spacing between building centers)*
Street width, w = 12m *(this is the spacing between building faces)*
Street orientation = 90degrees
Average rooftop height, hroof = 25m

An alternative is using WINNER model, which is given as:

$PL(d)=38.4+35\log_{10}(d)$ dB *for* 50m < d < 5km

where d is the distance in meter and the carrier frequency is 5GHz.

2.1.2.4 Type-F: LOS (BRT-to-BRT)

Both LOS and NLOS models are provided separately below for this case.

2.1.2.4.1 LOS version

For this scenario we assume that both node antennas are located below the rooftop, and that they are located on the same street.

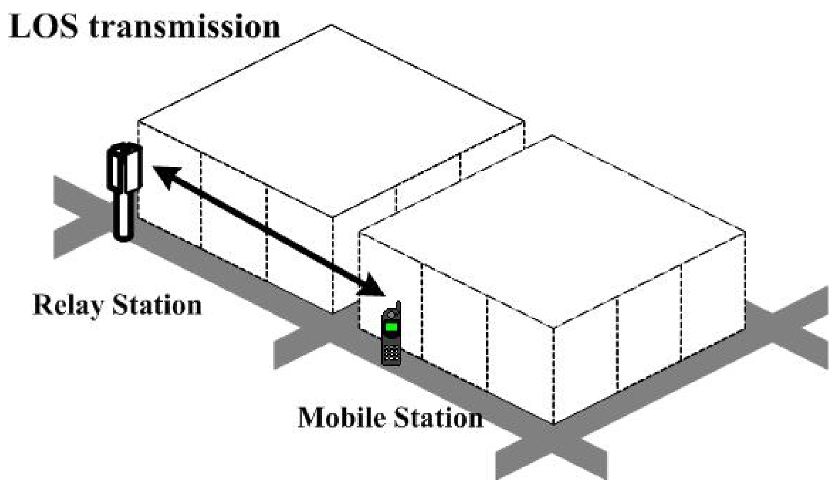


Figure 6. RS-MS LOS Scenario

For this case an advanced LOS model is recommended. This 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:-

$$PL(dB) = 20\log\left(\frac{e^{sr}\,4\pi r D(r)}{\lambda}\right)$$

where,

$r =$ distance between Tx and Rx

$e^{sr} =$ Visibility factor $(s = 0.002)$

$\lambda =$ Wavelength

$$D(r) = \begin{cases} 1 & r \le r_{bp} \\ \dfrac{r}{r_{bp}} & r > r_{bp} \end{cases}$$

$$r_{bp} = \frac{4(h_t - h_0)(h_r - h_0)}{\lambda}$$

$h_t =$ 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.

For this scenario, the alternative WINNER path-loss model can be used:

$PL(d)$=22.7 $\log_{10}(d)$+41.0 dB for 10m < d <650m

where d is the distance in meter and the carrier frequency is 5 GHz

2.1.2.4.2 Type-F: NLOS (BRT-to-BRT)

For this scenario both nodes antenna heights are below rooftop and they are located on different streets.

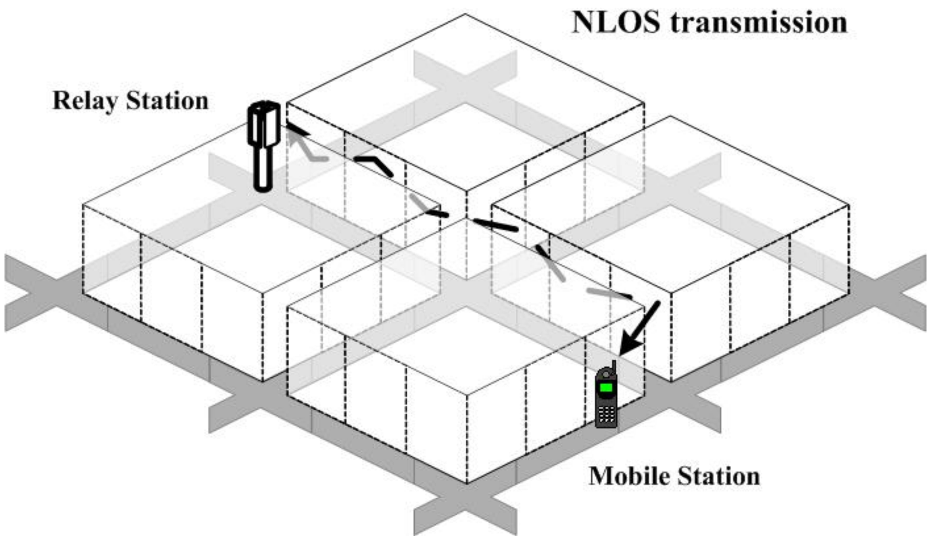


Figure 7. RS-MS NLOS scenario

For this case, the model takes the 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 (see section 2.1.2.4), such that the visibility factor is included, and the effective road height to give the correct breakpoint in the first street section. The full model is shown below:

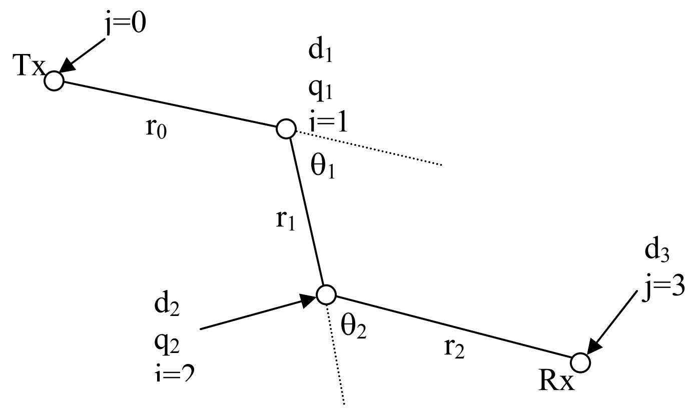


Figure 8. Geometry of street sections used for Berg model

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

$R = \sum_{j=1}^{n} r_{j-1} =$ 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)

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

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

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 $d_0 = 0$

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

$\theta_j =$ 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 this scenario the alternative WINNER path-loss model (for 5GHz) can be used:

$PL = 65 + 0.096 \cdot d_1 + (28 - 0.024 \cdot d_1) \cdot \log_{10}(d_2)\ dB$ *for* 10m < d_1 < 550m and $w/2 < d_2 <$ 450m

where d_1 is the distance along the main street in meters, d_2 is the distance for perpendicular street, w is the street width, and the carrier frequency is 5 GHz

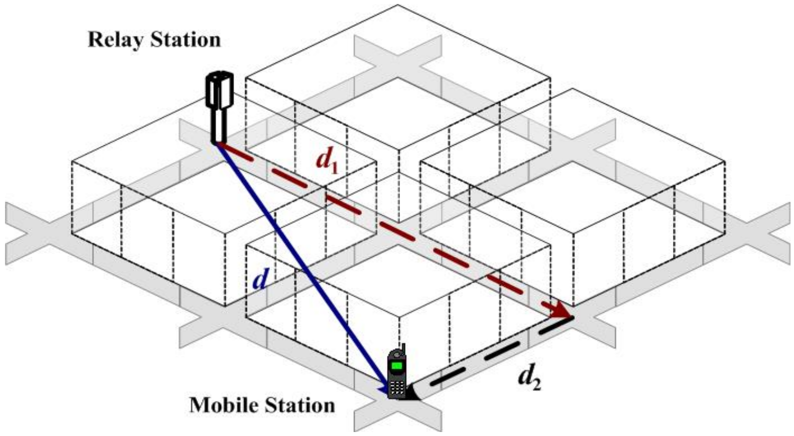


Figure 9. The alternative model for RS-MS NLOS scenario

2.1.2.5 Type-G Indoor Office Environment path-loss Model

The path-loss model for indoor environment is
$PL = 37 + 30 \cdot \log_{10}(d) + 18.3 \cdot n^{((n+2)/(n+1)-0.46)}\ dB$
where d is the distance in meters and n is the number of floors in the path.

For Type-G Indoor Office Environment scenario the alternative WINNER path-loss model can be used:
For LOS case:

$PL(d)$=18*log10 (d)+46.8 dB for 3m < d < 100m

For NLOS case:

$PL(d)$=36.8*log (d)+38.8 dB for 3m < d < 100m

Where d is in meters and the carrier frequency is 5GHz.

2.1.2.6 Type H Urban ART to ART model

The model for Type H environments is TBD. A suggestion for a model can be found in document C802.16j-06/262. A modified version is to be proposed in an upcoming meeting.

2.1.2.7 Type J Outdoor to Indoor, tunnels, in-vehicle and subway station model

The model for Type J environments is TBD. A suggestion for a model can be found in document C802.16j-06/262. A modified version is to be proposed in an upcoming meeting.

2.1.2.8 LOS Probability

In path-loss Type-F and Type-G, the radio link may be either LOS (Line-Of-Sight) or NLOS (Non Line-Of-Sight).

For Type-F, both node-antennas are below rooftop. Therefore, the following equation for LOS probability [15] can be considered in simulation.

$$P_{LOS}(d) = \begin{cases} 1 & d \le 15m \\ 1-\left(1-\left(1.56-0.48\cdot\log_{10}(d)\right)^3\right)^{1/3} & d > 15m \end{cases}$$

where $d = \sqrt{d_1^2 + d_2^2}$, and d_1 and d_2 are like in Figure 9.

For Type-G, indoor office environment, the following equation for LOS probability [15] should be considered when simulation.

$$P_{LOS}(d) = \begin{cases} 1 & d \le 2.5m \\ 1-0.9\cdot(1-(1.24-0.61\cdot\log_{10}(d))^3)^{1/3} & d > 2.5m \end{cases}$$

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 based on WINNER models of the standard deviation for lognormal shadowing is listed in Table 4,

Table 4. Standard Deviation Values

	Type-A	Type-B	Type-C	Type-D	Type-E	Type-F		Type-G	
						LOS	NLOS	LOS	NLOS
Std (dB)	10.6	9.6	8.2	3.4	8.0	2.3	3.1	3.1	3.5

2.2.1.1 Correction factor for standard deviation of the 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.

$$\sigma(r) = \sigma_\mu \left[1 - e^{\frac{-\left|P(r)-P_{fs}(r)\right|}{4}}\right] + 1.5$$

Where,

σ_{μ} is the maximum standard deviation
$P(r)$ 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 (σ_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 10.

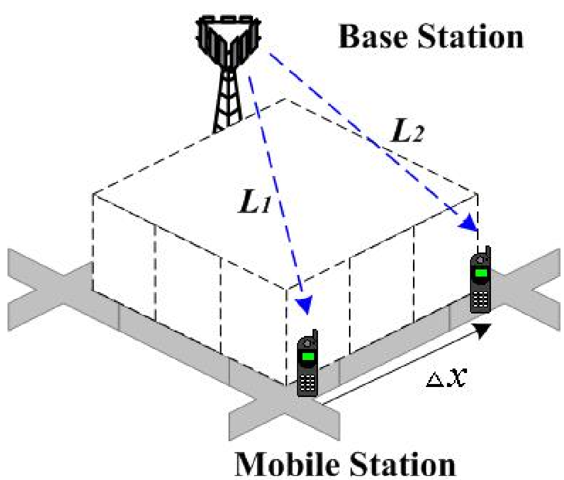


Figure 10. Auto-correlation of shadow fading

A popular model is:

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

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 11(a) should be lower than the one in Figure 11(b).

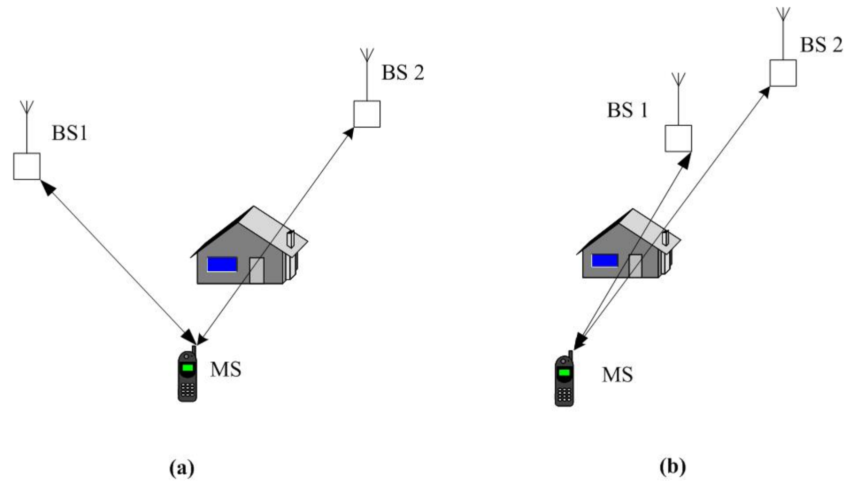


Figure 11. Cross-correlation of shadow fading

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 is recommended based on one proposed by Saunders. The geometry for the model is shown in Figure 12.

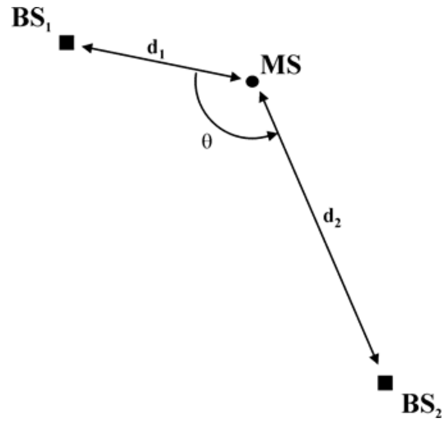


Figure 12. Geometry for correlation between two BSs

The correlation is then calculated using the following equations:

$$\rho = \begin{cases} \dfrac{d_1}{d_2} & \text{for } 0^\circ \le \theta \le \theta_T \text{ and } d_1 \ge \dfrac{d_c}{2} \\ \left(\dfrac{\theta_T}{\theta}\right)^\gamma \sqrt{\dfrac{d_1}{d_2}} & \text{for } \theta_T \le \theta \le \pi \text{ and } d_1 \ge \dfrac{d_c}{2} \\ \sqrt{\dfrac{d_c}{2d_2}} & \text{for } d_1 < \dfrac{d_c}{2} \end{cases}$$

where,

$\rho =$ Correlation

$d_c = \dfrac{1}{e}$ decorrelation distance

$\theta_T = 2\sin^{-1}\left(\dfrac{d_c}{2d_1}\right)$

$\gamma =$ 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 $\mathbf{R_{xx}}$.

$$\mathbf{y} = \mathbf{T}\mathbf{x}$$

where,

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

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

$\mathbf{T}$ = Transformation matrix

To calculate T : -

$$\begin{aligned}\mathbf{R_{xx}} &= E\left[\mathbf{y}\mathbf{y}^{\mathbf{H}}\right] = E\left[\mathbf{T}\mathbf{x}\left[\mathbf{T}\mathbf{x}\right]^{\mathbf{H}}\right]\\ &= \mathbf{T}E\left[\mathbf{x}\mathbf{x}^{\mathbf{H}}\right]\mathbf{T}^{\mathbf{H}}\\ &= \mathbf{T}\mathbf{I}\mathbf{T}^{\mathbf{H}}\\ &= \mathbf{T}\mathbf{T}^{\mathbf{H}}\end{aligned}$$

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

$$\begin{aligned}\mathbf{R_{xx}} &= \mathbf{U}\mathbf{D}\mathbf{U}^{\mathbf{H}}\\ \mathbf{D} &= \mathbf{D}^{1/2}\mathbf{D}^{1/2}\\ \mathbf{R_{xx}} &= \left(\mathbf{U}\mathbf{D}^{1/2}\right)\left(\mathbf{D}^{1/2}\mathbf{U}^{\mathbf{H}}\right)\\ &= \left(\mathbf{U}\mathbf{D}^{1/2}\right)\left(\mathbf{U}\mathbf{D}^{1/2}\right)^{\mathbf{H}}\\ &= \mathbf{T}\mathbf{T}^{\mathbf{H}}\end{aligned}$$

where : -

$\mathbf{A}^{\mathbf{H}}$ = 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:

$$L(dB) = \sum_{n=1}^{N}\left(a\cos\left(k_{n\_1}x + \phi_n\right)\cos\left(k_{n\_2}y + \psi_n\right)\right)$$

$$a = \sqrt{\frac{4\sigma^2}{N}}$$

ϕ_n and ψ_n are random phase terms uniformly distributed between 0 - 2π

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, 2π/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) multipath fading model parameters are summarized in Table 4 and other details regarding the channel models can be found in [14]. The SUI-1, SUI-2 and SUI-3 models are applicable for LOS condition, and SUI-4, SUI-5 and SUI-6 models are applicable for NLOS condition.

Table 5. 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	μs
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	μs
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	μs
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	μ 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	μ 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	μ 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:

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

where

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

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

θ_{3db} = 70° is the 3 dB beam width for 3 sector antenna, 35° 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

This section describes the traffic models in detail. A major objective of multihop 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 multihop 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.

The traffic models proposed in this document apply only to the MMR-BS and SS.

3.1 Traffic Modeling for IEEE802.16j Services

The required traffic models and their corresponding sections where they are defined are listed in Table 6.

Table 6. Services to be considered

#	Application	Traffic Category	Definition
1	Full buffer		Section 3
2	HTTP (UL and DL)	Interactive	Section 3.1.1
3	FTP (UL and DL)	Best effort/ Non real-time	Section 3.1.2
4	Near Real Time (NRT) Video Streaming (UL and DL)	Streaming	Section 3.1.3
5	VoIP	Real-time	Section 3.1.4
6	Gaming (UL and DL)	Real-time	Section 3.1.5
7	Live Video	Interactive Real-time	TBD

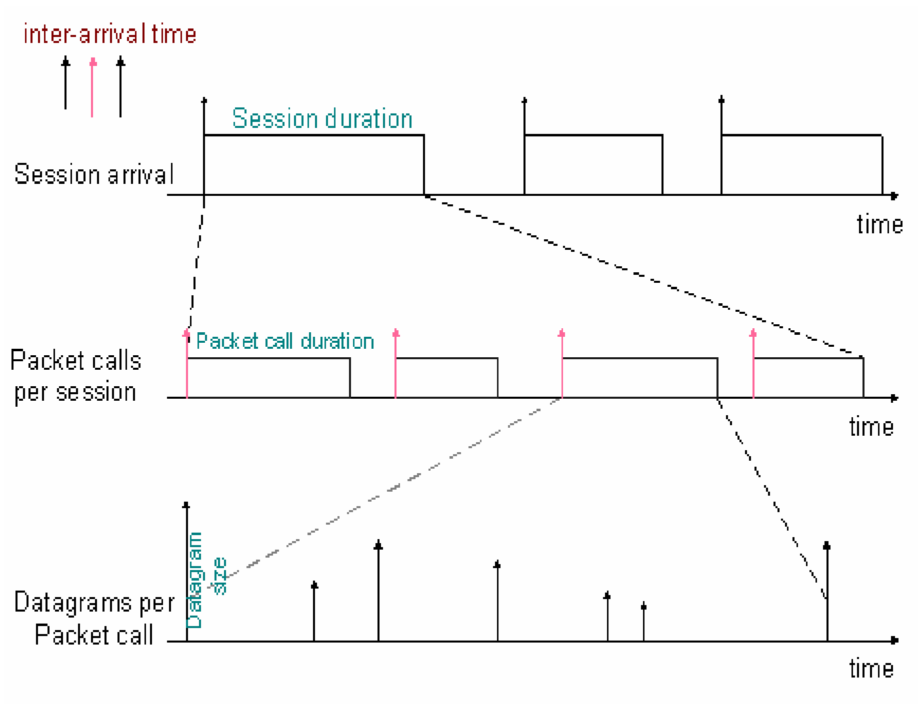
For a simulation with HTTP, FTP and NRT video streaming traffic models, if simulation is for DL (or UL) traffic only, UL (or DL) traffic modeling (e.g. HTTP/FTP requests) can be neglected for the simplicity as the bandwidth requirements for these messages are small compared to the data traffic.

The FTP and HTTP traffic models listed in Table 6 can be generated using the bursty traffic generation model described in Figure 13. For each traffic source, the following characteristics are modeled:

1. Session arrival in terms of session inter-arrival time and session duration.
2. Packet call arrival in terms of packet call inter-arrival time and packet call duration within a session. Within a packet call, there are periods of active traffic generation and periods of no activity.
3. Finally, datagram inter-arrival times and datagram size within a packet call.

We consider that a single session stays from the beginning of the simulation till the end of the simulation, i.e., the whole simulation time. Therefore, packet call and datagram inter-arrival times, packet call duration and datagram size distributions for these bursty traffic models will be described in the next sections.

Figure 13. Bursty traffic generation model

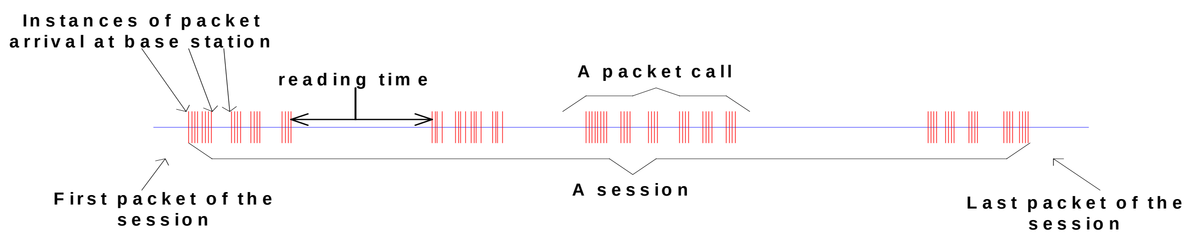


3.1.1 HTTP model (UL and DL) [1][16]

3.1.1.1 HTTP traffic model characteristics

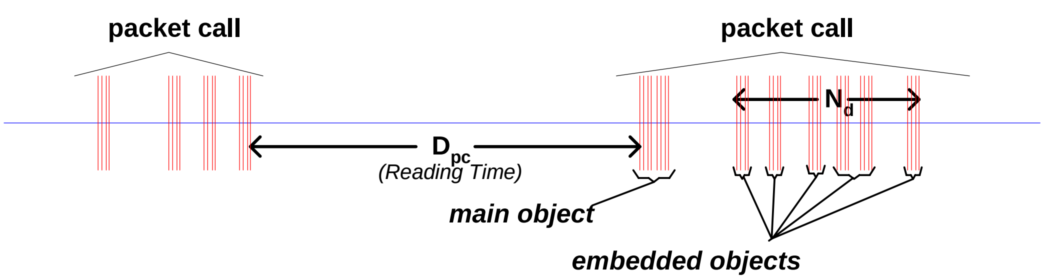
Figure 14 shows a typical web browsing session. Each session is divided into ON/OFF periods representing web-page downloads and intermediate reading times. Each web-page download is referred to as packet calls in Figure 14. During an ON period (packet call), users are requesting information. During an OFF period, user is reading/digesting the web-page.

Figure 14. Packet trace of a typical web browsing session



The activity within each packet call can be found in Figure 15. Note the similarity of the distribution for the packet calls within a session in Figure 14 and the datagram arrivals within a packet call in Figure 15. This can possibly be a result of self-similarity in web-browsing traffic.

Figure 15. Contents of a packet call



There are ON and OFF periods within a packet call. During an ON period, objects are being retrieved. Parsing time and protocol overhead are represented by the OFF periods within a packet call. During a packet call, the initial HTML page (referred to as the main object) is first downloaded. However, within the initial HTML page, there can be additional references to embedded object files such as graphics and buttons. After parsing the information on the embedded objects, the embedded objects will be loaded next as indicated in Figure 15.

3.1.1.2 HTTP traffic model parameters

The parameters for web browsing traffic are:

- No of pages per session;

- S_M: size of the main object in a packet call;
- S_E: size of an embedded object in a packet call;
- N_d: number of embedded objects in a packet call;
- D_{pc}: reading time;
- T_p: parsing time for main page

Table 7. HTTP Traffic Model Parameters

Component	Distribution	DL Parameters	UL Parameters	
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes, Std. Dev = 25032 bytes, Minimum = 100 bytes; Maximum = 2Mbytes, $\sigma = 1.37, \mu = 8.35$	Mean = 9055 byes, Std. dev. = 13265 bytes, Minimum = 100 bytes, Maximum = 100Kbytes $\sigma = 1.37, \mu = 8.35$	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right]$ $x \geq 0$
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758bytes, Std. dev. = 126168bytes, Minimum = 50bytes, Maximum = 2Mbytes $\sigma = 2.36, \mu = 6.17$	Mean = 5958bytes, Std. dev. = 11376bytes, Minimum = 50bytes, Maximum=100kbytes $\sigma = 1.69, \mu = 7.53$	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right]$ $x \geq 0$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64, Maximum = 53 $\alpha = 1.1, k = 2, m = 55$	Mean = 4.229, Maximum = 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 generated random value to obtain N_d.
Reading Time (D_{pc})	Exponential	Mean = 30seconds	Mean = 30seconds	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.033$
Parsing time (T_p)	Exponential	Mean = 0.13second	Mean = 0.13second	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 7.69$

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.

3.1.1.3 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 [1] for details.

3.1.1.4 HTTP and TCP interactions for UL HTTP traffic

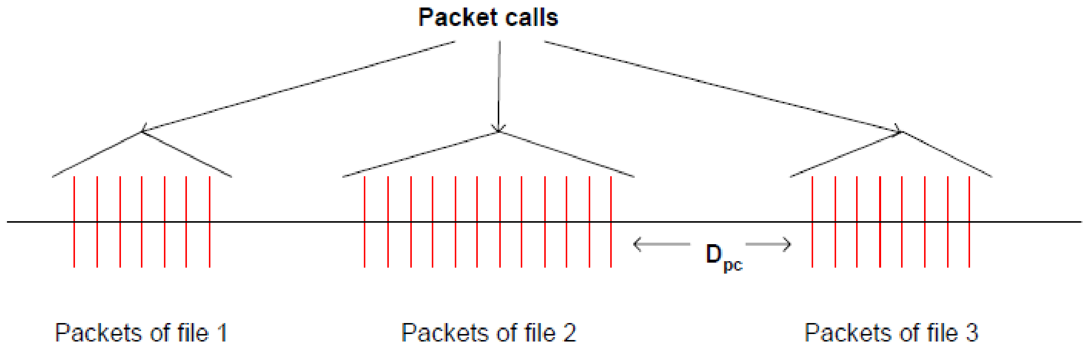
HTTP/1.1 is used for UL HTTP traffic. For details regarding the modeling of the interaction between HTTP traffic and the underling TCP connection, refer to 4.2.4.1, 4.2.4.2 of [1].

3.1.2 FTP model (UL and DL) [1][16]

3.1.2.1 DL FTP traffic model characteristics

For DL FTP, activities within a FTP session can be found in Figure 16. A typical FTP session consists of a sequence of file transfers separated by reading time. Each file transfer can be treated as a packet call. Reading time can be treated as the OFF period within a session. Within each packet call, only the file size is randomly generated.

Figure 16. Packet trace in a typical DL FTP session



3.1.2.2 DL FTP traffic model parameters

Hence, there are two main parameters for a DL FTP session:

- S: size of file to be transferred;
- D_{pc}: reading time. This is the time interval between end of download of the previous file and the user request for the next file.

The parameters distribution and values can be found in Table 8.

Table 8. DL FTP traffic model parameters

Component	Distribution	Parameters	PDF

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

3.1.2.3 UL FTP traffic model characteristics

FTP traffic in the UL direction is generated mainly from file upload and email attachment upload. Each FTP upload user stays in the system until it finishes the transmission of its file. The FTP upload user leaves the system immediately after it finishes the transmission of its file.

Hence, for UL FTP traffic, each FTP session consists of 1 packet call. Within the packet call, only the file size is randomly generated.

3.1.2.4 UL FTP traffic model parameters

The only traffic model parameter is the upload file size and can be found in Table 9.
For UL FTP traffic, users shall arrive according to a Poisson process with arrival rate λ.

Table 9. UL FTP traffic model parameter

Arrival of new users	Poisson with parameter λ
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, discard it and regenerate a new value. The resulting truncated lognormal distribution has a mean = 19.5 kbytes and standard deviation = 46.7 kbytes

3.1.2.5 FTP and TCP interactions

To model the FTP and TCP interactions, please refer to 4.1.4.2 of [1] for details.

3.1.3 Near real time video streaming (NRT video streaming) (UL and DL) [1][16]

A video streaming session is defined as the entire video streaming call time. It is equal to the simulation time for this model. Hence, a video streaming session occurs during the whole simulation period. No session inter-arrival time is needed. It is originally modeled for DL direction. However, the same model is proposed to be used for UL direction.

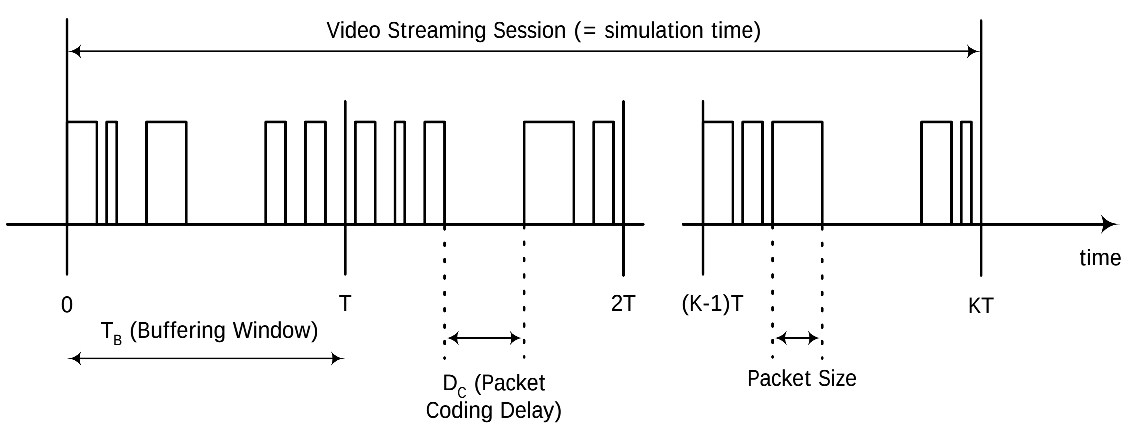
3.1.3.1 NRT video streaming traffic model characteristics

Figure 17 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.

Each frame of video data arrives at a regular interval T. Each frame can be treated as a packet call and there will be zero OFF duration within a session. Within each frame (packet call), packets (or datagrams) arrive randomly and the packet sizes are random as well.

To counter the jittering effect caused by the random packet arrival rate within a frame at the MS, the MS uses a de-jitter buffer window to guarantee a continuous display of video streaming data. The de-jitter buffer window for video streaming service is 5 seconds. At the beginning of simulation, the MS de-jitter buffer shall be full with video data. During simulation, data is leaked out of this buffer at the source video data rate and filled as DL traffic reaches the MS from the BS. As a performance criterion, the simulation shall record the length of time, if any, during which the de-jitter buffer runs dry.

Figure 17. Video streaming traffic model



3.1.3.2 NRT video streaming traffic model parameters

The packet sizes and packet inter-arrival rate can be found in Table 10 when using a source rate of 64 kbps.

Table 10. 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 $\alpha = 1.2$	K=2.5ms $\alpha = 1.2$

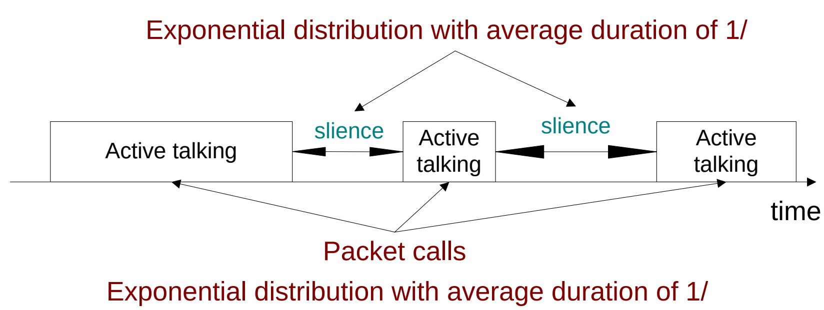
3.1.4 VoIP model [1][17][18][19]

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.

3.1.4.1 VoIP traffic model characteristics

A typical phone conversation is marked by periods of active talking interleaved by silence/listening period as shown in Figure 18.

Figure 18. Typical phone conversation profile



A two state Markov process (active-inactive) is used to model a VoIP source in Figure 19. The alternating periods of activity and silence are exponentially distributed with average durations of $1/\beta$ and $1/\alpha$ respectively. Hence, the fraction of time the voice source is active is $\alpha/(\alpha+\beta)$. For a voice activity factor of 40%, $1/\beta = 1$s and $1/\alpha = 1.5$s. Each active state period can be treated as a packet call and inactive period as the OFF period within a session.

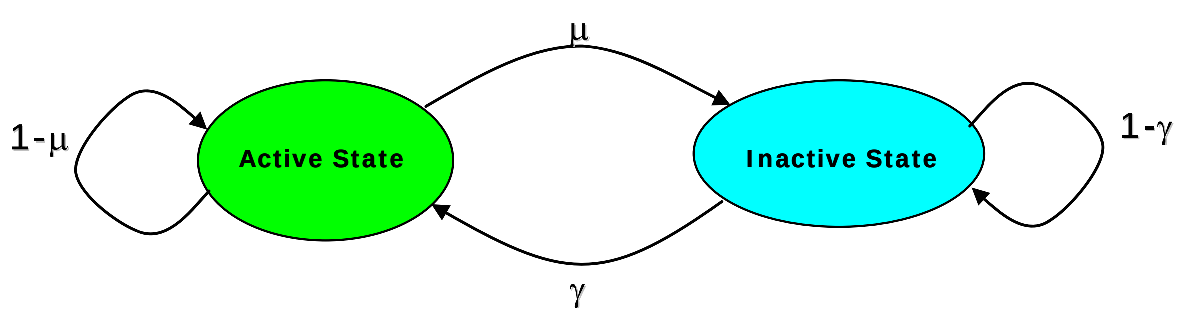
During the active state, packets of fixed sizes are generated at a regular interval. During the inactive state, we have chosen to generate comfort noise with smaller packet sizes at a regular interval instead of no packet transmission. The size of packet and the rate at which the packets are sent depends on the corresponding voice codecs and compression schemes. Table 11 provides information on some common vocoders.

Table 11. Information on various vocoders

Vocoder	EVRC	AMR	G.711	G.723.1		G729A
Source Bit rate [Kb/s]	0.8/2/4/8.55	4.75-12.2	64	5.3	6.3	8
Frame duration [ms]	20	20	10	30	30	10
Information bits per frame	16/40/80/171	95-244	640	159	189	80

Among the various vocoders in Table 11, 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.

Figure 19. Markov chain model of a VoIP source



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. Table 12 shows the VoIP packet size calculation for simplified AMR with or without header compression when using IPv4 or IPv6.

Table 12. VoIP packet size calculation for simplified AMR and G. 729

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 (inactive) 33 bytes (active)	7bytes (inactive) 33 bytes (active)	0 bytes (inactive) 20 bytes (active)	0 bytes (inactive) 20 bytes (active)
Protocol Headers	40 bytes / 60 bytes	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	6 bytes	6 bytes	6 bytes	6 bytes
CRC	4 bytes	4 bytes	4 bytes	4 bytes
Total VoIP packet size	57 bytes/ 77 bytes (inactive) 87 bytes / 103 bytes (active)	19 bytes/ 21 bytes (inactive) 45 bytes/ 47 bytes (active)	0 bytes (inactive) 70 bytes / 90 bytes (active)	0 bytes (inactive) 32 bytes/ 34 bytes (active)

3.1.4.2 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. Parameters associated with the VoIP traffic model can be found in Table 13.

Table 13. VoIP traffic model parameters specification

Component	Distribution	Parameters	PDF
Active state duration	Exponential	Mean = 1 second	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 1/Mean$
Inactive state duration	Exponential	Mean = 1.5 second.	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 1/Mean$
Probability of transition from active to inactive state	N/A	μ (=0.6)	N/A

Probability of transition from inactive to active state	N/A	γ (=0.4)	N/A

3.1.5 Gaming model (UL and DL) [1][20]

Gaming traffic is generated by users engaged in interactive gaming of multiple users in different locations via the internet. A gaming session is defined as the time duration that a user plays a game and a gaming session occurs during the whole simulation period.

3.1.5.1 Gaming traffic model characteristics

The packet arrival time and the frame boundary are random and shall be simulated. Gaming packets are relatively small in size. Due to the interactive nature of gaming, packet delay must be short. Any packets that are generated and not transmitted at the PHY layer within 160ms shall be dropped.

3.1.5.2 Gaming traffic model parameters

Gaming traffic model parameters for DL and UL can be found in Table 14[20]. Largest Extreme Value distribution is used for random packet size generation. Since packet size has to be an integer, the largest integer less than or equal to X is used as the actual packet size.

Table 14. Gaming traffic model parameters

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	
Initial packet arrival	Uniform	Uniform	a=0, b=40ms	a=0, b=40ms	$f(x)=\frac{1}{b-a}, a \le x \le b$
Packet inter-arrival time	Extreme	Extreme	a=48ms, b=4.5ms	a=40ms, b=6ms	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=\lfloor a-b\ln(-\ln Y)\rfloor, Y\in U(0,1)$
Packet size	Extreme	Extreme	a=330bytes, b=82bytes	a=45bytes, b=5.7	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=\lfloor a-b\ln(-\ln Y)\rfloor+2, Y\in U(0,1)$ Addition of 2 in the equation is due to 2 bytes of UDP header size after header compression.

3.2 Traffic mix proposal

To test various aspect of the system, we propose the following traffic mixes:
1. Five cases of HTTP, FTP, NRT Video Streaming, Gaming, or Voice only.
2. Three cases of mixed traffic from Mix -1 to Mix -3 referenced in Table 15. The percentage of the traffic mix in these 3 cases is expressed in terms of data capacity (i.e., bps) of a given targeted cell.

Table 15. Proposed traffic mixes

	VoIP	FTP	HTTP	NRT video	Gaming
Voice Only	100% #users = Nv	0%	0%	0%	0%
FTP only	0%	100%	0%	0%	0%
HTTP only	0%	0%	100%	0%	0%
NRT Video only	0%	0%	0%	100%	0%
Gaming only	0%	0%	0%	0%	100%
Traffic Mix 1	0.5 Nv	Remaining Capacity for Data Users 100%	0%	0%	0%
Traffic Mix 2	0.5 Nv	Remaining Capacity for Data Users 30%	30%	30%	10%
Traffic Mix 3	0.75 Nv	Remaining Capacity for Data Users 30%	30%	30%	10%

Nv is the system voice capacity that satisfy outage criteria at system and user level.

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 RS to BS, RS to MS and BS to MS links separately. 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 when the relays are deployed. Link budget template from ITU-R M.1225 [21] is modified, which are provided in detail in Appendix B.

Since relays can be used to extend the range covered by a cell under noise limited environment (i.e. negligible 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.

4.2 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 (Rmin), and the coverage (allowable outage probability).

4.2.1 Combined Coverage and Capacity Index (*cc*)

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 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.

4.2.2 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 Rmin. The first step in evaluating *cc* is to sort the MSs in descending or der 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) < Rmin, $cc = 0$ (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.

4.2.3 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, 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 $R\min$ is achieved. Bandwidth is shared by the BS and RSs, while the BS and each RS have their own power resource. 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

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

4.3 Definitions of Performance Metric

4.3.1 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.

4.3.2 Packet call throughput:

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

$$\text{Packet Call Throughput} = \frac{1}{K}\sum_{k=1}^{K}\frac{\text{bits in packet call k}}{(t_{end\_k} - t_{arrival\_k})}$$

4.3.3 Effective system spectral efficiency

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

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

4.3.4 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 received divided by the amount of time the user was actively involved in data packet transfer.

4.3.5 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.

4.3.5.1 Maximum MMR Packet Latency

The maximum MMR packet latency 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 MMR system for a given packet call duration.

4.3.5.2 X%-tile MMR Packet Latency

The x%-tile MMR packet latency is simply the packet latency number in which x% of packets have latency below this number.

4.3.5.3 Average MMR Packet Latency

The average MMR packet latency 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 an MMR system for a given packet call duration.

4.3.5.4 Jitter

This parameter defines the maximum delay variation (jitter) for the packets of a given packet call duration in an MMR system.

4.3.6 Packet Loss Ratio

The packet loss ratio per user is defined as:

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

Typically for a VoIP application, 2% packet loss ratio is tolerable. For gaming and video streaming applications, packet loss ratio is typically less than 1%. Both the single link packet latency and the packet loss ratio per user are important performance metrics for assessing different QoS schemes.

4.4 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 for best effort traffic, a measure of fairness under best effort assumption is 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

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

4.4.1 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

4.4.2 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.

4.4.3 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 Table 16.

Table 16. Criterion CDF

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

4.4.4 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.

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Appendices

A.1 Multi-Cell Layout

In Figure 20, 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.

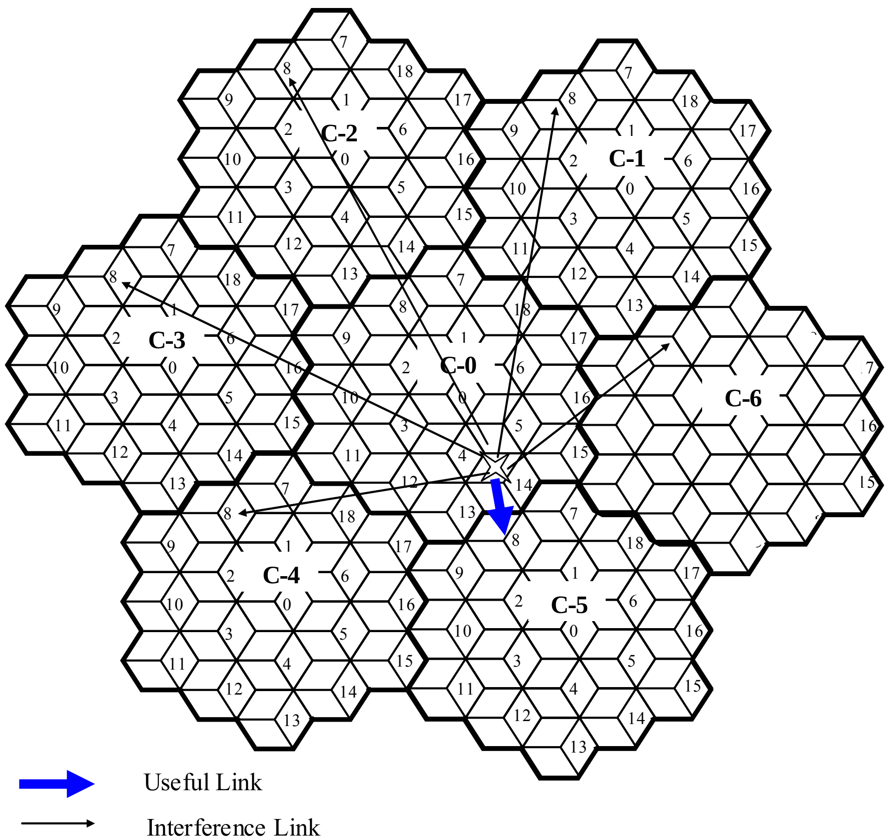

Figure 20. Multi-cell Layout and Wrap-around Example

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

The link budget can be divided into two parts: The system gain reflects the performance of the transmitter and receiver, including aspects such as antenna gain and receiver sensitivity. The following link budget template in ITU-R M.1225 [21] with slight modifications is given in Table below. Entries that have explicit example numerical values in the table (such as power levels, cable losses, etc) should be used to support system level simulations. The values provided for RS antenna gain are just as an example and should be adjusted based on the antenna used in simulation.

Item	Downlink	Uplink
(a1)	dBm	dBm
(b)	3 dB for BS 1 dB for RS	0 dB for MS 1 dB for RS
Body Losses	0 dB for both RS and BS	3 dB for MS 0 dB for RS
(c)	17 dBi for BS 11 dBi for RS	0 dBi for MS 11 dBi for RS
(d1)	dBm	dBm
Penetration Loss (Ref: 3GPP2) [Determine how to use these numbers for different environments, revisit if 20dB is a reasonable value for building penetration)]	20 dB (Building) 10 dB (Vehicular)	20 dB (Building) 10 dB (Vehicular)
(e)	0 dBi e.g., 11 dBi for RS	17 dBi for BS 11 dBi for RS
(f)	0 dB for MS 1 db for RS	3 dB for BS 1 dB for RS
Body Losses	3 dB for BS 0 dB for RS	0 dB for both RS and BS
(g)	Refer to Equation (149b) in 802.16e-2005	Refer to Equation (149b) in 802.16e-2005
(h)	dB	dB
(i)	dB	dB
(j)	dB	dB
(k)	dB	dB

(l)	dB	dB
(m)	m	m