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Abstract	
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Contents

1.0 Link Level Simulation Modeling

Link level simulation reflects link-performance of the proposed system and provides link performance curves for system level simulation.

1.1 Link Level Simulation Assumptions

Table 1 provides a list of the link-level simulation assumptions that are relevant to the OFDM evaluation case.

Table 1: OFDM link-level simulation assumptions

Parameter	Assumption
Carrier frequency	450 MHz, 800 MHz, 3.4 ~ 4.2 GHz, and 4.4 ~ 5.0 GHz, which are considered candidate frequency bands [1]
Fast fading model	Jakes model or others
Pilot power	The pilot power remains fixed, regardless of the number of data channels in use.
Power for other common channels	The other channel power remains fixed, regardless of the number of data channels in use.
Power for data transmission	The total data transmission power will depend on the number of data channels in use.
Channel estimation	Ideal: Ideal channel information is assumed to be available at the receiver. Perfect timing and frequency estimation is also assumed. Real: Real estimation from pilot sub-carriers
Bandwidth	Scaleable bandwidth 5 to 20 MHz. And other bandwidth shall be considered as necessary to meet operator and ITU requirements.
Number of subcarriers	It depends on OFDM system parameter
OFDM sampling frequency	It depends on OFDM system parameter
FFT size	It depends on OFDM system parameter
Subcarrier MCS levels	Assume one MCS per MS in one frame or more

The performance of the OFDM system is depended on the extent of frequency selective fading of the channel. In OFDMA system, frequency selective scheduling can be applied by channel band selection for MSs. For this mode of scheduling with a contiguous permutation of subcarriers (as the Band Adaptive Modulation and Coding (AMC) mode in IEEE 802.16 systems), the sub-channels may experience different attenuation. A Channel Quality Indicator (CQI) channel is utilized to provide channel-state information from the user terminals to the base station scheduler. Relevant channel-state information can be fed back by the CQICH including: Physical CINR, effective CINR, MIMO mode selection and frequency selective sub-channel selection. With TDD implementations, link adaptation can also take advantage of channel reciprocity to provide a more accurate measure of the channel condition (such as sounding). The base station scheduler determines the appropriate data rate (or burst profile) for each burst allocation based on the buffer size, from the MSs, etc.

1.2 Performance Metrics

1.2.1 Packet error rate (PER) (vs. SNR or SINR)

$$PER = \frac{n_{erroneous\_packets}}{n_{packets}}$$

where $n_{erroneous\_packets}$ is the total number of erroneous packets, and $n_{packets}$ is the total number of packets transmitted.

1.2.2 Link Throughput [bps]

$$R = \frac{b}{T}$$

where R is link throughput, b is the total number of correctly received data bits (If HARQ is applied, b is the total number of correctly received data bits after all retransmission) over the whole simulation time, T.

2.0 System Level Simulation Modeling

System level simulation shall reflect performance of the proposed system by mimicking characteristics of communication environment as well as the proposed system itself. Characterizing communication environment in exact manner is not possible to simulate. So, we shall adopt some assumptions to reduce simulation time.

2.1 System Level Simulation Assumptions

System level simulation setup is based on channel model.

The test environments for IMT-Advanced are the following [2] :

- Base coverage urban : an urban macro-cellular environment targeting to continuous coverage for pedestrian up to fast vehicular users in built-up areas.
- Microcellular : an urban micro-cellular envirionment with higher user density focusing on pedestrian and slow vehicular users.
- Indoor : an indoor hotspot environment targeting isolated cells at home or in small offices based on stationary and pedestrian users.
- High speed : macro cells environment with high speed vehicular and trains.

shows the layout for a standard 2-tier, 19-cell model.

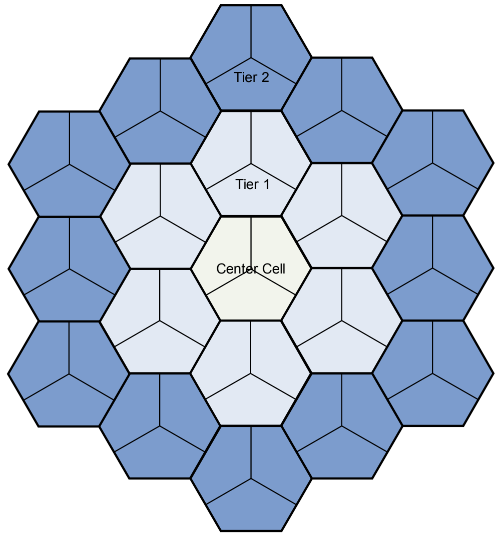

Fig. 1 A standard 2-tier, 19-cell, 57-sector layout

For three sector cell, the base station antenna pattern used for each sector, is specified as

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

where $-180 \le \theta \le 180$, and $\theta_{3dB} = 70$, the 3dB beam width, and A_m=20 [dB] is the maximum attenuation. [3]

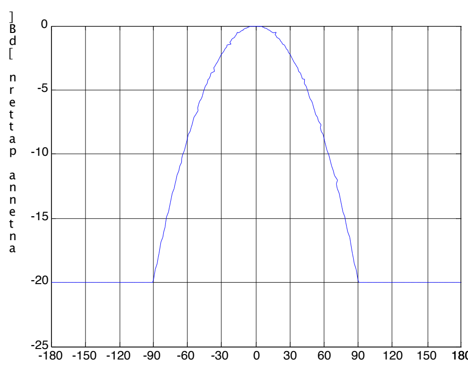

Fig. 2 Antenna pattern used for three sector base station transmit antenna

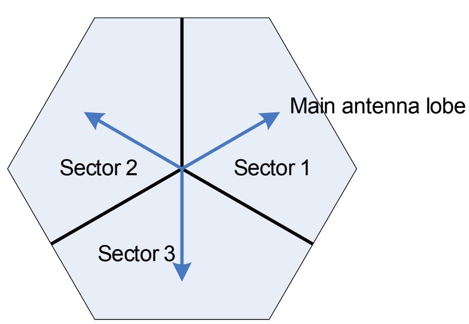

Fig. 3 Antenna bearing orientation

Mobile stations must be uniformly dropped onto the system layout for each simulation run. For mixed channel model, each mobile station decides it's channel model and mobility according to the assignment probability specified. A mobile station must be redropped if it is placed within 35m of the closest base station. Except for handover simulation, each mobile station must remain stationary (no changes in path loss, shadowing or antenna gain); only short term channel fading effects are updated through simulation time. Short term fading does not count when a mobile station decides it's serving base station, but only long term fading such as shadowing, path loss, and antenna gain counts.

The shadow fading is modeled as a Gaussian distributed random variable with zero mean and standard deviation σ . The shadow fading may be expressed as the weighted sum of a common component, Z_i, to all cell sites, and an independent component, Z_j, from each cell site. In other words, Z_i is generated based on local shadowing point where a mobile station is dropped, in the mean time, Z_j is generated based on local shadowing point where a base station is dropped.
The shadow fading value between each mobile station, *i*, from each base station, *j*, is $X_{ij} = aZ_i + bZ_j$. Typical values for *a* and *b* are $a^2 = b^2 = 1/2$. That is, the correlation is 0.5 between sectors from different cells and 1.0 between sectors of the same cell. And Z_i is calculated as following procedures.

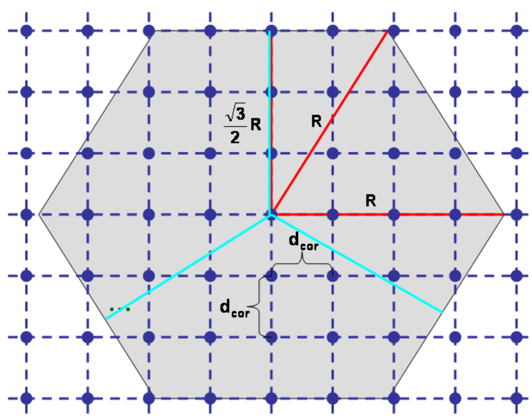

Fig. 4 Lattice structure for a cell

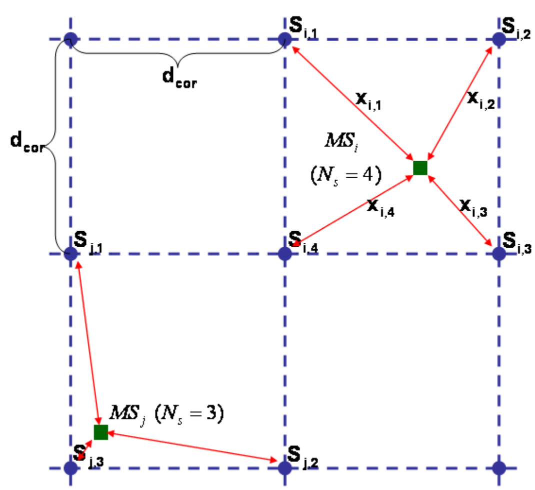

Fig. 5 Location based shadowing calculation

- Step 1. Lattice square zone selection for each MS position

- Step 2. Distance $x_{i,k}$ calculated from a MS to 4 shadowing points for its selected lattice square zone
- Step 3. Selection of valid shadowing point candidates and Ns with $x_{i,k} \quad d_{COR}$
- Step 4. Calculate $R(x_{i,k}) = \exp\left(\frac{x_{i,k}}{d_{COR}} \ln 2\right)$
- Step 5. $Z_i = \frac{\sum_{k=1}^{N_s}\left\{ R\left(x_{i,k}\right) \; S_{i,k}\right\}}{\sum_{p=1}^{N_s} R\left(x_{i,p}\right)}$

To emulate link level PHY performance in system level simulation, exponential effective SIR mapping (EESM) might be used. EESM method is described in Appendix session. Before using link to system interface method, one must show the verification of the method.

Following table shows common system level simulation assumptions.

Table 2 Common system level simulation assumptions

Parameter	Assumptions	Comments
Cellular Layout	Hexagonal grid, 19 cells sites, 3 sectors per site	
Distance-dependent path loss	L=128.1 + 37.6log$_{10}$(.R), R in kilometers [4]	Macro cell
		Micro cell
Inter site distance		
Penetration loss	20dB for under 10km/h mobile 10dB for over 10km/h mobile	[4]
Shadowing standard deviation	8.9 [dB]	[3]
Shadowing correlation distance	50 [m]	[3]
Max. base station power	43dBm (5MHz)	[4]
Max. mobile station power	23dBm (5MHz)	[4]
Other cell interference	All base station always on at full power	Full buffer
	Depend on actual base station activity	Traffic model
	Mobile station's power depends on power control mechanism	
Mobile station feedback delay	depends on frame structure	
Base station antenna height	32 [m]	[3]
Mobile station height	1.5 [m]	[3]
Base station thermal noise figure	5 [dB]	[3]
Mobile station noise figure	7 [dB]	[3]
Thermal noise density (N_0)	-174 [dBm/Hz]	[3]

Antenna gain with cable loss	14 [dB]	For directional antenna
	0 [dB]	For omni antenna

2.2 Performance Metrics

2.2.1 General output metrics

2.2.1.1 Average cell/sector throughput [kbps/cell or kbps/sector]

The average cell/sector throughput is calculated as

$$R = \frac{b}{k \quad T}$$

where R is throughput, b is the total number of correctly received information data bits in all data mobile stations in the simulated system over the whole simulated time, k is the number of cells or sectors in the simulation, and T is the simulation time.

2.2.1.2 Average cell/sector spectral efficiency [bps/Hz/cell or kbps/Hz/sector]

The average cell/sector spectral efficiency is defined as

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

Where R is the average cell/sector throughput, BW_{eff} is the effective channel bandwidth. The effective channel bandwidth is defined as

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

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

2.2.1.3 Packet service session packet error rate (PER)

The packet error rate is calculated as

$$PER_{session} = \frac{n_{erroneous\_packets}}{n_{pakcets}}$$

where $n_{erroneous\_packets}$ is the total number of erroneous packets in the packet service session, and $n_{pakcets}$ is the total number of packets in the packet service session.

2.2.1.4 Residual PER

The residual PER is calculated as

$$PER_{residual} = \frac{n_{dropped\_pakcets}}{n_{packets}}$$

where $n_{dropped\_packets}$ is the total number of dropped packets in the packet service session after the maximum number of HARQ retransmission, and $n_{packets}$ is the total number of packets in the packet service session.

2.2.1.5 Cell edge user throughput

The cell edge user throughput can be observed by 5%tile point of CDF of user throughput.

2.2.1.6 Fairness

The fairness is evaluated by cumulative distribution function (CDF) of the average user throughput and the three points, {(0.1, 0.1), (0.2, 0.2), (0.5, 0.5)}. The CDF shall lie to the right of the line given by the three points.

2.2.2 Output metric for traffic model

[editor's note : this section shall be modified to reflect the IEEE 802.16m requirement document, section 5.3.]

2.2.2.1 Average packet call throughput [kbps]

The average packet call throughput per user i is defined as

$$R_{pktcall}(i) = \frac{\sum_{k} \text{good bits in packet call } k \text{ of user } i}{\sum_{k} \left(t_{end\_k} - t_{arrival\_k} \right)}$$

where $t_{arrival\_k}$ is the time when the first packet of packet call k arrives in queue, and t_{end_k} is the time when the last packet of packet k is received by mobile station.

2.2.2.2 The averaged packet delay

The average packet delay per sector is calculated as

$$D = \frac{\sum \text{delay from all packets of all MSs}}{\text{number of packets}}.$$

The delay for an individual packet is defined as the time between when the packet enters the queue at the transmitter and when the packet is received successively.

2.2.2.3 System outage

A user is in outage if more than a given percentage of packets (blocks) experience a delay of greater than a certain time. The system is considered to be in outage if any individual users are in outage.

3.0 Application Traffic Models

[editor's note : this section shall be modified to reflect the IEEE 802.16m requirement document, section 5.3.]

IMT-Advanced services are classified into 4 'user- experience' classes, conversational, interactive, streaming, and background. IMT-Advanced application traffic models shall be tested.
IMT-Advanced application traffics are:

- *Low Multimedia*: up to 144 kbit/s.
  - VoIP, video telephony and file sharing
- *Medium multimedia*: up to 2 Mbit/s.
  - video conference, mobile TV, broadcast IP TV, video/audio streaming, photo messages and business intranet/extranet.
- *High multimedia*: up to 30 Mbit/s.
  - high quality video conference, video streaming and messaging, application sharing, mobile internet/intranet/extranet and navigation.
- *Super high multimedia*: up to 100 Mbit/s or even 1 Gbit/s.
  - high volume streaming, e-newspaper and game data download, and mobile internet/ intranet/ extranet.

Application traffic models including the list above shall be considered in the evaluation methodology.
[Note: Modeling of each application traffic is FFS]

Some examples of the application traffic models, such as internet game, VoIP, Video streaming, HTTP, FTP, etc are presented in the appendix.

4.0 Appendix

4.1 Link to System Interface

The methodology for evaluation of proposals of IEEE 802.16m performance at the system level is shown in Fig. 6.

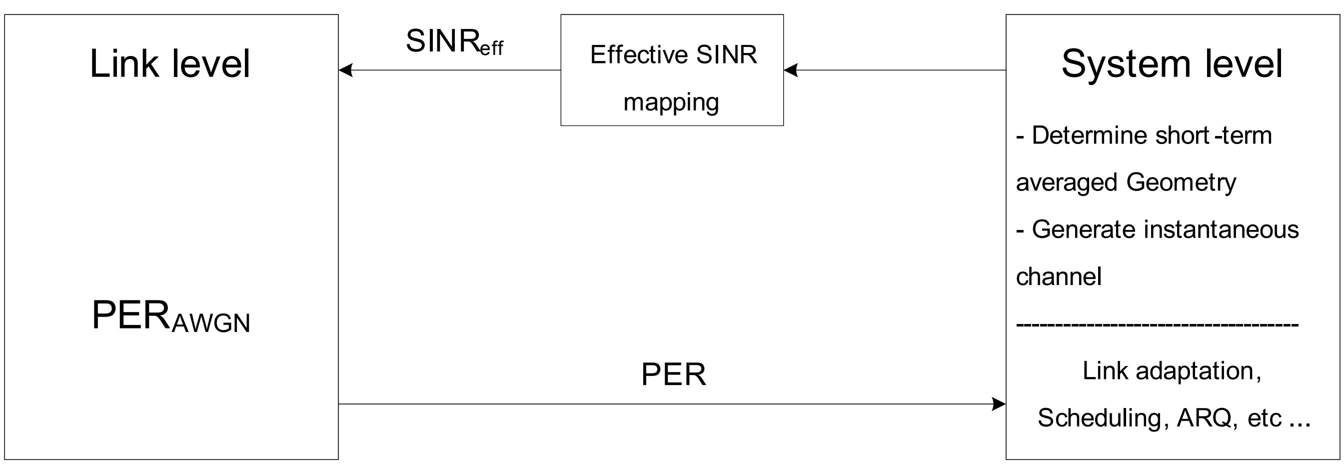

Fig. 6 System level methodology

The effective signal to interference and noise ratio ($SINR_{eff}$) is used to find the packet error rate (PER) for different modulation and coding set (MCS) from AWGN link level performance curves. $SINR_{eff}$ is calculated by Effective SINR mapping block with inter-cell interference, noise, intra-cell interference. After getting PER, the actual packet error is decided by coin tossing.

4.1.1 Effective SINR mapping functions

Effective SINR can be calculated using these values;

γ_k : SINR for k_{th} subcarrier

For single antenna transmission, $\gamma_k = P(k) \cdot G \cdot \left(\frac{N}{N + N_p}\right) \cdot \left(\frac{R_D}{N_{SD} / N_{ST}}\right)$.

$P(k)$: channel power for k_{th} subcarrier, if received signal at the k_{th} subcarrier is $y_k = h_k x_k + n_k$, where x_k is transmitted signal, n_k is noise signal, and h_k is the channel at k_{th} subcarrier, then $P(k) = |h_k|^2$.

G : the current geometry

N : the FFT size

N_p : the cyclic prefix length

R_d : the percentage of maximum total available transmission power allocated to the data subcarriers

N_{SD} : the number of data subcarriers per frame

N_{ST} : the number of total useful subcarriers per frame

4.1.1.1 Exponential Effective SIR Mapping (EESM)

$$SINR_{eff} = -\beta \ln\left(\frac{1}{N_u}\sum_{k=1}^{N_u} \exp\left(\frac{\gamma_k}{\beta}\right)\right)$$

where N_u is the number of transmitted subcarriers, β is a value for optimization.

The following is an example of the verification of EESM method.

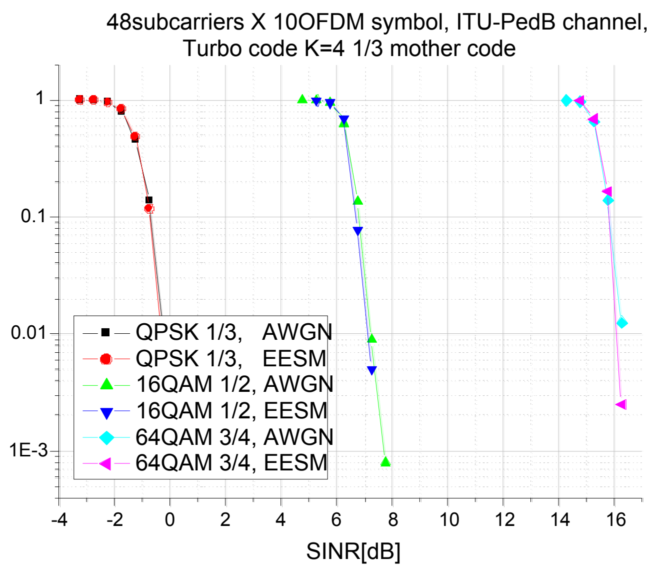

Fig. 7 AWGN curve vs. link curve by EESM method

For 3GPP Rel. 6 turbo code, the following beta shall be used [8]

- QPSK b = 0.413 * ER + 1.3661 (1.45 to 1.71 for ER=1/5 to 5/6)
- 16QAM b = 4.4492 * ER * ER + 4.5655 * ER + 1.2982 (2.39 to 8.19 for ER=1/5 to 5/6)
- 64QAM b=4.1182 * exp(2.4129 * ER) (6.67 to 30.76 for ER=1/5 to 5/6)

where ER is effective encoding rate.

4.2 Application traffic models

This section describes the traffic models in detail. The application traffic models are categorized in 5 type, these models are representative of internet service. They are Internet game, VoIP, Video streaming, Web browsing, FTP. These traffic models are defined as that aspect of network performance optimization of IEEE 802.16m network. Therefore, the traffic generated by a service should be accurately modeled in order to find out the performance of the system.

In the following section, we will explain the detailed traffic model. In addition, the interaction of the generated traffic with the higher layer protocol stack suck as TCP is not included here. However, we will provide reference to document which provide the detailed TCP transport layer implementation and its interaction with the various traffic models.

4.2.1 Internet Game Traffic Model[3][6]

Internet Game is computer games where multiple players simultaneously participate in a game session over Internet. In general, Internet Games can be classified into first-person-shooter (FPS) games (e.g., Counter-strike, Half-life, and Unreal Tournament), real-time strategy games (e.g., StarCraft and Age of Empire) and massive multi-player role-playing games (e.g, Lineage II, World of Warcraft).
Most of Internet Games are implemented using a client-server architecture. Also the most important requirement of a Internet Game implementation is to provide the game players with a consistent view of the game environment in a timely manner. Achieving so in a distributed environment involves the transmission of state update messages between the game clients and the game server. Therefore, on-line gaming requires low-latency point-to-point communication as well as directed broadcast channels to facilitate its real-time game logic.
Additionally different types of games may have different quality-of-service (QoS) demand on the underlying network. Among them, FPS games often have the most stringent requirements on network delay and loss ratio.
To analyze Internet Game Traffic, it is needed to classify games according to game type and analyze the traffic behavior in term of packet size distribution, packet inter-arrival time distribution and bandwidth usage.

4.2.1.1 Internet Game 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.

4.2.1.2 Internet Game traffic model parameters

4.2.1.2.1 MMORPG (e.g. World of Warcraft) Traffic Model

Table 3 MMORPG traffic model

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	
Packet inter-arrival time	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln(\ \ln Y), Y \quad U(0,1)$
Packet size	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln(\ \ln Y) \quad 2, Y \quad U(0,1)$ Addition of 2 in the equation is due to 2bytes of UDP header size after header compression

4.2.1.2.2 Online Strategic Simulation (e.g. Starcraft) Traffic Model

Table 4 online stratege game traffic model

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	
Packet inter-arrival time	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln(\ \ln Y), Y \quad U(0,1)$
Packet size	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln(\ \ln Y) \quad 2, Y \quad U(0,1)$ Addition of 2 in the equation is due to 2bytes of UDP header size after header compression

4.2.1.2.3 FPS (e.g. Counter Strike) Traffic Model

Table 5 FPS traffic model

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	
Packet inter-arrival time	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b \ln(\ \ln Y), Y \quad U(0,1)$

Packet size	Extreme	Extreme	TBD	TBD	$f(x) \quad \frac{1}{b} e^{\frac{x \quad a}{b}} e^{e^{\frac{x \quad a}{b}}}, b \quad 0$ $X \quad a \quad b\ln(\quad \ln Y) \quad 2, Y \quad U(0,1)$ Addition of 2 in the equation is due to 2bytes of UDP header size after header compression

4.2.2 VoIP Traffic Model[3][6][7]

VoIP refers to real-time delivery of packet voice across networks using the Internet protocols. A VoIP session is defined as the entire user call time and VoIP session occurs during the whole simulation period. There is a variety of encoding schemes for voice (i.e., G.711, G.722, G.722.1, G.723.1, G.728, G.729, and AMR) that result in different bandwidth requirements. Including the protocol overhead, it is very common for a VoIP call to require between 5 Kbps and 64 Kbps of bi-directional bandwidth.

4.2.2.1 VoIP traffic model characteristics

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

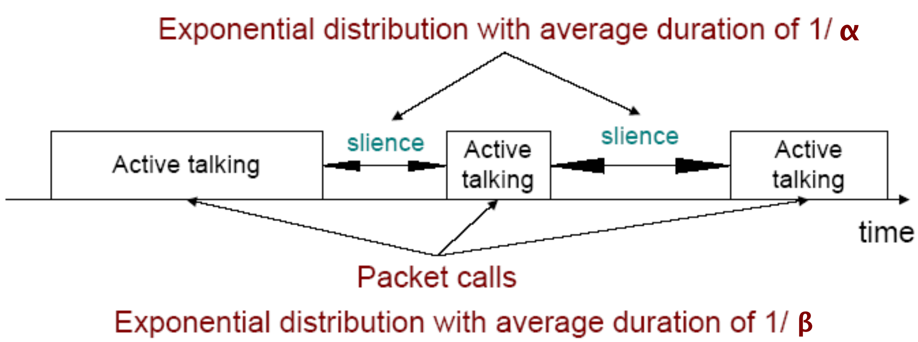

Fig. 8 Typical phone conversation profile

A two state Markov process (active-inactive) is used to model a VoIP source in Figure 7. The alternating periods of activity and silence are exponentially distributed with average durations of 1/β and 1/α respectively.

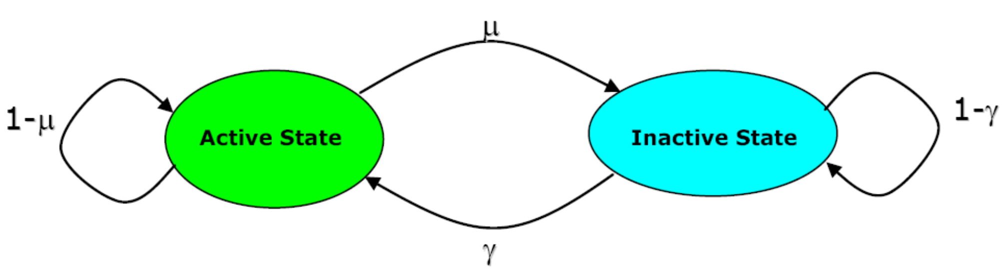

Fig. 9 Markov chain model of a VoIP source

Hence, the fraction of time the voice source is active is α/(α+β). For a voice activity factor of 40%, 1/β = 1s and $1/\alpha$ = 1.5s. 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. provides information on some common vocoders.

Table 6 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 , a simplified AMR (adaptive multi-rate) audio data compression can be used to simplify the VoIP modeling process.

The Adaptive multi Rate (AMR) codec is the most important vocoder in wireless applications being the newest vocoder for the existing GSM networks and it has been adopted as a mandatory speech codec speech processing function in UMTS.
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. shows the VoIP packet size calculation for simplified AMR with or without header compression when using IPv4 or IPv6.

Table 7 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 / 60bytes	2 bytes / 4 bytes	40 bytes / 60 bytes	2 bytes / 4 bytes
RTP	12 bytes		12 bytes	
UDP	8 bytes		8 bytes	
IPv4 / IPv6	20 bytes / 40 bytes		20 bytes / 40 bytes	
802.16 Generic MAC Header	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)

4.2.2.2 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. shows parameters associated with the VoIP traffic model.

Table 8 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

4.2.3 Video Streaming Traffic Model[3][6]

This section describes a model for streaming video traffic for DL direction. Figure 8 describes the steady state of video streaming traffic from the network, as seen by the BS. Latency at call startup is not considered in this steady-state model.

A video streaming session is defined as the entire video 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. Each frame is decomposed into a fixed number of slices, each transmitted as a single packet.

The parameter T_B is the length (in seconds) of de-jitter buffer window in the MS, and the MS uses a de-jitter buffer window to guarantee a continuous display of video streaming data. This parameter is not relevant for generating the traffic distribution, but it is useful for identifying periods when the real-time constraint of this service is not met. At the beginning of the simulation, the MS de-jitter buffer shall be full with source video 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 mobile station can record the length of time, if any, during which the de-jitter buffer runs dry. The de-jitter buffer window for the video streaming service is 5 seconds.

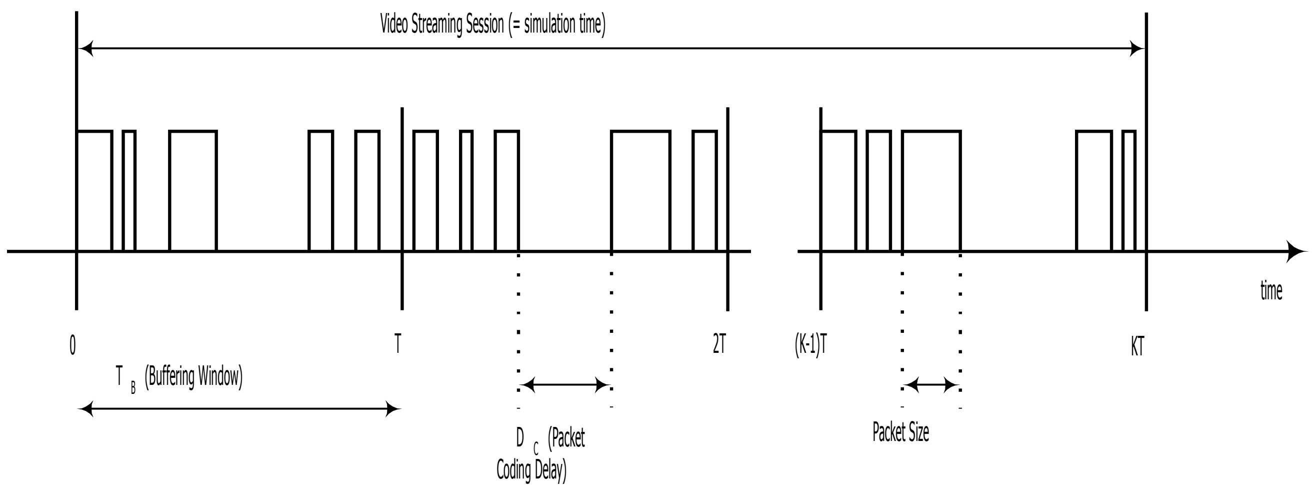

Fig. 10 Video Streaming Traffic Model

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

Table 9 Video Streaming 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= 250bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution Parameters	100ms	8	K = 40 bytes = 1.2	K = 2.5ms = 1.2

Only system-level simulations with homogenous traffic mixes are to be conducted. That is, for a particular simulation, all users will either have all FTP traffic, all HTTP traffic, or all NRTV traffic. There is no mixing of different traffic types within a single simulation.

4.2.4 Web Browsing (HTTP) Traffic Model[3][6]

4.2.4.1 HTTP traffic model characteristics

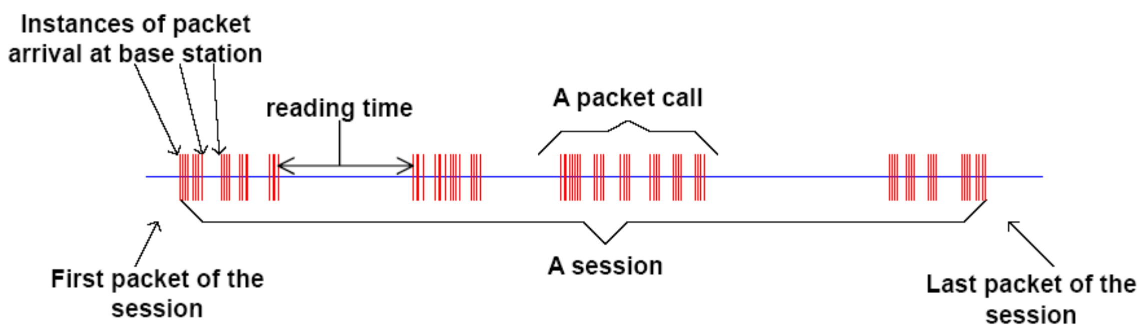

Fig. 11 Packet Trace of a Typical Web Browsing session

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 , 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. 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. 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".

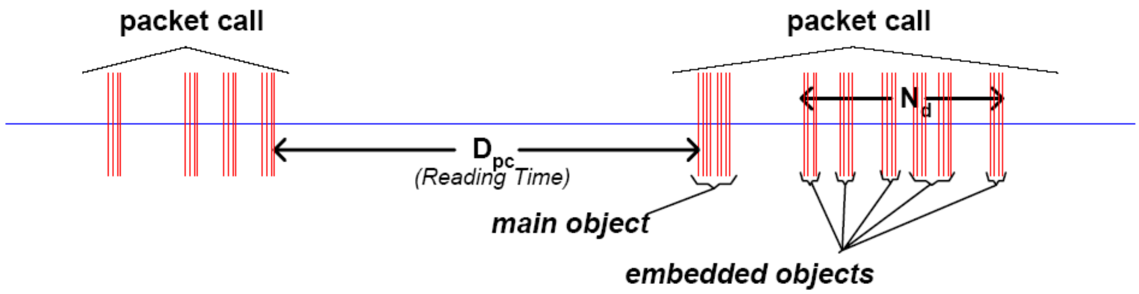

Fig. 12Contents in a Packet Call

4.2.4.2 HTTP traffic model parameters

The parameters for web browsing traffic are as follows:
SM: Size of the main object in a packet call;
SE: Size of an embedded object in a packet call;
Nd: Number of embedded objects in a packet call;
Dpc: Reading time;
Tp: Parsing time for main page

The packet traffic characteristics within a packet call will depend on the version of HTTP used by the web servers and browsers. Currently two versions of the protocol, HTTP/1.0 and HTTP/1.1, are widely used by the servers and browsers. These two versions differ in how the transport layer TCP connections are used for the transfer of the main and the embedded objects as described below.

Table 10 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	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 7.69$

4.2.5 FTP Traffic Model[5][6]

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 parameters for the FTP application session are described in .

Table 11 FTP traffic model parameters

Component	Distribution	Parameters
File size (S)	Truncated Lognormal	Mean = 2 Mbytes Std. Dev. = 0.7222 Mbytes Maximum = 5 Mbytes
Reading time (Dpc)	Exponential	Mean = 180s

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

References

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