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CONTENTS

1 Introduction 1
1.1 Study Objective and Scope 1
1.2 Simulation Description Overview 1
2 Evaluation Methodology for the Forward Link 1
2.1 System Level Setup 1
2.1.1 Antenna Pattern 1
2.1.2 System Level Assumptions 2
2.1.3 Dynamical Simulation of the Forward Link Overhead Channels 7
2.1.4 Reverse Link Modeling in Forward Link System Simulation 8
2.1.5 Signaling Errors 8
2.1.6 Fairness Criteria 9
2.1.6.1 A Generic Proportional Fair Scheduler 11
2.1.7 C/I Predictor Model for System Simulation 11
2.2 Link Level Modeling 12
2.2.1 Channel Models 14
2.3 Simulation Flow and Output Matrices 16
2.3.1 Simulation Flow for the Center Cell Method 16
2.3.2 Simulation Flow for the Iteration Method 19
2.3.3 Output Matrices 21
2.3.3.1 General output matrices 22
2.3.3.2 Data Services and Related Output Matrices 23
2.3.3.3 1xEV-DV Systems Only 24
2.3.3.3.1 Voice Services and Related Output Matrices 24
2.3.3.3.2 Mixed Voice and Data Services 25
2.3.3.4 Mixed Rev. 0 and Rev. A Mobiles (1xEV-DO Systems Only) 25
3 Evaluation Methodology for the Reverse Link 26
3.1 System Level Setup 26
3.1.1 Antenna Pattern 26
3.1.2 System Level Assumptions 26
3.1.3 Call Setup Model 31
3.1.4 Packet Scheduler 32
3.1.5 Backhaul Overhead Modeling in Reverse Link System Simulations 32

CONTENTS

3.1.6 Simulation of Forward Link Overheads for Reverse Link System Simulation 33
3.1.6.1 Static Modeling Method......33
3.1.6.2 Dynamic Modeling Method......34
3.1.6.3 Quantification of Forward Link Overhead as a Data Rate Cost (1xEV-DV Systems Only)......35
3.1.7 Signaling Errors 35
3.1.8 Fairness Criteria......36
3.1.9 FER Criterion 36
3.2 Link Level Modeling 36
3.2.1 Link Level Parameters and Assumptions 36
3.2.1.1 Frame Erasures 36
3.2.1.2 Target FER......38
3.2.1.3 Channel Models 38
3.2.2 Forward Link Loading......39
3.2.3 Reverse Link Power Control 40
3.3 Simulation Requirements......41
3.3.1 Simulation Flow 41
3.3.1.1 Soft and Softer Handoff......41
3.3.1.2 Simulation Description 41
3.3.2 Outputs and Performance Metrics 43
3.3.2.1 General Output Matrices 43
3.3.2.2 Data Services and Related Output Matrices 44
3.3.2.3 1xEV-DV Systems Only......45
3.3.2.3.1 Voice Services and Related Output Matrices......45
3.3.2.3.2 Mixed Voice and Data Services......46
3.3.2.4 Mixed Rev. 0 and Rev. A Mobiles (1xEV-DO Systems Only) 46
3.3.2.5 Link Level Output 47
3.4 Calibration Requirements 47
3.4.1 Link Level Calibration......47
3.4.2 System Level Calibration 47
3.4.2.1 1xEV-DV System Calibration 47
3.4.2.2 1xEV-DO System Calibration 48

CONTENTS

3.5 1xEV-DO Baseline Simulation Procedures 48
3.5.1 Access Terminal Requirements and Procedures: 48
3.5.2 Access Network Requirements and Procedures: 49
3.5.3 Simulation Procedures 49
4 Traffic Service Models 50
4.1 Forward Link Services 50
4.1.1 Service Mix (1xEV-DV Systems Only) 50
4.1.2 TCP Model 50
4.1.3 HTTP Model 56
4.1.3.1 HTTP Traffic Model Characteristics 56
4.1.3.2 HTTP Traffic Model Parameters 58
4.1.3.2.1 Packet Arrival Model for HTTP/1.0-Burst Mode 60
4.1.3.2.2 Packet Arrival Model for HTTP/1.1-Persistent Mode 62
4.1.4 FTP Model 65
4.1.4.1 FTP Traffic Model Characteristics 65
4.1.4.2 FTP Traffic Model Parameters 65
4.1.5 WAP Model 67
4.1.6 Near Real Time Video Model 68
4.1.7 Voice Model (1xEV-DV Systems Only) 70
4.1.8 Delay Criteria 70
4.1.8.1 Performance Criteria for Near Real Time Video 71
4.1.8.2 Delay Criterion for WAP Users 71
4.2 Reverse Link Services 71
4.2.1 Service Mix (1xEV-DV Systems Only) 71
4.2.1.1 Data Model 71
4.2.1.2 Traffic Model 72
4.2.2 TCP Modeling 73
4.2.3 FTP Upload / Email 78
4.2.4 HTTP Model 79
4.2.4.1 HTTP Traffic Model Parameters 80
4.2.4.2 Packet Arrival Model for HTTP 82
4.2.4.3 Forward Link Delay Model for HTTP Users 84

CONTENTS

4.2.5 WAP Users 85
4.2.6 Reverse Link Delay Criteria for HTTP/WAP 87
4.2.7 Mobile Network Gaming Model 88
4.2.8 Voice Model (1xEV-DV Systems Only) 89
5 References 89
Appendix A: Lognormal description 92
Appendix B: Antenna Orientation 93
Appendix C: Definition of System Outage and Voice Capacity 95
Appendix D: Formula to define various throughput and Delay Definitions 96
Appendix E: Link budget 99
Appendix F: Quasi-static Method for Link Frame Erasures Generation and Dynamically Simulated Forward Link Overhead Channels 106
Appendix G: Equalization 117
Appendix H: Max-Log-Map Turbo Decoder Metric 120
Appendix I: 19 Cell Wrap-Around Implementation 122
Appendix J: Link Level Simulation Parameters 126
Appendix K: Joint Technical Committee (JTC) Fader 129
Appendix L: Largest Extreme Value Distribution 133
Appendix M: Reverse Link Output Matrices 134
M.1 Output Matrix 134
M.1.1 1xEV-DV Systems 134
M.1.2 1xEV-DO Systems 141
M.2 Definitions 146
Appendix N: Link Prediction Methodology for Uplink System Simulations 149
N.1 Definition of Required Terms 152
N.1.1 Channel Estimation SNR, $\Gamma_{i,p}$ 152
N.1.2 Non-Gaussian Penalty, γ_{NG} 153
N.1.3 Reference Curves 153
Appendix O: Reverse Link Hybrid ARQ: Link Error Prediction Methodology Based on Convex Metric 155
O.1 Convex Metric based on Channel Capacity Formula 155
O.2 Equivalent SNR Method based on Convex Metric (ECM) 158

CONTENTS

O.2.1 Overview of the Procedure 158
O.2.2 Detailed Procedure 159
O.2.3 Combining Procedure for H-ARQ 161
Appendix P: Pilot SINR Estimation For Power-Control Command Update in Link-Level Simulations 162
Appendix Q: 1xEV-DV Reverse Link Simulation and Scheduler Procedures 163
Q.1.1 Mobile station Requirements and Procedures 163
Q.1.2 Base Station Requirements and Procedures 166
Q.1.3 Scheduler Requirements and Procedures 168
Q.1.3.1 Scheduling, Rate Assignment and Transmission Timeline 169
Q.1.3.2 Scheduler Description and Procedures 171
Q.2 Baseline specific simulation parameters 174
Appendix R: Modeling of D_RL(request) and D_FL(Assign) 177

No text.

FIGURES

Figure 2.1.1-1 Antenna Pattern for 3-Sector Cells 2

Figure 2.3.1-1 Simulation Flow Chart 19

Figure 3.1.3-1: Simplified Call Setup Timeline for 1xEV-DV. (Timeline for 1xEV-DO is the same by modifying 320 ms to 427 ms) 32

Figure 4.1.2-1 Control Segments in TCP Connection Set-up and Release.......... 51

Figure 4.1.2-2 TCP Flow Control During Slow-Start; τ_l = Transmission Time over the Access Link; τ_{rt} = Roundtrip Time 52

Figure 4.1.2-3 Packet Arrival Process at the Base Station for the Download of an Object Using TCP; PW = the Size of the TCP Congestion Window at the End of Transfer of the Object; T_c=τ_c (Described in Figure 4.1.2-2).......... 55

Figure 4.1.3-1 Packet Trace of a Typical Web Browsing Session 56

Figure 4.1.3-2 Contents in a Packet Call 57

Figure 4.1.3-3 A Typical Web Page and Its Content 57

Figure 4.1.3-4 Modeling a Web Page Download.......... 60

Figure 4.1.3-5 Download of an Object in HTTP/1.0-Burst Mode 62

Figure 4.1.3-6 Download of Objects in HTTP/1.1-Persistent Mode.......... 64

Figure 4.1.4-1 Packet Trace in a Typical FTP Session 65

Figure 4.1.4-2 Model for FTP Traffic 67

Figure 4.1.5-1 Packet Trace for the WAP Traffic Model.......... 68

Figure 4.1.6-1 Video Streaming Traffic Model 69

Figure 4.2.2-1: Modeling of TCP three-way handshake 73

Figure 4.2.2-2: TCP Flow Control During Slow-Start; τ_l = Transmission Time over the Access Link (RL); τ_{rt} = Roundtrip Time.......... 75

Figure 4.2.2-3 Packet Arrival Process at the mobile Station for the Upload of a File Using TCP.......... 78

Figure 4.2.4-1: Example of events occurring during web browsing.......... 80

Figure 4.2.5-1: Packet Trace for the WAP Traffic Model.......... 86

Figure B-1 Center Cell Antenna Bearing Orientation diagram.......... 93

Figure B-2 Orientation of the Center Cell Hexagon 93

Figure B-3 Mobile Bearing orientation diagram example.......... 94

Figure D-1: Description of arrival and delivered time for a packet and a packet call.......... 98

Figure F-1 Flowchart for QPSK modulation 110

Figure F-2 Prediction methodology for higher order modulations without pure Chase combining.......... 111

FIGURES

Figure F-3 Prediction methodology for higher order modulations with pure Chase combining (this corresponds to Case 1) 112

Figure F-4 Obtaining sample values of $\Sigma E_s / N_t$ and indicators of packet errors............... 113

Figure F-5 Determining the Doppler penalty by evaluating the predictor performance 114

Figure H-1 QAM Receiver Block Diagram... 121

Figure I-1 Wrap-around with '9' sets of 19 cells showing the toroidal nature of the wrap-around surface.. 123

Figure I-2: The antenna orientations to be used in the wrap-around simulation. The arrows in the Figure show the directions that the antennas are pointing. 125

Figure K-1: I and Q Fade Multiplier Generation 129

Figure N-1: Outline of Equivalent SNR Method. 149

Figure O-1. Approximate channel capacities at BPSK and QPSK signaling (Gaussian signaling case is plotted as a reference)... 157

Figure Q-1: Set point adjustment due to rate transitions on R-SCH............................... 168

Figure Q-2 Scheduling Delay Timing 169

Figure Q-3: Parameters associated in mobile station scheduling on RL.......................... 170

Figure R-1: PDF of FL transmission delays of ESCAM on F-PDCH 180

Figure R-2: ESCAM delays on F-PDCH 181

TABLES

Table 2.1.2-1 Forward System Level Simulation Parameters 3
Table 2.1.2-2 Details of Self-Interference Values Resulting in 17.8 dB of Maximum C/I 6
Table 2.1.5-1 Signaling Errors 9
Table 2.1.6-1 Criterion CDF 10
Table 2.1.6-2 Web Browsing Model Parameters 11
Table 2.2.1-1 Channel Models 14
Table 2.2.1-2 Fractional Recovered Power and Fractional UnRecovered Power 14
Table 2.3.3-1 Required 1xEV-DV Simulation Evaluation Comparison Cases Table 22
Table 3.1.2-1 Reverse Link System Level Simulation Parameters 27
Table 3.1.5-1 Backhaul bandwidth used by signaling and measurement messages 33
Table 3.1.8-1 CDF Criterion for FTP Upload MS 36
Table 3.4.2.2-1 Default 1xEV-DO RL MAC Transition Probabilities 48
Table 4.1.3.2-1 HTTP Traffic Model Parameters 59
Table 4.1.4.2-1 FTP Traffic Model Parameters 66
Table 4.1.5-1 WAP Traffic Model Parameters 68
Table 4.1.6-1 Video Streaming Traffic Model Parameters 70
Table 4.2.1.2-1: Traffic Configurations 72
Table 4.2.2-1 Delay components in the TCP model for the RL upload traffic 76
Table 4.2.3-1: FTP Characteristics 79
Table 4.2.4.1-1: HTTP Traffic Model Parameters 82
Table 4.2.4.3-1 Points to obtain the average transmission rate (ATR) given the geometry and channel model of a user 85
Table 4.2.5-1: WAP Traffic Model Parameters 87
Table 4.2.6-1 Reverse link delay criteria for HTTP request 87
Table 4.2.7-1 Mobile network gaming traffic model parameters 88
Table E-1 Link-Budget Template for the Reverse Link 100
Table E-2 Link-Budget Template for the Forward Link 102
Table E-3 Propagation Index and Log-Normal Sigma Values from [20] 105
Table J-1 Link Level Simulation Parameters for Forward Link 126
Table J-2 Link Level Simulation Parameters for Reverse Link 127
Table K-1: Coefficients of the 6-tap FIR Filter 129
Table K-2: Coefficients of the 8-tap FIR Filter 129

TABLES

Table K-3: Coefficients of the 11-tap Filter 130
Table K-4: Coefficients of the 28-tap Filter 130
Table K-5: Jakes (Classic) Spectrum IIR Filter Coefficients 131
Table M-1 Required statistics output in excel spread sheet for the base station side 134
Table M-2 Required statistics output in excel spread sheet for the base station side 140
Table M-3 Required statistics output in excel spread sheet for the base station side 141
Table N-1. Notations used. 150
Table R-1: D_RL(request) delay for Method a 177
Table R-2: D_RL(request) delay for Method b 178
Table R-3: D_FL(assign) delay for Method a 178
Table R-4: D_FL(assign) delay for Method b (excluding F-PDCH scheduling delay) 179
Table R-5: Reference table for mean transmission times vs Geometry 181

1 INTRODUCTION

1.1 Study Objective and Scope

The objective of this document is to explain the set of definitions, assumptions, and a general framework for simulating cdma2000®[1] systems (e.g., 1xEV-DV and 1xEV-DO) to arrive at system wide voice, data, or both voice and data performance on the forward and reverse links.

Revision 0 of this document was used in the evaluation and analysis leading to the development of the following cdma2000 system specifications: cdma2000 Revision C (1xEV-DV), cdma2000 Revision D (1xEV-DV), and cdma2000 High Rate Packet Data Air Interface Revision A (1xEV-DO).

This document also defines the necessary framework for simulating the performance of a cdma2000 system with proposed enhancements that are not part of the current cdma2000 family of specifications. The proponent(s) of any proposal shall provide the details required so that other companies can evaluate the proposal independently. The proponent(s) of any simulation results shall provide the details required so that other companies can repeat the simulation independently. The information about the simulations will include the predictors being used, and the reported results will include the prediction errors (bias and standard deviation).

1.2 Simulation Description Overview

Determining voice and high rate packet data system performance requires a dynamic system simulation tool to accurately model feedback loops, signal latency, protocol execution, and random packet arrival in a multipath-fading environment. The packet system simulation tool will include Rayleigh and Rician fading and evolve in time with discrete steps (e.g., time steps of 1.25 ms or 1.67 ms). The time steps need to be small enough to correctly model feedback loops, latencies, scheduling activities, and measurements of the proposed system.

2 EVALUATION METHODOLOGY FOR THE FORWARD LINK

2.1 System Level Setup

2.1.1 Antenna Pattern

The antenna pattern used for each sector, reverse link and forward link, is plotted in Figure 2.1.1-1 and is specified by

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[1] cdma2000® is the trademark for the technical nomenclature for certain specifications and standards of the Organizational Partners (OPs) of 3GPP2. Geographically (and as of the date of publication), cdma2000® is a registered trademark of the Telecommunications Industry Association (TIA-USA) in the United States.

$$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3dB}}\right)^2, \ A_m\right], \quad \text{where} \quad -180 \le \theta \le 180. \qquad (2.1.1\text{-}1)$$

θ_{3dB} is the 3 dB beamwidth, and $A_m = 20dB$ is the maximum attenuation.

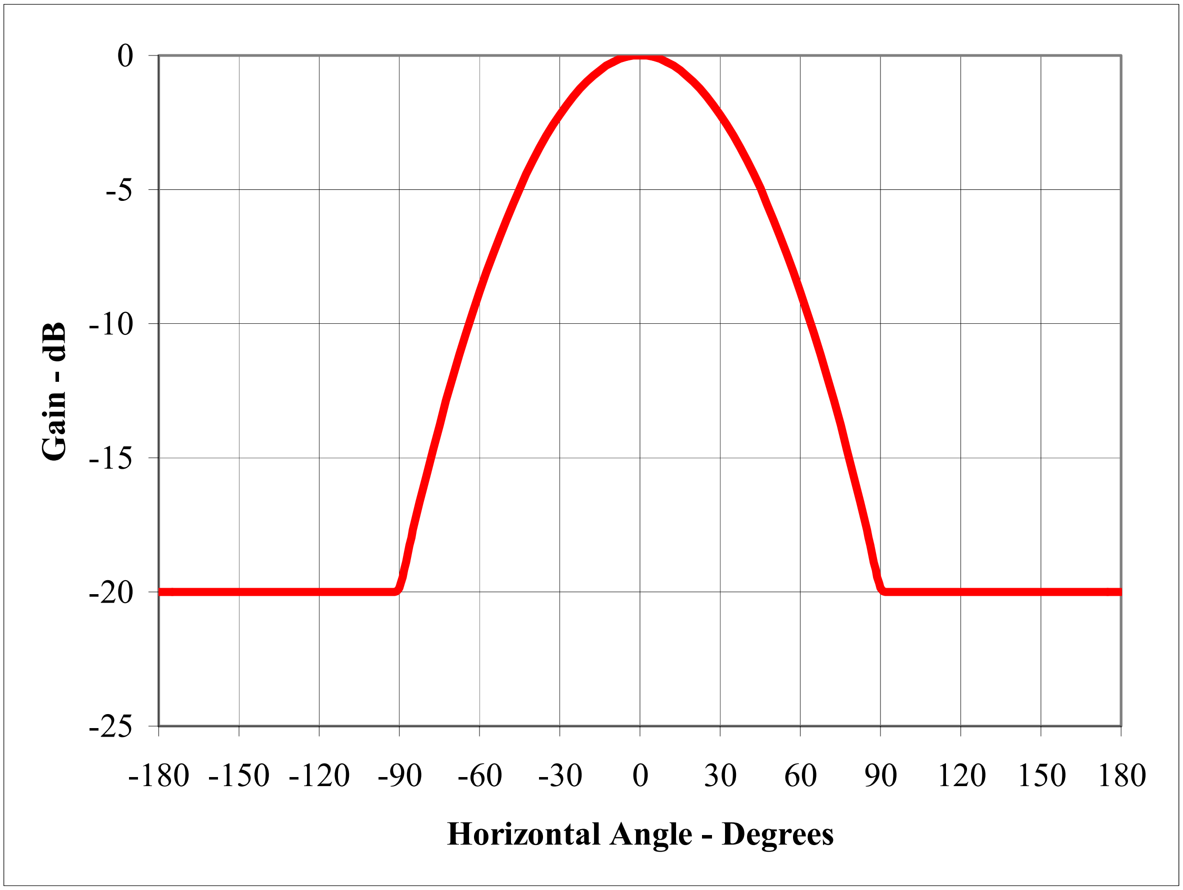

Figure 2.1.1-1 Antenna Pattern for 3-Sector Cells

2.1.2 System Level Assumptions

The parameters used in the simulation are listed in Table 2.1.2-1. Where values are not shown, the values and assumptions used shall be specified in the simulation description.

Table 2.1.2-1 Forward System Level Simulation Parameters

Parameter	Value	Comments
Number of Cells (3 sectored)	19	2 rings, 3-sector system, 57 sectors.
Antenna Horizontal Pattern	70 deg (-3 dB) with 20 dB front-to-back ratio	See section 2.1.1
Antenna Orientation	0 degree horizontal azimuth is East (main lobe)	No loss is assumed on the vertical azimuth. (See Appendix B)
Propagation Model (BTS Ant Ht=32m, MS=1.5m)	28.6+ 35log10(d) dB, d in meters	Modified Hata Urban Prop. Model @1.9GHz (COST 231). Minimum of 35 meters separation between MS and BS.[2]
Log-Normal Shadowing	Standard Deviation = 8.9 dB	Independently generate lognormal per mobile and use the method described in Appendix A. This shadowing is constant for each MS in each simulation run. The same shadowing amount shall be used for all the sector antennas of a BS to a given MS. The correlation coefficient between the BS's Tx antennas and a given MS and the BS's RX antennas and a given MS is 1.
Base Station Shadowing Correlation	0.5	See Appendix A

[2] In this document the word "modified" represents a difference from the COST231-Hata model wherein the path loss has been reduced by 3 dB [33]. If a mobile is dropped within 35 meters of a base station, it shall be redropped until it is outside the 35-meter circle.

Parameter		Value	Comments
Forward Link Overhead Channel Resource Consumption	Circuit switched and packet switched data systems (e.g., 1xEV-DV)	Pilot, Paging and Sync overhead: 20%.	Any additional overhead needed to support other control channels (dedicated or common) must be specified and accounted for in the simulation
	Packet switched data systems (e.g., 1xEV-DO)	FL MAC, Preamble and Pilot channel overhead shall be considered. The portion of times the Control Channel (CC) (38.4 kbps or 76.8 kbps) is sent shall be set as a fixed TDM overhead.	CC portion is assumed to be 6.25% of the total time. Any additional overhead must be specified and accounted for in the simulation.
Mobile Noise Figure		10.0 dB	
Thermal Noise Density		-174 dBm/Hz	
Carrier Frequency		2 GHz	
BS Antenna Gain with Cable Loss		15 dB	17 dB BS antenna gain; 2 dB cable loss
MS Antenna Gain		-1 dBi	
Other Losses		10 dB	Applicable to all fading models
Fast Fading Model		Based on Speed	See Table 2.2.1-1. The fading model is specified in Appendix K.
Active Set Membership	Circuit switched and packet switched data systems (e.g., 1xEV-DV)		Up to 3 members are in the Active Set if the pilot Ec/Io is larger than T_ADD = -18 dB (=9 dB below the FL pilot Ec/Ior) based on the FL evaluation methodology
	Packet switched data systems (e.g., 1xEV-DO)		Up to 3 members are in the Active Set if the pilot Ec/Io is larger than T_ADD = -9 dB based on the FL evaluation methodology
Delay Spread Model			See Table 2.2.1-1 and Table 2.2.1-2

<table>
<tr><th colspan="2">Parameter</th><th>Value</th><th>Comments</th></tr>
<tr><td colspan="2">Fast Cell Site Selection</td><td></td><td>Disable. The overhead shall be accounted for if it is used in the proposal.</td></tr>
<tr><td rowspan="2">Forward Link Power Control</td><td>Circuit switched and packet switched data systems (e.g., 1xEV-DV)
(If used on dedicated channel)</td><td>Power Control loop delay: two PCGs[3]</td><td>Update Rate: Up to 800Hz
PC BER: 4%</td></tr>
<tr><td>Packet switched data systems (e.g., 1xEV-DO)
(If used on MAC channels)</td><td>Power Control loop delay: two slots[3]</td><td>Update Rate: Up to 600 Hz
PC BER: 4%,
Based on DRC feedback</td></tr>
<tr><td colspan="2">BS Maximum PA Power</td><td>20 Watts</td><td></td></tr>
<tr><td colspan="2">Site to Site distance</td><td>2.5 km</td><td></td></tr>
<tr><td colspan="2">Maximum C/I achievable, where C is the instantaneous total received signal from the serving base station(s) (usually also referred to as $rx\_I_{or}(t)$, or $\hat{I}_{or}(t)$), and I is the instantaneous total interference level (usually also referred to as $N_t(t)$).</td><td>13 dB and 17.8 dB</td><td>13 dB for typical current subscriber designs for IS-95 and cdma2000 1x systems; 17.8 dB for improved subscriber designs for 1xEV-DV and 1xEV-DO systems. The details on how the 17.8 dB value is derived are in Table 2.1.2-2.</td></tr>
</table>

[3] One PCG/slot delay in link level modeling (measured from the time that the SIR is sampled to the time that the BS changes TX power level.)

Table 2.1.2-2 Details of Self-Interference Values Resulting in 17.8 dB of Maximum C/I

Contribution of Self-Interference	$\hat{I}_{or} / \hat{I}_{self}^{(i)}$	**Note**
Base-band pulse shaping waveform	24 dB	IS-95 Tx filter with 64-tap Rx filter
Radio noise floor	20 dB	For Tx RHO increased to 99%
ADC quantization noise	31.9 dB	6-bit A/D converter
Adjacent channel interference	27 dB	1.25 MHz spacing

The maximum C/I achievable in the subscriber receiver is limited by several sources, including inter-chip interference induced by the base-band pulse shaping waveform, the radio noise floor, ADC quantization error, and adjacent carrier interference.

In the system level simulation, the noise floor associated with the maximum C/I limitation can be characterized by the parameter α , given by

$$\alpha = \frac{1}{(C/I)_{\max}} , \tag{2.1.2-1}$$

where $(C/I)_{\max}$ denotes the maximum achievable C/I for the subscriber receiver. As indicated in Table 2.1.2-1, $(C/I)_{\max}$ is assumed to be 13 dB for the current IS-95 and cdma2000 1X subscriber receivers, and 17.8 dB for improved 1xEV-DV/1xEV-DO designs. Thus, α = 0.05 and α = 0.0166 for maximum C/I values of 13 dB and 17.8 dB, respectively.

In the system level simulation, the effective C/I shall be given by

$$(C/I)_{\text{effective}} = \frac{1}{\dfrac{1}{(C/I)_{\text{combined}}} + \alpha} , \tag{2.1.2-2}$$

where $(C/I)_{\text{combined}}$ is the instantaneous signal-to-interference ratio after pilot-weighted combining of the Rake fingers (see section 2.2.1 for detail). The effective signal-to-interference ratio, $(C/I)_{\text{effective}}$, accounts for the interference sources associated with the maximum C/I limitation, and shall be used as the C/I observed by the mobile station receiver.

The channel between the serving cell and the subscriber is modeled using the channel models defined in section 2.2.1. The channel between any interfering cell and the subscriber is modeled as a one-path Rayleigh fading channel, where the Doppler of the fading process is randomly chosen based on the velocities specified in Table 2.2.1-1 and its corresponding probabilities.

If transmit diversity is used (e.g., in cdma2000 1x and 1xEV-DV systems), the transmit diversity PA size shall be the same as the main PA size, 12.5% of the main PA power shall be used for the Pilot Channel and 7.5% for the Paging Channel and Sync Channel. The Transmit Diversity Pilot Channel power is half the power of the Pilot Channel. For example, if the main PA size is 20 W, then the transmit diversity PA size is 20 W, 2.5 W of the main PA is for the Pilot Channel, 1.5 W of the main PA for the Paging Channel and Sync Channel, and 1.25 W of the transmit diversity PA is for the Transmit Diversity Pilot Channel.

2.1.3 Dynamical Simulation of the Forward Link Overhead Channels

Dynamically simulating the overhead channels for 1xEV-DV or 1xEV-DO systems is essential to capture the dynamic nature of power and code space allocation to these channels. The simulations shall be done as follows:

1) The performance of the new overhead channels (other than the Pilot, Sync, and Paging Channels for 1xEV-DV systems or the Pilot and control channels for 1xEV-DO systems) must be included in the system level simulations. The Pilot Channel, Sync Channel, and Paging Channel are taken into account as part of the fixed overhead (power and code space) in 1xEV-DV systems. For 1xEV-DO systems, the Pilot, preamble, and the total FL MAC shall be transmitted at full BTS power (20 W), and the 38.4 kbps and 76.8 kbps Control Channels are taken into account as part of the fixed overhead (as a fixed percentage of the total transmission time).

2) There are two types of these new overhead channels: static and dynamic. A static overhead channel requires fixed base station power. A dynamic overhead channel requires dynamic base station power.

3) The system level simulations do not directly include the coding and decoding of these new overhead channels. There are two aspects that are important for the system level simulation: the required Ec/Ior during the simulation interval (e.g., a power control group or slot) and demodulation performance (detection, miss, and error probability — whatever is appropriate).

4) The link level performance is evaluated off-line by using separate link-level simulations. A quasi-static approach shall be used to conduct the link-level simulation. The performance is characterized by curves of detection, miss, false alarm, and error probability (whatever is appropriate) versus Eb/No.

5) For static overhead channels, the system simulation should compute the received Eb/No.

6) For dynamic overhead channels with open-loop control only, the simulations should take into account the estimate of the required forward link power that needed to be transmitted to the mobile station. For dynamic overhead channels that use closed loop feedback, the base station allocates forward link power based upon the combination of open-loop and closed-loop feedback. During the reception of overhead information, the system simulation should compute the received Eb/No.

7) Once the received Eb/No is obtained, then the various miss error events should be determined. The impact of these events should then be modeled. The false alarm

events are evaluated in link-level simulation, and the simulation results will be included in the evaluation report. The impact of false alarm, such as delay increases and throughput reductions for both the forward and reverse links, will be appropriately taken into account in system-level simulation.

8) The Walsh space utilization shall be modeling dynamically for 1xEV-DV systems.

9) All new overhead channels shall be modeled.

10) If a proposal adds messages to an existing channel (overhead or otherwise), the proponent shall justify that this can be done without creating undue loading on this channel. If a proposal requires an additional overhead channel of the type that is already in the system under evaluation, then the proposal shall include the power required for this channel. The system level and link level simulation required for this modified overhead channel as a result of the new messages shall be performed according to 3) and 4), respectively.

2.1.4 Reverse Link Modeling in Forward Link System Simulation

The proponents shall only model feedback errors (e.g., power control, acknowledgements, rate indication, etc.) and measurements (e.g., C/I measurement) without explicitly modeling the reverse link and reverse link channels. In addition to supplying the feedback error rate average and distribution, the measurement error model and selected parameters, the estimated power level required for the physical reverse link channels will be supplied (including those used for fast cell selection even though it is not going to be explicitly modeled for the 1xEV-DV or 1xEV-DO system simulations).

2.1.5 Signaling Errors

Signaling errors shall be modeled and specified as in Table 2.1.5-1.

Table 2.1.5-1 Signaling Errors

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

Proponents shall quantify and justify the signaling errors and their impacts in the evaluation report. As an example, if an ACK is misinterpreted as a NACK (duplicate transmission), the packet call throughput will be scaled down by (1-p_{ACK}), where p_{ACK} is the ACK error probability.

2.1.6 Fairness Criteria

Because maximum system capacity may be obtained by providing low throughput to some users, it is important that all mobile stations be provided with a minimal level of throughput. This is called fairness. The fairness is evaluated by determining the normalized cumulative distribution function (CDF) of the user throughput, which meets a predetermined function in two tests (seven test 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 proponent(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

$$\widetilde{T}_{put}[k] = \frac{T_{put}[k]}{\underset{i}{\mathrm{avg}}\, T_{put}[i]} . \qquad (2.1.6\text{-}1)$$

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 2.1.6-1.

Table 2.1.6-1 Criterion CDF

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

This CDF shall be met for the seven test conditions given in the following two tests:

Test 1 – for FTP, six test conditions

- Single path Rayleigh fading
- 3, 30, 100 km/h
- All FTP users, with buffers always full – Note that this model differs from the FTP traffic model specified in section 4.1.4
- 10, 20 users dropped uniformly in a sector
- 80% (for cdma2000 1x and 1xEV-DV systems) or 100% (for 1xEV-DO systems) of BS power available for data users; max. BS power = 20 w
- Full BS power from other cells
- The 6 test conditions are the combinations 3, 30, and 100 km/h with 10 and 20 FTP users per sector

Test 2 – for HTTP, one test condition

- Single path Rayleigh fading
- 3 km/h
- HTTP users, with traffic model provided in Table 2.1.6-2 – Note that this traffic model differs from the HTTP traffic model specified in section 4.1.3
- 44 users dropped uniformly in a sector
- 70% (for cdma2000 1x and 1xEV-DV systems) or 100% (for 1xEV-DO systems) of BS power available for data users; max. BS power = 20 w
- Full BS power from other cells

Table 2.1.6-2 Web Browsing Model Parameters

Process	**Random Variable**	**Parameters**
Packet Calls Size	Pareto with cutoff	α=1.2, k=4.5 Kbytes, m=2 Mbytes, μ = 25 Kbytes
Time Between Packet Calls	Geometric	μ = 5 seconds

2.1.6.1 A Generic Proportional Fair Scheduler

Although the proponent of a proposal is free to use any scheduler, a generic proportional-fair scheduler [31,32], for full-buffer traffic model, with a priority function $P_i(k)$ is given below for reference:

$$P_i(k) = \frac{R_i(k)}{\left[T_i(k)\right]^{\alpha}}, \qquad 2.1.6.1\text{-}1$$

where k is the slot index, $R_i(k)$ is the data rate potentially achievable for the i-th mobile station based upon the reported C/I and the power available to the F-PDCH, $T_i(k)$ is the average "fairness throughput" of the i-th mobile station up to time k, and α is the fairness exponent factor with the default value chosen as 0.75. Users with the highest priority are selected for service. The number of users selected is dependent upon the number of users to be serviced simultaneously. The average "fairness throughput" can be calculated as follows:

$$T_i(k) = \begin{cases} \beta T_i(k-1) & \text{if the i-th MS was not scheduled at time } k-1 \\ \beta T_i(k-1) + (1-\beta) N_i(k-1) & \text{if the i-th MS was scheduled at time } k-1 \end{cases}, \qquad 2.1.6.1\text{-}2$$

where

$$\beta = \begin{cases} 1 - \dfrac{1.25\times 10^{-3}}{t} & \text{for 1x EV-DV Systems} \\ 1 - \dfrac{1.67\times 10^{-3}}{t} & \text{for 1xEV-DO Systems} \end{cases}, \qquad 2.1.6.1\text{-}3$$

t is set to 1.5s, $N_i(k\text{-}1)$ is the number of bits delivered to the MS at time k-1 and T_i should be initialized to a small value greater than zero.

2.1.7 C/I Predictor Model for System Simulation

Each company shall use their own prediction methodology and describe the prediction method in enough detail so other companies can replicate the simulations. This shall include the timing diagram from measurements at the mobile to scheduling decisions at the base station based on those measurements. Furthermore, this delay shall be explicitly modeled in the system level simulator.

2.2 Link Level Modeling

The performance characteristics of individual links used in the system simulation are generated a priori from link level simulations. Link level simulation parameters are specified in Appendix J.

Turbo Decoder Metric and Soft Value Generation into Turbo Decoder shall be as specified in Appendix H.

The quasi-static approach with fudge factors or with short term FER shall be used to generate the frame erasures for both the 1xEV-DV packet data channel and the 1xEV-DO Forward Traffic Channel (FTC), dynamically simulated forward link overhead channels, voice and SCH (applicable only to 1xEV-DV), as described below.

Brief Description of Quasi-Static Approach with Fudge Factors:

Quasi-static approach with fudge factors shall be used for 1xEV-DV Packet Data Channel, 1xEV-DO FTC, and Dynamically Simulated Forward Link Overhead Channel.

The aggregated Es/Nt is computed over a transmission period and mapped to an FER using AWGN curves. The proponent shall select one of two possible methods to determine the FER:

a) Map the aggregated Es/Nt directly to the AWGN curve corresponding to the given modulation and coding.

b) Adjust the aggregated Es/Nt for the given modulation and coding and lookup a curve obtained using a reference modulation and coding.

Furthermore the proponents shall account for an additional Es/Nt loss at higher Dopplers for either method.

Full details of the quasi-static frame error modeling with fudge factors are given in the Appendix F.

Description of Quasi-Static Approach with Short Term FER:

The quasi-static approach with short term FER may be used to generate frame erasures for voice, SCH, and F-PDCH for 1xEV-DV systems. The quasi-static approach with short term FER may be used to generate frame (i.e. physical-layer packet) erasures for the Forward Traffic Channel (FTC) for 1xEV-DO systems.

A full set of short term FER vs. average Eb/Nt per frame curves is generated as a function of radio configurations, transmission diversity schemes (if applicable), channel models, different ways of soft hand-off (SHO), different SHO imbalances, and geometries. The number of curves should be reduced if possible, provided that this won't unduly affect the validity of this quasi-static approach.

All companies shall use the same set of short term FER vs. average Eb/Nt per frame.

In the system-level simulation, the average Eb/Nt per frame is computed as follows. First, the average Eb/Nt is calculated in a PCG (slot). The short-term average Eb/Nt per frame is defined as the average of the average Eb/Nt for all N_S PCG's (slots) in a frame (physical layer packet), i.e.,

$$\frac{E_b}{N_t} = \frac{1}{N_s}\sum_{n=1}^{N_s}\left(\frac{E_b}{N_t}\right)_n , \qquad (2.2\text{-}1)$$

where $(Eb/Nt)_n$ is the average Eb/Nt in the n-th PCG (slot) in a frame (physical-layer packet). Note that finger combining and self-interference are applied before the average Eb/Nt in a PCG (slot) is calculated. Once the Eb/Nt is calculated as in the above equation, it is used to look up the corresponding link level short term FER vs. average Eb/Nt per frame curves for the specific condition (i.e., radio configuration, transmission diversity (if applicable) scheme, channel model, way of soft hand-off (SHO), SHO imbalance(s), and geometry). A frame erasure event is then generated based on the FER value.

If a short term FER vs. average Eb/Nt per frame curve is not available for a condition, the curve should be computed by interpolating those curves for similar conditions (e.g., between the factors for closest geometries available).

The short term FER vs. average Eb/Nt per frame curves shall be generated as follows:

1. The link-level simulation is conducted for a specific condition. The average Eb/Nt in a frame and the frame erasure indicator for the frame are recorded. For 1xEV-DV systems, the average Eb/Nt per frame is computed as follows in the link-level simulation

$$\frac{E_b}{N_t} = \frac{1}{16}\sum_{n=1}^{N_s}\left(\frac{m\sum_k \left(S_b^{(n,k)}\right)^2}{\sum_k \left(n_t^{(n,k)}\right)^2}\right), \qquad (2.2\text{-}2)$$

2. where n is the index of PCG in a frame and k is the index of symbols within a PCG. $S_b^{(n,k)}$ is the signal component in the k-th received coded symbol in the n-th PCG, $n_t^{(n,k)}$ is the noise and interference component in the k-th received symbol in the n-th PCG in a frame, and m is the inverse of the code rate (i.e., 4 for RC3 and 2 for RC4, etc). For 1xEV-DO systems, the average Eb/Nt per slot is computed as follows in the link-level simulation

$$(\frac{E_b}{N_t})_n = \frac{1}{M}\sum_{k=1}^{K}\sum_{l=1}^{L}\frac{E_s^{(n,k,l)}}{N_t^{(n,k,l)}} , \qquad (2.2\text{-}3)$$

   where M equals to the number of information bits per packet; n is the slot index and k is the symbol index within a slot; l is the path index, where the total number of captured paths is denoted by L; the total number of symbols per slot is K; $E_s^{(n,k,l)}$ denotes the signal energy in the k-th received symbol in the n-th slot in the physical-layer packet at the l-th RAKE finger, and $N_t^{(n,k,l)}$ denotes the noise and interference variance in the k-th received symbol in the n-th slot in the physical-layer packet seen at the l-th RAKE finger.

3. Generate the histogram of FER vs. the average Eb/Nt per frame, i.e., the range of Eb/Nt is divided into many bins, and the FER in each bin is computed based on the outputs mentioned in step 1. The size of each bin is 0.25 dB.

2.2.1 Channel Models

A channel model corresponds to a specific number of paths, path delay and power profile (ITU multi-path models), and Doppler frequencies for the paths.

Table 2.2.1-1 Channel Models

Channel Model	Multi-path Model	# of Fingers	Speed (kmph)	Fading	Assignment Probability
Model A	Pedestrian A	1	3	Jakes	0.30
Model B	Pedestrian B	3	10	Jakes	0.30
Model C	Vehicular A	2	30	Jakes	0.20
Model D	Pedestrian A	1	120	Jakes	0.10
Model E	Single path	1	0, f_D=1.5 Hz	Rician Factor K = 10 dB	0.10

The channel models are randomly assigned to the various users according to the probabilities of Table 2.2.1-1 at the beginning of each drop and are not changed for the duration of that drop. The assignment probabilities given in Table 2.2.1-1 are interpreted as the percentage of users with that channel model in each sector. The JTC fader (see Appendix K) shall be used to generate the Jakes fading samples.

The Fractional Recovered Power (FRP) and Fractional UnRecovered Power (FURP) are given in Table 2.2.1-2. FURP shall contribute to the interference of the finger demodulator outputs as an independent fader. The power on all fingers (including FURP) for each channel model shall be normalized so that the total power for that channel model adds up to unit one.

Table 2.2.1-2 Fractional Recovered Power and Fractional UnRecovered Power

Model	Finger1 (dB)	Delay	Finger2 (dB)	Delay (Tc)	Finger3 (dB)	Delay (Tc)	FURP (dB)
Ped-A	-0.06	0.0					-18.8606
Ped-B	-1.64	0.0	-7.8	1.23	-11.7	2.83	-10.9151
Veh-A	-0.9	0.0	-10.3	1.23			-10.2759

The delay values given in Table 2.2.1-2 are for information purposes and do not need to be accounted for in the system simulation.

Each channel model shall be modeled in the system level simulation as follows:

In the system level simulation, the interference due to unrecovered power shall be modeled as an additional ray that is not demodulated by the Rake receiver. Let J denote the number of rays used in a particular channel model, excluding the ray used to model FURP. The average power assigned to each of the rays is given in Table 2.2.1-2. The average power assigned to the ray used to model FURP is also given in Table 2.2.1-2. The recovered rays and the additional ray used to model the unrecovered power all fade independently of each other.

Let $\{\gamma_i\}_{i=1}^{J}$ denote the samples of the fading processes, for a particular PCG, of the J recovered rays. Let λ denote the sample of the fading process for the additional ray used to model interference due to the unrecovered power, for a particular PCG. Let $\{(C/I)_i\}_{i=1}^{J}$ denote the signal-to-interference ratio for each of the Rake fingers, which can be expressed as

$$(C/I)_i = \frac{\|\gamma_i\|^2}{G^{-1} + \|\lambda\|^2 + \sum_{1\le k\le J, k\ne i} \|\gamma_k\|^2} \ , \tag{2.2.1-1}$$

where G denotes the subscriber geometry, given by

$$G = \frac{\hat{I}_{or}}{N_0 + \sum_{n=1}^{N} I_{oc}(n)\|\rho_n\|^2} \ , \tag{2.2.1-2}$$

N is the number of interfering sectors, ρ_n is the fading process of the ray between the receiver and the n-th interfering sector for a particular PCG, N_0 is the variance of the thermal noise including the mobile station noise figure defined in Table 2.1.2-1, $\hat{I}_{or}$ is the total energy per chip averaged over fading and received from the serving sector, and $I_{oc}(n)$ is the total energy per chip averaged over fading and received from the n-th interfering sector.

In the system level simulation, the Rake fingers shall be combined using pilot-weighted combining. The signal-to-interference ratio at the output of the pilot-weighted combiner is given by

$$(C/I)_{\text{combined}} = \frac{\left(\sum_{i=1}^{J}\|\gamma_i\|^2\right)^2}{\sum_{j=1}^{J}\|\gamma_j\|^2\left(G^{-1} + \|\lambda\|^2 + \sum_{1\le k\le J, k\ne j}\|\gamma_k\|^2\right)} \ . \tag{2.2.1-3}$$

This combined C/I shall further be limited by a C/I ceiling as described in section 2.1.2.

For system level simulations including transmit diversity (STS), the channels between the two transmit antennas and the subscriber are assumed to fade independently of each other. The channel models are taken from Table 2.2.1-1. For a particular PCG, let $\{\gamma_i\}_{i=1}^{J}$ and $\{\tilde{\gamma}_i\}_{i=1}^{J}$, respectively, denote the samples of the fading processes for the J recovered rays of the first and second antennas. Let λ and $\tilde{\lambda}$, respectively, denote the sample of the fading process for the additional rays used to model interference due to the unrecovered power for the first and second antennas.

Let $\left\{(C/I)_{1,i}\right\}_{i=1}^{J}$ denote the signal-to-interference ratio of Rake fingers demodulating symbols transmitted from the first antenna. For transmit diversity using STS, one-half of the energy of a given code symbol is transmitted on each of the antennas. If all code channels are transmitted using transmit diversity, the signal-to-interference ratio of the i-th Rake finger can be expressed as

$$(C/I)_{1,i} = \frac{\|\gamma_i\|^2/2}{G^{-1} + \frac{1}{2}\left[\left(\|\lambda\|^2 + \|\tilde{\lambda}\|^2\right) + \sum_{1\le k\le J, k\ne i}\left(\|\gamma_k\|^2 + \|\tilde{\gamma}_k\|^2\right)\right]} \qquad (2.2.1\text{-}4)$$

Let $\left\{(C/I)_{2,i}\right\}_{i=1}^{J}$ denote the signal-to-interference ratio of Rake fingers demodulating symbols transmitted from the second antenna, which can be expressed as

$$(C/I)_{2,i} = \frac{\|\tilde{\gamma}_i\|^2/2}{G^{-1} + \frac{1}{2}\left[\left(\|\lambda\|^2 + \|\tilde{\lambda}\|^2\right) + \sum_{1\le k\le J, k\ne i}\left(\|\gamma_k\|^2 + \|\tilde{\gamma}_k\|^2\right)\right]} \qquad (2.2.1\text{-}5)$$

The signal-to-interference ratio for STS with pilot-weighted combining of the Rake fingers in both delay and diversity is given by

$$(C/I)_{\text{STS,combined}} = \frac{\frac{1}{2}\left(\sum_{i=1}^{J}\left(\|\gamma_i\|^2 + \|\tilde{\gamma}_i\|^2\right)\right)^2}{\sum_{j=1}^{J}\left(\|\gamma_j\|^2 + \|\tilde{\gamma}_j\|^2\right)\left(G^{-1} + \frac{1}{2}\left[\left(\|\lambda\|^2 + \|\tilde{\lambda}\|^2\right) + \sum_{1\le k\le J, k\ne j}\left(\|\gamma_k\|^2 + \|\tilde{\gamma}_k\|^2\right)\right]\right)} \qquad (2.2.1\text{-}6)$$

This combined C/I shall further be limited by a C/I ceiling as described in section 2.1.2.

2.3 Simulation Flow and Output Matrices

Either the center cell method or the iteration method shall be used.

The total simulation time per drop shall be long enough to guarantee residual FER to be 10^{-2} with certain confidence level. The required upper layers to guarantee the TCP input FER to be 10^{-4} is not modeled.

2.3.1 Simulation Flow for the Center Cell Method

The simulation will make the following assumptions:

1. The system consists of 19 hexagonal cells. Six cells of the first tier and 12 cells of the second tier surround the central cell. Each cell has three sectors.

2. Mobiles are first dropped uniformly throughout the system. Each mobile corresponds to an active user session. A session runs for the duration of the drop. Mobiles are assigned channel models described in Table 2.2.1-1 with the given probabilities.

3. **Applicable to 1xEV-DV only:** For simulation of systems loaded only with voice-only mobiles, the first run is done with five mobile stations per sector and every run following that is done with five more mobile stations per sector until the outage criteria are violated, after which every run is done with one less mobile until the outage criteria are satisfied. The maximum number of users per sector of N_{max} (voice capacity) is achieved if N_{max} + 1 users per sector would not satisfy the outage criteria and N_{max} users per sector would.

4. For simulation of the system loaded only with data-only mobiles, the runs are done with an increment of two mobile stations per sector.

5. **Applicable to 1xEV-DV only:** With voice-only and data-only mobile stations in the same simulated system, the number of voice-only mobile stations is fixed at $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ per sector. The number of data-only mobile stations is incremented by two mobile stations per sector for each successive simulation run.

6. Mobile stations are randomly dropped over the 57 sectors such that each sector has the required numbers of voice (1xEV-DV only) and data users. Although users may be in soft-handoff, each user is assigned to only one sector for counting purposes. A data user shall be assigned to a sector if the sector is its primary server. To simplify the 1xEV-DV simulation, only two-way or three-way handoff is used for voice users. A 1xEV-DV voice user on a two-way or three-way soft handoff counts as ½ or 1/3 a user on each of the sectors in the Active Set, respectively. All sectors of the system shall continue accepting users until the desired fixed number of data/voice users per sector is achieved everywhere.

7. Fading signal and fading interference are computed from each data/voice mobile station into each sector for each PCG (slot) or equivalent power control related time interval.

8. The total simulation time per drop will be 10 minutes excluding any time required for initialization (~10 sec). The total number of drops per run is 12 for a total simulation time of 2 hours per condition.

9. Packet calls arrive as per the HTTP model of section 4.1.3. Packets are not blocked when they arrive into the system (i.e. queue depths are infinite).

10. WAP is modeled as per section 4.1.5.

11. FTP is modeled as per section 4.1.4. FTP results presented shall be from a stable system. That is a system in which the average rate of FTP users exiting the system is equal to the average rate of FTP users entering the system.

12. Near real time video is modeled as per section 4.1.6.

13. Packets are scheduled with a packet scheduler.

14. The ARQ process is modeled by explicitly rescheduling a packet as part of the current packet call after a specified ARQ feedback delay period.

15. Simulation flow with data-only mobiles or voice-only and data-only mobiles simultaneously is shown in Figure 2.3.1-1:

For $n = 0$, $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ voice-only mobiles per sector (1xEV-DV only),

For each k data-only users (to be incremented by 2),

a. Place n voice users in each sector of all cells (1xEV-DV only).

b. Keep adding data users until the quality of service criteria are not met.

c. Collect results according to the output matrix.

16. Only statistics of the mobiles of the center cell are collected.
17. All 57 sectors in the system shall be dynamically simulated.

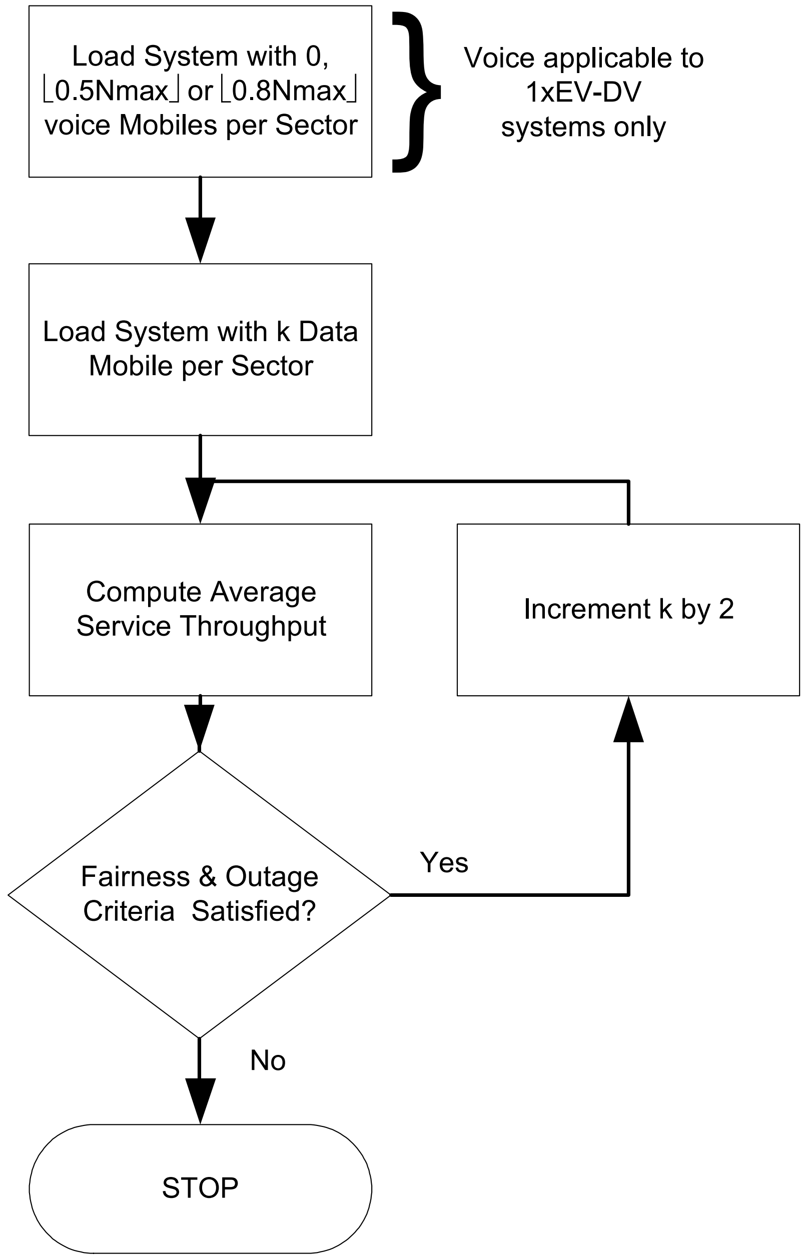

Figure 2.3.1-1 Simulation Flow Chart

2.3.2 Simulation Flow for the Iteration Method

The cells are classified into two types: active and passive cells. Active cells are fully simulated and monitored, whereas passive cells only model the interference created by neighboring cells.

Terminology:

Active cell – the central cell of the 19 cell layout which handles all the data traffic, runs a scheduler, keeps track of the voice users (when applicable), generates the transmission power profile, and collects all the statistics.

Passive cells – the other 18 cells, which follow the power profile of the active cell as found in the previous iteration. In order to break the temporal correlations 18 equally spaced offsets are introduced, one for each passive cell. For example, for 600 second run each offset is 33 seconds, which is sufficient to compensate for any correlation. For 1xEV-DV systems, voice calls can be in soft handoff with passive cells, in which case Ec/Ior for a given call is assumed to be the same at all the base stations in the active set. Passive cells are present in the system in order to model the inter-cell interference, and therefore data users associated with them do not need to be modeled.

Iteration – A simulation run where passive cells follow a well-specified power profile obtained from the active cell on a previous iteration. Iteration 0 starts with passive cells transmitting with the maximum power.

Description:

The system consists of 19 hexagonal cells. The central cell is an active cell and is surrounded by six passive cells of the first tier and 12 passive cells of the second tier cells. Each cell has three sectors.

Applicable to 1xEV-DV only: For simulation of systems loaded only with voice-only mobiles, the first run is done with five mobile stations per sector and every run following that is done with five more mobile stations per sector until the outage criteria are violated, after which every run is done with one less mobile until the outage criteria are satisfied. The maximum number of users per sector of N_{max} (voice capacity) is achieved if $N_{max} + 1$ users per sector would not satisfy the outage criteria and N_{max} users per sector would.

For simulation of the system loaded only with data-only mobiles, the runs are done with an increment of two mobile stations per sector.

Applicable to 1xEV-DV only: With voice-only and data-only mobile stations in the same simulated system, the number of voice-only mobile stations is fixed at $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ per sector. The number of data-only mobile stations is incremented by two mobile stations per sector for each successive simulation run.

Voice and data users are randomly dropped over the 57 sectors such that each cell has the required numbers of voice and data users. To simplify the 1xEV-DV simulation, only two-way or three-way handoff is used for voice users. A 1xEV-DV voice user on a two-way or three-way soft handoff counts as ½ or 1/3 a user on each of the sectors in the Active Set, respectively. A data user shall be assigned to a sector if the sector is its primary server. Users whose Active Set does not contain a sector of the active (center) cell shall be discarded. The sectors of the active cell shall continue accepting users until the desired fixed number of data/voice users per sector is achieved.

Fading signal and fading interference are computed for each data/voice user for each PCG or equivalent power control related time interval.

Iterations are performed in the following order:

- **Iteration 0**: Passive cells radiate at maximum power. Power statistics of the active (central) cell is collected for use in the next iteration.
- **Iteration n (n>0)**: Run the system forcing passive cells to follow the active's cell power profile found on the iteration (n-1). Time offsets are introduced to break the correlation, as described previously.

Convergence criterion: stability of the per sector throughput and of the power profile second order statistics

The total simulation time per drop will be 10 minutes excluding any time required for initialization (~10 sec). The total number of drops per run is 12 for a total simulation time of 2 hours per condition.

Packet calls arrive as per the HTTP model of section 4.1.3. Packets are not blocked when they arrive into the system (i.e. queue depths are infinite).

WAP is modeled as per section 4.1.5.

FTP is modeled as per section 4.1.4.

Near real time video is modeled as per section 4.1.6.

Packets are scheduled with a packet scheduler.

The ARQ process is modeled by explicitly rescheduling a packet as part of the current packet call after a specified ARQ feedback delay period.

Simulation flow with data-only mobiles or voice-only and data-only mobiles simultaneously is shown in Figure 2.3.1-1:

For n = 0, $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ voice-only mobiles per sector (1xEV-DV only),

For each k data-only users (to be incremented by 2),

a. Place n voice users in each sector of all cells.

b. Keep adding data users until the quality of service criteria are not met.

c. Collect results according to the output matrix.

2.3.3 Output Matrices

The performance report shall contain a pdf of the forward link C/I observed in one of the sectors of the center cell under the assumption of full power transmitted by all sectors. Three curves shall be generated. First one includes path loss, shadowing, sectorization, but not Rayleigh fading; the second one and the third one are the first one with the restriction of maximum C/I to be 13 dB and 17.8 dB, respectively.

Table 2.3.3-1 summarizes all the cases to be simulated for 1xEV-DV systems.

Table 2.3.3-1 Required 1xEV-DV Simulation Evaluation Comparison Cases Table

	Loading Scenarios	Tx Diversity	no Tx Diversity	Max C/I 13.0 dB	Max C/I 17.8 dB	RC3	RC4
1	voice only 100% (Nmax) load	x		x		x	
2			x	x		x	
3		x		x			x
4			x	x			x
5		x			x	x	
6			x		x	x	
7		x			x		x
8			x		x		x
9	1xEVDV data only	x		x			
10			x	x			
11		x			x		
12			x		x		
13	50%voice + 1xEVDV data	x		x		x	
14			x	x		x	
15		x		x			x
16			x	x			x
17		x			x	x	
18			x		x	x	
19		x			x		x
20			x		x		x
21	80%voice + 1xEVDV data	x		x		x	
22			x	x		x	
23		x		x			x
24			x	x			x
25		x			x	x	
26			x		x	x	
27		x			x		x
28			x		x		x

The voice capacity numbers generated from run #1 to run #8 should be approximately the same across different companies, which can be used as calibration of different simulators.

1xEV-DV data only with transmit diversity scenarios (run #9 and run #11) are optional for proposals that do not support transmit diversity.

Transmit diversity (STS with independent fading assumption) will be used for framework selection.

2.3.3.1 General output matrices

The following matrices shall be provided:

1. All link-level results used in system-level simulator,
2. The histogram of C/I used in system-level simulation, where the C/I includes path loss, shadowing, and sectorization:
    a. Without any limitation on the maximum C/I
    b. With a limit of 13 dB on the maximum C/I, as specified in section 2.1.2
    c. With a limit of 17.8 dB on the maximum C/I, as specified in section 2.1.2

3. The curve of geometry vs. the distance from a user's location to its closest serving cell, where geometry is solely a function of the distance and excludes fading and shadowing factors.

4. The histogram of the distance between a user and its closest serving cell/sector.

5. The performance of any estimator(s) or predictor(s) that are required by the proposal. For instance, if a channel predictor is used in the proposal, the details of the predictor/estimator, its bias, and standard deviation shall be provided by the proponent(s).

6. The performance of any forward channels that are not simulated by the link-level or system-level simulator shall be justified by the proponent. For example, if the forward signaling channels are not simulated by the system-level simulator, the proponent companies shall specify the performance of these channels and justify their claims (see section 2.1.5).

2.3.3.2 Data Services and Related Output Matrices

The following statistics related to data traffics shall be generated and included in the evaluation report.

1. **Data throughput per sector**. The data throughput of a sector is defined as the number of information bits per second that a sector can deliver and are received successfully by all data users it serves, using the scheduling algorithm validated in section 2.1.6, and that certain number of voice users can be maintained with certain GOS.

2. **Averaged packet delay per sector**. The averaged packet delay per sector is defined as the ratio of the accumulated delay for all packets it delivers to all users and the total number of packets it delivers. The delay for an individual packet is defined as the time between when the packet enters the queue at transmitter and the time when the packet is received successively by the mobile station. If a packet is not successfully delivered by the end of a run, its ending time is the end of the run.

3. **The histogram of data throughput per user**. The throughput of a user is defined as the ratio of the number of information bits that the user successfully receives during a simulation run and the simulation time. Note that this definition is applicable to all data users.

4. **The histogram of packet call throughput for users with packet call arrival process.** The packet call throughput of a user is defined as the ratio of the total number of information bits that an user successfully receives and the accumulated delay for all packet calls for the user, where the delay for an individual packet call is defined as the time between when the first packet of the packet call enters the queue for transmission at transmitter and the time when the last packet of the packet call is successively received by the receiver. If a packet call is not successfully delivered by the end of a run, its ending time is the end of the run, and none of the information bits of the packet call shall be counted. Note that this definition is applicable only to a user with packet call arrival process.

5. **The histogram of averaged packet delay per user**. The averaged packet delay is defined as the ratio of the accumulated delay for all packets for the user and the total number of packets for the user. The delay for a packet is defined as in 2. Note that this definition is applicable to all data users.
6. **The histogram of averaged packet call delay for users with packet call arrival process**. The averaged packet call delay is defined as the ratio of the accumulated delay for all packet calls for the user and the total number of packet calls for the user. The delay for a packet call is defined as in 4. Note that this definition is applicable only to a user with packet call arrival process.
7. **The scattering plot of data throughput per user vs. the distance from the user's location to its serving sector.** In case of SHO or sector switching, the distance between the user and the closest serving sector shall be used. The data throughput for a user is defined as in 3.
8. **The scattering plot of packet call throughputs for users with packet call arrival processes vs. the distance from the users' locations to their serving sectors.** In case of SHO or sector switching, the distance between the user and the closest serving sector shall be used. The packet call throughput for a user is defined as in 4.
9. **The scattering plot of averaged packet delay per user vs. the distance from the mobile's location to its serving sector.** In case of SHO or sector switching, the distance between the user and its closest serving sector shall be used. The averaged packet delay per user is defined as in 2.
10. **The scattering plot of averaged packet call delays for users with packet call arrival processes vs. the distance from the mobiles' locations to their serving sectors.** In case of SHO or sector switching, the distance between the user and its closest serving sector shall be used. The averaged packet call delay per user is defined as in 4.
11. **The scattering plot of data throughput per user vs. its averaged packet delay**. The data throughput and averaged packet delay per user are defined as in 3 and 2, respectively.
12. **The scattering plot of packet call throughputs for users with packet call arrival processes vs. their averaged packet call delays**. The packet call throughput and averaged packet call delay per user are defined as in 4.

Appendix D provides formulas of the above definitions.

The channel model and speed of a data user are randomly chosen according to the pre-determined distributions specified in 2.2.1.

2.3.3.3 1xEV-DV Systems Only

2.3.3.3.1 Voice Services and Related Output Matrices

The following statistics related to voice traffic shall be generated and included in the evaluation report.

1. Voice capacity, where the voice capacity is defined as the maximum number of voice users that the system can support within a sector with certain maximum system outage probability. The details on how to determine the voice capacity of a sector are described in Appendix C.
2. The histogram of voice data rates (for a frame) per user and for all users.
3. The scattering plot of the outage probability vs. the distance from the mobile to the serving cell. In case of soft hand-off (SHO), the distance from the mobile to the closest serving cell shall be used.
4. The curve of outage indicator vs. time for each voice user. The outage indicator equals to one when the voice user is in outage, and zero otherwise. The speed, channel model and the distance of the voice user to the serving cell shall also be included in the curve. In case of SHO, the distance from the mobile to the closest serving cell shall be used.
5. The outage probability for each user. Note that this value can be calculated from the curve described in the previous item.

The channel model and speed of a voice user are randomly chosen according to the pre-determined distributions specified in Table 2.2.1-1.

2.3.3.3.2 Mixed Voice and Data Services

In order to fully evaluate the performance of a proposal with mixed data and voice services, simulations shall be repeated with different loads of voice users. The following outputs shall be generated and included in the evaluation report.

1. The following cases shall be simulated: no voice users (i.e., data only), voice users only (i.e., the number of voice users equals to voice capacity), and average $\lfloor 0.5N_{max} \rfloor$ and $\lfloor 0.8N_{max} \rfloor$ voice users per sector.
2. For each of the above case, all corresponding output matrices defined for voice and data services shall be generated, whenever they are applicable.

In addition to the output matrices described in the previous two sections, the following output matrix shall also be generated and included in the evaluation report.

1. A curve of cell/sector data throughput vs. the number of voice users, where the cell/sector data throughput is defined as above.

2.3.3.4 Mixed Rev. 0 and Rev. A Mobiles (1xEV-DO Systems Only)

For proposals that support backward compatibility with 1xEV-DO (Rev. 0) ATs, the following mixed Rev. 0 and Rev. A user tests shall be simulated using the standard system layout and channel mix specified in section 2.2.1:

a. 8 full-buffer Rev. 0 ATs and 8 full-buffer Rev. A ATs per sector.

b. 16 Rev. 0 full-buffer ATs per sector.

c. 16 Rev. A full-buffer ATs per sector.

The statistics required by the output matrices specified in appendix M shall be generated and presented for the purpose of evaluation, as well as the associated statistics required for the evaluation report specified in 2.3.3.2. In addition, the following statistic shall be reported:

a. Avg_TP_Rev0_Mix: Average throughput of Rev. 0 ATs in scenario a.

b. Avg_TP_RevA_Mix: Average throughput of Rev. A ATs in scenario a.

c. $\left(\frac{Avg\_TP\_\mathrm{Re}v0\_Mix}{Avg\_TP\_\mathrm{Re}v0\_Only}, \frac{Avg\_TP\_\mathrm{Re}vA\_Mix}{Avg\_TP\_\mathrm{Re}vA\_Only}\right):$

where Avg_TP_Rev0_Only denotes the average throughput per AT from scenario b, Avg_TP_Rev_A_Only denotes the average throughput per AT from scenario c.

3 EVALUATION METHODOLOGY FOR THE REVERSE LINK

3.1 System Level Setup

3.1.1 Antenna Pattern

Antenna pattern shall be as specified in 2.1.1.

3.1.2 System Level Assumptions

The parameters used in the simulation are listed in Table 3.1.2-1. Where values are not shown, the values and assumptions used in the simulation shall be specified in the simulation description.

Table 3.1.2-1 Reverse Link System Level Simulation Parameters

Parameter	Value	Comments
Number of 3-sector Cells	19	2 ring, 3-sector system, 57 sectors total. These cells are on a "wrap-around" model where the signal or interference from any MS to a given cell is treated as if that MS is in the first 2 rings of neighboring cells. MSs are uniformly dropped over the 19 cells. Simulation is done with the desired number of MSs for each sector of each cell. Throughput and capacity are collected from all cells
Antenna Horizontal Pattern	70 degree (-3 dB) with 20 dB front-to-back ratio	see Section 2.1.1
Antenna Orientation	0 degree horizontal azimuth is East (main lobe)	No loss is assumed on the vertical azimuth. (See Appendix B)
Propagation Model (BTS Ant Ht=32m, MS=1.5m)	28.6+ 35log10(d) dB, d in meters	Modified Hata Urban Prop. Model @1.9GHz (COST 231). Minimum of 35 meters separation between MS and BS. [4]

[4] In this document the word "modified" represents a difference from the COST231-Hata model wherein the path loss has been reduced by 3 dB [33]. If a mobile is dropped within 35 meters of a base station, it shall be redropped until it is outside the 35-meter circle.

Parameter	Value	Comments
Log-Normal Shadowing	Standard Deviation = 8.9 dB for both FL and RL	Independently generate lognormal per mobile-sector pair and use the method described in Appendix A. This shadowing is constant in each simulation run. The same shadowing amount shall be used for all Rx antennas of a BS (up to six) to a given MS. The correlation coefficient between the BS's Tx antennas and a given MS and the BS's RX antennas and a given MS is 1.
Maximum RL Total Path Loss	146 dB	This term includes the MS and BS antenna gains, cable and connector losses, other losses, and shadowing, but not fading.
Base Station Shadowing Correlation	0.5	See Appendix A
Overhead Channel Reverse Link Power Usage		Any additional overhead needed to support other control channels (dedicated or common) for the forward link or the reverse link must be specified and accounted for in the simulation
Base Noise Figure	5.0 dB	
Thermal Noise Density	-174 dBm/Hz	
Carrier Frequency	2 GHz	
BS Antenna Gain w Cable Loss	15 dB	17 dB BS antenna gain; 2 dB cable loss
MS Antenna Gain	-1 dBi	
Other Losses	10 dB	Applicable to all fading models
Maximum MS EIRP	23 dBm	

Parameter		**Value**	**Comments**
Fast Fading Model		Based on Speed	The fading processes on the paths from a given MS to the two BS antennas are mutually independent. The fading model is specified in Appendix K.
Active Set Membership	Circuit switched and packet switched data systems (e.g., 1xEV-DV)		Up to 3 members are in the Active Set if the pilot Ec/Io is larger than T_ADD = -18 dB (=9 dB below the FL pilot Ec/Ior) based on the FL evaluation methodology
	Packet switched data systems (e.g., 1xEV-DO)		Up to 3 members are in the Active Set if the pilot Ec/Io is larger than T_ADD = -9 dB based on the FL evaluation methodology
Delay Spread Model			See Table 2.2.1-1 and Table 2.2.1-2
Reverse Link Scheduling			System specific. Proponents need to declare the scheme and the associated MAC delay and reliability.
Active Set Change			System specific. Proponents need to declare the scheme and the associated signaling delay and reliability.

Parameter		Value	Comments
Reverse Link Power Control	Circuit switched and packet switched data systems (e.g., 1xEV-DV)	Closed-loop power control delay: two PCGs[5]	Update Rate: Dependent on proposal. Power control feedback: BER = 4% for a BS-MS pair. Different values shall be specified and accounted for in the simulation Ec/Nt measurement error at the BS: additive in dB, log normal, zero-mean random variable with a 2 dB standard deviation.
	Packet switched data systems (e.g., 1xEV-DO)	Closed-loop power control delay: two slots[5]	Update Rate: Up to 600 Hz, dependent on proposal. Power control feedback: BER = 4% for a BS-MS pair. Different values shall be specified and accounted for in the simulation Ec/Nt measurement error at the BS: additive in dB, log normal, zero-mean random variable with a 2 dB standard deviation.
MS PA Size		200 mW	
Site to Site distance		2.5 km	

[5] The MS transmit power changes in PCG/slot i+2 in response to measurement made in PCG/slot i. One PCG/slot delay for link level modeling (measured from the last chip that the reverse pilot is measured to the time that the mobile changes TX power level).

Parameter	Value	Comments
Rise over Thermal (Reverse Received Power Normalized by Thermal Noise Level)	7 dB	Histogram of this parameter with a 1.25 (1xEV-DV) or 1.67 (1xEV-DO) ms time resolution shall be provided with the mean rise-over-thermal. The percentage of time the rise over thermal above the 7 dB target shall not exceed 1%. Rise over thermal for the default two receiving antenna mode is ½[(Io1+No)/No + (Io2 + No)/No], where the total received signal power at antenna i is defined as Ioi, i=1,2.

3.1.3 Call Setup Model

The following is the method to simulate a call setup on the RL, regardless of the traffic time:

1. MS gets in the system at $t = t_0$[6]
2. At $t = t_0$, MS starts transmitting pilot only for 320 ms[7] or 427 ms[7] for 1xEV-DV or 1xEV-DO systems respectively
    a. Closed loop power control is active but outer loop has a fixed target pilot Ec/Io
    b. Starting pilot transmission power is –50 dBm
    c. BS Set Point is fixed at – 22.5 dB or –20.5 dB for 1xEV-DV or 1xEV-DO systems respectively
3. At $t = t_0$ + 320 ms, together with the pilot, MS begins transmission of the null traffic at 1.5 kbps using the traffic to pilot ratio of -5.875 (= -47/8) dB for 1xEV-DV systems. At $t = t_0$ + 427 ms, together with the pilot, the MS starts the DRC transmission using a specified DRC-to-pilot ratio for 1xEV-DO systems. (For 1xEV-

[6] t_0 is a relative time, i.e. it is not the absolute system time but the moment when a MS gets in the system. For HTTP and WAP users, t_0 is at the very beginning of the simulation, while for FTP upload MS, t_0 occurs during the simulation, as these MSs arrive in the system according to the Poisson arrival process.

[7] This step is meant to replace the probing process. Although it does not represent the real system events, it approximately models the adjustment of MS transmit power, delay and loading on RL

DO Rev. 0, the DRC-to-pilot ratio equals to -1.5 dB for non-handoff ATs (DRC Length = 2 slots) and –3 dB for ATs in soft-handoff (DRC length = 4 slots).)

   a. Outer loop power control is active

4. At $t = t_0 + 580$ ms, call setup procedure is finished

   a. Statistic collection for the MS starts at the end of the call setup

   b. MS can now start using R-FCH and request R-SCH

The timeline diagram is shown in Figure 3.1.3-1 below.

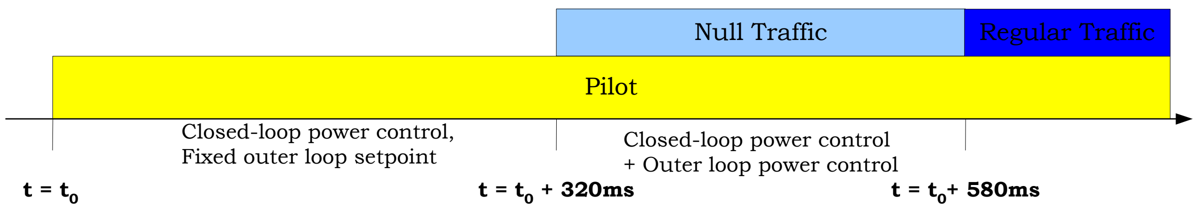

Figure 3.1.3-1: Simplified Call Setup Timeline for 1xEV-DV. (Timeline for 1xEV-DO is the same by modifying 320 ms to 427 ms)

3.1.4 Packet Scheduler

The voice users' (when simulated together with the data users) transmissions are not scheduled in 1xEV-DV systems. The data users can be scheduled or allowed to transmit in a random fashion. The exact procedure, and its delay and reliability, with which a mobile station gains the right to transmit, shall be specified in detail. Proponents shall state whether the reverse scheduling signaling is sent from the entire Active Set of the MS or its subset. As a baseline, the scheduler shall be unaware of the application(s) running on the MS's and shall not use the information provided by the QoS BLOB. Baseline simulation results shall be generated accordingly and be presented. The proponent may, however, present additional results with a more sophisticated scheduler in the system.

3.1.5 Backhaul Overhead Modeling in Reverse Link System Simulations

The backhaul bandwidth used by signaling and measurement messages is provided in the format shown in Table 3.1.5-1. It is assumed that the messages sent over the backhaul links use the TCP/IP protocol. Therefore, a TCP/IP packet overhead of 320 bits (40 bytes) is added to each signaling or measurement message sent over the backhaul. The backhaul overhead on the FL is given by:

$$\text{FL Backhaul overhead} = \frac{1}{T_{run}} \sum_{i=1}^{N_{FL}} S_i \text{ b/s} \qquad (3.1.5\text{-}1)$$

Where S_i is the signaling message size in bits (including 320 bits TCP/IP header) and N_{FL} is the total number of messages (signaling, measurements etc.) sent on the FL during a simulation run. T_{run} is the simulation time in seconds for a given run. The FL backhaul overhead is averaged over all the simulation runs.

The backhaul overhead on the RL is given by:

$$\text{RL Backhaul overhead} = \frac{1}{T_{run}} \sum_{j=1}^{N_{RL}} S_j \text{ b/s} \qquad (3.1.5\text{-}2)$$

Where S_j is the signaling message size in bits (including 320 bits TCP/IP header) and N_{RL} is the total number of messages sent on the RL during a simulation run. The RL backhaul overhead is averaged over all the simulation runs.

Table 3.1.5-1 Backhaul bandwidth used by signaling and measurement messages

FL backhaul overhead [b/s]	RL backhaul overhead [b/s]
Under study	Under study

3.1.6 Simulation of Forward Link Overheads for Reverse Link System Simulation

For reverse-link enhancement proposal envisioning communicating schedule grants (rate assignments), rate adjustments, acknowledgements (to support H-ARQ operation) from one or more base station(s) to the data mobile, on new or existing Forward Link channels, the following shall apply when modeling the impact of these forward-link (FL) channels on RL performance:

1. The impact of imperfect RPC on reverse-link power-control shall be modeled by assuming a 4% error rate on the RPC channel for each BS-MS pair. The closed-loop power-control delay shall be 1 slot/PCG.
2. For proposals with hybrid-ARQ scheme, the impact of imperfect FL ARQ signal shall be modeled in the network simulation with dynamic FL modeling in the simulations. The dynamic FL modeling shall assume the same multipath and Doppler channel model on FL and RL (i.e. channel A, B, C, D, and E). The fading process on FL and RL shall be assumed as independent.
3. Any additional FL channel whose error performance may impact the performance on RL shall be modeled either via dynamic FL modeling (assuming the same multipath and Doppler channel model but independent fading on FL and RL) or via static modeling, both of which are described below.

Both static and dynamic simulation methods are allowed for modeling these Forward Link Overhead channels that support Reverse Link operation.[8] Static simulation results are required to be provided; dynamic simulation results may be provided. These two methods are described below.

3.1.6.1 Static Modeling Method

1. Long-term (FER versus average Eb/No) error curves are to be generated, over each channel model, for each of the proposed overhead channels[9] on the Forward Link.

[8] If channel sensitive scheduling is used on the forward link, the static method may be pessimistic.

[9] These curves are generated for each frame format allowed for the overhead channels.

2. For users receiving the overhead channels in handoff, long term error curves are to be generated for each channel model, way of soft hand-off (SHO), and a range of SHO imbalance(s), and Geometry.

3. A desired target FER shall be specified for each of these channels for proper system operation.

4. For each user dropped into a system simulation run, the overhead channel frame format is determined, the Geometry is determined, and the Eb/No required to meet the target FER is translated into an average fractional power allocation at each serving base station. Each mobile is assigned the same channel model on the Forward Link as on the Reverse Link.

The following methods apply in addition to the above to 1xEV-DV systems only:

1. The fractional power allocations for all the users in each drop are summed at each base station. This is the average fractional power cost on the forward link, for that drop at each base station.

2. Average these fractional power cost values across the required number of RL system simulation drops, at each base station. This is the average power cost on the forward link at each base station.

3. Similarly for each drop, determine the number of Walsh channels used to support the reverse link at each base station. Average the number of Walsh channels used across the required number of RL system simulation drops, at each base station. This is the average number of Walsh channels used on the forward link at each base station.

Reverse Link System Simulation Assumptions and Modeling

Independent error events should be generated in the Reverse Link system simulation at time instants corresponding to the reception of any Forward Link overhead channel. An error shall occur on each reception with a probability equal to the specified target FER for the corresponding channel. The resulting impact (schedule grant miss, error in rate assignment, ACK/NACK feedback error, etc.) shall be modeled and modeling mechanism specified.

3.1.6.2 Dynamic Modeling Method

The dynamic modeling of Forward Link overheads is done by running fading on the Forward Link (operating only the Forward Link overhead channels) in conjunction with the Reverse Link in the Reverse Link system simulation.

1. Short-term (FER Vs average Eb/No) error curves are to be generated, over each channel model, for each of the proposed overhead channels on the Forward Link.

2. For users receiving the overhead channels in handoff, short term error curves are to be generated for each channel model, way of soft hand-off (SHO), and a range of SHO imbalance(s), and instantaneous Geometry.

3. A desired target FER shall be specified for each of these channels for proper system operation.

4. The Forward Link fades independently of the Reverse Link at each connected base station. A connected base station is one that sends schedule grants, rate adjustments, or acknowledgements to a mobile. Each mobile is assigned the same channel model on the Forward Link as on the Reverse Link.

5. Power allocation for each of the overhead channels, in the system simulation, is done dynamically and logged. The power allocations should be done for all members of the active set. The proponent of the proposal should explain how the power allocations are done. [For example, for the serving base station, the CQI value is fed back to the base station by the mobile station and the forward link power is set based upon the returned CQI value.]

The following apply in addition to the above for 1xEV-DV systems only:

1. The instantaneous power allocations for each user are averaged over the drop for every base station. Then these average power allocations are summed over all users at each base station. This is the average fractional power cost on the forward link, for that drop at each base station.

2. Average these fractional power cost values across the required number of RL system simulation drops, at each base station. This is the average power cost on the forward link at each base station.

3. Similarly for each drop, determine the number of Walsh channels used to support the reverse link at each base station. Average the number of Walsh channels used across the required number of RL system simulation drops, at each base station. This is the average number of Walsh channels used on the forward link at each base station.

3.1.6.3 Quantification of Forward Link Overhead as a Data Rate Cost (1xEV-DV Systems Only)

An appropriate metric is needed for specifying the cost (in power and bandwidth) on the Forward Link associated with the Forward Link overhead channels supporting Reverse Link operation. For both the static and dynamic methods, the average amount of forward link power and the number of required Walsh channels are determined.

The Forward Link (Revision C) system simulations are run with a new amount of overhead power and a new number of available Walsh channels. Specifically, the amount of overhead power is increased by the average amount of overhead power determined from either the static or dynamic system level simulation. Similarly the number of available Walsh channels is reduced by the number of Walsh channels used to support the reverse link as determined by either the static or dynamic method. In determining the resulting capacity reduction, 40 forward link users should be used in a data-only simulation.

3.1.7 Signaling Errors

Signaling errors shall be as specified in section 2.1.5.

For modelling the signalling error in TCP three-way handshake protocol as shown in Figure 4.2.2-1 in FTP upload traffic model, failed RL handshake packets will be re-transmitted in the physical layer up to a number of times per proposal, if physical layer ARQ is used. If it still fails after physical layer retransmission, it will be assumed error free from TCP protocol viewpoint, but the error is modeled from throughput counting viewpoint. The handshake and ACK packets on the FL are assumed to be error free. However, the delay element of the FL handshake and ACK packets is modeled.

3.1.8 Fairness Criteria

The CDF of the normalized throughputs with respect to the average user throughput for all FTP upload MS in the same sector is determined. This CDF shall lie to the right of the curve given by the three points in Table 3.1.8-1.

Table 3.1.8-1 CDF Criterion for FTP Upload MS

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

3.1.9 FER Criterion

The following outage criterion is defined for the final FER after physical layer retransmissions:

- For each traffic channel type (e.g., R-FCH, R-DCCH, R-SCH, RTC or other new RL traffic channels proposed), the mean FER across all users in the system shall be sufficient to sustain services.
- For each traffic channel type (e.g., R-FCH, R-DCCH, R-SCH, RTC or other new RL traffic channels proposed), the percentage of users with FER greater than 5% shall be low enough to insure adequate service delivery and coverage.

3.2 Link Level Modeling

3.2.1 Link Level Parameters and Assumptions

The performance characteristics of individual links used in the system simulation are generated a priori from link level simulations. Link level simulation parameters are specified in Appendix J.

Turbo Decoder Metric and Soft Value Generation into Turbo Decoder shall be as specified in Appendix H.

3.2.1.1 Frame Erasures

Data for the link-level performance shall be presented and agreed upon so that one set of data is used in all simulations.

The following method describes the Quasi-Static approach for modeling link performance. Additional link level modeling methods can be found in Appendix N and Appendix O:

Description of Quasi-Static Approach with Short Term FER:

The quasi-static approach with short term FER shall be used to generate the frame erasures for reverse link channels with fixed traffic to pilot power (T/P) ratio in a frame such as R-FCH, R-DCCH, R-SCH, RTC, other new proposed RL data channels, and RL overhead channels.

A full set of short term FER vs. average traffic E_b/N_t per frame curves is generated as a function of radio configurations, data rates, T/P ratios, and channel models. The number of curves should be reduced if possible, provided that this won't unduly affect the validity of this quasi-static approach.

In the system-level simulation, the average traffic E_b/N_t per frame is computed as follows. First, the traffic E_b/N_t is calculated in a PCG/slot as specified in section 3.2.1.3. The short-term average traffic E_b/N_t per frame is defined as the average of the traffic E_b/N_t for all N_{PCG} PCGs/slots in a frame, i.e.,

$$Avg(E_b / N_t) = \frac{1}{N_{PCG}} \sum_{n=1}^{N_{PCG}} \left(\frac{E_b}{N_t} \right)_n , \qquad (3.2.1.1\text{-}1)$$

where $(E_b / N_t)_n$ is the traffic E_b/N_t in the n-th PCG in a frame. Once the $Avg(E_b / N_t)$ is calculated as in the above equation, it is used to look up the corresponding link level short term FER vs. average E_b/N_t per frame curve for the specific condition. A frame erasure event is then generated based on the FER value.

In HARQ scheme, a full set of short term FER vs. total traffic E_b/N_t curves is generated as a function of the number of transmission frames for the specific condition. The total traffic E_b/N_t is defined as the summation of $Avg(E_b / N_t)$ of transmitted frames. It is used to look up the corresponding link level short term FER vs. total traffic E_b/N_t curve for the number of transmitted frames.

The short-term FER curves shall be generated as follows:

1. The link-level simulation is conducted for the same condition as in system level simulation as much as possible. The average traffic E_b/N_t in a frame and the frame erasure indicator for the frame are recorded. The average traffic E_b/N_t per frame is computed as follows in the link-level simulation

$$Avg(E_b / N_t) = \frac{1}{N_{PCG} \cdot N_{Symbol}} \sum_{n=1}^{N_{PCG}} \sum_{k=1}^{N_{Symbol}} \left(\frac{\left(\sum_a \sum_j \sqrt{E_c E_p PG} \left| \alpha^{(n,k,a,j)} \right|^2 \right)^2}{\sum_a \sum_j E_p \left| \alpha^{(n,k,a,j)} \right|^2 N_t^{(n,k,a,j)}} \right), \qquad (3.2.1.1\text{-}2)$$

where n is the index of PCG/slot in a frame, k is the index of symbols within a PCG/slot, N_{Symbol} is the number of symbols in a PCG/slot, a is the index of antennas in a receiver, and j is the index of paths in an antenna. E_c is the transmitted chip energy of traffic channel, E_p is the transmitted chip energy of pilot channel, and

PG is the processing gain defined by the ratio of traffic channel data rate to chip rate. $\alpha^{(n,k,a,j)}$ is the attenuation factor in the *j*-th path of the *a*-th antenna for the *k*-th received coded symbol in the *n*-th PCG/slot. $N_t^{(n,k,a,j)}$ is the total variance of noise and non-orthogonal interference component in the *j*-th path of the *a*-th antenna for the *k*-th received symbol in the *n*-th PCG/slot.

2. In HARQ scheme, the total traffic E_b/N_t is calculated as the summation of $Avg(E_b/N_t)$ of transmitted frames. To generate short term FER curve after *R*-th transmissions of frame, the total traffic E_b/N_t after *R*-th transmissions and the frame erasure indicator are recorded only when the frame is erased after (*R*-1)-th transmission.

3. Generate the histogram of short term FER, i.e., the range of E_b/N_t is divided into many bins, and the FER in each bin is computed based on the outputs mentioned in step 1. The size of each bin is 0.5 dB.

3.2.1.2 Target FER

The operating frame erasure rate or FER will be 1% for voice for 1xEV-DV systems.

The FER for the data-only mobile stations shall be:
1. The final FER after retransmissions should be less than or equal to 1%

2. For 1xEV-DV systems, if the R-FCH is present, the pilot level shall be at least as high as what is required to obtain 1% FER on a 9.6 kbps channel, Otherwise the pilot level should not be below the minimum searcher requirements and must be explicitly stated.

3. The pilot level should not be below the minimum searcher requirements - the pilot level used must be explicitly stated in such cases.

4. Error rates on the F-ACK channel should be specified by the proponent of each RL design. The necessary FL cost is to be shown as in section 3.2.2.

3.2.1.3 Channel Models

Channel models shall be as Table 2.2.1-1 and Table 2.2.1-2 specified in section 2.2.1. When modeling the fading on a path between a MS and a sector that is not in that MS's Active Set, an alternative method may be used as follows where a single path is used to replace the channel models in 2.2.1 to speed up the simulation. In this alternative method,

- For Channel Models A through D, the replacement path has Rayleigh fading with a Doppler speed of the corresponding channel model
- For Channel Model E, a non-fading replacement path is used.

The received traffic E_b/N_t for the i-th mobile station at the base station is given by

$$\left(\frac{E_b}{N_t}\right)_{traffic,i} = \frac{PG \cdot E_{c\_traffic,i}\left(\sum_{s=1}^{N_{SL}} g_{i,s} \sum_{r=1}^{N_{ANT}} \sum_{j=1}^{J_i} \left|\gamma_{i,s,r,j}\right|^2\right)^2}{\sum_{s=1}^{N_{SL}} g_{i,s} \sum_{r=1}^{N_{ANT}} \sum_{j=1}^{J_i} \left|\gamma_{i,s,r,j}\right|^2 N_{t,i,s,r,j}} \qquad (3.2.1.3\text{-}1)$$

$$N_{t,i,s,r,j} = E_{c,i} \cdot g_{i,s} \left(\sum_{\substack{f=1 \\ f \neq j}}^{J_i} \left| \gamma_{i,s,r,f} \right|^2 + \left| \lambda_{i,s,r} \right|^2 \right) + \sum_{\substack{m=1 \\ m \neq i}}^{m=N_{AS}(s)} E_{c,m} \cdot g_{m,s} \left(\sum_{n=1}^{J_m} \left| \gamma_{m,s,r,n} \right|^2 + \left| \lambda_{m,s,r} \right|^2 \right) + \sum_{k=N_{AS}(s)+1}^{N_{MS}} E_{c,k} \cdot g_{k,s} \left| \rho_{k,s,r} \right|^2 + N_o \quad (3.2.1.3\text{-}2)$$

where

N_{SL}	The number of the softer handover legs
N_{ANT}	The number of receiving antennas
$N_{AS}(s)$	The number of MS's that communicate with the s-th sector
N_{MS}	The number of total MS in the system
$E_{c,i}$	Transmitted total (traffic + pilot) chip energy of the i-th MS
$E_{c\_traffic,i}$	Transmitted traffic chip energy of the i-th MS
J_i	The number of rays used in a particular channel model of the i-th MS, excluding the ray used to model FURP.
$g_{i,s}$	The average link gain between the i-th MS and the s-th sector. This term includes the MS and BS antenna gains, cable and connector losses, other losses, and shadowing, but not fading.
$\gamma_{i,s,r,j}$	Samples of the fading processes of the j-th recovered rays between the i-th MS and the r-th antenna of the s-th sector. It is used only for the MS's that have the s-th sector as an active set member.
$\lambda_{i,s,r}$	Samples of the fading process between the i-th MS and the r-th antenna of the s-th sector, for the additional ray used to model interference due to the FURP. It is used only for the MS's that have the s-th sector as an active set member.
$\rho_{k,s,r}$	Samples of the fading process between the k-th MS and the r-th antenna of the s-th sector. It is used for the MS's that do not have the s-th sector as an active set member. For Channel Model E, $\rho_{k,s,r}$ = 1.
N_0	The power of AWGN noise

The received traffic E_b/N_t shall be used to determine the frame erasures.

3.2.2 Forward Link Loading

The link level overhead channel performance (FER or BER) curves are to be simulated and submitted in the same set of fading models in 2.2.1. In addition, the loading (percentage of

BS transmit power and percentage of Walsh space) of these FL channels shall be evaluated and submitted for 1xEV-DV systems.

3.2.3 Reverse Link Power Control

The proponents of each reverse link proposal shall completely specify the algorithms for the open and closed loop power control if they are an integral part of the link. This includes the specification of how outer loop power control is accomplished. For a complete analysis of the proposal, it is preferable that the power control algorithms form part of the system simulation. If the proponents feel that it is impractical to simulate the power control algorithm as part of their system simulation and instead, model the effects of power control by changing the statistics of the mobile radio channel, the model shall be completely presented. Moreover, the proponents shall provide analysis to justify that their power control algorithm will modify the mobile radio channel statistics to that used in the system simulation.

For calibration purposes, the proponent shall simulate the following inner loop power control algorithm in system level simulation.

- Calculate the estimated received pilot power $\widetilde{E}_{p,i,s,r,j}$ for the j-th path between the i-th MS and the r-th antenna of the s-th sector, that is defined by

$$\begin{aligned}\widetilde{E}_{p,i,s,r,j} = &\left(\gamma_{i,s,r,j} \cdot \sqrt{g_{i,s} \cdot E_{c\_pilot,i}} + \eta_I(\sqrt{0.5N_{t,i,s,r,j} / N_{chip}})\right)^2 \\ &+ \left(\eta_Q(\sqrt{0.5N_{t,i,s,r,j} / N_{chip}})\right)^2\end{aligned} \quad (3.2.3\text{-}1)$$

  where $E_{c\_pilot,i}$ is the transmitted pilot chip energy of the i-th MS. $\eta_I(x)$ and $\eta_Q(x)$ are independent Gaussian random numbers with zero mean and standard deviation x. N_{chip} is the measurement duration in chips (i.e., 2048 chips for 1-slot duration in 1xEV-DO systems or for 1xEV-DV systems, 1152 in case of PCB puncturing and 1536 in case of no PCB puncturing in the reverse pilot channel). Other notations are specified in section 3.2.1.3.

- Calculate the estimated total received pilot power, that is defined by

$$\left(\widetilde{E}_c / N_t\right)_{pilot,i} = \sum_{s=1}^{N_{SL}} \sum_{r=1}^{N_{ANT}} \sum_{j=1}^{J_i} (\widetilde{E}_{p,i,s,r,j} / N_{t,i,s,r,j}) \,. \quad (3.2.3\text{-}2)$$

Compare the $\left(\widetilde{E}_c / N_t\right)_{pilot,i}$ with the target set point in order to make power control command.

For calibration purposes, the proponents shall simulate the following outer loop power control set point algorithm:

- Increase the power control set point by Δ_{up}=0.5 dB if the frame is decoded in error.
- Decrease the power control set point by Δ_{down}dB = Δ_{up} /(1/FER-1) dB, if the frame is correctly decoded

where power control set point is the comparison threshold for the received pilot Ec/Nt at a base station, and Ec is the received pilot energy and Nt is the total interference and thermal noise.

- The adjustment of the set point in response to the reception of frame i takes place at the beginning of the frame i+2.

For 1xEV-DV systems, the outer loop power control set point algorithm shall be constrained to the following range for voice-only operation:

- Minimum power control set point shall be –26 dB
- Maximum power control set point shall be –17 dB.

3.3 Simulation Requirements

3.3.1 Simulation Flow

There are 19 3-sectored cells in the simulated system. All cells are fully simulated and monitored for the throughput or voice capacity. See Appendix I for details.

3.3.1.1 Soft and Softer Handoff

Calls can be in soft or softer handoff, depending on the path losses and their distributions. When in soft handoff, the target Eb/(Io+No) for a given call is assumed to be the same at all the sectors in the Active Set while the actual received Eb/(Io+No) values differ by the path loss difference. For 1xEV-DV systems, the same voice link level curves are to be used in soft handoff for the reverse link as in the non-soft handoff case. One power control data stream including power control errors is to be used in the case of softer handoff.

3.3.1.2 Simulation Description

1. The system consists of 19 cells, each with an imaginary[10] hexagonal coverage area. Each cell has three sectors.
2. Mobile stations are uniformly dropped into the 19-cell system.[11] A MS dropped within 35 meters of a base station is redropped. One fixed path loss (used for both forward link and reverse link) and one randomized shadowing component (for the forward link and for the reverse link) are computed for each BS-MS pair for the duration of the simulation run. Two independent reverse link fading components are computed once every PCG for the two BS receiving antennas in each sector for each MS.
3. The sector with the smallest total path loss is the serving sector of the MS. When evaluating per-sector data (number of MSs, sector throughput, average throughput

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[10] The actual coverage areas are determined by propagation, fading, antenna patterns, and other factors.

[11] Any method that is equivalent to the following method is allowed: For each of the19 possible cells, the MS is uniformly dropped over an imaginary hexagonal coverage area.

for MSs in a sector, etc.), the MS is counted towards its serving sector. For 1xEV-DV systems, each MS shall be assigned to a sector if the sector is in the Active Set (with a maximum size of three) of the voice user. A sector is in the Active Set of a mobile station only if the FL pilot Ec/Io[12] of that sector computed without fading is above T_ADD (in Table 3.1.2-1) at the mobile station. A MS dropped into the system shall be redropped if there are no members in its Active Set or if the total path loss to the serving sector is above a threshold defined in Table 3.1.2-1. All sectors of the system shall continue to accept mobile stations until the desired fixed number of MSs per sector is achieved everywhere.

4. Fading signal and fading interference are computed from each mobile station into each sector for each PCG or equivalent power control related time interval. See Section 3.2.1.3.

5. All reverse links from every MS to all Active Set members (sectors in communication with the MS) are to be modeled for frame erasure, outage, and throughput.

6. **Applicable to 1xEV-DV Systems Only:** For simulation of systems loaded only with voice-only mobiles, the first run is done with five mobile stations per sector and every run following that is done with five more mobile stations per sector until the outage criteria (defined in Appendix C) or the rise-over-thermal limit is violated, after which every run is done with one less mobile until the outage criteria and the rise-over-thermal limit are satisfied. The maximum number of users per sector of N_{max} (voice capacity) is achieved if N_{max} + 1 users per sector would not satisfy the outage criteria or rise-over-thermal limit and N_{max} users per sector would satisfy both.

7. For simulation of the system loaded with data-only mobiles, the runs are done with the desired number of HTTP and WAP MSs in each sector as well as a number of FTP upload users that arrive at the system according to a pseudo-random arrival process. To evaluate the system capacity, the inter-arrival parameter controlling the arrival process is decreased until the stopping criteria in step 9 below are met

8. **Applicable to 1xEV-DV Systems Only:** With voice-only and data-only mobile stations in the same simulated system, the number of voice-only mobile stations is fixed at $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ per sector for the center cell. The numbers of HTTP and the WAP MSs per sector are also fixed, as specified in Section 4.2.1.2. Throughput from all data users at all Active Set members is to be counted for each category separately.

9. Run-stopping criterion: The increase of FTP upload user arrival rate is stopped when one of the following conditions happens:
   A. For 1xEV-DV systems, voice-only users in the whole system as a group are in outage according to Appendix C.
   B. Fairness criteria are violated for FTP.

[12] The forward link pilot Ec/Io is the ratio of the pilot energy per chip and the sum of thermal noise density and power spectral density of all forward link sectors in the system, assuming all sectors are transmitting at full power.

C. Packet call throughput criteria are violated for FTP.

D. Delay criteria are violated, in one of the traffic categories: HTTP, WAP, TCP ACK and Gaming.

E. FER criterion for data users in the whole system is violated.

F. The rise-over-thermal limitation is violated. (For this purpose, the rise-over-thermal statistics are collected from all sectors of the system. The RoT outage is computed from those system-wide statistics.). System throughput is valid from simulations runs with none of these possible violations.

G. The system becomes unstable. In other words, the average rate of FTP users entering the system is larger than the average rate of FTP users exiting the system.

Note: To simplify the simulation, only two-way or three-way handoff is used. A mobile station on a two-way or three-way soft handoff counts as one user on the sector with the smallest total path loss in the Active Set.

3.3.2 Outputs and Performance Metrics

3.3.2.1 General Output Matrices

The following matrices shall be generated and included in the evaluation report.

1. All link-level results used in system-level simulator.
2. The histogram of C/I that a base station observed for all users in system-level simulation, where the C/I includes path loss and shadowing.
3. The performance of any estimator(s) or predictor(s) that are required by the proposal. For instance, if a channel predictor is used in the proposal, the proponent(s) shall provide the details of the predictor/estimator, its bias, and standard deviation.
4. The proponent shall justify the performance of any reverse channels that are not simulated by the link-level or system-level simulator. For example, if the system-level simulator does not simulate the reverse signaling channels, the proponent companies shall specify the performance of these channels and justify their claims (see section 2.1.5).
5. In order to evaluate the impact of the proposed reverse or forward link modifications to the system performance, the proponent shall specify all additional resources that are consumed in the forward and reverse links. These resources are not limited to the additional overhead channels needed to support the reverse link. It should also encompass parts of existing channels that are consumed (e.g., layer 3 messaging on an existing channel). Furthermore, the proponent shall present an analysis of the system impact due to the consumption of these resources. [Pending decision on dynamic/static simulation]
6. The histogram and CDF of rise-over-thermal in 1.25 ms or $1.6\overline{6}$ ms (depending on system) time resolution for all sectors.

3.3.2.2 Data Services and Related Output Matrices

The following statistics related to data traffics shall be generated and included in the evaluation report

1. **Data throughput per sector**. The data throughput of a sector is defined as the number of information bits per second that a sector can receive successfully from all data users it serves, provided that all data users satisfy certain fairness criterion, including fairness in terms of per user throughput as well as delay, and that certain number of voice users (for 1xEV-DV systems) can be maintained with certain GOS.

2. **Averaged packet delay per sector**. The averaged packet delay per sector is defined as the ratio of the accumulated delay for all packets for all users served by the sector and the total number of packets. The delay for an individual packet is defined as the time between when the packet enters the queue at transmitter and the time when the packet is received successively by the base station. If a packet is not successfully delivered by the end of a run, its ending time is the end of the run.

3. **The histogram 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 delivers during a simulation run and the simulation time. Note that this definition can be applied to all data users.

4. **The histogram of packet call throughput for users with packet call arrival process.** The packet call throughput of a user is defined as the ratio of the total number of information bits that an user successfully delivers and the accumulated delay for all packet calls for the user, where the delay for an individual packet call is defined as the time between when the first packet of the packet call enters the queue at transmitter and the time when the last packet of the packet call is successively received by the receiver.

5. **The histogram of averaged packet delay per user**. The averaged packet delay is defined as the ratio of the accumulated delay for all packets for the user and the total number of packets for the user. The delay for a packet is defined as in 2. Note that this definition is applicable to a data user without packet call arrival process.

6. **The histogram of averaged packet call delay for users with packet call arrival process**. The averaged packet call delay is defined as the ratio of the accumulated delay for all packet calls for the user and the total number of packet calls for the user. The delay for a packet call is defined as in 4. Note that this definition is applicable to a user with packet call arrival process.

7. **The scattering plot of data throughput per user vs. the distance from the user's location to its serving sector.** In case of SHO or sector switching, the distance between the user and the closest serving sector shall be used. The data throughput for a user is defined as in 3.

8. **The scattering plot of packet call throughputs for users with packet call arrival processes vs. the distance from the users' locations to their serving sectors.** In case of SHO or sector switching, the distance between the user and the closest

serving sector shall be used. The packet call throughput for a user is defined as in 4.

9. **The scattering plot of averaged packet delay per user vs. the distance from the mobile's location to its serving sector.** In case of SHO or sector switching, the distance between the user and its closest serving sector shall be used. The averaged packet delay per user is defined as in 2.

10. **The scattering plot of averaged packet call delays for users with packet call arrival processes vs. the distance from the mobiles' locations to their serving sectors.** In case of SHO or sector switching, the distance between the user and its closest serving sector shall be used. The averaged packet call delay per user is defined as in 4.

11. **The scattering plot of data throughput per user vs. its averaged packet delay**. The data throughput and averaged packet delay per user are defined as in 3 and 2, respectively.

12. **The scattering plot of packet call throughputs for users with packet call arrival processes vs. their averaged packet call delays**. The packet call throughput and averaged packet call delay per user are defined as in 4.

In order to understand more easily these definitions, some mathematical formula and figures are provided in Appendix D.

The channel model and speed of a data user are randomly chosen according to the pre-determined distributions.

3.3.2.3 1xEV-DV Systems Only

The applicable statistics required by the output matrices specified in Appendix M shall be generated and presented for the purpose of evaluation. In addition, the following statistics shall also be reported.

3.3.2.3.1 Voice Services and Related Output Matrices

The following statistics related to voice traffics shall be generated and included in the evaluation report.

1. **Voice capacity**. Voice capacity is defined as the maximum number of voice users that the system can support within a sector with certain maximum outage probability. The details on how to determine the voice capacity of a sector are described in Appendix C.

2. **The histogram of voice data rates (for a frame) per user and for all users**.

3. **The scattering plot of the outage probability vs. the distance from the mobile to the serving sector.** In case of soft hand-off (SHO), the distance from the mobile to the closest serving sector shall be used.

4. **The curve of outage indicator vs. time for each user.** The outage indicator equals to one when the voice user is in outage, and zero otherwise. The speed, channel model and the distance of the voice user to the serving sector shall also

be included in the curve. In case of SHO, the distance from the mobile to the closest serving sector shall be used.

5. **The outage probability for each user.** Note that this value can be calculated from the curve described in previous item.

The channel model and speed of a voice user are randomly chosen according to the pre-determined distributions specified in Table 2.2.1-1.

3.3.2.3.2 Mixed Voice and Data Services

In order to fully evaluate the performance of a proposal with mixed data and voice services, simulations are repeated with different loads of voice users. The following outputs shall be generated and included in the evaluation report.

1. The following cases shall be simulated: no voice users (i.e., data only), voice users only (i.e., number of voice users equal to voice capacity), and $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ voice users with data users, where N_{max} is the voice capacity.
2. For each of the above case, all corresponding output matrices defined previously are generated, whenever it is applicable.

In addition, the following output shall also be generated and included in the evaluation report:

1. A curve of sector data throughput vs. the number of voice users is generated, where the sector data throughput is defined as above.

3.3.2.4 Mixed Rev. 0 and Rev. A Mobiles (1xEV-DO Systems Only)

For proposals that support backward compatibility with 1xEV-DO (Rev. 0) ATs, the following mixed Rev. 0 and Rev. A user tests shall be simulated using the standard system layout and channel mix specified in section 2.2.1:

1. 8 full-buffer Rev. 0 ATs and 8 full-buffer Rev. A ATs per sector.
2. 16 Rev. 0 full-buffer ATs per sector.
3. 16 Rev. A full-buffer ATs per sector.

The applicable statistics required by the output matrices specified in Appendix M shall be generated and presented for the purpose of evaluation, as well as the associated statistics required for the evaluation report specified in 3.3.2.2. In addition, the following statistic shall be reported:

i. Avg_TP_Rev0_Mix: Average throughput of Rev. 0 ATs in scenario 1.

ii. Avg_TP_RevA_Mix: Average throughput of Rev. A ATs in scenario 1.

iii. $$\left(\frac{Avg\_TP\_Rev0\_Mix}{Avg\_TP\_Rev0\_Only}, \frac{Avg\_TP\_RevA\_Mix}{Avg\_TP\_RevA\_Only}\right):$$

where Avg_TP_Rev0_Only denotes the average throughput per AT from scenario 2, Avg_TP_Rev_A_Only denotes the average throughput per AT from scenario 3.

3.3.2.5 Link Level Output

All link level curves depicting the Eb/(No+Io) (from both receiving antennas of the sector) vs. FER generated from reverse link level simulator are to be provided.

3.4 Calibration Requirements

Proponents and evaluators of any proposed RL design shall follow the following calibration steps to ensure a consistent evaluation process.

3.4.1 Link Level Calibration

All link-level data (in the form of Excel spreadsheet) shall be submitted. One set of data for use by all evaluators.

3.4.2 System Level Calibration

Each evaluator of the reverse link proposal(s) shall submit the following calibration data for the simulator.

3.4.2.1 1xEV-DV System Calibration

The capacity of the RL shall be simulated for voice-only users. The simulation environment and assumptions shall be the same as those defined herein. Under these assumptions the capacity of RL N_{max} shall be computed and compared for voice-only users according to the definition of outage in Appendix C and the rise-over-thermal criteria. Besides the capacity number N_{max}, the outages and the rise-over-thermals shall be included for various system loads, from 5 mobiles per sector to N_{max} + 3 MSs per sector with an increment of 5 mobiles or less.

The following cases shall be simulated for a Rev. C system with F-FCH scheduling loaded with data-only MSs. Scheduler operation is described in Appendix Q.

- A single full buffer MS in each sector of the whole system in each of the fading conditions at the following two locations, where the location of the MS relative to the sector is the same across the sectors:
  1. On the line between the sector and its closest neighboring cell at 0.3125 km from the cell. (1/8 the way to the neighbor cell);
  2. On the line between the sector and its closest neighboring cell at 1.25 km from the cell (half way between these two cells).
  Shadowing is zero dB for all path loss computations for this step.

Two full-buffer MSs positioned at the two fixed locations in the preceding step, respectively, in each sector in the system. These MSs in the whole system for a simulation run shall have the same fading model and the results for each of the fading models shall be submitted. Shadowing is zero dB for all path loss computations for this step.

3.4.2.2 1xEV-DO System Calibration

The throughput performance for 1xEV-DO Rev. 0 reverse-link (RL) shall be simulated with data-only users. The simulation shall be done for 4 and 16 full-buffer access terminals (ATs) per sector[13]. The DRC gain shall be –1.5 dB for non-handoff ATs and –3 dB for ATs in soft-handoff as mentioned in Table 3.1.2-1. For the purpose of calibration, the target rise-over-thermal (ROT) threshold that is used to set the RAB shall be 5 dB[14]. The received ROT shall be updated per slot as the maximum per antenna ROT processed through a first-order IIR filter with a time constant of 13.3333... ms (8 slots). The data-rate transition probabilities in Table 3.4.2.2-1 shall be used with the default 1xEV-DO RL MAC algorithm:

Table 3.4.2.2-1 Default 1xEV-DO RL MAC Transition Probabilities

Transition009k6_019k2	0.5020
Transition019k2_038k4	0.2510
Transition038k4_076k8	0.1255
Transition076k8_153k6	0.0314
Transition019k2_009k6	0.0314
Transition038k4_019k2	0.0627
Transition076k8_038k4	0.1255
Transition153k6_076k8	1.0000

Unless otherwise specified, all channel structures and attributes shall follow the default setting in 1xEV-DO Specification in document 3GPP2 C.S0024, Version 4.0. For the specified mix of the five standard channel models in Section 2.2.1, the statistics required by the output matrices specified in appendix M shall be generated and presented for the purpose of evaluation, as well as the associated statistics required for the evaluation report specified in Section 3.3.2.2.

3.5 1xEV-DO Baseline Simulation Procedures

3.5.1 Access Terminal Requirements and Procedures:

1xEV-DO Revision 0 access terminal (AT) shall follow procedures specified in 1xEV-DO Specification in document 3GPP2 C.S0024, version 4.0 unless otherwise specified below:

1. Default setting of DRC channel gain: The DRC channel shall be modeled as a continuous channel with a gain of –1.5 dB w.r.t. pilot for terminals that are not in

[13] The choice of 16 ATs is used only for calibration purpose. It does not serve as an indication of the maximum attainable 1xEV-DO RL capacity.

[14] The purpose of this threshold is for calibration purpose only. This may or may not meet the requirement of exceeding 7 dB ROT less than 1% of the time.

soft-handoff (DRC Length = 2 slots) and –3 dB w.r.t. pilot for terminals that are in soft-handoff (DRC Length = 4 slots).

2. Rate transition rules: The data rate of each AT shall be adjusted according to the default 1xEV-DO RL MAC rules and the received RAB information using the rate transition probabilities specified in Table 3.4.2.2-1.

3. The effect of RL ACK channel shall be ignored due to its negligible impact on RL capacity.

4. For baseline purposes, the procedure for determining whether a specific data-rate is power-limited for the AT shall be carried out using the default MAC algorithm by assuming a fixed transmit power headroom of 3 dB. The estimate for pilot transmit power used in this process shall be the received pilot power averaged over the most recent frame (16 slots).

3.5.2 Access Network Requirements and Procedures:

1xEV-DO Revision 0 access network (AN) shall follow procedures specified in 1xEV-DO Specification in document 3GPP2 C.S0024, version 4.0 unless otherwise specified below:

1. The target rise-over-thermal (ROT) threshold that is used to set the RAB bit shall be set to meet the criterion that the percentage of time the received ROT above 7 dB does not exceed 1% over the entire simulation run for simulation involving 16 full-buffer users per sector. This specific ROT threshold shall be used for all simulation scenarios specified in 3.5.3.

2. The received ROT shall be measured and updated per slot as the maximum per antenna ROT processed through a first-order IIR filter with a time constant of 13.3333… ms (8 slots).

3. The base-station compares the measured ROT (at the output of the IIR filter) with the target threshold every frame (16 slots). If the ROT exceeds the threshold, the RAB bit status shall be set to busy, and otherwise it shall be set to not busy. The updated RAB is transmitted over one frame (16 slots) to all ATs within the sector.

4. The impact of imperfect RAB channel shall be modeled in the network simulation using a RAB decoder with dynamic FL modeling. The dynamic modeling shall assume the same multipath and Doppler channel model on RL and FL. The fading process on FL and RL shall be assumed as independent.

3.5.3 Simulation Procedures

The simulation shall be carried out using the following settings based on the layout configuration and method specified in section 3.3[15]:

1. 16 full-buffer ATs per sector.

2. 4 full-buffer ATs per sector.

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[15] Settings 3, 4, and 5 apply standard traffic mixes specified in 4.2.1.2.

3. 4 HTTP users, 10 WAP users, 3 MNG users, and FTP users arrive in a Poisson manner described in section 4.2.
4. 8 HTTP users, 20 WAP users, 6 MNG users, and FTP users arrive in a Poisson manner describe in section 4.2.
5. No HTTP, WAP or MNG users, and FTP users arrive in a Poisson manner described in section 4.2.

For each setting, the statistics according to the output matrices in appendix M shall be generated and presented as well as the associated statistics required for the evaluation report specified in section 3.3.2. The simulation time for each run in scenarios 3, 4, and 5 shall be 100 seconds (60000 slots). FTP user arrival rate shall be determined in the same way as that specified in section 3.3.1.2. In addition, the link-level PER vs. Eb/Nt curves used in the network simulator shall be presented. The link-level curve shall be run with a target FER of 5% or less.

4 TRAFFIC SERVICE MODELS

4.1 Forward Link Services

4.1.1 Service Mix (1xEV-DV Systems Only)

A configurable fixed number of voice calls are maintained during each simulation run. Data sector throughput is evaluated as a function of the number of voice users supported. The following cases shall be simulated: no voice users (i.e., data only), voice users only (i.e., the number of voice users equals to voice capacity), and average $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ voice users per sector plus data users, where N_{max} is the voice capacity defined in Appendix C.

The data users in each sector shall be assigned one of the four traffic models: WAP (56.43%), HTTP (24.43%), FTP (9.29%), near real time video (9.85%), with the respective probabilities in parentheses.

4.1.2 TCP Model

Since FTP and HTTP use TCP as their transport protocol, a TCP traffic model is introduced to more accurately represent the distribution of TCP packets for the FTP and HTTP traffic models described in the next sections.

The TCP connection set-up and release protocols use a three-way handshake mechanism as described in Figure 4.1.2-1 [16]. The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size and the receiver window size. After the connection establishment is complete, the transfer of data starts in slow-start mode with an initial congestion window size of 1 segment. The congestion window increases by one segment for each ACK packet received by the sender regardless of whether the packet is correctly received or not, and regardless of whether the packet is out of order or not. This results in an exponential growth of the congestion window. This process is illustrated in Figure 4.1.2-2.

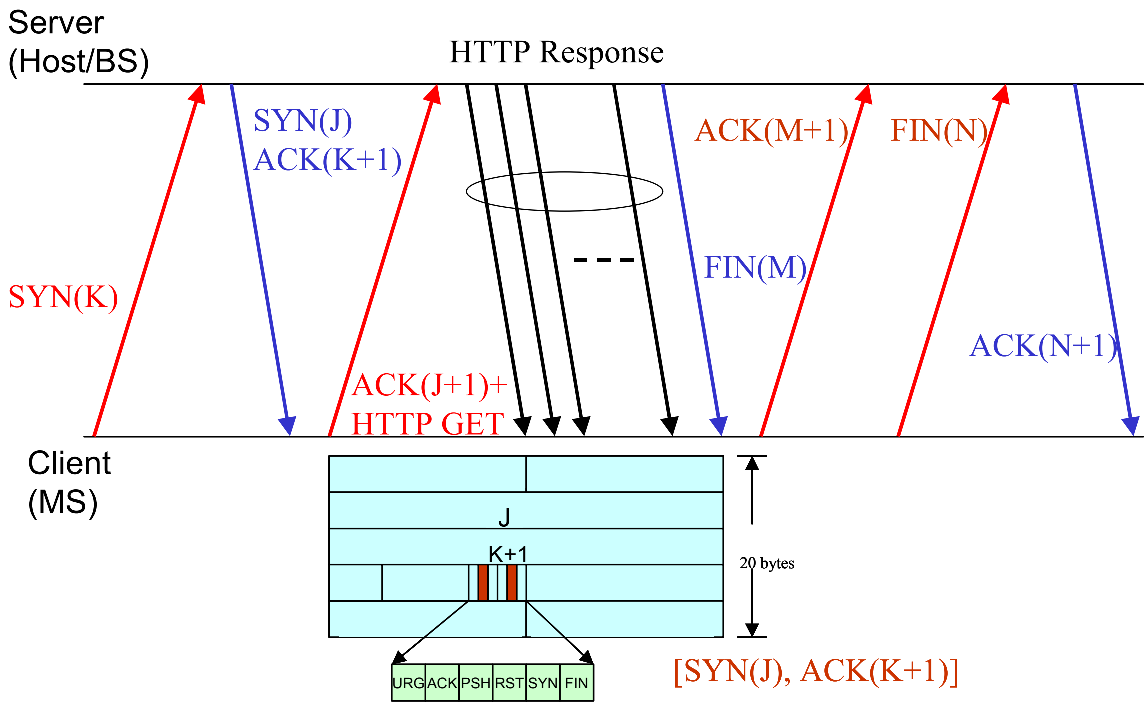

Figure 4.1.2-1 Control Segments in TCP Connection Set-up and Release

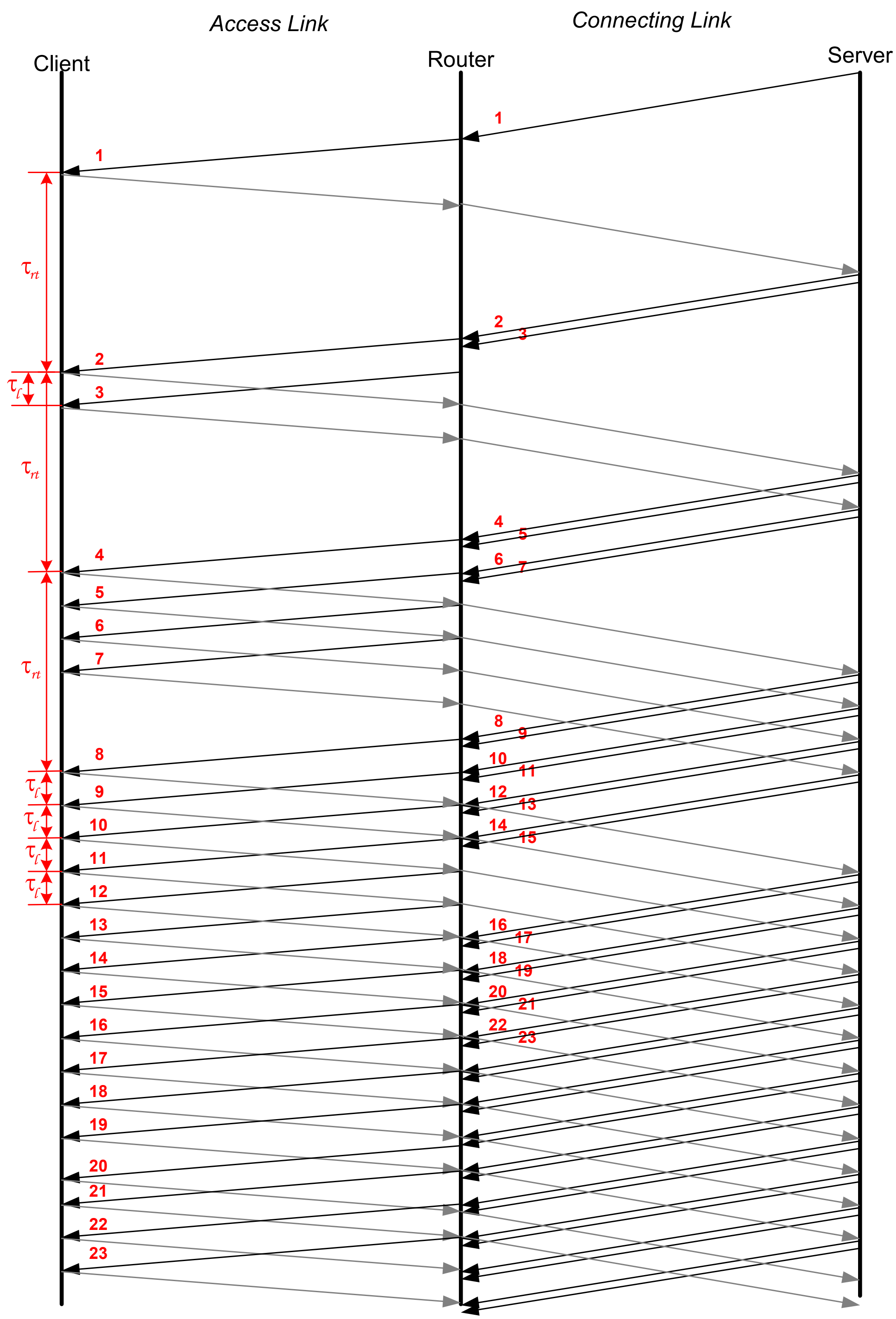

Figure 4.1.2-2 TCP Flow Control During Slow-Start; τ_l = Transmission Time over the Access Link; τ_{rt} = Roundtrip Time

The round-trip time in Figure 4.1.2-2, τ_{rt}, consists of two components:

$$\tau_{rt} = \tau_c + \tau_l \qquad (4.1.2\text{-}1)$$

where τ_c = the sum of the time taken by an ACK packet to travel from the client to the server and the time taken by a TCP data segment to travel from the server to the base station router; τ_l = the transmission time of a TCP data segment over the access link from the base station router to the client.

In this model of TCP, τ_c is modeled as an exponentially distributed random variable with a mean of 50 ms; τ_l is determined by the available access link throughput. Also, it should be mentioned that the detailed specifics of congestion control and avoidance has not been modeled; only the slow-start part has been modeled. It is also assumed that, the receiver window size is very large and hence is not a limitation.

From Figure 4.1.2-2, it can be observed that, during the slow-start process, for every ACK packet received by the sender two data segments are generated and sent back to back. Thus, at the base station, after a packet is successfully transmitted, two segments arrive back-to-back after an interval τ_c. Based on this observation, the packet arrival process at the base station for the download of an object is shown in Figure 4.1.2-3. It is described as follows:

1. Let S = size of the object in bytes. Compute the number of packets in the object, $N = \lceil S/(MTU-40) \rceil$. Let W = size of the initial congestion window of TCP[16].

[16] Compressed TCP header (5 bytes from 40 bytes) and PPP framing overhead (7 bytes + ceiling (size(payload + compressed TCP header + 6)/128)) shall be transmitted over the air in addition to the payload (MTU – 40) bytes.

For the 1500 MTU size:

1460 Payload (MTU – 40)

+ 5 Compressed TCP header

+ 7 PPP headers and CRC

+ 12 PPP escaping = ceiling ((1460 + 5 + 6)/128)

1484 bytes above the multiplex sublayer

For the 576 MTU size:

536 Payload (MTU – 40)

+ 5 Compressed TCP header

+ 7 PPP headers and CRC

+ 5 PPP escaping = ceiling ((536 + 5 + 6)/128)

553 bytes above the multiplex sublayer

2. If N>W, then W packets are put into the queue for transmission; otherwise, all packets of the object are put into the queue for transmission in FIFO order. Let P=the number of packets remaining to be transmitted. If P=0, go to step 6.

3. Wait until a packet of the object in the queue is transmitted over the access link.

4. Schedule arrival of next two packets (or the last packet if P=1) of the object after an interval of τ_c. If P=1, then P=0, else P=P-2.

5. If P>0 go to step 3.

6. Preserve PW = N+W, as the size of the congestion window to be used by persistent TCP connections.

7. Return.

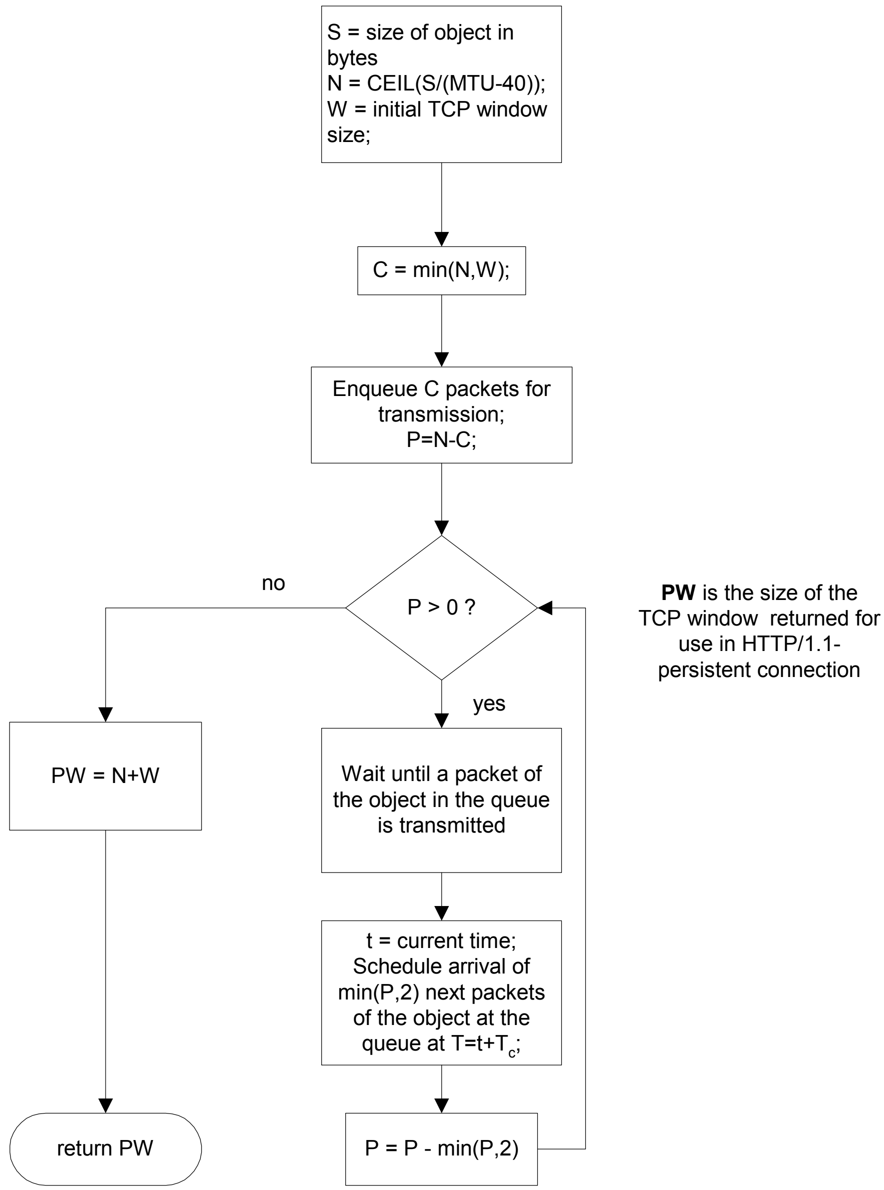

Figure 4.1.2-3 Packet Arrival Process at the Base Station for the Download of an Object Using TCP; PW = the Size of the TCP Congestion Window at the End of Transfer of the Object; $T_c=\tau_c$ (Described in Figure 4.1.2-2)

4.1.3 HTTP Model

4.1.3.1 HTTP Traffic Model Characteristics

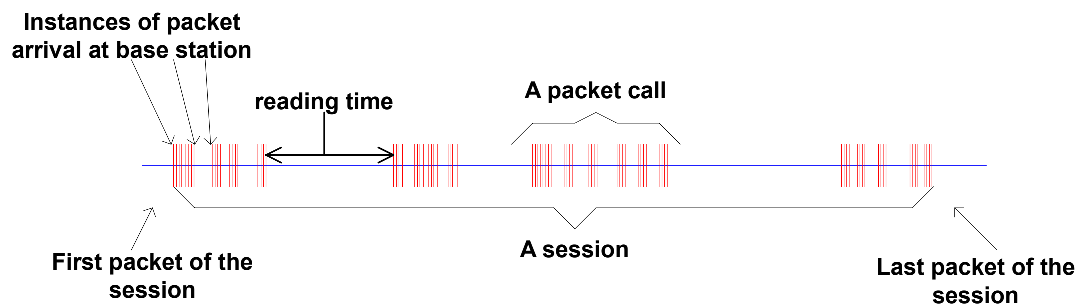

Figure 4.1.3-1 Packet Trace of a Typical Web Browsing Session

Figure 4.1.3-1 shows the packet trace of a typical web browsing session. The session is divided into ON/OFF periods representing web-page downloads and the intermediate reading times. In Figure 4.1.3-1, the web-page downloads are referred to as packet calls and are denoted as such in [18]. 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 Figure 4.1.3-2. Unlike a packet session, the ON/OFF periods within a packet call are attributed to machine interaction rather than human interaction. As an example, consider a typical web-page from the Wall Street Journal (WSJ) Interactive edition depicted in Figure 4.1.3-3. This web-page is constructed from many individually referenced objects. A web-browser will begin serving a user's request by fetching the initial HTML page using an HTTP GET request. After receiving the page, the web-browser will parse the HTML page for additional references to embedded image files such as the graphics on the tops and sides of the page as well as the stylized buttons. The retrieval of the initial page and each of the constituent *objects* is represented by ON period within the packet call while the parsing time and protocol overhead are represented by the OFF periods within a packet call. For simplicity, the term "page" will be used in this paper to refer to each packet call ON period. As a rule-of-thumb, a page represents an individual HTTP request explicitly initiated by the user. The initial HTML page is referred to as the "main object" and the each of the constituent objects referenced from the main object are referred to as an "embedded object".

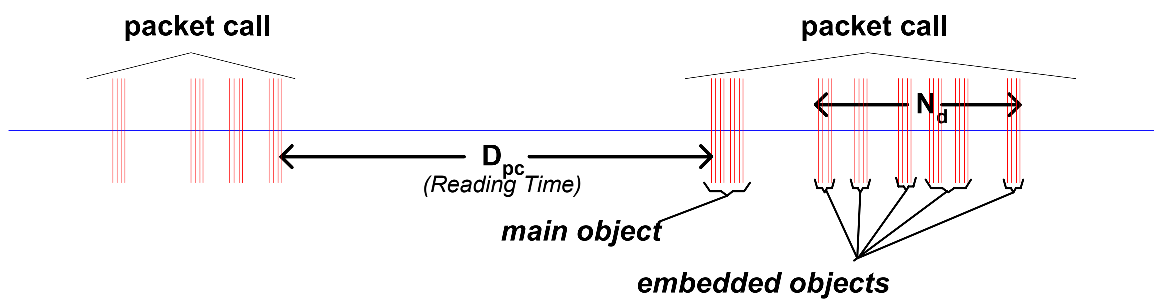

Figure 4.1.3-2 Contents in a Packet Call

Figure 4.1.3-3 A Typical Web Page and Its Content

The parameters for the web browsing traffic are as follows:

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

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[7,8,11,12,13], 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.

In HTTP/1.0, a distinct TCP connection is used for each of the main and embedded objects downloaded in a web page. Most of the popular browser clients download the embedded objects using multiple simultaneous TCP connections; this is known as *HTTP/1.0-burst mode transfer*. The maximum number of such simultaneous TCP connections, N, is configurable; most browsers use a maximum of 4 simultaneous TCP connections. If there are more than N embedded objects, a new TCP connection is initiated when an existing connection is closed. The effects of slow-start and congestion control overhead of TCP occur on a per object basis.

In HTTP/1.1, persistent TCP connections are used to download the objects, which are located at the same server and the objects are transferred serially over a single TCP connection; this is known as *HTTP/1.1-persistent mode transfer*. The TCP overhead of slow-start and congestion control occur only once per persistent connection.

4.1.3.2 HTTP Traffic Model Parameters

The distributions of the parameters for the web browsing traffic model were determined based on the survey of the literature on web browsing traffic characteristics [12,13]. These parameters are described in Table 4.1.3.2-1[17]

[17] Truncated lognormal distribution for the main and embedded object size parameters shall be used. The truncation points are as follows:

For main object: Maximum = 2,000,000 bytes, minimum = 100 bytes

For embedded objects: Maximum = 2,000,000 bytes, minimum = 50 bytes

Table 4.1.3.2-1 HTTP Traffic Model Parameters

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

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

From the literature [13], it was found that, most of the web pages are downloaded in HTTP/1.0-burst mode or HTTP/1.1-persistent mode. From the statistics presented in these literatures, it was concluded that, a 50%-50% distribution of the web pages between HTTP 1.0-burst mode and HTTP 1.1-serial mode will closely approximate the web browsing traffic behavior in the Internet in the short term (in the time-frame of 1xEV-DV and 1xEV-DO deployments). Also, based on some of the studies on packet size properties in the Internet [10,19], it was observed that, the MTU sizes most prominently used by the TCP connections in the Internet are 576 bytes and 1500 bytes. A distribution of 24%-76% of all web pages for transfer using an MTU of 576 bytes and 1500 bytes[18] along with the TCP control segments of 40 bytes (as described in Figure 4.1.2-1) will closely approximate the packet size characteristics for the downlink. Thus, the web traffic generation process can be described as in Figure 4.1.3-4.

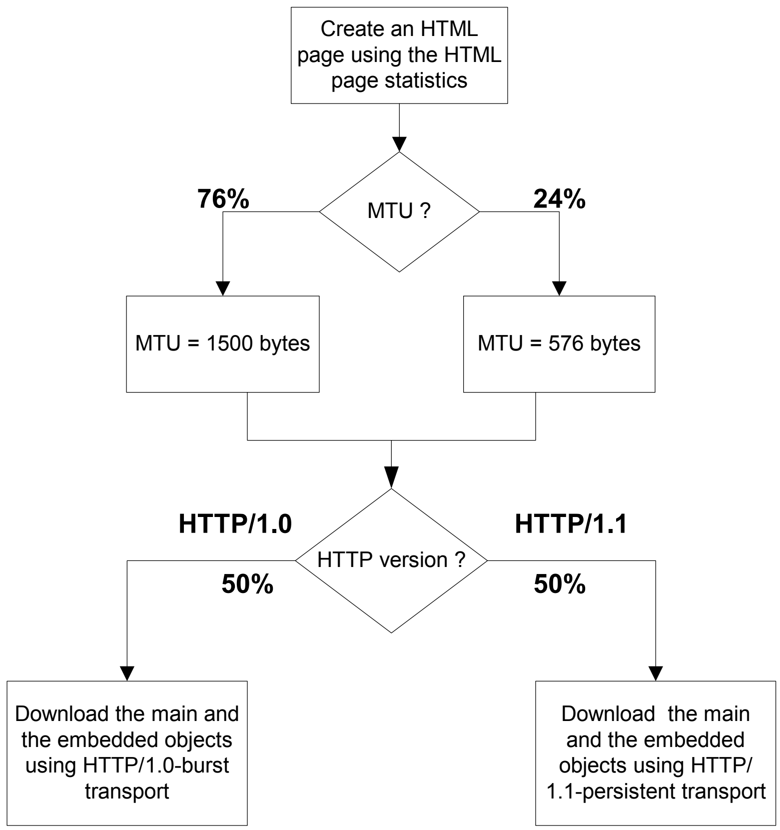

Figure 4.1.3-4 Modeling a Web Page Download

In the rest of this section, the packet arrival models are described which approximate the packet arrival process on the downlink in HTTP/1.0-burst mode and HTTP/1.1-persistent mode.

4.1.3.2.1 Packet Arrival Model for HTTP/1.0-Burst Mode

The download of the web page is modeled as follows:

1. The main object is downloaded with a new TCP connection whose initial congestion window size is 1.
2. The embedded objects are partitioned into groups of four objects; each group is called a *composite object*.
3. A composite object, made of b (1<=b<=4) embedded objects, is transferred using a new TCP connection whose initial window size is b.

[18] The MTU size remains fixed for a packet call or page. In other words, all objects in a page (both the main object and the embedded objects) are transferred using the same MTU size.

4. The transfer of the 1st composite object is initiated $T_p+\tau_c$ second after the transmission of the main object is completed. The transfer of subsequent composite object begins τ_c second after the transmission of the previous composite object is completed.

The transfer of the main object, or of a composite object made of *b* embedded objects, is modeled as shown in Figure 4.1.3-5. The process is described as follows:

1. Let S = size of the object in bytes. If it is the main object, set the initial TCP window size m=1; if it is a composite object, set m=b, where b=the number of embedded objects in the composite object.
2. Put m 40-byte SYN+ACK packets for the connection establishment into the queue for transmission and wait until it is transmitted.
3. Transmit another set of m 40-byte packets [9].
4. Begin transfer of the object (using the flowchart of Figure 4.1.2-3) τ_c second later with an initial TCP window size m.
5. When transfer of the object is completed, transmit m 40-byte FIN segment to initiate closing of connection.
6. Wait for τ_c second.
7. Transmit m 40-byte ACK packets to complete connection close.

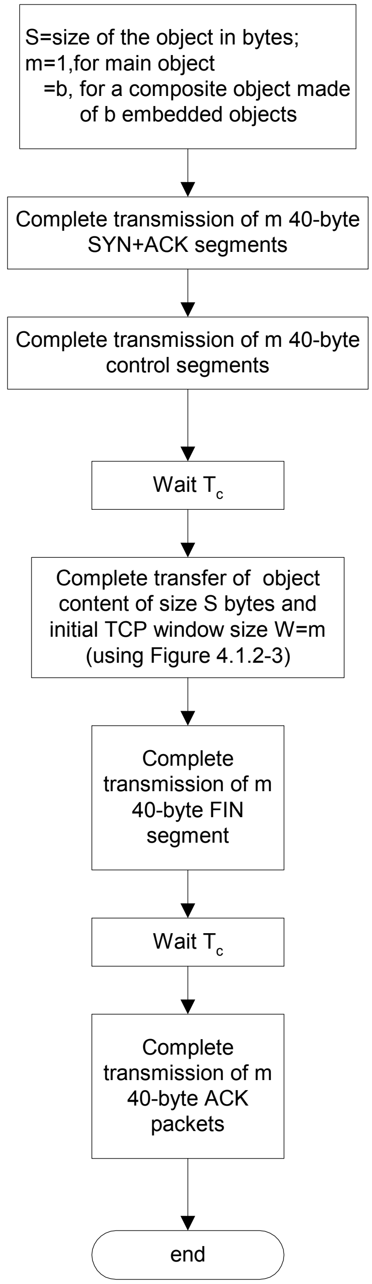

Figure 4.1.3-5 Download of an Object in HTTP/1.0-Burst Mode

4.1.3.2.2 Packet Arrival Model for HTTP/1.1-Persistent Mode

The download of the web page is modeled as follows:

1. The main object is transferred using a new TCP connection whose initial window size is 1; this TCP connection and its congestion window size is preserved.
2. All of the embedded objects are transferred, serially one after another, using the preserved TCP connection.

3. The transfer of embedded objects begins $T_p+\tau_c$ second after the completion of the transfer of the main object.
4. The TCP congestion window size is preserved at the end of transmission of each embedded object.
5. Transfer of next embedded object begins τ_c second after the transmission of the previous embedded object is completed.
6. At the end of transfer of all embedded objects, the client initiates release of the TCP connection.

The transfer of the main and embedded objects are shown in Figure 4.1.3-6. It is described as follows:

1. Let S = size of the main object in bytes.
2. Transmit a 40-byte SYN+ACK packet for the connection establishment.
3. Transmit another 40-byte packet.
4. Begin transfer of the main object (using the flowchart of Figure 4.1.2-3) τ_c second later with an initial TCP window size 1.
5. At the end of transfer of the main object, preserve the congestion window size PW;
6. Wait for T_p sec (the parsing time).
7. If N_d = 0 (there are no embedded objects), go to step 16.
8. Let I=1.
9. Transmit four 40-byte packets.
10. If I>N_d (the number of embedded objects) go to step 16
11. Wait for τ_c second (for initiation of transfer of embedded object).
12. S = size of the I-th embedded object in bytes.
13. Begin transfer of the I-th embedded object (using the flowchart in Figure 4.1.2-3) with initial TCP congestion window size of PW (the preserved congestion window).
14. At the completion of transfer of the I-th object, preserve the congestion window size PW.
15. I = I+1; go to step 10 to initiate transfer of next embedded object.
16. Wait for τ_c (delay for client initiated TCP close).
17. Transmit a 40-byte ACK packet for connection close.
18. Transmit a 40-byte FIN packet for connection close.

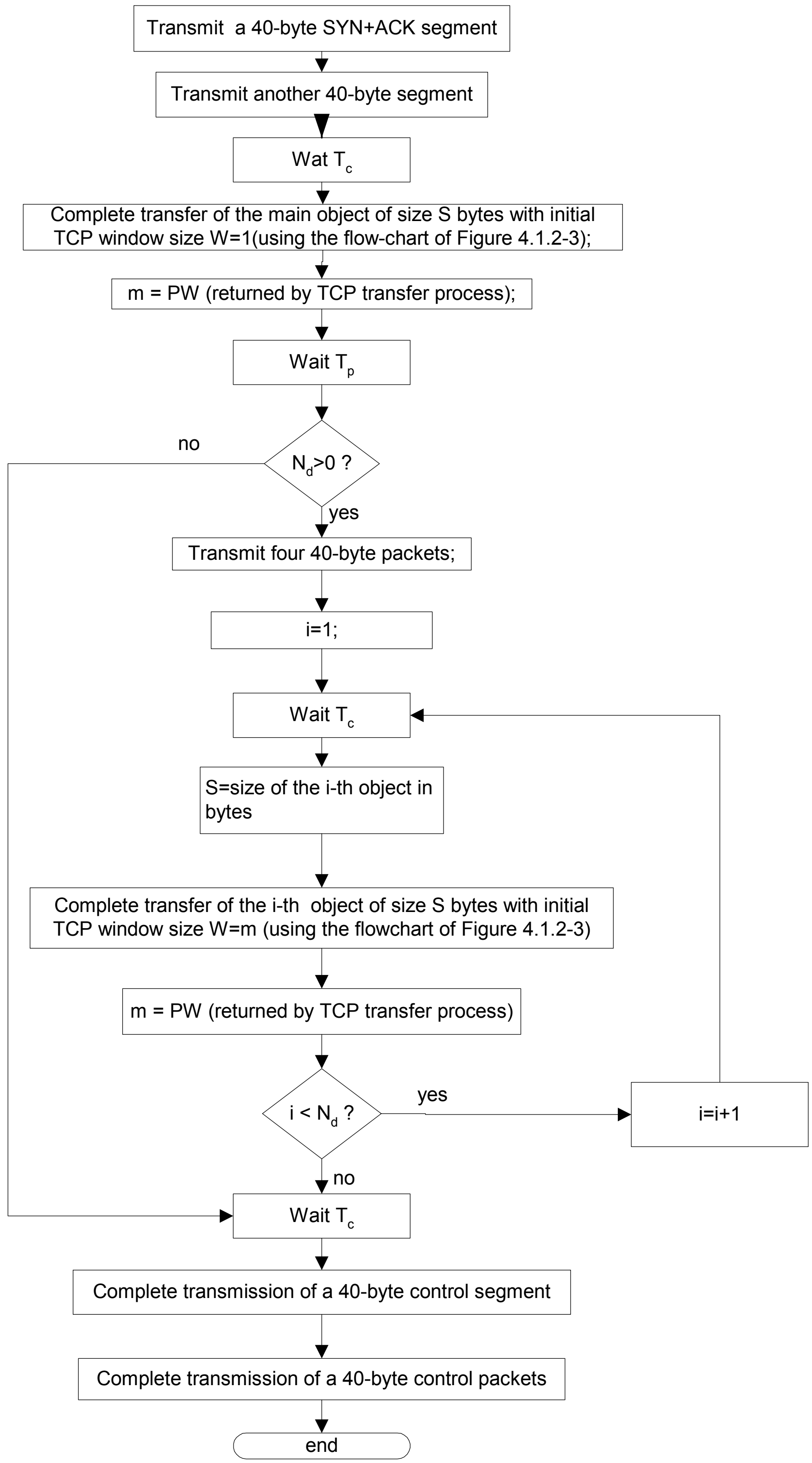

Figure 4.1.3-6 Download of Objects in HTTP/1.1-Persistent Mode

4.1.4 FTP Model

4.1.4.1 FTP Traffic Model Characteristics

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:

1. S : the size of a file to be transferred
2. 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 model of TCP connection described in section 4.1.2 will be used to model the FTP traffic. The packet trace of an FTP session is shown in Figure 4.1.4-1.

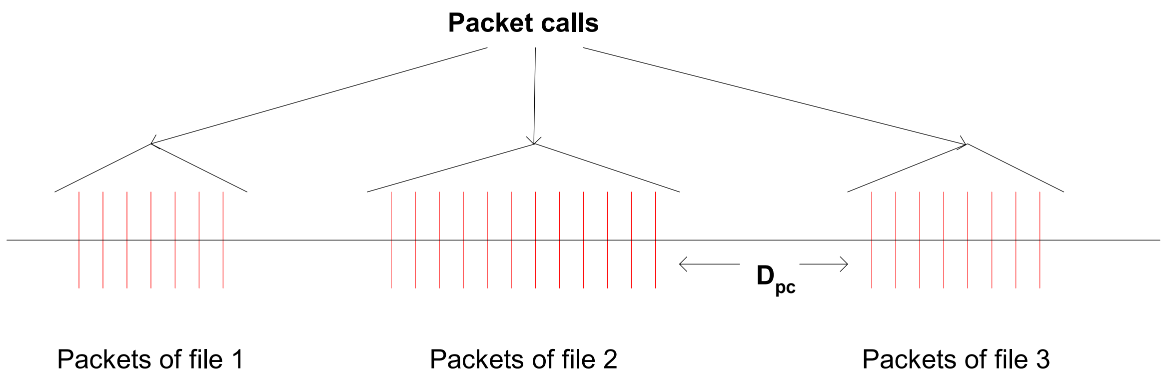

Figure 4.1.4-1 Packet Trace in a Typical FTP Session

4.1.4.2 FTP Traffic Model Parameters

The parameters for the FTP application sessions are described in Table 4.1.4.2-1.

Table 4.1.4.2-1 FTP Traffic Model Parameters

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

Based on the results on packet size distribution [10,19] (described in section 4.1.3.2), 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. For each file transfer a new TCP connection is used whose initial congestion window size is 1 segment (i.e. MTU). The packet arrival process at the base station is described by the flowchart in Figure 4.1.2-3 (in which the object represents the file being transferred). The process for generation of FTP traffic is described in Figure 4.1.4-2.

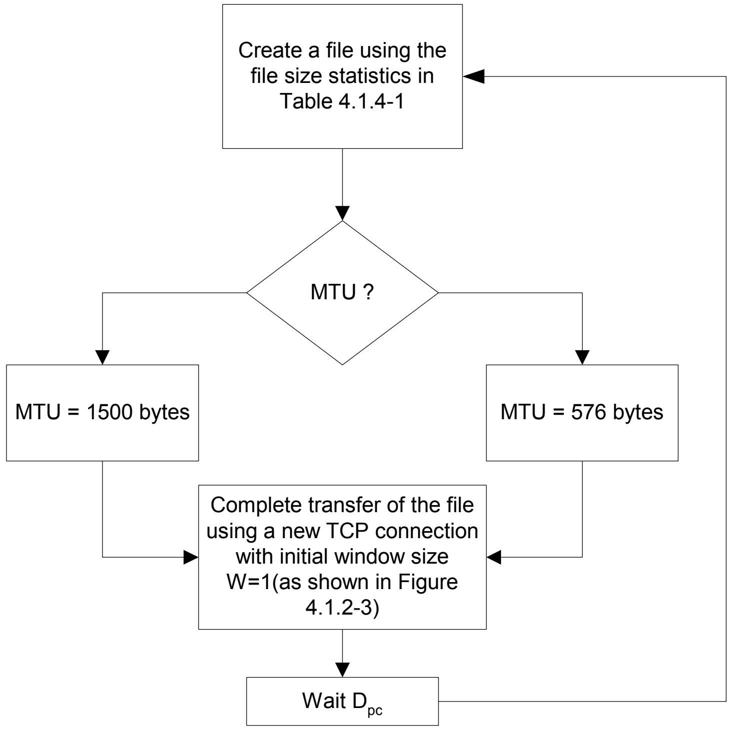

Figure 4.1.4-2 Model for FTP Traffic

4.1.5 WAP Model

Each WAP request from the browser is modeled as having a fixed size and causes the WAP server to send back a response with an exponentially distributed response time. The WAP gateway response time is the time between when the last octet of the request is sent and when the first octet of the response is received from the WAP server. The response itself is composed of a geometrically distributed number of objects, and the inter-arrival time between these objects is exponentially distributed. Once the last object is received, the exponentially distributed reading time starts, and it ends when the WAP browser generates the next request. Figure 4.1.5-1 illustrates the data flow for the WAP traffic model and Table 4.1.5-1 describes the distribution of the model parameters.

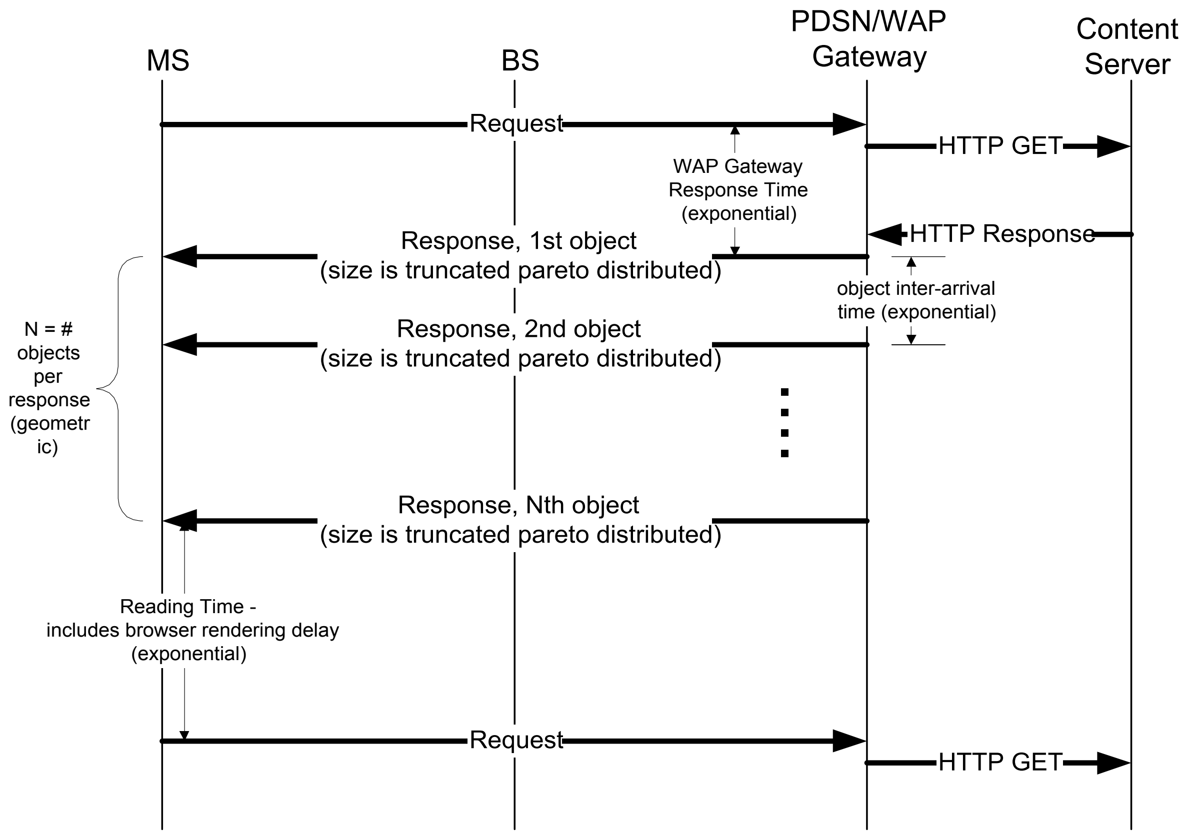

Figure 4.1.5-1 Packet Trace for the WAP Traffic Model

During the simulation period, the model assumes that each WAP user is continuously active, i.e., making WAP requests, waiting for the response, waiting the reading time, and then making the next request.

Table 4.1.5-1 WAP Traffic Model Parameters

Packet based information types	**Size of WAP request**	**Object size**	**# of objects per response**	**Inter-arrival time between objects**	**WAP gateway response time**	**Reading time**
Distribution	Deterministic	Truncated Pareto (Mean= 256 bytes, Max= 1400 bytes)	Geometric $P(k) = 0.5^k$, $k>=1$	Exponential	Exponential	Exponential
Distribution Parameters	76 octets	K = 71.7 bytes, $\alpha = 1.1$	Mean = 2	Mean = 1.6 s	Mean = 2.5 s	Mean = 5.5 s

4.1.6 Near Real Time Video Model

The following section describes a model for streaming video traffic on the forward link.

Figure 4.1.6-1 describes the steady state of video streaming traffic from the network as seen by the base station. Latency of starting up the call is not considered in this steady state model.

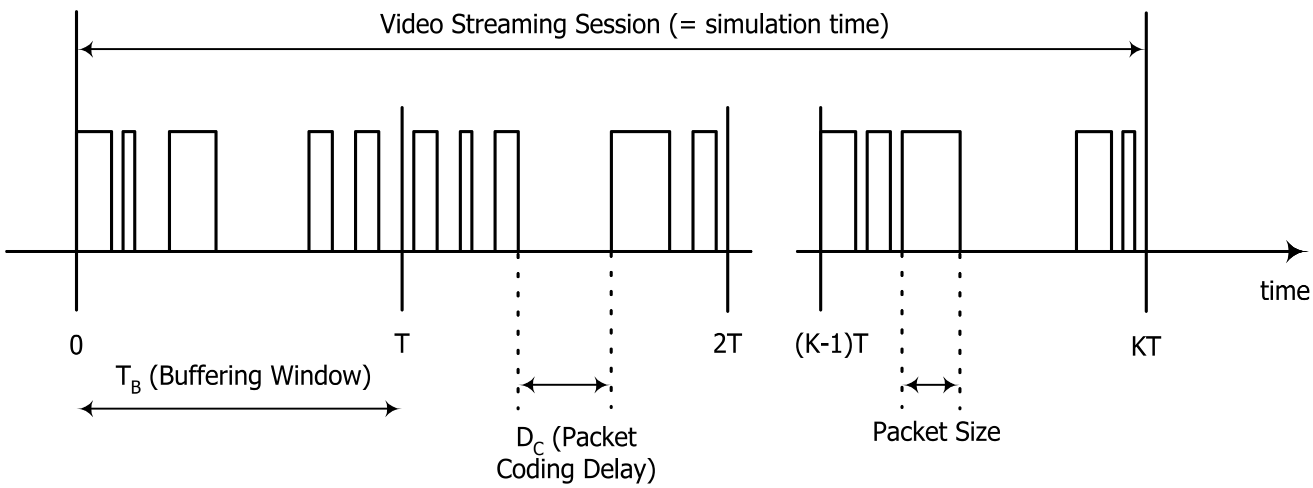

Figure 4.1.6-1 Video Streaming Traffic Model

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 (fps). Each frame is decomposed into a fixed number of slices, each transmitted as a single packet. The size of these packets/slices is distributed as a truncated Pareto. Encoding delay, Dc, at the video encoder introduces delay intervals between the packets of a frame. These intervals are modeled by a truncated Pareto distribution.

The parameter T_B is the length (in seconds) of the de-jitter buffer window in the mobile station used to guarantee a continuous display of video streaming data. This parameter is not relevant for generating the traffic distribution but is useful for identifying periods when the real-time constraint of this service is not met. At the beginning of the simulation, it is assumed that the mobile station de-jitter buffer is full with (T_B x source video data rate) bits of data. Over the simulation time, data is “leaked” out of this buffer at the source video data rate and “filled” as forward link traffic reaches the mobile station. As a performance criterion, the 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.

Using a source video rate of 32 kbps, the video traffic model parameters are defined in Table 4.1.6-1.

Table 4.1.6-1 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= 125bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution Parameters	100ms	8	K = 20bytes α = 1.2	K = 2.5ms α = 1.2

4.1.7 Voice Model (1xEV-DV Systems Only)

Voice follows a standard Markov source model. The voice activity factor is 0.403 with 29% full rate, 60% eighth rate, 4% half rate, and 7% quarter rate. The corresponding transition probabilities are defined in C.S0025 (TIA/EIA/IS-871).

Voice capacity shall be obtained based on three outage criteria (Short Term FER 15%, Per User Outage 1%, and System Outage 3%) defined in Appendix C.

Ignoring other effects, energy consumption per bit for the four data rates in the voice service is the same. That implies, for example, the required traffic Ec for the 4800 bps is 3 dB less than that required for 9600 bps.

Voice is modeled as C.S0002 (IS-2000) RC3 and RC4, power controlled, with and without STS. Four possible combinations of RC3 and RC4, non-transmit-diversity, and STS shall be simulated independently. Power control subchannel power consumption shall be specified. The forward power control subchannel is sent at the same power level as the full rate (9.6 kbps) frame when the MS is not in soft handoff. That subchannel is sent at a level 3 and 5 dB higher than the 9.6 kbps level when in two-way and three-way soft handoff, respectively. The reverse link power control subchannel is sent on the Reverse Pilot Channel in a TDM fashion with the same power level as the pilot. Any other arrangements in the proposals shall be properly accounted for with clear definitions for power consumption. The power consumption by all reverse link overhead channels, including the Reverse Pilot Channel shall be specified and justified.

4.1.8 Delay Criteria

Except for WAP users, all 1xEV-DV and 1xEV-DO packet data (HTTP, FTP, or near real time video) users shall satisfy the following delay criterion: no more than 2% of the users shall get less than 9600 bps throughput (goodput). The throughput will be the user's packet call throughput, except in the case where there is no arrival process (FTP users are persistent) in which case it will be the throughput averaged over the simulation time.

Neal real time video users shall also satisfy the performance criteria defined in section 4.1.8.1.

WAP users shall satisfy the delay criterion defined in section 4.1.8.2.

4.1.8.1 Performance Criteria for Near Real Time Video

Video playout buffers introduce a delay between receipt of frames and the frame playout. This absorbs variations in the data arrival pattern and permits a continuous playout of the frames. The actual design of these playout buffers involves a number of factors (including reset policies when the buffer runs dry) and is specific to the mobile. To avoid modeling such implementation details, we focus on what the BS scheduler must do to generally accommodate this continuous playout. Therefore, the scheduler should transmit an entire video frame within 5 second of receipt of the entire frame (i.e., receipt of the last octet of the last slice of the frame). If a frame exceeds the 5-second requirement, the scheduler discards the remainder of the frame that has not yet been transmitted. The size and arrival statistics for the video frames are defined in section 4.1.6.

Therefore, the performance requirement is that the fraction of video frames that are not completely transmitted within 5 seconds of their arrival at the scheduler shall be less than 2% for each user. All users shall meet the above performance requirement.

4.1.8.2 Delay Criterion for WAP Users

No more than 2% of the users shall get less than 4800 bps throughput (goodput). The throughput will be the user's packet throughput. The packet throughput of a user is defined as the ratio of the total number of information bits that an user successfully receives and the accumulated delay for all packets for the user, where the delay for an individual packet is defined as the time between when the packet enters the queue at transmitter and the time when the packet is received successively by the mobile station. If a packet is not successfully delivered by the end of a run, its ending time is the end of the run. Using the terminology defined in Appendix D, the packet throughput for user(m, n) can be obtained as

$$\text{Packet throughput for user(m,n)} = \frac{\sum_{k=1}^{K(m,n)} \sum_{l=1}^{L(m,n,k)} B(m,n,k,l)}{\sum_{k=1}^{K(m,n)} \sum_{l=1}^{L(m,n,k)} \left(TD(m,n,k,l) - TA(m,n,k,l)\right)} \qquad (4.1.8\text{-}1)$$

4.2 Reverse Link Services

4.2.1 Service Mix (1xEV-DV Systems Only)

A configurable fixed number of voice calls are maintained during each simulation run. Data sector throughput is evaluated as a function of the number of voice users supported. The following cases shall be simulated:

- No voice users (i.e., data only)
- Voice users only
- $\lfloor 0.5N_{max} \rfloor$ or $\lfloor 0.8N_{max} \rfloor$ voice users per sector plus data users.

4.2.1.1 Data Model

The throughput contribution of an individual MS_i is:

$$\text{Ave Throughput for MS}_i = \frac{\text{Correctly Received Bits from MS}_i}{\text{Simulation Time of MS}_i} \quad (4.2.1.1\text{-}1)$$

where

- The simulation time of a WAP, HTTP, or MNG MS is the entire duration of the simulation run.
- The simulation time of an FTP upload mobile station is the duration that mobile station is on the system, excluding the initial setup period.

4.2.1.2 Traffic Model

Data traffic is divided into two categories:

- Forward link supporting traffic: HTTP/WAP requests and TCP ACKs[19]. Among these users, 70% corresponds to WAP users and 30% corresponds to HTTP users. The traffic configurations to be simulated are given in Table 4.2.1.2-1, where N_{max} is the voice capacity per sector, as defined in Appendix C[20].

Table 4.2.1.2-1: Traffic Configurations

HTTP	WAP	MNG	Voice[21]	FTP
0	0	0	0	Poisson arrival
4	10	3	0	Poisson arrival
8	20	6	0	Poisson arrival
4	10	3	$\lfloor 0.5N_{\max} \rfloor$	Poisson arrival
2	5	0	$\lfloor 0.8N_{\max} \rfloor$	Poisson arrival

- Reverse link specific traffic: FTP upload users. FTP Upload users represent mobiles uploading a file with FTP, or sending an email attachment. The arrival of FTP upload users is modeled as a Poisson arrival process with arrival rate λ. The Poisson arrival process is defined per sector. Arrival of a mobile is accounted as a part of the arrival process of the serving sector of that mobile. The sector with the smallest total path loss is the serving sector of the mobile.

[19] The maximum number of users in this class is obtained according to the FL traffic mix that needs support on the RL. The maximum number of users on the FL is assumed to be 42 users (approximately the FL capacity for data-only users, mix of channel models and mix of traffic models). Since WAP+HTTP compose 80.86% of the FL traffic, the maximum number of users to support on RL is 34.

[20] The concept of voice capacity in the current context is not applicable to 1xEV-DO systems.

[21] Traffic configurations requiring voice users are not applicable to 1xEV-DO systems.

4.2.2 TCP Modeling

The TCP connection set-up and release protocols use a three-way handshake mechanism as described in Figure 4.2.2-1.

After the call setup process is completed at time $t=t_0+580$ms, the procedure for MS to set up a TCP session is as follows:

1. MS sends a 47-byte[22] SYNC packet and wait for ACK from remote server.
2. MS starts TCP in slow-start mode (The ACK flag is set in the first TCP segment). The MS starts this process by generating the first TCP packet for FTP transfer. The ACK flag is embedded in this TCP packet along with FTP data.

The procedure for MS to release the TCP session is as follows:

1. MS sets the FIN flag in the last TCP segment. The FIN flag is embedded in the TCP packet along with FTP data.
2. MS receives ACKs for all TCP segments from the remote server and terminates the session (MS leaves the system).

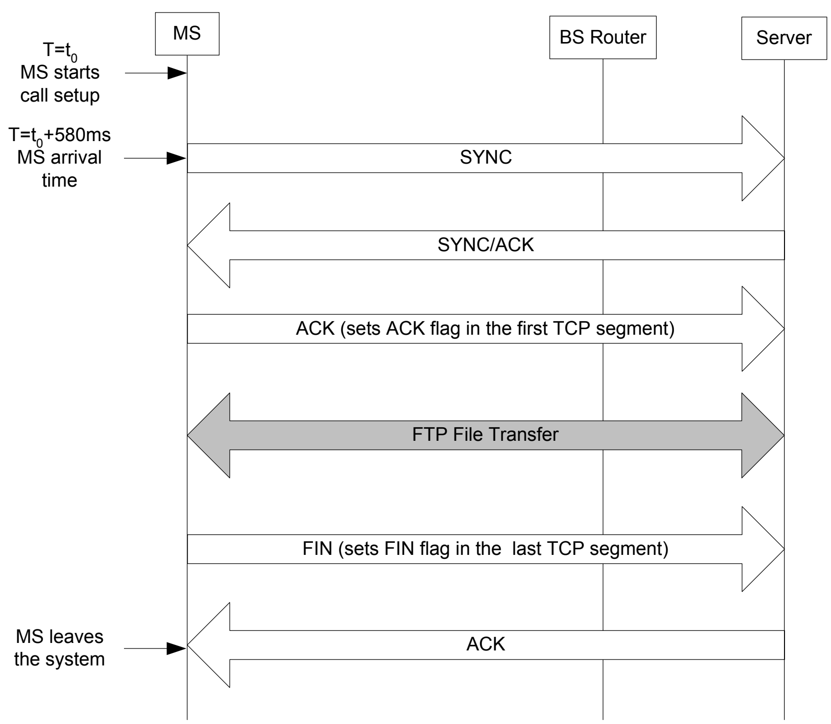

Figure 4.2.2-1: Modeling of TCP three-way handshake

[22] The TCP/IP header of 40 bytes + 7 bytes PPP framing overhead = 47 bytes for the SYNC packet.

The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size of the transmitter and the receiver window size. After the connection establishment is completed, the transfer of data starts in slow-start mode with an initial congestion window size of 1 segment. The congestion window increases by one segment for each ACK packet received by the sender regardless of whether the packet is correctly received or not, and regardless of whether the packet is out of order or not. This results in an exponential growth of the congestion window. This process is illustrated in Figure 4.2.2-2.

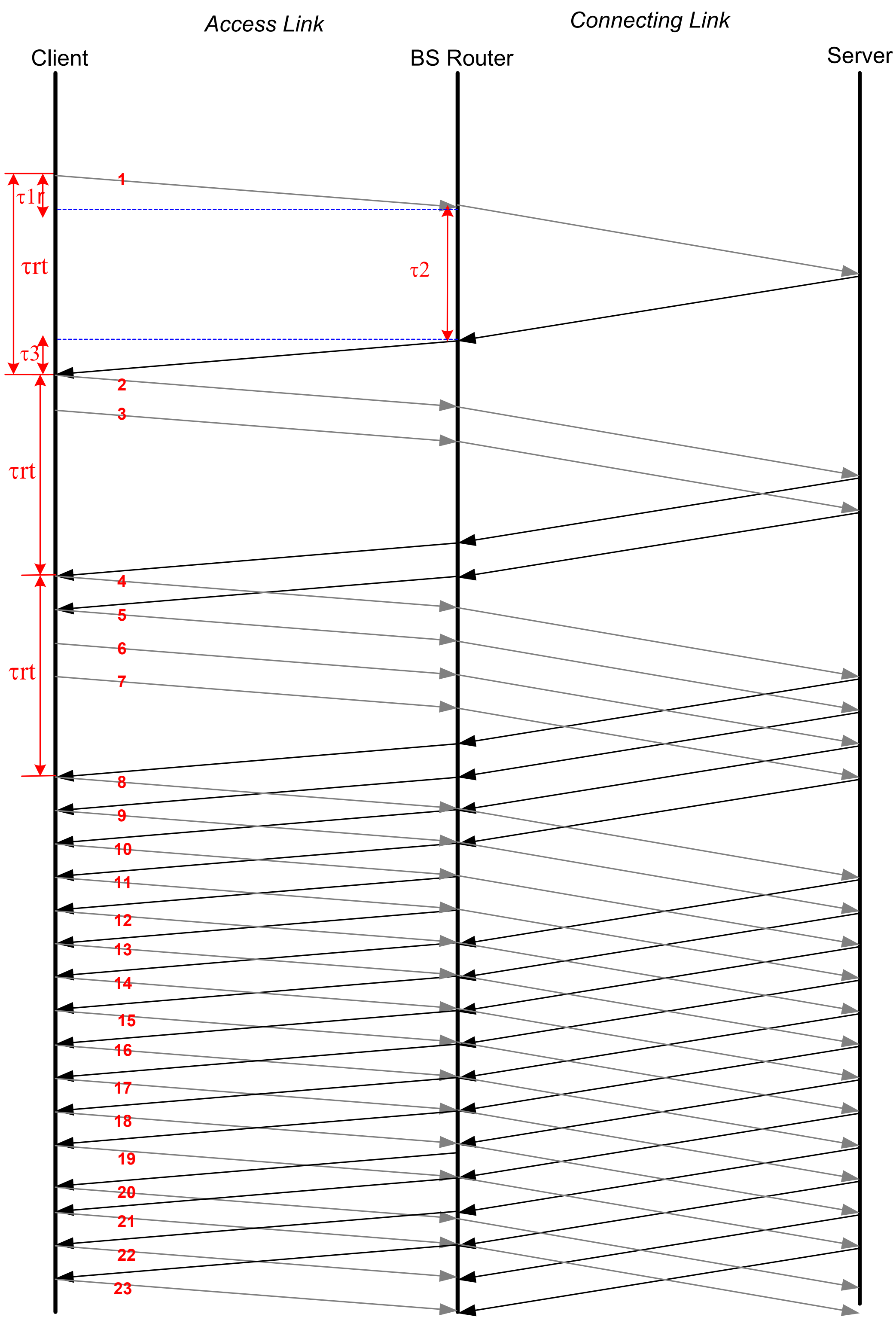

Figure 4.2.2-2: TCP Flow Control During Slow-Start; τ_1 = Transmission Time over the Access Link (RL); τ_{rt} = Roundtrip Time

The round-trip time in Figure 4.2.2-2, $\tau_{rt,}$ consists of two components:

$$\tau_{rt} = \tau_{cr} + \tau_{lr} \tag{4.2.2-1}$$

where τ_{cr} = the sum of the time taken by a TCP data segment to travel from the base station router to the server plus the time taken by an ACK packet to travel from the server to the client; τ_{lr} = the transmission time of a TCP data segment over the access link from the client to the base station router. τ_{cr} is further divided into two components; τ_2 = the time taken by a TCP data segment to travel from the base station router to the server plus the time taken by an ACK packet to travel from the server back to the base station router and τ_3 = the time taken by the ACK packet to travel from the base station router to the client.

τ_{cr} (=τ_2 +τ_3), which accounts for the delay on the connecting link and the TCP ACK transmission on the F-PDCH is modeled as sum of two random variables, τ_2 is modeled as an exponentially distributed random variable while the time to transmit a TCP ACK on the F-PDCH (τ_3) is modeled as a lognormal distributed random variable with the same mean and standard deviation, as defined in 4.2.4.3. The values for the different delay components are given in Table 4.2.2-1.

Table 4.2.2-1 Delay components in the TCP model for the RL upload traffic

Delay component	Symbol	Value
The transmission time of a TCP data segment over the access link from the client to the base station router.	τ_{1r}	Determined by the access link throughput
The sum of the time taken by a TCP data segment to travel from the base station router to the server and the time taken by an ACK packet to travel from the server to the base station router.	τ_2	Exponential distribution Mean=50ms.
The time taken by a TCP data segment to travel from the base station router to the client.	τ_3	Lognormal distribution Mean = 50ms Standard deviation=50ms

From Figure 4.2.2-2, it can be observed that, during the slow-start process, for every ACK packet received by the sender two data segments are generated and sent back to back. Thus, at the mobile station, after a packet is successfully transmitted, two segments arrive back-to-back after an interval $\tau_{cr} = \tau_2 + \tau_3$. Based on this observation, the packet arrival process at the mobile station for the upload of a file is shown in Figure 4.2.2-3. It is described as follows:

1. Let S = size of the FTP upload file in bytes. Compute the number of packets in the file, N = $\lceil S/(MTU\text{-}40) \rceil$. Let W = size of the initial congestion window of TCP[23]. The MTU size is fixed at 1500 bytes
2. If N>W, then W packets are put into the queue for transmission; otherwise, all packets of the file are put into the queue for transmission in FIFO order. Let P=the number of packets remaining to be transmitted beside the W packets in the window. If P=0, go to step 6
3. Wait until a packet of the file in the queue is transmitted over the access link
4. Schedule arrival of next two packets (or the last packet if P=1) of the file after the packet is successfully ACKed. If P=1, then P=0, else P=P-2
5. If P>0 go to step 3
6. End.

---

23 Compressed TCP header (5 bytes from 40 bytes) and PPP framing overhead (7 bytes + ceiling (size(payload + compressed TCP header + 6)/128)) shall be transmitted over the air in addition to the payload (MTU – 40) bytes.

For the 1500 MTU size:

1460 Payload (MTU – 40)

+ 5 Compressed TCP header

+ 7 PPP headers and CRC

+ 12 PPP escaping = ceiling ((1460 + 5 + 6)/128)

1484 bytes above the multiplex sublayer

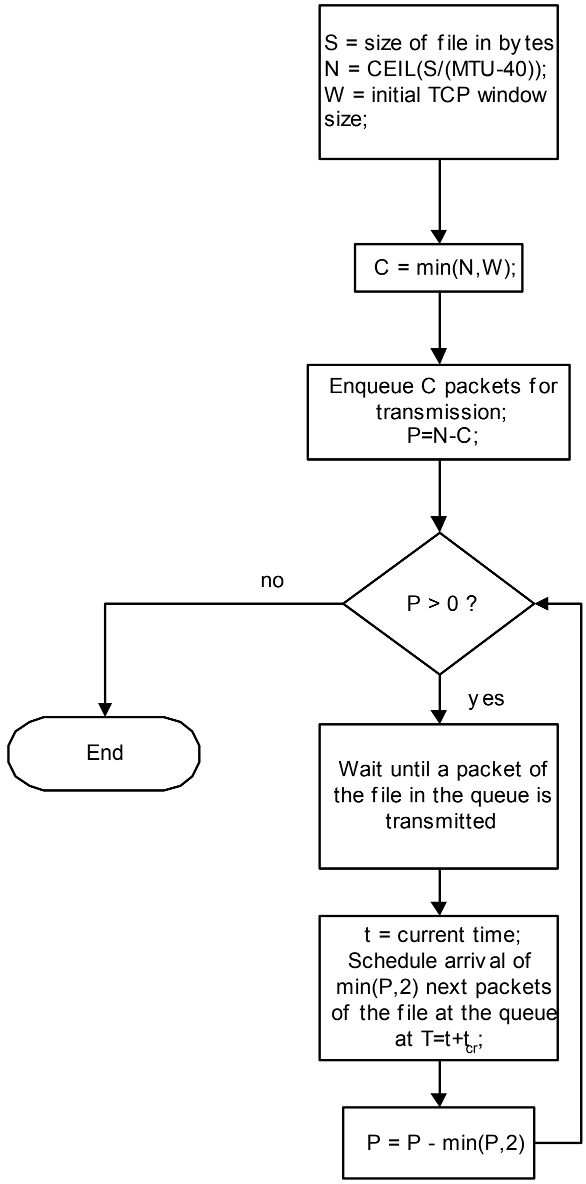

Figure 4.2.2-3 Packet Arrival Process at the mobile Station for the Upload of a File Using TCP

4.2.3 FTP Upload / Email

Since FTP uses TCP as its transport protocol, the TCP traffic model described in Section 4.1.2 is used to represent the distribution of TCP packets for the FTP upload traffic on the RL.

The file upload and email attachment upload are modeled as in Table 4.2.3-1.

Table 4.2.3-1: FTP Characteristics

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

The FTP traffic is simulated as follows:

- At the beginning of the simulation there are 5 FTP users[24] waiting to transmit.
    - Before transmitting, call setup is performed for each user

Afterwards, FTP upload users arrive according to the Poisson arrival process, as defined in Table 4.2.3-1.

- For each new FTP upload user coming into the system, call setup is performed
- Each FTP upload user stays in the system until it finishes the transmission of its file
- After an FTP upload user finishes the transmission of its file, it immediately leaves the system.

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

4.2.4 HTTP Model

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

[24] In order to skip the transient period, the number of 5 initial FTP users is taken to represent the number of users at steady state.

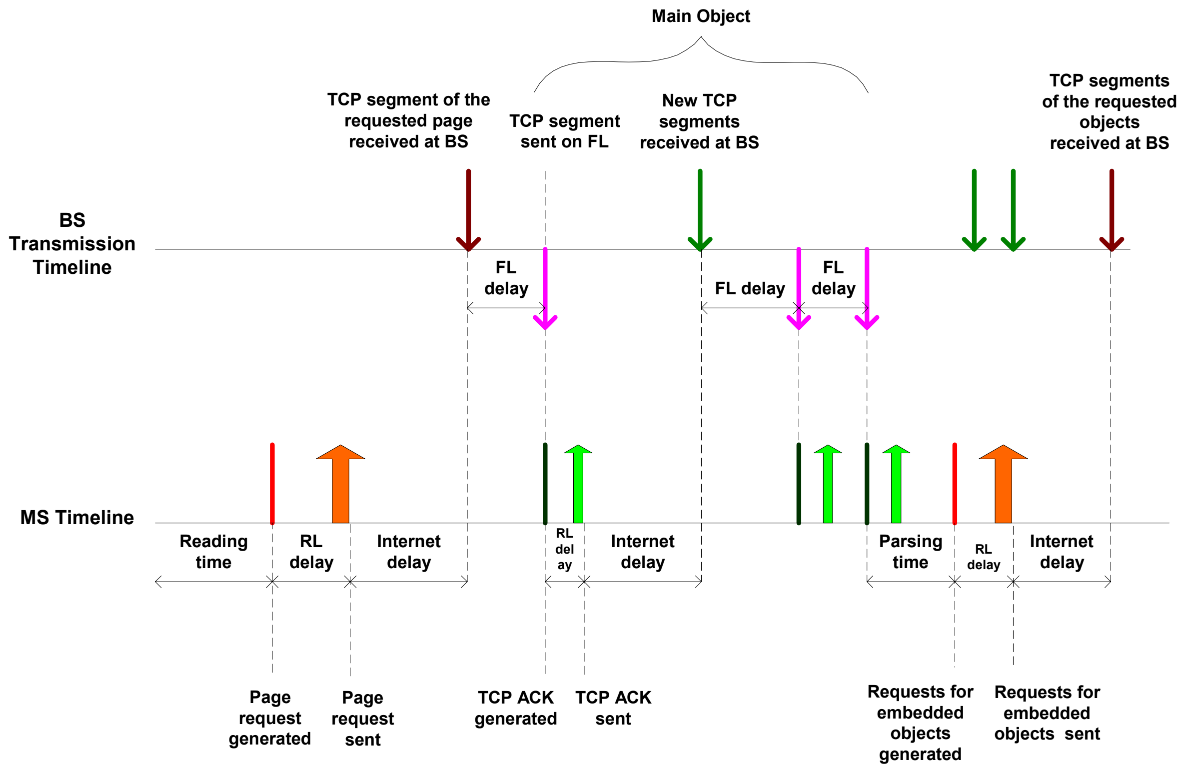

Figure 4.2.4-1: Example of events occurring during web browsing.

4.2.4.1 HTTP Traffic Model Parameters

- Reading time (D_{pc}): modeled as in Table 4.2.4.1-1
- Internet delay (D_I): modeled as an exponentially distributed random variable with a mean of 50ms
- Parsing time (T_p): modeled as in Table 4.2.4.1-1
- RL delay: specific for the implemented system. Includes RL packet transmission delay and scheduling delay (if scheduled)
- FL delay (D_{FL}): defined as the time a TCP segment is first in the queue for transmission until it finishes transmission on forward link. The delay includes transmission delay and forward link scheduling delay. If there are multiple packets, each packet has its own additional contribution to the overall $D_{FL.}$ The model is given in 4.2.4.3.
- Number of TCP segments in the main object (N_M). $N_M = \lceil S_M /(MTU\text{-}40) \rceil$. The main object size, S_M, is generated according to Table 4.2.4.1-1
- Number of TCP segments in embedded object (N_E). $N_E = \lceil S_E /(MTU\text{-}40) \rceil$. The embedded object size, S_E, is generated according to Table 4.2.4.1-1
- Number of embedded objects (N_d). Modeled according to Table 4.2.4.1-1
- HTTP1.1 mode

- The opening and the closing of the TCP connections is not modeled[25]
- HTTP request size = 350 bytes
- Requests for embedded objects are pipelined – all requests are buffered together
- MTU size = 1500 bytes
- ACK size = 12 bytes[26]
- Every received TCP segment is acknowledged.

[25] This does not have much influence since in HTTP1.1 persistent TCP connections are used to download the objects (located at the same server) and the objects are transferred serially over a single TCP connection.

[26] Compressed TCP/IP header (5 bytes from 40 bytes) and HDLC framing and PPP overhead (7 bytes).

Table 4.2.4.1-1: HTTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 9055 bytes Std. dev. = 13265 bytes Minimum = 100 bytes Maximum = 100 Kbytes	If x > max or x < min, then discard and re-generate a new value for x. $f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 1.37, \quad \mu = 8.35$
Embedded object size (S_E)	Truncated Lognormal	Mean = 5958 bytes Std. dev. = 11376 bytes Minimum = 50 bytes Maximum = 100 Kbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 1.69, \quad \mu = 7.53$ If x > max or x < min, then discard and re-generate a new value for x.
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 4.229 Max. = 53	$f_x = \frac{a_k^\alpha}{x^{\alpha+1}}, k \le x < m$ $\alpha = 1.1, k = 2, m = 55$ Note: Subtract k from the generated random value to obtain N_d If x > max, then discard and re-generate a new value for x
Reading time (D_{pc})	Exponential	Mean = 30 sec	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.033$
Initial reading time (D_{ipc})	Uniform	Range [0, 10] s	$f_x = \frac{1}{b-a}, \quad a \le x \le b$ $a = 0, b = 10$
Parsing time (T_p)	Exponential	Mean = 0.13 sec	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 7.69$

4.2.4.2 Packet Arrival Model for HTTP

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

1. Generate an initial reading time D_{ipc}.[27] Wait D_{ipc} seconds.
2. Initiate the TCP window size W=1
3. Generate a request for the main page
4. Wait for the requests to go through the RL and reach the bases station (RL delay):
   - In case these are requests for embedded objects, wait until all requests reach the base station.
5. Generate an Internet delay D_I. Wait D_I seconds.
6. Generate random delays, which define the time instances when each of the TCP segment transmission is completed the FL. The number of these instances is:
   a. For the main page:
      i. At the very beginning of the packet call: 1.
      ii. Afterwards: min(2n, #of outstanding TCP segments on FL), where n is the number of ACKs received in the last physical layer packet (from the step 9.a.i)
   b. For embedded objects:
      i. At the very beginning of the transmission of embedded objects: min(W, $\sum_{i=1}^{N_d} N_E^i$).
      ii. Afterwards: min(2n, #of outstanding TCP segments on FL), where n is the number of ACKs received in the last physical layer packet (from the step 9 a.i.
7. Every time instance of the completed TCP segment transmission on FL generates an ACK on RL
8. Continue RL simulation – when ACK is generated, reduce the number of outstanding TCP packets by 1
9. Examine if the transmission of the very last TCP segment of the HTTP object is completed:
   a. If no:
      i. Proceed with simulation until next ACK or a group of n ACKs within a single physical layer packet is transmitted
      ii. Increase W:=W+n
      iii. Go to step 5
   b. If yes, for main page:

[27] The initial reading time is defined differently from subsequent reading times in order to ensure that all HTTP users finish the reading time within a limited period.

i. Generate T_p (parsing time)

ii. Generate requests for embedded objects

iii. Continue RL simulation - transmit outstanding ACK(s) for the main page and accordingly increment W:=W+n for each group of n ACKs transmitted, until requests for embedded objects are generated

iv. Go to step 4

c. If yes, for embedded objects:

i. Generate D_{pc} (reading time)

ii. Continue RL simulation - transmit outstanding ACK(s) for the embedded objects

iii. Go to step 2 when reading time expires or until all ACKs are transmitted, whichever is longer

4.2.4.3 Forward Link Delay Model for HTTP Users

Forward link delay (D_{FL}) is defined as a time needed for transmission of a TCP segment that is first in the queue for transmission. The delay includes transmission delay and forward link scheduling delay (waiting for other users to use the FTC/F-PDCH). It is modeled according to a distribution obtained from the forward link simulation.

The forward link delay is simulated as follows:

1. The time to transmit a TCP packet is simulated as a lognormal distributed random variable with the same mean and standard deviation
2. The mean and standard deviation of the time to transmit a TCP packet (in PCGs) for a given user has to be computed with the following expression:

$$T = \frac{PS}{DRC} * U * \left(\frac{Spkts}{Pkt}\right) * 800\ PCGs/\sec \qquad \text{(1xEV-DV Systems)} \quad (4.2.4.3\text{-}1)$$

or

$$T = \frac{PS}{DRC} * U * \left(\frac{Spkts}{Pkt}\right) * 600\ slots/\sec \qquad \text{(1xEV-DO Systems)} \quad (4.2.4.3\text{-}2)$$

where,

- PS (Packet Size) is 12,000 bits (for MTU=1500 bytes)
- U (average number of users who have data to transmit) = 6.28 (for data only case)
- $\left(\frac{Spkts}{Pkt}\right)$ (average number of subpackets per packet) = 1.32

3. The ATR (average transmission rate) in the expression above is computed based on the geometry (see below) and channel model of a given user. The DRC for a user is

obtained by linear interpolation/extrapolation between the points in Table 4.2.4.3-1 for the appropriate channel model. If the DRC resulting from extrapolating between the points in Table 4.2.4.3-1 is 0 bps, the user is discarded and replaced.

- Geometry = 1 / ((Ioc+No)/Ior + 0.05)
- For geometry computation, all BTSs are assumed to be transmitting at full power (20 W)
- Ior is the total energy per chip received from the serving sector, which is assumed to be transmitting at max transmit power (20 Watts)
- Ioc is the sum of total energy per chip from all other sectors, each assumed to be transmitting at max transmit power (20 Watts)
- No is the thermal noise spectral density

Table 4.2.4.3-1 Points to obtain the average transmission rate (ATR) given the geometry and channel model of a user

Channel Model A		Channel Model B		Channel Model C		Channel Model D		Channel Model E	
Geometry (dB)	Average Transmission Rate (kbps)	Geometry (dB)	Average Transmission Rate (kbps)	Geometry (dB)	Average Transmission Rate (kbps)	Geometry (dB)	Average Transmission Rate (kbps)	Geometry (dB)	Average Transmission Rate (kbps)
-8	200	-7	280	-6	35	-4	150	-5	290
11	2,250	11	1,500	11	1,100	0	330	3	980
NA	NA	NA	NA	NA	NA	11	1,650	11	2,500

4.2.5 WAP Users

Figure 4.2.5-1 illustrates the data flow for the WAP traffic model, and Table 4.2.5-1 describes the distribution of the model parameters.

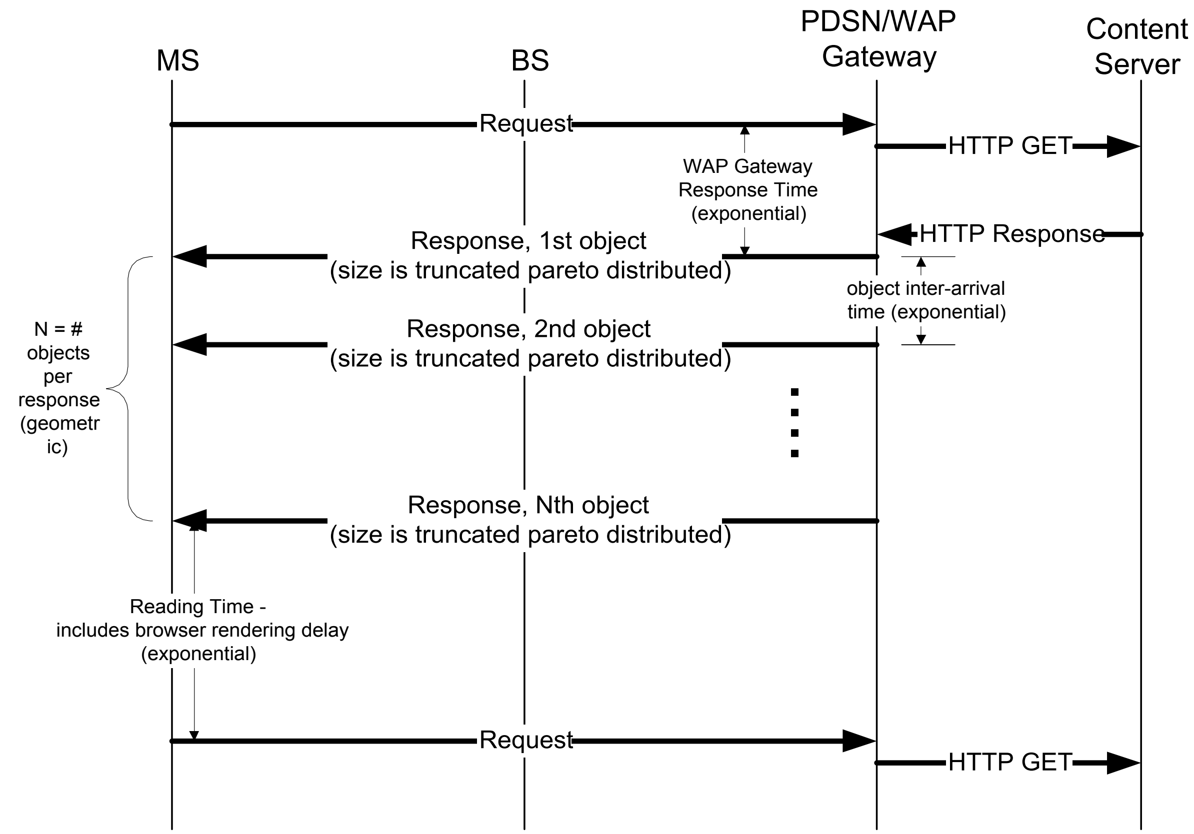

Figure 4.2.5-1: Packet Trace for the WAP Traffic Model

During the simulation period, the model assumes that each WAP user is continuously active, i.e., making a WAP request, waiting for the response, waiting the reading time, and then making the next request.

WAP traffic on the RL simply represents the WAP requests used by the mobile to download the WAP pages.

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

1. Generate the reading time T_{ri}. Wait T_{ri} seconds.
2. Generate N_W (number of objects per response)
3. Initialize $N:=N_W$
4. At the end of the reading time generate WAP request of 76 bytes
5. Generate the gateway response time T_g. Wait T_g seconds.
6. Decrement N:=N-1
7. Examine if N > 0
   - If yes
     - Generate an object inter-arrival time T_{IA}. Wait T_{IA} seconds.

        - Decrement N:=N-1
        - Go to step 7
    - If no
        - Wait T_r (reading time)
        - Go to step 2

Table 4.2.5-1: WAP Traffic Model Parameters

Packet based information types	Size of WAP request	Object size	# of objects per response N_W	Inter-arrival time between objects Tia	WAP gateway response time Tg	Reading time T_r	Initial reading time T_{ri}
Distribution	Deterministic	Truncated Pareto (Mean= 256 bytes, Max= 1400 bytes)	Geometric $P(k) = 0.5^k$, k>=1	Exponential	Exponential	Exponential	Uniform
Distribution Parameters	76 octets	K = 71.7 bytes, α = 1.1	Mean = 2	Mean = 1.6 s	Mean = 2.5 s	Mean = 5.5 s	Range [0, 5] s

4.2.6 Reverse Link Delay Criteria for HTTP/WAP

Reverse link delay for a TCP ACK, defined as the time from the moment the TCP ACK is generated at the mobile station until it is received at the base station, shall not be more than 160 ms 90% of the time.

Reverse link delay for a HTTP request, defined as the time from the moment the HTTP request is generated at the mobile station until it is received at the base station, shall satisfy the following criteria. The CDF curve of the reverse link HTTP request delays shall lie to the left of the curve given by the three points in Table 4.2.6-1.

Table 4.2.6-1 Reverse link delay criteria for HTTP request

HTTP request delay [s]	CDF
1.0	0.8
0.6	0.5
0.4	0.2

Reverse link delay for a WAP request, defined as the time from the moment the WAP request is generated at the mobile station until it is received at the base station, shall not be more than 300 ms 90% of the time.

4.2.7 Mobile Network Gaming Model

This section describes a model for mobile network gaming traffic on the reverse link. Table 4.2.7-1 describes the parameters for the mobile network gaming traffic on the reverse link.

Table 4.2.7-1 Mobile network gaming traffic model parameters

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

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

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

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

4.2.8 Voice Model (1xEV-DV Systems Only)

Voice follows the standard Markov source model. The voice activity factor is 0.403 with 29% full rate, 60% eighth rate, 4% half rate, and 7% quarter rate. The corresponding transition probabilities are defined in C.S0025 (TIA/EIA/IS-871). For the received frame in the system level simulation of voice traffic, lookup_Eb/Nt(rate) is computed in order to lookup the FER on the 9600 bps FER versus Eb/Nt curve for all 4 rates.

Lookup_Eb/Nt(rate) is computed as follows:

lookup_Eb/Nt(rate) = T/P(9600) * 1228800/9600 * Ec/Nt(pilot)

where Ec/Nt(pilot) is the average pilot power received over the frame.

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[30] Ramakrishna, S., Holtzman, J.M., "*A scheme for throughput maximization in a dual-class CDMA system* ", Selected Areas in Communications, IEEE Journal on, Vol. 16 Issue: 6, page 830 –844, Aug. 1998.

[31] F. Kelly, "Charging and rate control for elastic traffic", *European Trans. On Telecommunications*, vol. 8, pp. 33-37, 1997.

[32] L3NQS, "Results of L3NQS Simulation Study", 3GPP2-C50-20010820-011, August 2001.

[33] Doug Reed, "'Modified' Hata path loss model used in 3GPP2 ", 3GPP2 Contribution C30-20040920-012, Motorola.

APPENDIX A: LOGNORMAL DESCRIPTION

The attenuation between a mobile and the transmit antenna of the i-th cell site, or between a mobile and each of the receiving antennas of the i-th cell site is modeled by

$$L_i = k_o D_i^{-\mu} 10^{X_i/10} R_i^2 \quad \text{(A-1)}$$

where D_i is the distance between the mobile and the cell site, μ is the path loss exponent and X_i represents the shadow fading which is modeled as a Gaussian distributed random variable with zero mean and standard deviation σ. X_i may be expressed as the weighted sum of a component Z common to all cell sites and a component Z_i that is independent from one cell site to the next. Both components are assumed to be Gaussian distributed random variables with zero mean and standard deviation σ independent from each other, so that

$$X_i = aZ + bZ_i \text{ such that } a^2 + b^2 = 1 \quad \text{(A-2)}$$

Typical parameters are $\sigma = 8.9$ and $a^2 = b^2 = 1/2$ for 50% correlation. The correlation is 0.5 between sectors from different cells, and 1.0 between sectors of the same cell.

APPENDIX B: ANTENNA ORIENTATION

Antenna Bearing is the angle between the main antenna lobe center and a line directed due east given in degrees. The Bearing Angle increases in a clockwise direction. Figure B-1 below shows the 3-sector 120-degree center cell site with a sector 1 bearing angle of zero degrees.

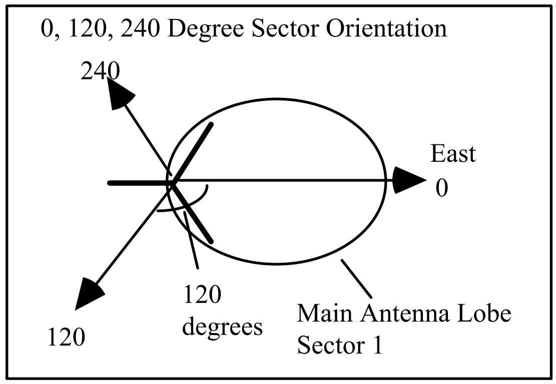

Figure B-1 Center Cell Antenna Bearing Orientation diagram

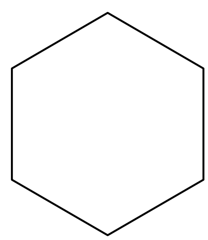

Figure B-2 Orientation of the Center Cell Hexagon

Figure B-2 shows the orientation of the center cell hexagon corresponding the antenna bearing orientation diagram of Figure B-1. The main antenna lobe center directions shall point to the sides of the hexagon. The main antenna lobe center directions of the other 18-surrounding cell shall be parallel to those of the center cell.

Antenna downtilt is the angle between the main antenna lobe center and a line directed perpendicular from the antenna face. As the downtilt angle increases (positively) the antenna main lobe points increasingly toward the ground.

The Antenna gain that results for a given bearing and downtilt angle for a given cell/sector 'i' with respect to a mobile 'k' is characterized by the equations given below (B-1,2). A geometric representation is given in Figure B-3 below.

$$\theta(i,k) = \gamma(i,k) - \xi(i) \quad \text{(B-1)}$$

$$\phi(i,k) = \omega(i,k) - \zeta(i,k) \quad \text{(B-2)}$$

where

$\theta(i,k)$ = angle between antenna main lobe center and line connecting cell 'i' and mobile 'k' in radians in horizontal plane.

$\phi(i,k)$ = angle between antenna main lobe center and line connecting cell 'i' and mobile 'k' in radians in vertical plane.

$\xi(i)$ = antenna bearing for cell 'i' in radians.

$h(i)$ = antenna height for cell 'i' in meters.

$\delta(i)$ = antenna downtilt in radians .

$\zeta(i,k)$ = corrected downtilt angle for given horizontal offset angle ($\theta(i,k)$) in radians.

= ATAN(COS($\theta(i,k)$)TAN($\delta(i)$))

$\omega(i,k)$ = antenna-mobile line of site angle in radians

= ATAN(h(i)/d(i,k)) .

$d(i,k)$ = (mobile(k)_ypos - cell[i]_ypos)^2 + (mobile(k)_xpos - cell[i]_xpos)^2 = distance

$\gamma(i,k)$ = mobile bearing is the angle between the line drawn between the cell and mobile and a line directed due east from the cell.

= ATAN2((mobile(k)_ypos - cell[i]_ypos, (mobile(k)_xpos - cell[i]_xpos))

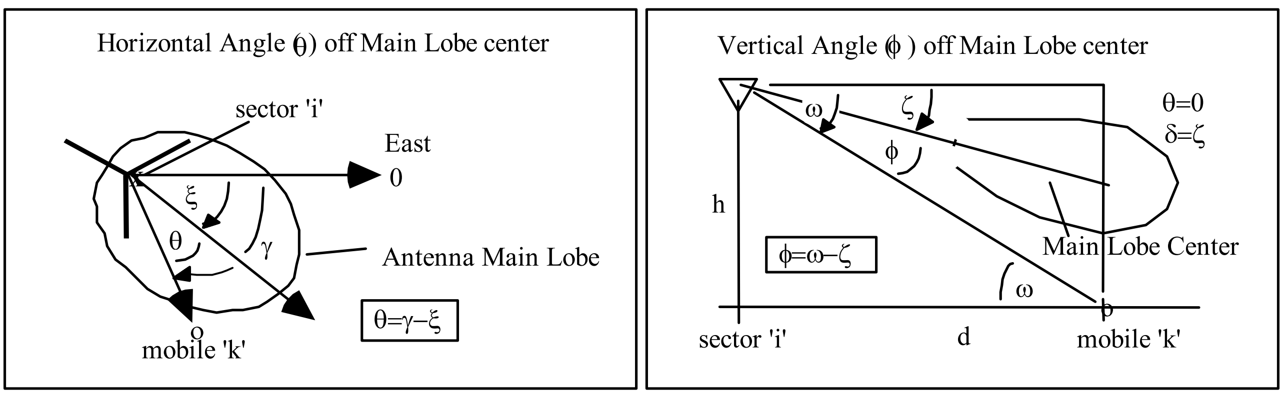

Figure B-3 Mobile Bearing orientation diagram example.

APPENDIX C: DEFINITION OF SYSTEM OUTAGE AND VOICE CAPACITY

The voice capacity N_{max} is defined as the maximum number of voice users that a sector can support while the system outage is below certain probability. System outage for voice users is to be evaluated based on the percentage of voice users in per-link outage. To get the per-link forward or reverse outage for a given voice user, one shall evaluate the short-term FER for a voice link by measuring the FER over windows of 400 ms (20 20-ms frames).

The proposed DV systems shall not be in system outage for more than 3% of the time. This is defined as follows:

Assume the short-term FER for user i is $FER_{i,j}$, where i = 1,…,N, is the user index for all cells and j=1,…, M, is the time index spanning a simulation run.

T_{system} outage, the system outage target, is the limit:

Prob.(Per-user outage among all N users in all runs) < $T_{system\ outage}$ = 3%

Per-user outage is defined as the event where a user's voice connection in either direction has short-term FER higher than 15% more often than $T_{per\ link}$ = 1% of the time. That is,

(Per-user outage for user i) = [$(\Sigma_{j=1\ to\ M}\ (I_{i,j})/M > 1\%$ for either forward or reverse]

where the indicator function $I_{i,j}$ is defined by: $I_{i,j} = 1$ if $FER_{i,j} >= 15\%$ and 0 if $FER_{i,j} < 15\%$

Note that the actual user in outage would perceive the quality to be low when either forward or reverse link is in outage.

To simplify the simulations, no voice calls will be dropped during each simulation run and the same number of voice calls is maintained throughout each simulation run.

APPENDIX D: FORMULA TO DEFINE VARIOUS THROUGHPUT AND DELAY DEFINITIONS

For each fixed simulation condition (e.g., voice load, distribution of different traffic, number of data users, etc.), simulation is run for multiple independent runs using Monte Carlo approach. Let

- T = simulation time.
- M = total number of independent runs (for a specific configuration).
- N= total number of data users in each run (for a sector).
- m= index of the simulation runs, i.e., m = 1,2,3, …, M.
- n = index of a data user within a simulation run, i.e., n = 1,2,…, N.

Therefore, the n-th data user in the m-th simulation run can be specified by *user(m,n)*.

Let

- K(m,n) = total number of packet calls generated for user(m,n).
- k = index of packet calls for a user. For user(m,n),, k = 1, 2, …, K(m,n).
- L(m,n,k) = total number of packets generated for the k-th packet call of user(m,n).
- l = index of packet within a packet call. For the k-th packet call of user(m,n), l = 1,2, …, L(m,n,k).
- B(m,n,k,l) = number of information bits contained in the l-th packet of the k-th packet calls for user(m,n) . If the packet is not successfully delivered by the end of the simulation run, B(m,n,k,l) = 0.
- TA(m,n,k,l) = arrival time of the l-th packet of the k-th packet calls for user(m,n). it is the time when the packet arrives at the transmitter side and is put into a queue.
- TD(m,n,k,l) = delivered time of the l-th packet of the k-th packet calls for user(m,n) . It is the time when the receiver successfully receives the packet. Due to fixed simulation time, there may be packets waiting to be completed at the end of a simulation run. For these packets, the delivered time is the end of the simulation.
- PCTA(m,n,k) = arrival time of the k-th packet call for user(m,n), it is the time when the first packet of the packet call arrives at the transmitter side and is put into a queue.
- PCTD(m,n,k) = delivered time of the k-th packet call for user(m,n). It is the time when the receiver successfully receives the last packet of the packet call. Due to fixed simulation time, there may be packet calls waiting to be completed at the end of a simulation run. For these packet calls, the delivered time is the end of the simulation.

The arrival time of a packet call is the time when the first packet of the packet call arrives at the transmitter side and is put into a queue, and the delivered time of a packet call is the time when the last packet of the packet call is successfully received by the receiver, i.e., PCTA(m,n,k) = TA(m,n,k,1) and PCTD(m,n,k) = TA(m,n,k,L(m,n,k)). Due to fixed simulation time, there may be packet calls waiting to be completed at the end of a simulation run. For these packet calls, the delivered time is the end of the simulation. Figure D-1 demonstrates the arrival and delivered times for a packet and a packet call.

With the above notation, we can now define various throughputs and delays as follows.

$$\text{Data throughput per sector} = \frac{\sum_{m=1}^{M}\sum_{n=1}^{N}\sum_{k=1}^{K(m,n)}\sum_{l=1}^{L(m,n,k)} B(m,n,k,l)}{MT}, \quad \text{(D-1)}$$

$$\text{Averaged delay per sector} = \frac{\sum_{m=1}^{M}\sum_{n=1}^{N}\sum_{k=1}^{K(m,n)} \left(PCTD(m,n,k) - PCTA(m,n,k)\right)}{\sum_{m=1}^{M}\sum_{n=1}^{N} K(m,n)}, \quad \text{(D-2)}$$

$$\text{Data throughput for user(m,n)} = \frac{\sum_{k=1}^{K(m,n)}\sum_{l=1}^{L(m,n,k)} B(m,n,k,l)}{T}, \quad \text{(D-3)}$$

$$\text{Packet call throughput for user(m,n)} = \frac{\sum_{k=1}^{K(m,n)}\sum_{l=1}^{L(m,n,k)} B(m,n,k,l)}{\sum_{k=1}^{K(m,n)} \left(PCTD(m,n,k) - PCTA(m,n,k)\right)}, \quad \text{(D-4)}$$

$$\text{Averaged packet delay per sector} = \frac{\sum_{m=1}^{M}\sum_{n=1}^{N}\sum_{k=1}^{K(m,n)}\sum_{l=1}^{L(m,n,k)} \left(\mathrm{TD}(m,n,k,l) - \mathrm{TA}(m,n,k,l)\right)}{\sum_{m=1}^{M}\sum_{n=1}^{N}\sum_{k=1}^{K(m,n)} L(m,n,k)}, \quad \text{(D-5)}$$

$$\text{Averaged packet delay for user(m,n)} = \frac{\sum_{k=1}^{K(m,n)}\sum_{l=1}^{L(m,n,k)} \left(TD(m,n,k,l) - TA(m,n,k,l)\right)}{\sum_{k=1}^{K(m,n)} L(m,n,k)}, \quad \text{(D-6)}$$

$$\text{Averaged packet call delay for user(m,n)} = \frac{\sum_{k=1}^{K(m,n)} \left(PCTD(m,n,k) - PCTA(m,n,k)\right)}{K(m,n)}. \quad \text{(D-7)}$$

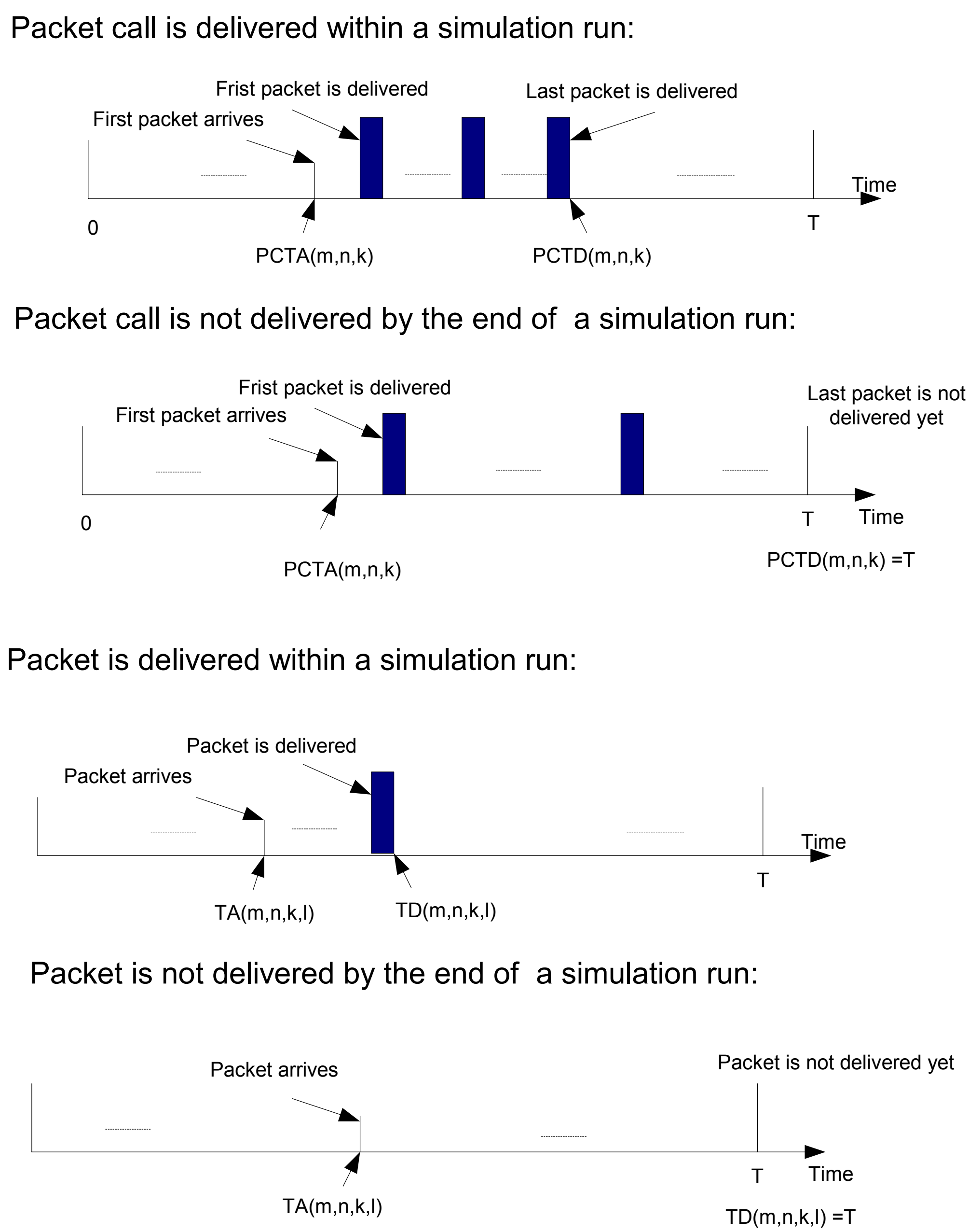

Figure D-1: Description of arrival and delivered time for a packet and a packet call.

APPENDIX E: LINK BUDGET

This appendix contains the forward and reverse link budget templates for 1xEV-DV and 1xEV-DO evaluation. These templates, when filled out in the evaluation process, provide maximum sustainable path loss and maximum range for different data rates or throughput levels.

The forward link templates differ from the typical approach where a fixed data rate is used to determine the range. This is because some system proposals use adaptive modulation and coding schemes to change the data rate on the forward link to take advantage of the changing channel conditions. Three fixed per-sector forward throughput levels (for four statistically identical mobile stations) are used to determine the maximum supportable path loss for data users. The way to fill out this forward link budget for data is:

- Determine the per-sector forward link throughput to be used in the link budget. This shall be at a level all proposals can achieve.
- Set the geometry/location of the mobile station (interior or cell edge) and per-sector multi-path fading profile (Pedestrian A, 1-path 3 km/hr, Pedestrian B, 3-path, 10 km/hr, Vehicular A, 2-path, 30 km/hr, Pedestrian A, 1-path, 120 km/hr, or Single path Rician).
- Simulate the link performance and obtain the FER vs Eb/Nt plot for each of the possible data rates the adaptive scheme uses
- Simulate sector throughput with the set of performance curves for the possible data rates for one mobile station. If applicable, scheduling is simulated, too. With all other link budget parameters fixed, adjust the path loss so that the throughput reaches the specified level above.
- To account for the additional range the adaptive schemes can achieve, a “multi-user diversity gain” is included. This is obtained by running the sector simulator with scheduler among four mobile stations. These four mobile stations have the same path loss to the base station and independent but statistically identical fading process. When the simulated sector throughput is the same as the single-user case above, the range increase (in dB) over the single-mobile case is the multi-user diversity gain.

Note for 1xEV-DV systems: Since the forward link channelization (e.g., Walsh) code space is limited, the forward link overhead channels and control channels have to be accounted for. The available code space is to be specified and justified by the proponents of the DV proposals. The sector throughput simulation above has to be carried out within the available code space for traffic channels.

Also, to make the comparison easy, no transmit diversity is to be used in these link budgets on either link.

Link-Budget Templates

Tables E-1 and E-2 give the proposed link-budget templates for the reverse link and the forward link, respectively.

Table E-1 Link-Budget Template for the Reverse Link

Reverse-Link Item		Value	Comments
Test Environment			Input (per antenna) Pedestrian A, 1-path 3 km/hr Pedestrian B, 3-path, 10 km/hr Vehicular A, 2-path, 30 km/hr Pedestrian A, 1-path, 120 km/hr Single path Rician
Test Service			Descriptive Text
Type of Traffic			Input: Voice Only, Data Only, or Data with Voice at a power division that gets the same range for 9.6 kbps and the data rate under consideration.
Chip Rate	(Mcps)	1.2288	Fixed Value
Transmitter Power	(dBm)	23	Fixed Value: Maximum available total power after any peak-to-average backoff has been taken out.[28]
Fraction of the Power Used for the Traffic Channel	(dB)		Input: Varies by Approach. Traffic Channel E_c/I_{or}.
Traffic Channel Transmitter Power	(dBm)		Calculated
Cable, Connector, and Combiner Losses	(dB)	0	Fixed Value
Transmitter Antenna Gain	(dBi)	–1	Fixed Value:
Transmitter EIRP per Traffic Channel	(dBm)		Calculated
Receiver Antenna Gain	(dBi)	17	Fixed Value
Cable and Connector Losses	(dB)	2	Fixed Value
Receiver Noise Figure	(dB)	5	Fixed Value
Thermal Noise Density	(dBm/Hz)		Calculated: –174 + Receiver Noise Figure
Rise Over Thermal, $(I_0 + N_0)/N_0$	(dB)	7	Fixed Value
Noise Plus Interference Density	(dBm/Hz)		Calculated

[28] In other words, the actual PA has to be larger than 23 dBm since it has to take into account the peak to average ratio.

Reverse-Link Item	**Value**	**Comments**
FER without Retransmissions		Fixed Value: For voice, it is 0.10 when in HO and 0.01 when not in HO. For data, it is 0.20 when in HO and 0.04 when not in HO.
Data Rate (kbps)		Fixed Value: For voice, it is 9.6. For data, link budgets should be provided for data rates of 9.6, 38.4, and the highest supported rate.
Required per Antenna Traffic $E_b/(N_0 + I_0)$ with Two Antennas (dB)		Input: Varies by Approach. Value with power control off since the mobile is at the cell edge. Determined by a link simulation.
Receiver Sensitivity per Antenna (dBm)		Calculated
Building Penetration Loss for Outdoor-to-Indoor or Vehicular Penetration Loss for Vehicular (dB)		20 for the Building Penetration Loss and 10 for the Vehicular Penetration Loss.
Log-Normal Fade Margin (dB)		Fixed Value: 11.2 for Pedestrian and Rician channel, 11.4 for Vehicular.
Handoff Gain (dB)		Fixed Value: Isolated cell fade margin – fade margin with best-of-two cell-edge coverage. 5.0 for Pedestrian, Rician channel and Vehicular.
Maximum Path Loss (dB)		Calculated
Maximum Range (m)		Calculated from the maximum path loss (see Note 1).

1 **Table E-2 Link-Budget Template for the Forward Link**

Forward-Link Item		Value	Comments
Test Environment			Input: Pedestrian A, 1-path 3 km/hr Pedestrian B, 3-path, 10 km/hr Vehicular A, 2-path, 30 km/hr Pedestrian A, 1-path, 120 km/hr Single path Rician
Mobile Velocity	(km/h)		Input
Test Service			Descriptive Text
Type of Traffic to a Particular Mobile			Input: Voice Only, Data Only, or Data with Voice with 50-50 split of non-overhead power
Mobile Location			Cell Edge in HO, Cell Edge Not in HO, or Cell Interior Not in HO. See Note 3.
Chip Rate	(Mcps)	1.2288	Fixed Value
Maximum PA Power	(dBm)	43	Fixed Value
Peak-to-Average Backoff at 99.9 percentile for the used channel configuration	(dB)		Input: Varies by Approach
Total Transmitter Output Power	(dBm)		Calculated
Fraction of the Power Used for Voice Traffic			Total voice E_c/I_{or}, including power control subchannel. For Voice Only, Data Only, and Data with Voice.
Fraction of the Power Used for Data Traffic			Total data E_c/I_{or}. Fixed value of 0.0 for Voice Only Specify Data Only and Data with Voice.
Fraction of the Power Used for the Pilot, Synch, and Paging Channels			Total overhead E_c/I_{or}. Fixed value of 0.2
Fraction of the Power Used for the Other Control Channels[29]			Varies by approaches

[29] This and the 3 terms above it sum to 1

Forward-Link Item		Value	Comments
Fraction of the Code Space Used for Data Traffic			Fixed value of 0.0 for Voice Only. Input that varies by approach for Data Only and Data with Voice.
Fraction of the Code Space Used for the Pilot, Control Traffic, and Margin			Calculated. The sum of this value and the previous code space is limited by the total code space available.
Fraction of the Power Used for the Link-Budget Traffic Channel	(dB)		Traffic E_c/I_{or}. Input: Varies by Approach. This fraction is less than or equal to the total power fraction for voice or data noted above [30].
Fraction of Code Space Used for the Link-Budget Traffic Channel			Input: Varies by Approach. This fraction is less than or equal to the total channel resource fraction for voice or data noted above.[31]
Cable, Connector, and Combiner Losses	(dB)	2	Fixed Value
Transmitter Antenna Gain	(dBi)	17	Fixed Value
Traffic Channel EIRP	(dBm)		Calculated
Total Transmitter EIRP	(dBm)		Calculated
Receiver Antenna Gain	(dBi)	–1	Fixed Value
Cable and Connector Losses	(dB)	0	Fixed Value
Receiver Noise Figure	(dB)	10	Fixed Value
Thermal Noise Density	(dBm/Hz)		Calculated: –174 + Receiver Noise Figure
Voice FER			Fixed input of 0.01 for voice and N/A for data.
Voice Data Rate	(kbps)		Fixed input of 9.6 for Voice Only and N/A for Data Only and Data with Voice.

[30] For example, power control subchannel power consumption has to be excluded.

[31] This term does not reduce the amount of power available for the traffic channel but might limit the sector throughput.

Forward-Link Item		Value	Comments
Sector Data Throughput	(kbps)		Input: N/A for Voice Only and at least three different values for Data Only and Data with Voice. This is the throughput of just the data portion of the link. The throughput is without RLP packet retransmissions, where packets may consist of multiple slots and subpackets that are not transmitted contiguously.
Required $\hat{I}_{or}/N_0$ with a Single User	(dB)		Input: Determined by a link simulation and a network simulation.
Multiple-User Data $\hat{I}_{or}/N_0$ Gain	(dB)		Input: Gain with four scheduled users compared to that required for a single user for the same total sector throughputs and total (over the 4 users) average fractional powers. Determined by a network simulation with the specified channel conditions.
Receiver Interference Density, I_{oc}	(dBm/Hz)		Calculated based on the equations in Note 3.
Noise Plus Interference Density	(dBm/Hz)		Calculated
Receiver Sensitivity	(dBm)		Calculated
Building Penetration Loss for Outdoor-to-Indoor or Vehicular Penetration Loss for Vehicular	(dB)		20 for the Building Penetration Loss and 10 for the Vehicular Penetration Loss.
Log-Normal Fade Margin	(dB)		Fixed Value: 11.2 for Pedestrian and Rician channel, 11.4 for Vehicular.
Handoff Gain	(dB)		Fixed Value: the values for Pedestrian Rician channel, and Vehicular environments are 5.0 for best-of-two HO and 6.2 for sum-of-two HO[32]. Use 0 for no HO gain.
Maximum Path Loss	(dB)		Calculated
Maximum Range	(m)		Calculated from the maximum path loss (see Note 1).

[32] Sum of two is the typical soft handoff where multiple sectors transmit to the MS and the MS combines the traffic channels across sectors. The Best-of-two HO is closer to hard handoff in that only one of the sectors is transmitting to the MS at any given time.

The following are some notes for the link budgets.

Note 1: The maximum range in meters, R, is calculated from the maximum path loss in dB, L. In [20], the specified equations depend on the test environment as follows:

$$R = \begin{cases} \min(10^{(L-38.462)/20}, 10^{(L-37)/30}) & \text{for Indoor Office} \\ \min(10^{(L-38.462)/20}, 10^{(L-28.031)/40}) & \text{for Pedestrian and Outdoor-to-Indoor} \\ \min(10^{(L-38.462)/20}, 10^{(L-15.352)/37.6}) & \text{for Vehicular} \end{cases} \quad \text{(E-1)}$$

In [23], the range for all test environments is

$$R = 10^{(L-28.6)/35} \quad \text{(E-2)}$$

The range equations from [23] are recommended.

Note 2: The propagation index and log-normal fading sigma values specified in [20] are shown in Table E-3.

Table E-3 Propagation Index and Log-Normal Sigma Values from [20]

Test Environment	Propagation Index	Log-Normal Sigma (dB)
Indoor Office	3	12.0
Pedestrian	4	10.0
Outdoor-to-Indoor with the Building Penetration Loss	4	14.4
Vehicular	3.76	10.0

Note 3: When the mobile is at the cell/sector edge, assume, as in [24], that it is receiving a signal from three cells/sectors with two of the signals having the same average received power density and the other having 6 dB less. When the mobile is in the cell/sector interior, assume that it is receiving a signal from two cells/sectors with the average received power density from one being 6 dB less. If the signal density received from the target cell (the one where the link budget range is being calculated) is I_{tc}, the other-cell interference density is I_{oc}, and the total received signal power signal is $\hat{I}_{or}$, then

$$I_{oc} = \begin{cases} 0.25 I_{tc} & \text{for the cell edge in handoff (HO)} \\ 1.25 I_{tc} & \text{for the cell edge not in HO} \\ 0.25 I_{tc} & \text{for the cell interior not in HO} \end{cases} \quad \text{(E-3)}$$

and

$$\hat{I}_{or} = \begin{cases} 2 I_{tc} & \text{for the cell edge in HO} \\ I_{tc} & \text{for the cell edge not in HO} \\ I_{tc} & \text{for the cell interior not in HO} \end{cases} \quad \text{(E-4)}$$

Note 4: The frequency shall be 2 GHz, as in [20].

APPENDIX F: QUASI-STATIC METHOD FOR LINK FRAME ERASURES GENERATION AND DYNAMICALLY SIMULATED FORWARD LINK OVERHEAD CHANNELS

The following Appendix describes in more details the quasi-static method for modeling the link level performance of 1xEV-DV PDCH and 1xEV-DO FTC and dynamically simulated forward link overhead channels in the system level.

Definitions

Aggregate Es/Nt Metric for the AWGN Channel

The aggregate Es/Nt, denoted $\Sigma_{\mathrm{E_s/N_t}}$ is defined as

$$\Sigma_{\mathrm{E_s/N_t}} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n} N_j.(\mathrm{E_s/N_t})_j\right]\right), \tag{F-1}$$

where

1. N equals the number of information bits (i.e., the encoder packet size).
2. N_j equals the number of modulation symbols transmitted in slot j.
3. n is the number of slots over which the transmission occurs. This includes both the original transmission, and retransmissions, if any. For example, if the duration of the original transmission is 4 slots, and that of the retransmission is 2 slots, then n=6.
4. $(\mathrm{E_s/N_t})_j,\ j=1,\ldots,n,$ is the SNR per modulation symbol for slot j. These terms are *not* in dB.
5. Note that $\Sigma_{\mathrm{E_s/N_t}}$ =Eb/No because N equals the number of information bits.
6. $(\mathrm{E_s/N_t})_j,\ j=1,\ldots,n,$ is the Es/Nt observed *after* Rayleigh (or Jakes) fading.

The functional relationship[33] between $\Sigma_{\mathrm{E_s/N_t}}$ and BLER for the base 1/5 turbo code over the AWGN channel with m-ary modulation will be denoted by f_5^m. For the sake of convenience, the superscript in f_5^m is dropped for QPSK modulation. Thus, f_5 denotes the functional relationship between $\Sigma_{\mathrm{E_s/N_t}}$ and BLER for the base 1/5 turbo code over the AWGN channel, when using QPSK modulation. The following table, *which is for illustration purposes only,* is an example of what f_5 will look like.

[33] This functional relationship *will* depend on the encoder packet size. So, *the proponent will need to generate this relationship for every encoder packet size that has been defined.*

Σ_{E_s/N_t} **(in dB)**	**BLER**
-0.75	1
-0.55	0.986842
-0.35	0.707547
-0.15	0.205479
0.05	0.015538
0.25	0.00019

Effective Coding Rate

$$\text{Effective coding rate} = \frac{\text{Total number of information bits}}{\left(\begin{array}{l}\text{Total number of information bits +}\\ \text{Total number of unique parity bits received so far}\end{array}\right)}. \quad \text{(F-2)}$$

Clearly, the effective coding rate remains unchanged in the case of pure Chase combining. In the case of incremental redundancy based schemes, the effective coding rate continues to decrease with every retransmission, until it equals the base turbo coding rate.

Prediction Error Rate

Difference in the actual and predicted BLER at different average Es/Nt. In particular, one look at the prediction error percentage, defined as

$$\frac{(\text{Actual number of block errors } - \text{ Predicted number of block errors})}{\text{Total number of blocks transmitted}}.100 \quad \text{(F-3)}$$

The motivation behind this measure is as follows. The actual throughput seen at a given Es/Nt is

$$\left(1-\frac{\text{Actual number of block errors}}{\text{Total number of blocks transmitted}}\right)(\text{Peak rate}), \quad \text{(F-4)}$$

while the predicted throughput is

$$\left(1-\frac{\text{Predicted number of block errors}}{\text{Total number of blocks transmitted}}\right)(\text{Peak rate}). \quad \text{(F-5)}$$

Therefore, the difference in actual throughput and predicted throughput is

$$\frac{(\text{Actual number of block errors } - \text{ Predicted number of block errors})}{\text{Total number of blocks transmitted}}(\text{Peak rate}). \quad \text{(F-6)}$$

Normalizing with respect to peak rate, and taking a percentage yields the first expression.

Catastrophic Error Rate

Number or percentage of "Catastrophic Errors." For an AWGN channel and a given effective coding rate, the BLER is almost 0 if $\Sigma_{E_s/N_t} > T_0$ dB, and the BLER is 1 if $\Sigma_{E_s/N_t} < T_1$ dB. For turbo codes, $T_0 - T_1 \approx 1$ dB. For the fading channel, one declare a "catastrophic error" if one of the following two events occur

$\Sigma_{E_s/N_t} > T_0$ AND the block is actually in error, or

$\Sigma_{E_s/N_t} < T_1$ AND the block is actually NOT in error.

The significance of a catastrophic error is as follows. The aggregate Es/Nt metric only captures a first order statistic of the channel variations. If the second order variations of the channel were to play a significant role in determining BLERs, then Σ_{E_s/N_t} would prove to be insufficient in characterizing BLERs, and this, in turn, would lead to a large number of catastrophic errors. Few catastrophic errors, therefore, imply that the second order statistics of the channel are not important as long as $\Sigma_{E_s/N_t} > T_0$ or $\Sigma_{E_s/N_t} < T_1$. Since $T_0 - T_1 \approx 1\,\text{dB}$ (i.e., small), this would also imply that Σ_{E_s/N_t} is a sufficient for predicting BLERs.

Puncturing Penalty (or Coding Gain)

For a code with effective coding rate 1/M, where 1/M > 1/5, and modulation order *m*, the puncturing penalty is defined to be the additional Σ_{E_s/N_t} (or Eb/No) (in dB) required (with respect to the base 1/5 turbo code) to achieve a BLER of 0.001 over an AWGN channel. The puncturing penalty[34] is denoted by C_M^m. The superscript here refers to the modulation, while the subscript refers to the effective coding rate. For the sake of convenience, the superscript is dropped in the case of QPSK modulation. The following table, *which is for illustration purposes only,* shows the puncturing penalty for a few sample cases for an encoder packet size of 3072 bits *for QPSK modulation.* Σ_{E_s/N_t} required to achieve a BLER of 0.001 for the base turbo code is approximately 0.2 dB.

Effective coding rate	Σ_{E_s/N_t} **required**	**Puncturing penalty in dB**
¼	0.4	0.2
1/3	0.75	0.55
½	1.6	1.4

This penalty applies both for IR and Chase combining.

Doppler Penalty

$\beta_{D,M}$ is denoted as the Doppler penalty at Doppler = D Hz, for an effective coding rate of 1/M. Simulations show that $\beta_{D,M}$ =0, if D < 30Hz. For the values of *M* that were considered, at Dopplers around 100 Hz, the penalty lay between 0.2 and 0.6 dB. *This penalty applies both for IR and Chase combining.*

Demapping Penalty

Suppose *m*-ary (*m* > 4) modulation is being used for transmission over the channel. 2 cases are considered here.

[34] Note that C_M (expressed in dB) is always positive.

1. Pure Chase combining. *This implies that the modulation and coding rate does not change for retransmissions, AND the combining of the repeated symbols is done at the modulation symbol level.* These combined symbols are then demapped to soft bits (which are then input to the turbo decoder). In this case, f_5^m should be used for predicting BLER. Recall that f_5^m denotes the functional relationship between Σ_{E_s/N_t} and BLER for the base 1/5 turbo code over the AWGN channel with *m*-ary modulation. So, there is no need for explicitly introducing demapping penalties in this case[35]. *If the effective code rate of the transmission is greater than 1/5, coding penalties should also be applied before prediction.*

2. *For all schemes that do not fall in the category 1 above, demapping penalties apply.* Therefore this includes

    - *Schemes which use IR (pure or otherwise), and*
    - *Schemes that use Chase combining, BUT the modulation or the coding rate is different for retransmissions, OR the modulation symbols are first demapped to soft bits, and then combined.*

    Demapping penalties are a function of the modulation being used, and the $(E_s/N_t)_j$, which denotes the SNR for a modulation symbol in slot *j*. Precisely, Σ_{E_s/N_t} is given by

$$\Sigma_{E_s/N_t} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n}\frac{N_j.(E_s/N_t)_j}{\alpha_{(E_s/N_t)_j}}\right]\right), \qquad \text{(F-7)}$$

where α_x denotes the demapping penalty when $E_s/N_t = x$, where Es/Nt denotes the modulation symbol SNR. Note that the terms $(E_s/N_t)_j$, and α_x are not in dB, and $\alpha_x > 1$. Simulations indicate that α_x is a monotone decreasing function of *x*, and is typically greater than 2. Upon obtaining Σ_{E_s/N_t}, as defined in F-7, f_5 should be used for predicting BLER. Recall that f_5 denotes the functional relationship between Σ_{E_s/N_t} and BLER for the base 1/5 turbo code over the AWGN channel *with QPSK modulation.*

Flowcharts

The first flowchart illustrates the method when the QPSK modulation is used. As the figure illustrates, the following values are needed for aggregate Es/Nt method:

1. f_5 denotes the functional relationship between Σ_{E_s/N_t} and BLER for the base 1/5 turbo code.
2. Effective coding rates, and the corresponding puncturing penalties (or coding gains) for various effective rates. These puncturing penalties will, in general, depend on the size of the encoder packet, i.e., the number of information bits.

[35] In this case, demapping penalties are being implicitly accounted for in f_5^m.

3. The Doppler penalty for various Dopplers and encoder packet sizes.

Effective coding rate = 1/M

↓

Calculate $\Sigma_{\mathrm{E_s/N_t}} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n} N_j.(\mathrm{E_s/N_t})_j\right]\right)$.

↓

Introduce puncturing penalty:
$\Sigma_{\mathrm{E_s/N_t}} = \Sigma_{\mathrm{E_s/N_t}} - C_M$, where
C_M is in dB.

↓

Introduce Doppler penalty:
$\Sigma_{\mathrm{E_s/N_t}} = \Sigma_{\mathrm{E_s/N_t}} - \beta_{D,M}$, where
$\beta_{D,M}$ is in dB.

↓

Use the adjusted $\Sigma_{\mathrm{E_s/N_t}}$ and f_5 to determine whether the block is in error.

Figure F-1 Flowchart for QPSK modulation

The next flowchart illustrates the methodology for higher order modulations when IR is being used.

Effective coding rate = 1/M

$$\text{Calculate } \Sigma_{E_s/N_t} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n}\frac{N_j.(E_s/N_t)_j}{\alpha_{(E_s/N_t)_j}}\right]\right).$$

Introduce puncturing penalty: $\Sigma_{E_s/N_t} = \Sigma_{E_s/N_t} - C_M$, where C_M is in dB.

Introduce Doppler penalty: $\Sigma_{E_s/N_t} = \Sigma_{E_s/N_t} - \beta_{D,M}$, where $\beta_{D,M}$ is in dB.

Use the adjusted Σ_{E_s/N_t} and f_5 to determine whether the block is in error.

Figure F-2 Prediction methodology for higher order modulations without pure Chase combining

The next flowchart illustrates the prediction methodology for higher order modulations when pure Chase combining is being used. In particular, *m*-ary modulation is being used. The puncturing penalty used in step 3 of the flowchart is C_M^m and not C_M, while in the last step of the flowchart the table being used is f_5^m and not f_5. This distinction is important; recall that f_5 is the relationship between Σ_{E_s/N_t} and BLER with QPSK modulation, while f_5^m is the relationship between Σ_{E_s/N_t} and BLER with *m*-ary modulation ($m > 4$). Similarly, C_M^m is the puncturing penalty for the rate 1/M code with *m*-ary modulation.

Effective coding rate = 1/M

↓

Calculate $\Sigma_{\mathrm{E_s/N_t}} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n} N_j.(\mathrm{E_s/N_t})_j\right]\right).$

↓

Introduce puncturing penalty:

$\Sigma_{\mathrm{E_s/N_t}} = \Sigma_{\mathrm{E_s/N_t}} - C_M^m$, where

C_M^m is in dB.

↓

Introduce Doppler penalty:

$\Sigma_{\mathrm{E_s/N_t}} = \Sigma_{\mathrm{E_s/N_t}} - \beta_{D,M}$, where

$\beta_{D,M}$ is in dB.

↓

Use the adjusted $\Sigma_{\mathrm{E_s/N_t}}$ and f_5^m to determine whether the block is in error or not.

Figure F-3 Prediction methodology for higher order modulations with pure Chase combining (this corresponds to Case 1)

<u>Obtaining Coding Gains or Puncturing Penalties</u>

Let $\left(\Sigma_{\mathrm{E_s/N_t}}\right)_M$ denote the $\Sigma_{\mathrm{E_s/N_t}}$ required to achieve a BLER of 0.001 over an AWGN channel when the effective coding rate is 1/M. Then, the coding gain or puncturing penalty for the effective coding rate of 1/M is simply

$$\left(\Sigma_{\mathrm{E_s/N_t}}\right)_M - \left(\Sigma_{\mathrm{E_s/N_t}}\right)_5. \qquad \text{(F-8)}$$

- *Puncturing penalties depend on the encoder packet size N, and should be evaluated separately for each encoder packet size in use.*
- *In the case of pure Chase combining, puncturing penalties should be obtained for different modulations as well.* For cases without pure Chase combining, puncturing penalties should not depend on the modulation in use. However, one can determine them separately for each modulation also.

<u>Obtaining Doppler Penalties</u>

The Doppler penalty is obtained by simple curve fitting.

Fix 1/M (the effective coding rate), n (the duration of the transmission, i.e., number of slots), N (the encoder packet size), and the D (the Doppler). For various average geometries (from x dB to y dB, say), one run link level simulations to obtain sample values of $\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$ and an indicator of whether the packet is in error at the given value of $\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$.

For each collected sample pair ($\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$, indicator of block error), one applies the predictor, and chooses the value of the Doppler penalty that results in the best predictor performance. Precisely, one first fixes a value of $\beta_{D,M}$. Then, for each sample collected, one apply the Doppler penalty, and the puncturing penalty to obtain the modified $\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$, i.e.,

$$\Sigma_{\mathrm{E_s}/\mathrm{N_t}} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n} N_j.(\mathrm{E_s}/\mathrm{N_t})_j\right]\right) - \beta_{D,M} - C_M. \quad \text{(F-9)}$$

one then use f_5 and the modified $\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$ above to predict whether the block is in error. For each average geometry, one evaluates the performance of the predictor by computing the prediction error rates and catastrophic error rates. Finally, the value of Doppler penalty, which results in the best prediction performance is chosen to be the value of $\beta_{D,M}$.

The following flowcharts illustrate this method.

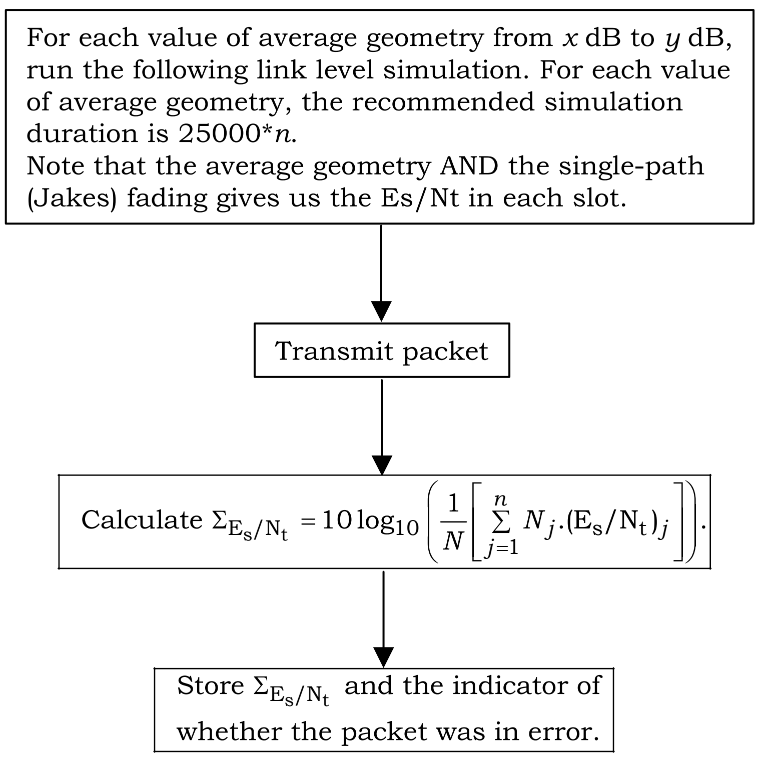

Figure F-4 Obtaining sample values of $\Sigma_{\mathrm{E_s}/\mathrm{N_t}}$ and indicators of packet errors

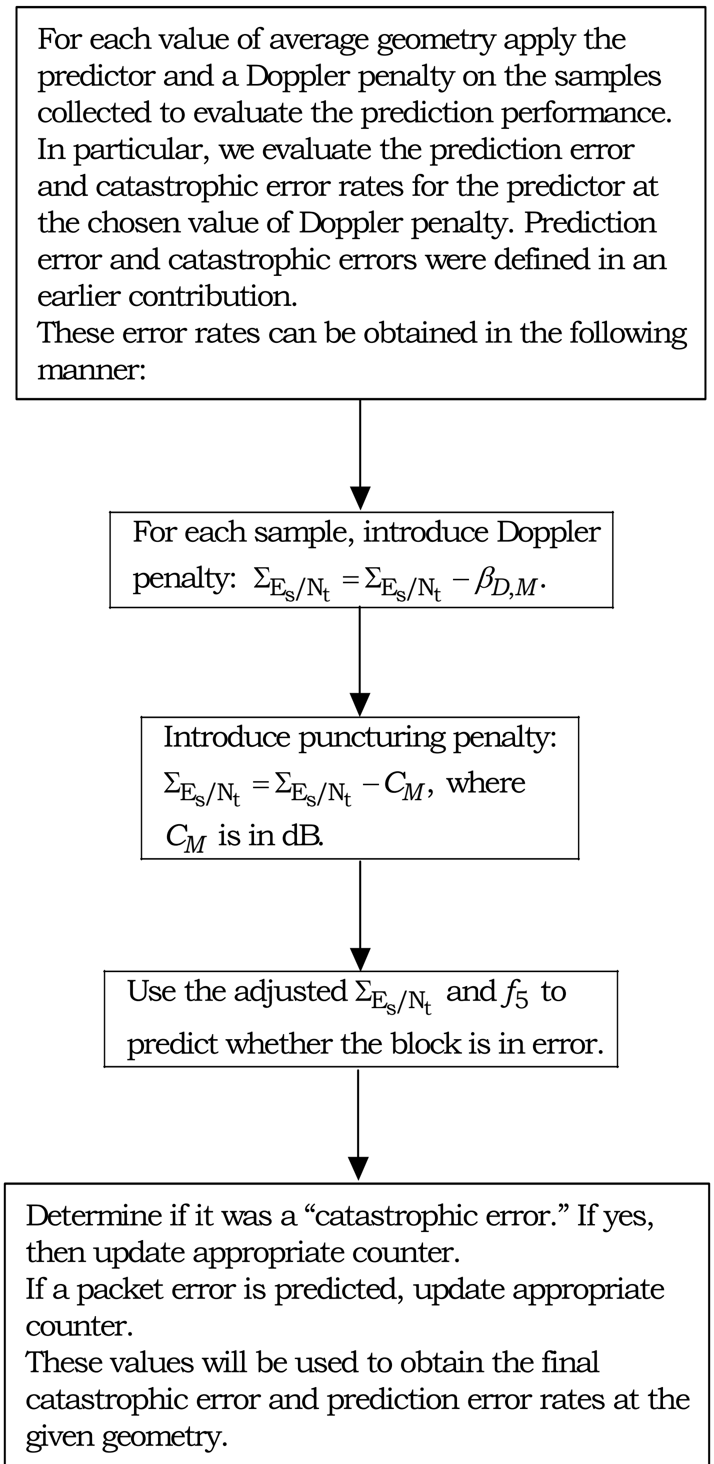

Figure F-5 Determining the Doppler penalty by evaluating the predictor performance

- *Doppler penalties depend on the encoder packet size and the effective coding rate.*
- However, to limit the simulation effort, extrapolation techniques may be used to obtain Doppler penalties for different effective coding rates and encoder packet sizes.

Obtaining Demapping Penalties

Fix N, the encoder packet size, and m, the modulation. Set the Doppler to 10 Hz. Let z dB be the Es/Nt required to achieve a BLER of 0.001 over an AWGN channel for the given encoder packet size and modulation when using the base 1/5 turbo code. Then, run the following link level simulations over the fading channel for average geometries ranging from z-12 dB to z-2 dB[36]. Transmit the encoder packet; schedule retransmissions, if necessary; continue retransmissions till the encoder packet is successfully decoded. Let $\tilde{n}$ denote the number of transmissions necessary for the packet to be successfully decoded, and $n(i),\ i=1,\ldots,\tilde{n},$ the number of slots used after the i-th transmission. For the given encoder packet, we collect the following samples: $(\mathrm{E_s}/\mathrm{N_t})_j,\ j=1,\ldots,n(\tilde{n})$.

Next, fix a set of demapping penalties α_x. Recall that the demapping penalties are a function of the modulation symbol SNR x. (one can assume that the demapping penalties are a piecewise linear function of modulation symbol SNR.) Then, calculate $\Sigma^i_{\mathrm{E_s}/\mathrm{N_t}},\ i=1,\ldots,\tilde{n},$ the aggregate SNR after applying the demapping penalties and the puncturing penalties, after the i-th transmission as follows:

$$\Sigma^i_{\mathrm{E_s/N_t}} = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n(i)}\frac{N_j.(\mathrm{E_s/N_t})_j}{\alpha_{(\mathrm{E_s/N_t})_j}}\right]\right) - C^i_M,\ i-1,\ldots,\tilde{n}, \tag{F-10}$$

where C^i_M is the puncturing penalty after the i-th transmission. Next, for each $i=1,\ldots,\tilde{n},$ use f_5 and $\Sigma^i_{\mathrm{E_s}/\mathrm{N_t}}$ to predict whether the packet was in error. *Note that for each* $i=1,\ldots,\tilde{n}-1,$ *the packet is actually in error.*

Finally, for each average geometry, evaluate the performance of the predictor by computing the prediction error rates and catastrophic error rates. The demapping penalties, which result in the best prediction performance, are then chosen to be the values of α_x.

- Obtaining the demapping penalties require a large number of samples. So, it is recommended that for each average geometry, one attempt at least 25000 transmissions of encoder packet transmissions.
- Demapping penalties should not depend on the encoder packet size. However, one can try to obtain separate demapping penalties for each such size defined.
- Demapping penalties do not depend on the effective coding rate.

[36] This range of average geometries may be reduced if it is expected that the higher order modulation is not going to be used at really low values of Es/Nt.

- *Different modulations will have different demapping penalties. So, demapping penalties need to be determined for each modulation being used.*

Recommendations

- Puncturing penalties (or coding gains) result in extremely accurate predictions if $1/M < 0.75$. If $1/M \geq 0.75$, then the use of the actual functional relationship between Σ_{E_s/N_t} and BLER for the corresponding value of M is recommended.

- Doppler penalties can be obtained by "intelligent" extrapolation. For example, for D=100Hz, if the Doppler penalty is 0.25, 0.5 dB and 1 dB for 1/M=0.2, 0.5, and 0.75, respectively, then suitable extrapolation can be used to obtain Doppler penalties for all values of M in between.

APPENDIX G: EQUALIZATION

The following Appendix describes a possible technique to reduce multi-path interference. This technique is optional; however, if equalization is used in a proposal, the technique discussed below shall be used.

Equalization (optional)

Equalization may be used to reduce the FURP. In the following text, a procedure is provided for evaluating the symbol signal-to-noise ratio of the ideal MMSE equalizer for channels randomly drawn according to the ITU models. The performance of the ideal MMSE equalizer is captured in equation (2) below. A second simulation procedure is described which accounts for the implementation losses associated with the use the MMSE equalizer. The performance of this non-ideal approximation of the MMSE equalizer is captured in equation (14) below.

The procedure for the implementation of equalization in the simulation will be the following:

i) A random draw is taken from the appropriate ITU channel model (Ped A, Ped B, Veh A).

ii) The random draw is convolved with the autocorrelation of the chip filter;

iii) Let the complex vector $\mathbf{f} = \{f_{L_1}, f_{L_1+1}, ..., f_{L_2}\}$ denote the vector of chip-spaced waveform samples. The vector $\mathbf{f}$ is normalized to unit energy.

iv) Let the complex vector $\mathbf{g} = \{g_{L_3}, g_{L_3+1}, ..., g_{L_4}\}$ of length $L_4 - L_3 + 1$ have coefficients given by

$$g_i = \begin{cases} f_i & \max\{L_1, L_3\} \le i \le \min\{L_2, L_4\} \\ 0 & else \end{cases} \quad . \tag{G-1}$$

In general, the index L_3 may be chosen to optimize the performance of an equalizer of length $L_4 - L_3 + 1$. Default values of L_3 and L_4 will be $L_3 = -1$ and $L_4 = 8$, so that the equalizer has a total of ten taps.

v) The symbol signal-to-noise ratio for the equalizer is given by

$$\left(\frac{E_s}{N_t}\right)_{MMSE} = N \frac{E_c}{I_{or}} \mathbf{g}^H \mathbf{\Omega}^{-1} \mathbf{g} \quad . \tag{G-2}$$

where N is the number of chips per symbol, E_c/I_{or} is the allocation for the given Walsh code, and I_{or}/I_{oc} is the geometry. The matrix $\mathbf{\Omega}$ is given by

$$\Omega_{l,m} = \sum_{\substack{L_1 \le k \le L_2 \\ k \ne l}} f_k f^*_{m-l+k} + \frac{I_{oc}}{I_{or}} \delta(m-l) \qquad L_3 \le l, m \le L_4 \ , \tag{G-3}$$

where

$$\delta(n) = \begin{cases} 1 & n = 0 \\ 0 & else \end{cases} . \tag{G-4}$$

Note that I_{or} is the intra-cell interference power for the given instantiation of the channel $\mathbf{f}$. It is not the intra-cell power averaged over the distribution of $\mathbf{f}$.

The procedure used to account for implementation losses associated with the use of MMSE equalization will be the following:

i) Let $\mathbf{\Delta f} = \{\Delta f_{L_1}, \Delta f_{L_1+1}, ..., \Delta f_{L_2}\}$ denote a zero-mean Gaussian random vector with covariance matrix $\mathbf{\Sigma}$ with elements

$$\Sigma_{l,m} = M^{-1}\left(\left(\frac{E_{c,p}}{I_{or}}\right)^{-1} \sum_{\substack{0 \le k \le L-1 \\ k \ne l}} f_k f^*_{m-l+k} + \left(\left(\frac{E_{c,p}}{I_{or}}\right) \frac{I_{or}}{I_{oc}} \right)^{-1} \delta(m-l) \right), \tag{G-5}$$

where $E_{c,p}/I_{or}$ is the pilot allocation. The integer M should be chosen as large as possible subject to the condition

$$M << \frac{\text{chip rate}}{\text{max Doppler}} = \frac{1{,}228{,}800}{\text{max Doppler}} . \tag{G-6}$$

In order to simplify the generation of samples of the random vector $\mathbf{\Delta f}$, the diagonal covariance matrix $\mathbf{\Gamma}$ with elements

$$\Gamma_{l,m} = \begin{cases} \Sigma_{l,l} & m = l \\ 0 & else . \end{cases} \tag{G-7}$$

may be used in place of the true covariance $\mathbf{\Sigma}$.

ii) A random draw is taken of the vector $\mathbf{\Delta f}$. Let $\hat{\mathbf{f}} = \mathbf{f} + \mathbf{\Delta f}$.

iii) Let x denote a Gaussian random variable with mean

$$E(x) = I_{or} + I_{oc} , \tag{G-8}$$

and variance

$$\text{var}(x) = \frac{1}{N}(I_{or} + I_{oc})^2 . \tag{G-9}$$

iv) Let y' denote a non-central chi-square random variable with two degrees of freedom, mean

$$E(y') = I_{or} + \frac{1}{N}\frac{1}{\|f_l\|^2}\left(\frac{E_{c,p}}{I_{or}}\right)^{-1}\left(\left(1-\|f_l\|^2\right)I_{or} + I_{oc}\right), \tag{G-10}$$

and variance

$$\begin{aligned}\mathrm{var}(y') = \frac{1}{N^2}\left(\frac{1}{\|f_l\|^2}\left(\frac{E_{c,p}}{I_{or}}\right)^{-1}\right)^2\left(\left(1-\|f_l\|^2\right)I_{or} + I_{oc}\right)^2 \\ + 2I_{or}\frac{1}{N}\frac{1}{\|f_l\|^2}\left(\frac{E_{c,p}}{I_{or}}\right)^{-1}\left(\left(1-\|f_l\|^2\right)I_{or} + I_{oc}\right).\end{aligned} \tag{G-11}$$

vi) Random draws are take of x and y'. Define

$$\left(\frac{\hat{I}_{or}}{I_{oc}}\right) = \begin{cases} \frac{y'}{x-y'} & x - y' > 0 \\ 100 & else \end{cases}. \tag{G-12}$$

vii) Define the matrix $\hat{\mathbf{\Omega}}$ with elements given by

$$\hat{\Omega}_{l,m} = \sum_{\substack{0\le k\le L-1 \\ k\ne l}} \hat{f}_k \hat{f}^*_{m-l+k} + \left(\frac{\hat{I}_{or}}{I_{oc}}\right)^{-1}\delta(m-l) \qquad L_3 \le l,m \le L_4 . \tag{G-13}$$

viii) The symbol signal-to-noise ratio of the MMSE equalizer based on the estimated channel $\hat{\mathbf{f}}$ and covariance $\hat{\mathbf{\Omega}}$, is given by

$$\left(\frac{E_s}{N_t}\right) = N\frac{E_c}{I_{or}}\frac{\left(\mathbf{g}^H\hat{\mathbf{\Omega}}^{-1}\hat{\mathbf{g}}\right)^2}{\hat{\mathbf{g}}^H\hat{\mathbf{\Omega}}^{-1}\,\mathbf{\Omega}\,\hat{\mathbf{\Omega}}^{-1}\hat{\mathbf{g}}} . \tag{G-14}$$

APPENDIX H: MAX-LOG-MAP TURBO DECODER METRIC

The following Appendix describes the MAX-LOG-MAP turbo decoder metric.

Turbo Decoder Metric and Soft Value Generation into Turbo Decoder

MAX-LOG-MAP shall be used as turbo decoder metric.

The basic block diagram of the mobile station receiver for the simulation is shown in Figure H-1. The M-ary QAM demodulator generates soft decisions as inputs to the Turbo decoder. For M-ary QAM, the soft inputs to the decoder are generated by an approximation[37] to the log-likelihood ratio function [3]. First define,

$$\Lambda^{(i)}(z) = K_f \left[\underset{j \in S_i}{Min}\left\{d_j^2\right\} - \underset{j \in \overline{S_i}}{Min}\left\{d_j^2\right\} \right],\ i = 0,1,2,\cdots,\log_2 M - 1 \qquad \text{(H-1)}$$

where M is the modulation alphabet size, i.e. 8, 16, 32 or 64 and

$$z = A_d A_p \alpha\, \hat{\alpha}\, e^{-j(\theta + \hat{\theta})} x + n\,, \qquad \text{(H-2)}$$

x is the transmitted QAM symbol, A_d is the traffic channel gain, A_p is the pilot channel gain, $\alpha\, e^{j\theta}$ is the complex fading channel gain, and $A_p \hat{\alpha}\, e^{j\hat{\theta}}$ is the fading channel estimate obtained from the pilot channel,

$$S_i = \left\{\forall j : i^{th} \text{ component of } y_j \text{ is "0"}\right\}, \qquad \text{(H-3)}$$

$$\overline{S_i} = \left\{\forall j : i^{th} \text{ component of } y_j \text{ is "1"}\right\} \qquad \text{(H-4)}$$

and K_f is a scale factor proportional to the received signal-to-noise ratio. The parameter d_j is the Euclidean distance of the received symbol z from the points on the QAM constellation in S or its complement (y_j). It is assumed that the value of A_p/A_d is known at the mobile station receiver. In this case the distance metric is computed as shown in Figure H-1 and is written as follows

$$d_j^2 = \left| A_p z - Q_j \beta \gamma^2 \right|^2 \quad Q_j \in S_i \ \textbf{or}\ \overline{S}_i \qquad \text{(H-5)}$$

where $\beta = A_d$ and $\gamma = A_p \hat{\alpha}$ is an estimate formed from the pilot channel after processing through the channel estimation filter as shown in Figure H-1.

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[37] The optimum LLR computation will also include the noise variance due to channel estimation [4].

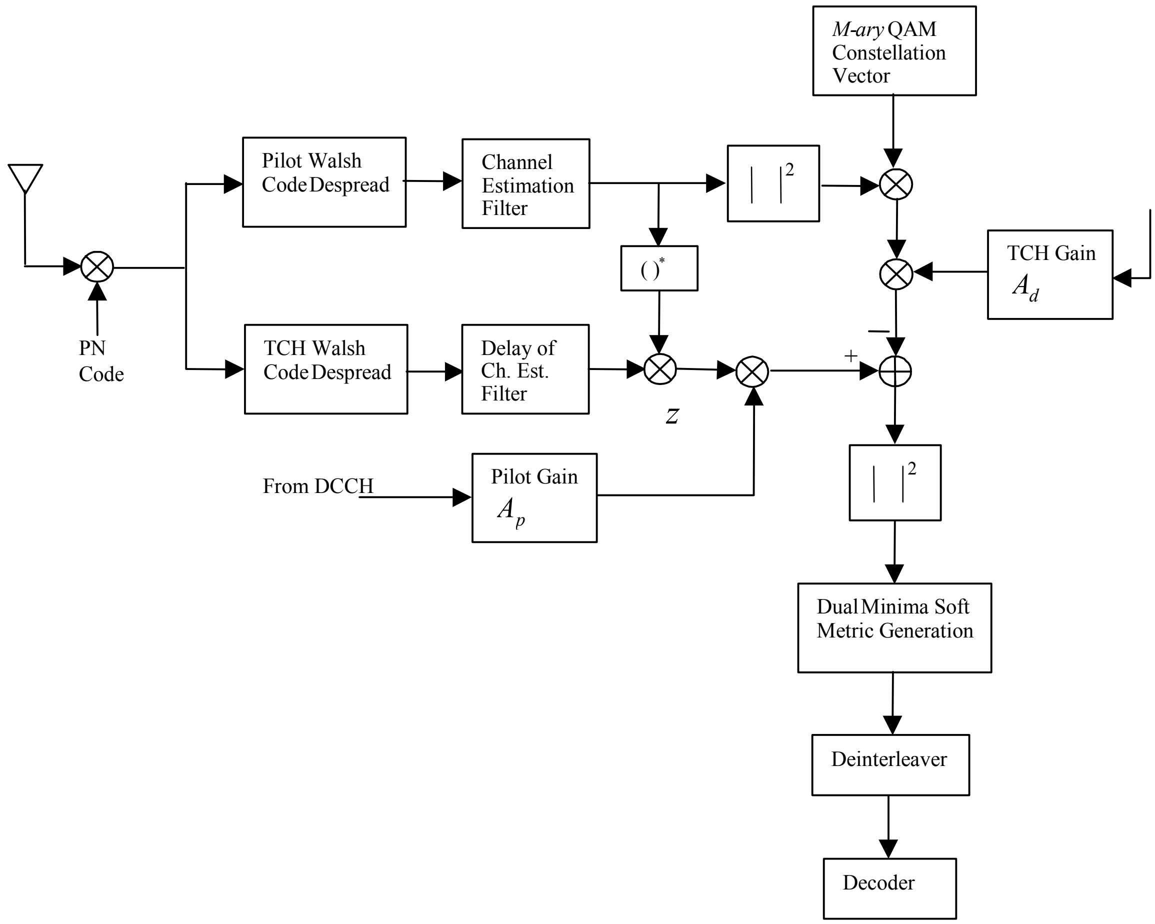


Figure H-1 QAM Receiver Block Diagram

APPENDIX I: 19 CELL WRAP-AROUND IMPLEMENTATION

The cell layout is wrap-around to form a toroidal surface to enable faster simulation run times. A toroidal surface is chosen because it can be easily formed from a rhombus by joining the opposing edges. To illustrate the cyclic nature of the wrap-around cell structure, this set of 19 cells is repeated 8 times at rhombus lattice vertices as shown in Figure I-1[38]. Note that the original cell set remains in the center while the 8 sets evenly surround this center set. From the figure, it is clear that by first cutting along the blue lines to obtain a rhombus and then joining the opposing edges of the rhombus can form a toroid. Furthermore, since the toroid is a continuous surface, there are an infinite number of rhombus lattice vertices but only a select few have been shown to illustrate the cyclic nature.

[38] Note that the set of 19 cells are only repeated for illustrating the cyclic nature of the wrap-around cell structure. The simulation only contains 19 cells and not 9 sets of 19 cells.

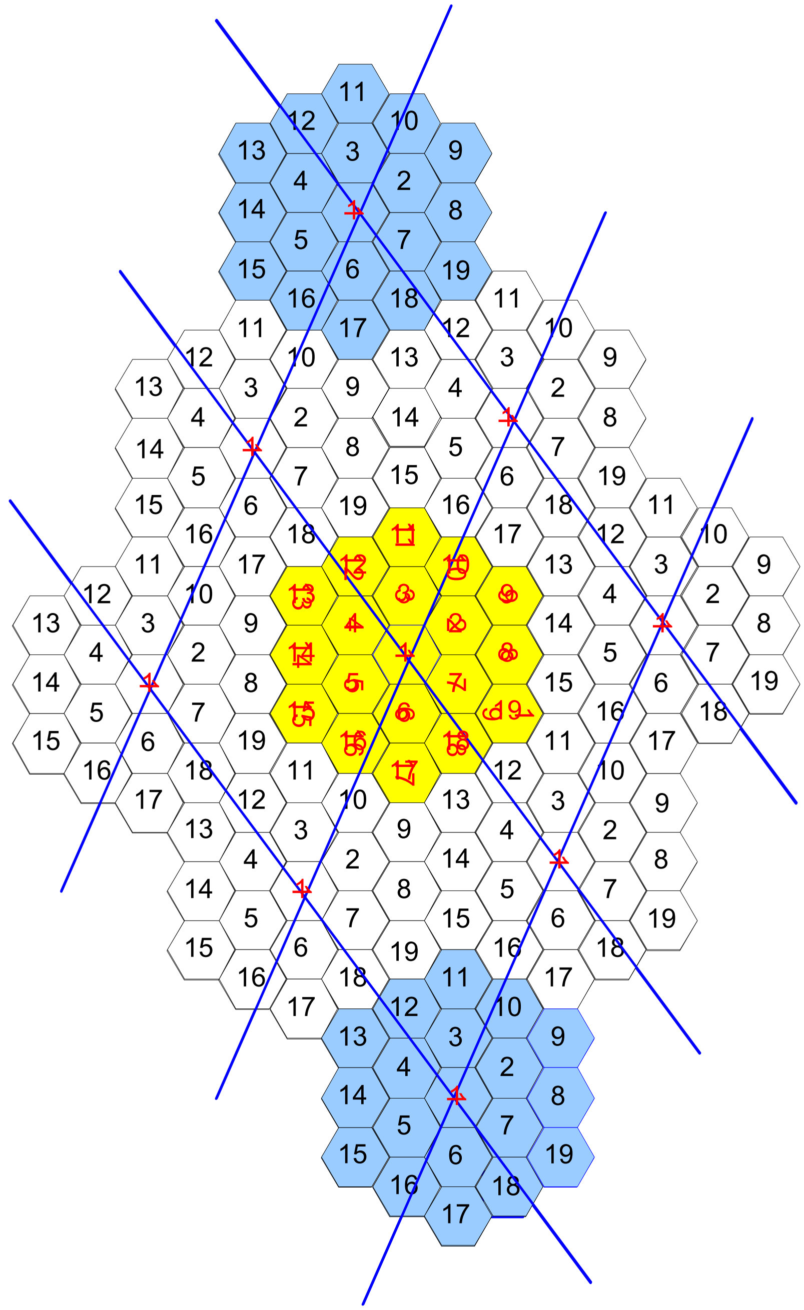


Figure I-1 Wrap-around with '9' sets of 19 cells showing the toroidal nature of the wrap-around surface.

The antenna orientations to be used in the simulation are defined in Figure 3.3.1.21-2. For simplicity, the clusters in blue from Figure 3.3.1.21-1 have been deleted in this Figure. The distance from any MS to any base station can be obtained from the following algorithm: Define a coordinate system such that the center of cell 1 is at (0,0). The path distance and angle used to compute the path loss and antenna gain of a MS at (x,y) to a BS at (a,b) is the minimum of the following:

a. Distance between (x,y) and (a,b);

b. Distance between (x,y) and $(a+3R, b+8\sqrt{3}R/2)$;

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

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

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

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

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

where R is the radius of a circle that connects the six vertices of the hexagon.

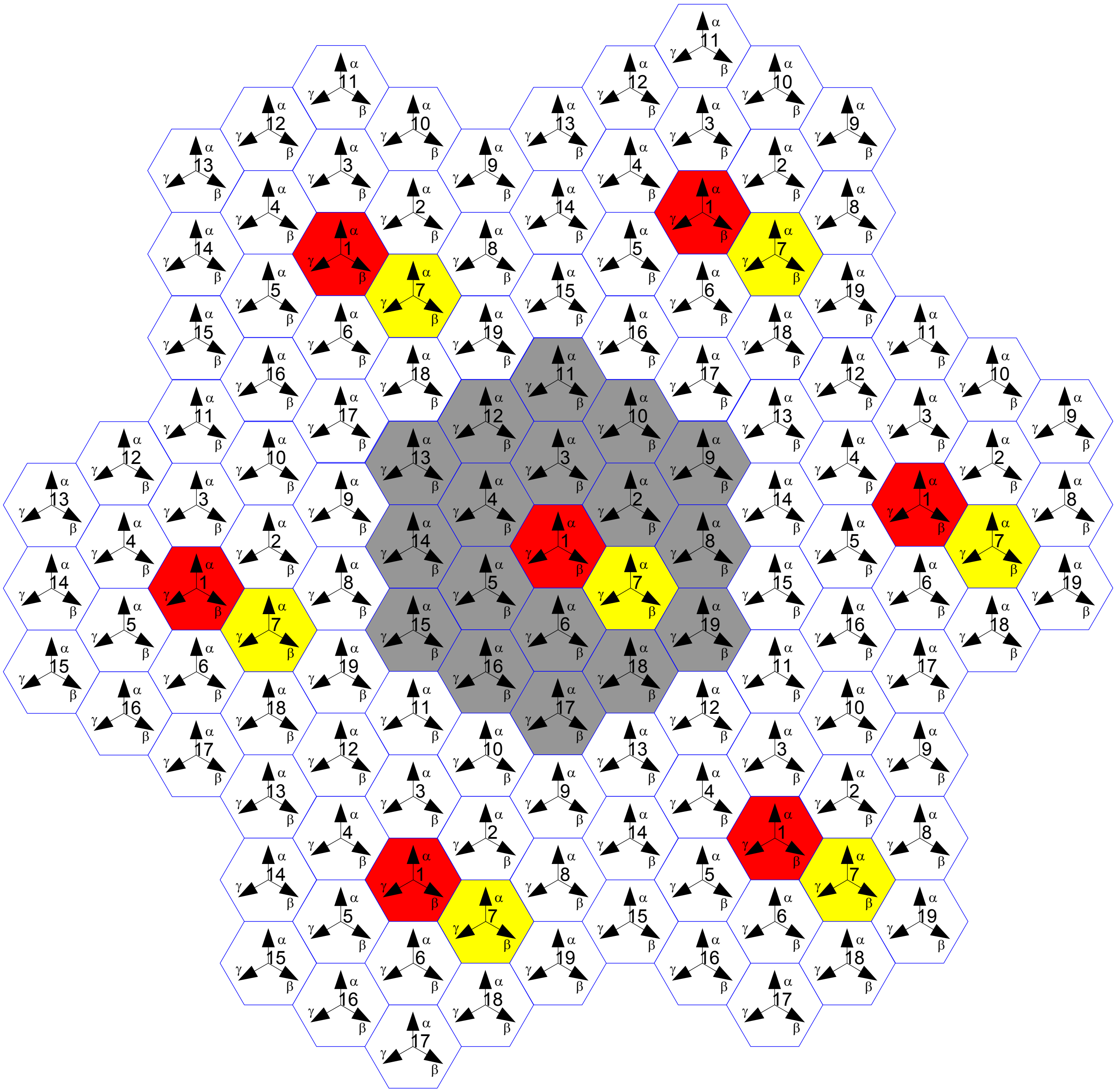


Figure I-2: The antenna orientations to be used in the wrap-around simulation. The arrows in the Figure show the directions that the antennas are pointing.

APPENDIX J: LINK LEVEL SIMULATION PARAMETERS

Table J-1 and Table J-2 summarize the default simulation parameters to use in generating the reference curves for system level evaluations.

Table J-1 Link Level Simulation Parameters for Forward Link

<table>
<tr><th colspan="2">Parameter</th><th>Value</th></tr>
<tr><td colspan="2">Chip rate</td><td>1.2288 Mcps</td></tr>
<tr><td colspan="2">Carrier frequency</td><td>2000 MHz</td></tr>
<tr><td colspan="2">Channel models</td><td>Table 2.2.1-1 and Table 2.2.1-2 specified in section 2.2.1.</td></tr>
<tr><td colspan="2">MS Receive diversity</td><td>1 RX antenna</td></tr>
<tr><td colspan="2">PC step size</td><td>0.5 dB</td></tr>
<tr><td colspan="2">PC command error rate</td><td>4%</td></tr>
<tr><td colspan="2">PC outer loop</td><td>0.5 dB for up-step</td></tr>
<tr><td colspan="2">PC delay</td><td>1 PCG/slot</td></tr>
<tr><td rowspan="2">Channel estimation for traffic channel</td><td>Circuit switched and packet switched data systems (e.g., 1xEV-DV)</td><td>1st order filtering of forward common pilot channel</td></tr>
<tr><td>Packet switched data systems (e.g., 1xEV-DO)</td><td>Based on two consecutive pilot bursts (96 chips each) with linear interpolation aligned to achieve zero group delay with respect to the traffic</td></tr>
<tr><td rowspan="2">Power estimation for power control</td><td>Circuit switched and packet switched data systems (e.g., 1xEV-DV)</td><td>1 PCG accumulation of forward PCB subchannel</td></tr>
<tr><td>Packet switched data systems (e.g., 1xEV-DO)</td><td>N/A</td></tr>
</table>

Table J-2 Link Level Simulation Parameters for Reverse Link

<table>
<tr><th colspan="2">Parameter</th><th>Value</th></tr>
<tr><td colspan="2">Chip rate</td><td>1.2288 Mcps</td></tr>
<tr><td colspan="2">Carrier frequency</td><td>2000 MHz</td></tr>
<tr><td colspan="2">Channel models</td><td>Table 2.2.1-1 and Table 2.2.1-2 specified in section 2.2.1.</td></tr>
<tr><td colspan="2">BS Receive diversity</td><td>2 RX antennas</td></tr>
<tr><td colspan="2">PC step size</td><td>1 dB</td></tr>
<tr><td colspan="2">PC command error rate</td><td>4%</td></tr>
<tr><td colspan="2">PC outer loop</td><td>0.5 dB for up-step</td></tr>
<tr><td colspan="2">PC delay</td><td>1 PCG/slot</td></tr>
<tr><td rowspan="2">Channel estimation for traffic channel</td><td>Circuit switched and packet switched data systems (e.g., 1xEV-DV)</td><td>Equal tap FIR filter (Length 2 PCG) of Reverse pilot channel</td></tr>
<tr><td>Packet switched data systems (e.g., 1xEV-DO)</td><td>Equal tap FIR (Length 1 slot) of Reverse pilot channel</td></tr>
<tr><td rowspan="2">Power estimation for power control</td><td>Circuit switched and packet switched data systems (e.g., 1xEV-DV)</td><td>1 PCG accumulation of Reverse pilot channel</td></tr>
<tr><td>Packet switched data systems (e.g., 1xEV-DO)</td><td>N slot(s) accumulation of Reverse pilot channel Ec/Nt for a power control update rate of 600/N Hz</td></tr>
</table>

The link level simulation for reverse link could be performed with one sample per symbol to expedite the simulations. In this case, the multi-path interference shall be modeled as AWGN. The total variance of noise and multi-path interference component in the j-th path of the a-th antenna for the k-th received symbol in the n-th PCG shall be[39]:

[39] The Gaussian approximation was considered reasonable tradeoff for simulation simplicity.

$$N_t^{(n,k,a,j)} = \sum_{\substack{i \\ i \neq j}} \left| \alpha^{(n,k,a,i)} \right|^2 E_{c\_total} + N_o \,, \tag{J-1}$$

where E_{c_total} is the total transmitted chip energy of all channels. $\alpha^{(n,k,a,j)}$ is the attenuation factor in the j-th path of the a-th antenna for the k-th received coded symbol in the n-th PCG.

APPENDIX K: JOINT TECHNICAL COMMITTEE (JTC) FADER

The JTC fader is best described as a filter bank, as depicted in Figure K-1. Each path of each antenna must be generated by independently. The fade multiplier output values change at the rate $64f_d$, where f_d denotes the Doppler frequency. The output power of the JTC fader is normalized to unity.

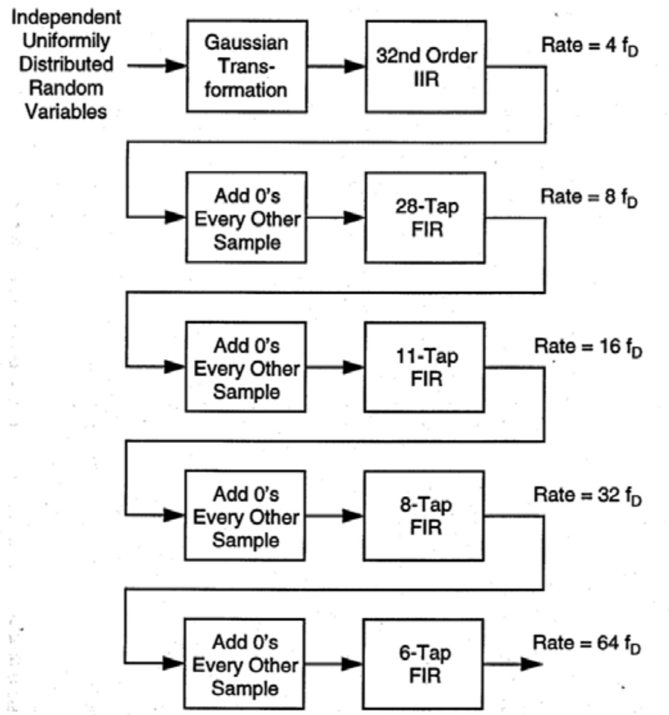


Figure K-1: I and Q Fade Multiplier Generation

A good uniform random generator must be used, and the filter clocked several times to "warm-up" before using the first fade sample.

Table K-1: Coefficients of the 6-tap FIR Filter

-0.040357044
0.086013602
0.454477919
0.454477919
0.086013602
-0.040357044

Table K-2: Coefficients of the 8-tap FIR Filter

-0.020403315

-0.027828149
0.137993831
0.410212183
0.410212183
0.137993831
-0.027828149
-0.020403315

Table K-3: Coefficients of the 11-tap Filter

0.003612442
-0.012663859
-0.043131662
0.041846468
0.289519221
0.440818845
0.289519221
0.041846468
-0.043131662
-0.012663859
0.003612442

Table K-4: Coefficients of the 28-tap Filter

0.000134449
-0.001035837
-0.001266781
0.003144467
0.005446165
-0.005705115
-0.015697801
0.005592858
0.035808300

0.004331282
-0.073120692
-0.044582508
0.174645729
0.412153786
0.412153786
0.174645729
-0.044582508
-0.073120692
0.004331282
0.035808300
0.005592858
-0.015697801
-0.005705115
0.005446165
0.003144467
-0.001266781
-0.001035837
0.000134449

Table K-5: Jakes (Classic) Spectrum IIR Filter Coefficients

Denominator Coefficients	Numerator Coefficients
1.0000000000000000	6.5248059900135200e-02
-1.2584602815172037e+01	-5.6908289014580038e-01
8.3781249094641240e+01	2.7480451166883220e+00
-3.8798703729842964e+02	-9.4773135180288293e+00
1.3927662726637102e+03	2.5786482996126544e+01
-4.1039030305379210e+03	-5.8241097311312117e+01
1.0278517997545167e+04	1.1247173657687033e+02
-2.2393748634049065e+04	-1.8904842233132774e+02
4.3133809439790406e+04	2.7936237305345003e+02

-7.4319282567554124e+04	-3.6418631194112885e+02
1.1554604041649372e+05	4.1715604202981109e+02
-1.6315680006218722e+05	-4.1320604132753033e+02
2.1026268214607492e+05	3.3901659663025242e+02
-2.4818342600838441e+05	-2.0059287960205506e+02
2.6898038693500403e+05	2.3734545818966293e+01
-2.6809721585952450e+05	1.5363912802007360e+02
2.4593366073473063e+05	-2.9424154728837402e+02
-2.0763108908648306e+05	3.7359596060374486e+02
1.6120527209223103e+05	-3.8642988435890055e+02
-1.1492103434104947e+05	3.4521505714177903e+02
7.5041686769138993e+04	-2.7265055759799253e+02
-4.4731841330872761e+04	1.9230535924562764e+02
2.4231115205405174e+04	-1.2153980630698008e+02
-1.1857508216082340e+04	6.8773930574859179e+01
5.2013837692697152e+03	-3.4696126060493945e+01
-2.0246855591971096e+03	1.5489134454590417e+01
6.9005516614518956e+02	-6.0495383196143626e+00
-2.0220131802145625e+02	2.0332679679817174e+00
4.9649188538197400e+01	-5.7404157101686004e-01
-9.8333304002079363e+00	1.3121847123296254e-01
1.4770279039919996e+00	-2.2867487042024594e-02
-1.5005452926258436e-01	2.7118486134987282e-03
7.7628588864503741e-03	-1.6371291227220021e-04

APPENDIX L: LARGEST EXTREME VALUE DISTRIBUTION

The Largest Extreme Value distribution is characterized by its probability density function (pdf) and cumulative distribution function (cdf) as follows:

$$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0 \tag{L-1}$$

$$F(x) = 1 - e^{-e^{-\frac{x-a}{b}}}, b > 0 \tag{L-2}$$

with the moments

$$\mu = a + \gamma b \tag{L-3}$$

$$\sigma^2 = \frac{1}{6}\pi^2 b^2 \tag{L-4}$$

where $\gamma \approx 0.57722$ is Euler's constant.

The random variable with an Extreme Value distribution, X, can be obtained from a random variable with a uniform distribution, Y, with the following transformation:

$$X = a - b \ln\left[-\ln(1 - Y)\right] . \tag{L-5}$$

APPENDIX M: REVERSE LINK OUTPUT MATRICES

M.1 Output Matrix

M.1.1 1xEV-DV Systems

Table M-1 Required statistics output in excel spread sheet for the base station side

Spreadsheet column ID	Name	Definition and comments.
1	Simulation run ID	Each simulation run is a new drop of users and a given simulation period.
2	MS ID	Mobile station identification for each simulation run.
3	Traffic type	HTTP =1, WAP =2, FTP = 3.
4	Channel model	Channel model is A, B, C, D, and E as defined in Table 2.2.1-1 of Section 2.2.1.
5,6	MS location (km)	Mobile station location, two columns (x, y) relative to the center of the center cell.
7	SHO Geometry (dB)	Forward link geometry for the active sectors. The SHO geometry is (Ior1+Ior2+…)/(No+Ioc), and does not have any maximum C/I limitation.
8	Geometry for primary sector (dB)	Forward link geometry for the primary (best serving) sector. This geometry is computed as Ior1/(No+Ioc), and does not have any maximum C/I limitation. Note that this is not the geometry used for generating the ESCAM delay on the F-PDCH in 1xEV-DV systems.
9	Number of active sectors	Number of sectors in the active set.
10,11,12	Active sector index	The indices of the serving sectors, best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector, are listed in the 3 respective columns.
13,14,15	Distance from MS (km)	Distance from MS to the center of each active sector. The considered sectors are the best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector, and are listed in the 3 respective columns.

Spreadsheet column ID	Name	Definition and comments.
16,17,18	FL pilot Ec/Io (dB)	Forward Link pilot Ec/Io for the best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector. They are listed in the 3 respective columns. This value does not have any maximum C/I limitation.
19,20,21	Average link gain (dB)	Average link gain for the best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector. They are listed in the 3 respective columns. Average link gain is the average propagation loss including distance dependent path loss, shadow fade, antenna gains and cable, connector and other losses.
22	MS average transmit power (dBm)	MS average transmit power per power control group (PCG), in dBm.
23	Power outage rate (%)	Ratio of the number of PCGs/slots MS doesn't follow the power control command to the total number of PCGs in simulation duration (expressed in percentage).
24, 25, 26	Average combined pilot received Ec/Nt (dB)	Combined RL pilot received Ec/Nt for the best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
27, 28, 29	Normalized average combined pilot received Ec/Nt (dB)	Normalized average combined pilot received Ec/Nt for the best serving, 2^{nd} best (if any) and 3^{rd} best (if any) sector. They are listed in the 3 respective columns. Ec/Nt is normalized by the Pilot Reference Level of the R-SCH relative to the Pilot Reference Level of the R-FCH. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
30,31,32	Average combined received	Average combined R-FCH received traffic

Spreadsheet column ID	Name	Definition and comments.
	R-FCH Ec/Nt(dB)	Ec/Nt for the best serving, 2nd best (if any) and 3rd best (if any) sector. This value is defined as accumulated R-FCH received traffic Ec/Nt divided by simulation time. Frames during which SCRM/SCRMM are sent are excluded from both the numerator and the denominator. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
33, 34, 35	Average combined received R-SCH Ec/Nt(dB)	Average combined R-SCH received traffic Ec/Nt for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
36, 37, 38, 39, 40, 41, 42	R-SCH data rate percentage (kbps)	Percentage of R-SCH data transmitted with the following rates: 0, 9.6, 19.2, 38.4, 76.8, 153.6, 307.2 [kbps]. Listed in seven columns. DTX and no-transmission are treated as rate 0 kbps.
43, 44, 45, 46, 47, 48	R-SCH FER (kbps)	R-SCH frame error rate when data is transmitted with the following rates: 9.6, 19.2, 38.4, 76.8, 153.6, 307.2 [kbps]. Listed in the six respective columns.
49	R-SCH probability of DTX	Probability of a MS being unable to transmit at the assigned rate either because it is in power outage or it doesn't have data. It is equal to (# of DTXed frames on R-SCH) / (# of R-SCH frames during the duration of the simulation when assigned rate at MS side is not 0 kbps) Frames that are zero rate because the ESCAM was not received or received late

Spreadsheet column ID	Name	Definition and comments.
		are not counted.
50	R-SCH average requested data rate [kbps]	Average R-SCH requested data rate measured at the MS, in kbps. Does not exclude CRC and tail bits.
51	R-SCH average assigned data rate (BTS) [kbps]	Average assigned R-SCH rate measured at BTS, in kbps. Does not exclude CRC and tail bits. Does not take into account whether this rate will be executed or not by the MS.
52	R-SCH average assigned data rate (MS) [kbps]	Average assigned R-SCH rate measured at MS, in kbps. Does not exclude CRC and tail bits. Differs from 51 in that assignment message may be lost by the physical layer or arrive too late.
53	R-SCH average transmitted data rate	R-SCH average transmitted data rate over the simulation time (column 63). Does not exclude CRC and tail bits.
54, 55, 56, 57	Percent of R-FCH data rate (kbps)	Ratio of the number of 9.6, 4.8, 2.7, 1.5 kbps R-FCH frames carrying data traffic (Excludes the frames for SCRM and the interrupted frames due to SCRMM) to the total number of 20ms frames in the simulation time. Listed in the 4 respective columns.
58	R-FCH FER	The frame error rate of R-FCH carrying data traffic. It is the ratio of the number of erased frames to the number of frames carrying data traffic (Excludes the frames for SCRM and the interrupted frames due to SCRMM).
59	R-FCH average transmitted data rate (kbps)	Average R-FCH transmitted data rate. It is the total transmitted bits in R-FCH frames carrying data traffic (Excludes the frames for SCRM and the interrupted frames due to SCRMM) divided by the simulation time. Does not exclude CRC and tail bits.
60	Total average transmitted data rate (kbps)	Sum of the average transmitted data rates of R-FCH and R-SCH. Does not exclude CRC and tail bits.

Spreadsheet column ID	Name	Definition and comments.
61	MS arrival time (s)	The arrival time of each MS (excluding the setup time).
62	MS departure time (s)	The departure time of each MS.
63	Simulation time (s)	Simulation time is the period between the arrival time and the departure time, excluding the setup time.
64	Total correctly received bits (kbits)	Total number of correctly received information bits per user on R-SCH and R-FCH combined. It excludes the physical layer overhead (CRC and tail bits).
65	Average physical layer data throughput (kbps)	User's total number of correctly received bits (column 64) divided by its simulation time, as defined in column 63. This throughput does not exclude zero-padding bits.
66	Total correctly received bits w overhead (kbits)	Total number of correctly received information bits per user on R-SCH and R-FCH combined. It includes the physical layer overhead (CRC and tail bits are not removed).
67	Average physical layer throughput w overhead (kbps)	User's total number of correctly received bits (column 66) divided by its simulation time, as defined in column 63.
68	Average packet call throughput per user (kbps)	Packet call for FTP, HTTP, WAP and Gaming users as defined in section 4.2. Average packet call throughput per user is defined in section 4.2.1.1.
69	Average packet delay per user (s)	Average packet delay per user is defined in Section 3.3.2.2, item 5. Note that packet time excludes the setup time.
70	Average TCP ACK delay (s)	Average TCP ACK delay is defined Section 3.3.2.2, item 4
71	Average packet call delay per user (s)	Average packet call delay per user is defined in Section 3.3.2.2, item 6. Note that packet call time excludes the setup time.

Spreadsheet column ID	Name	Definition and comments.
72	Correctly received bits on R-FCH channel (kbits)	Number of correctly received data bits on R-FCH. This does not include the CRC and tail bits.
73	Correctly received bits on R-SCH channel (kbits)	Number of correctly received data bits on R-SCH. This does not include the CRC and tail bits.
74	Average R-SCH FER	R-SCH FER averaged over all received frames.
75	Total transmitted R-SCH frames	Number of total transmitted R-SCH frames in each simulation run.
76	Total received R-SCH frames	Number of total received R-SCH frames in each simulation run.
77	Fraction of ESCAM messages arriving too late	Total number of ESCAMs received by MS after action time divided by total number of ESCAMs received by MS.
78	Percent of R-FCH for SCRM or SCRMM	Ratio of the number of R-FCH frames carrying SCRM or SCRMM to the total number of 20ms frames in the simulation time. The summation of columns 54, 55, 56, 57, and 77 should be one.
79	Total transmitted SCRM or SCRMM	Number of total transmitted SCRM or SCRMM in each simulation run.
80	SCRM or SCRMM error rate	The frame error rate of R-FCH carrying SCRM or SCRMM.
81, 82, 83	Average combined received R-CQICH Ec/Nt(dB)	Average combined R-CQICH received Ec/Nt for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
84	FTP file size [kbytes]	FTP upload file size.
85	Average queue size in MS [kbytes]	Averaged queue size in MS during the simulation time. This value should be updated every PCG.
86	Average estimated queue	Averaged estimated queue size in BSC

Spreadsheet column ID	**Name**	**Definition and comments.**
	size in BSC [kbytes]	during the simulation time. This value should be updated every PCG.
87	Average RLP layer throughput (kbit/s)	The average RLP layer throughput is computed as the ratio of the correctly received bits, excluding the physical layer CRC bits, tail bits, and any zero-padding bits, and the simulation time.
88	Gaming packet drop rate	The ratio between the total number of gaming packets dropped because they stay in MS buffer longer than 160ms, and the total number of gaming packets generated by the mobile station
89	Gaming outage indicator	“1” if this user in outage, “0” otherwise

Table M-2 Required statistics output in excel spread sheet for the base station side

Spreadsheet column ID	**Name**	**Definition and comments.**
1	Simulation run ID	Each simulation run is a new drop of users and a given simulation period.
2	Sector ID	Base Station Sector ID.
3	Average ROT(dB)	Average rise over thermal.
4	Prob(ROT_1.25ms > 7 dB)	Probability of short term (1.25ms) ROT exceeding 7 dB.
5	Average load	The average load of the sector of interest, where the load is computed each power control group (PCG) using the combined SINR per PCG per antenna. SINR used is pilot–weighted combined SINR computed over different fingers and both antennas of the sector of interest. Note that this load is not the estimated load that the scheduler uses. It is the actual load.
6	Number of voice MSs	Number of voice mobile stations that have the sector of interest as the primary sector.

Spreadsheet column ID	Name	Definition and comments.
7	Number of voice outage MSs	Number of voice mobile stations that have the sector of interest as the primary sector, and are in outage.
8	Number of data MSs	Number of data mobile stations that have the sector of interest as the primary sector.
9	Number of data outage MSs	Number of data mobile stations that have the sector of interest as the primary sector, and are in outage.

M.1.2 1xEV-DO Systems

Table M-3 Required statistics output in excel spread sheet for the base station side

Spreadsheet column ID	Name	Definition and comments.
1	Simulation run ID	Each simulation run is a new drop of users and a given simulation period.
2	MS ID	Mobile station identification for each simulation run.
3	Traffic type	HTTP =1, WAP =2, FTP = 3.
4	Channel model	Channel model is A, B, C, D, and E as defined in Table 2.2.1-1 of Section 2.2.1.
5,6	MS location (km)	Mobile station location, two columns (x, y) relative to the center of the center cell.
7	SHO Geometry (dB)	Forward link geometry for the active sectors. The SHO geometry is (Ior1+Ior2+…)/(No+Ioc), and does not have any maximum C/I limitation.
8	Geometry for primary sector (dB)	Forward link geometry for the primary (best serving) sector. This geometry is computed as Ior1/(No+Ioc), and does not have any maximum C/I limitation.
9	Number of active sectors	Number of sectors in the active set.
10,11,12	Active sector index	The indices of the serving sectors, best serving, 2nd best (if any) and 3rd best (if any) sector, are listed in the 3 respective

Spreadsheet column ID	Name	Definition and comments.
		columns.
13,14,15	Distance from MS (km)	Distance from MS to the center of each active sector. The considered sectors are the best serving, 2nd best (if any) and 3rd best (if any) sector, and are listed in the 3 respective columns.
16,17,18	FL pilot Ec/Io (dB)	Forward Link pilot Ec/Io for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. This value does not have any maximum C/I limitation.
19,20,21	Average link gain (dB)	Average link gain for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. Average link gain is the average propagation loss including distance dependent path loss, shadow fade, antenna gains and cable, connector and other losses.
22	MS average transmit power (dBm)	MS average transmit power per slot, in dBm.
23	Power outage rate (%)	Ratio of the number of slots MS doesn't follow the power control command to the total number of slots in simulation duration (expressed in percentage).
24, 25, 26	Average combined pilot received Ec/Nt (dB)	Combined RL pilot received Ec/Nt for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
27,28,29	Average combined received RTC Ec/Nt(dB)	Average combined RTC received traffic Ec/Nt for the best serving, 2nd best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.

Spreadsheet column ID	**Name**	**Definition and comments.**
30, 31, 32, 33, 34, 35	RTC data rate percentage (kbps)	Percentage of RTC data transmitted with the following rates: 0, 9.6, 19.2, 38.4, 76.8, 153.6[kbps]. Listed in six columns. More columns shall be added for any additional set of data rates.
36, 37, 38, 39, 40	RTC FER (kbps)	RTC frame error rate when data is transmitted with the following rates: 9.6, 19.2, 38.4, 76.8, 153.6 [kbps]. Listed in the five respective columns. More columns shall be added for any additional set of data rates.
41	RTC average requested data rate [kbps]	Average RTC requested data rate measured at the MS, in kbps. Does not exclude CRC and tail bits. Applicable only to proposals with centralized RL scheduler.
42	RTC average assigned data rate (BTS) [kbps]	Average assigned RTC rate measured at BTS, in kbps. Does not exclude CRC and tail bits. Does not take into account whether this rate will be executed or not by the MS. Applicable only to proposals with centralized RL scheduler.
43	RTC average assigned data rate (MS) [kbps]	Average assigned RTC rate measured at MS, in kbps. Does not exclude CRC and tail bits. Differs from 42 in that assignment message may be lost by the physical layer or arrive too late. Applicable only to proposals with centralized RL scheduler.
44	RTC average transmitted data rate	RTC average transmitted data rate over the simulation time (column 47). Does not exclude CRC and tail bits.
45	MS arrival time (s)	The arrival time of each MS (excluding the setup time).
46	MS departure time (s)	The departure time of each MS.
47	Simulation time (s)	Simulation time is the period between the arrival time and the departure time, excluding the setup time.
48	Total correctly received	Total number of correctly received

Spreadsheet column ID	Name	Definition and comments.
	bits (kbits)	information bits per user on RTC. It excludes the physical layer overhead (CRC and tail bits).
49	Average physical layer data throughput (kbps)	User's total number of correctly received bits (column 48) divided by its simulation time, as defined in column 47. This throughput does not exclude zero-padding bits.
50	Total correctly received bits w overhead (kbits)	Total number of correctly received information bits per user on RTC. It includes the physical layer overhead (CRC and tail bits are not removed).
51	Average physical layer throughput w overhead (kbps)	User's total number of correctly received bits (column 50) divided by its simulation time, as defined in column 47.
52	Average packet call throughput per user (kbps)	Packet call for FTP, HTTP, WAP and Gaming users as defined in section 4.2. Average packet call throughput per user is defined in section 4.2.1.1.
53	Average packet delay per user (s)	Average packet delay per user is defined in Section 3.3.2.2, item 5. Note that packet time excludes the setup time.
54	Average TCP ACK delay (s)	Average TCP ACK delay is defined Section 3.3.2.2, item 4
55	Average packet call delay per user (s)	Average packet call delay per user is defined in Section 3.3.2.2, item 6. Note that packet call time excludes the setup time.
56	Average RTC PER	RTC PER averaged over all received physical-layer packets.
57	Total transmitted RTC physical-layer packets	Number of total transmitted RTC physical-layer packets in each simulation run.
58	Total received RTC physical-layer packets	Number of total received RTC physical-layer packets in each simulation run.
59, 60, 61	Average combined received DRC (or equivalent)	Average combined DRC (or equivalent) received Ec/Nt for the best serving, 2nd

Spreadsheet column ID	Name	Definition and comments.
	Ec/Nt(dB)	best (if any) and 3rd best (if any) sector. They are listed in the 3 respective columns. If sectors are in softer-handoff, then the combined value should be entered in the columns corresponding to those sectors.
62	FTP file size [kbytes]	FTP upload file size.
63	Average queue size in MS [kbytes]	Averaged queue size in MS during the simulation time. This value should be updated every slot.
64	Average estimated queue size in BSC [kbytes]	Averaged estimated queue size in BSC during the simulation time. This value should be updated every slot.
65	Average RLP layer throughput (kbit/s)	The average RLP layer throughput is computed as the ratio of the correctly received bits, excluding the physical layer CRC bits, tail bits, and any zero-padding bits, and the simulation time.
66	Gaming packet drop rate	The ratio between the total number of gaming packets dropped because they stay in MS buffer longer than 160ms, and the total number of gaming packets generated by the mobile station
67	Gaming outage indicator	“1” if this user in outage, “0” otherwise

Table M-4 Required statistics output in excel spread sheet for the base station side

Spreadsheet column ID	Name	Definition and comments.
1	Simulation run ID	Each simulation run is a new drop of users and a given simulation period.
2	Sector ID	Base Station Sector ID.
3	Average ROT(dB)	Average rise over thermal.
4	Prob(ROT_1.67 ms > 7 dB)	Probability of short term (1.67 ms) ROT exceeding 7 dB.
5	Average load	The average load of the sector of interest, where the load is computed each power control group (slot) using the

Spreadsheet column ID	Name	Definition and comments.
		combined SINR per slot per antenna for all users in the system. SINR used is pilot–weighted combined SINR computed over different fingers and both antennas of the sector of interest. Note that this load is not the estimated load that the scheduler uses. It is the actual load considering all users in the system.
6	Average DV load	The average load of the sector of interest, where the load is computed each power control group (slot) using the combined SINR per slot per antenna for all users who consider the sector as a part of the active set. SINR used is pilot–weighted combined SINR computed over different fingers and both antennas of the sector of interest. Note that this load is not the estimated load that the scheduler uses. It is the actual load considering all users who considers the sector as a part of the active set.
6	Number of data MSs	Number of data mobile stations that have the sector of interest as the primary sector.
7	Number of data outage MSs	Number of data mobile stations that have the sector of interest as the primary sector, and are in outage.

M.2 Definitions

Average packet call throughput per user is computed as the ratio of the number of correctly received bits[40] (that exclude CRC, tail bits and zero padding bits) during the simulation and the sum of packet call delays during the simulation.

[40] The correctly received bits are considered and not the correctly received TCP segments because RLP is not modeled and therefore TCP segment loss rate may be very high.

FTP Traffic Model Packet call delay is measured as the time elapsed between the instant a user arrives in the system and the instant the user departs from the system.

WAP Traffic Model Packet call delay is measured as the time elapsed between the instant a WAP request is generated and the instant the next reading time begins.

HTTP Traffic Model Packet call delay is measured as the time elapsed between the instant a main page HTTP request is generated and the instant the next reading time begins.

Gaming Traffic Model Packet call delay is measured as the time elapsed between the instant a user arrives in the system and the instant the user departs from the system.

Due to the fixed simulation time, there may be outstanding not completed packet call at the end of a simulation run. For such a packet call, the end of the packet call is the end of the simulation.

Users that do not have even a portion of a packet call during the time statistics is collected (simulation time without warm-up period), are not taken into account for packet call statistics.

Average packet call delay per user is computed as the average of all packet call delays of that user during the simulation.

Since for each FTP and Gaming user, there is only one packet call, the average packet call delay is the same as the packet call delay.

Average packet delay per user is computed as the average of all packet delays during the simulation.

FTP Traffic Model Packet delay is defined as the time elapsed between the instant a TCP segment is passed to the transmitter physical layer buffer at the mobile station and the instant all physical layer packets that contain the TCP segment bits are received at a base station.

If HARQ operation is implemented, the TCP segment is considered "received" when all physical layer packets that contain the TCP segment bits are ACKed or after being transmitted the maximum number of times.

WAP Traffic Model Packet delay is defined as the time elapsed between the instant a WAP request is passed to the transmitter physical layer buffer at the mobile station and the instant all physical layer packets that contain the WAP request bits are received at a base station.

If HARQ operation is implemented, the WAP request is considered "received" when all physical layer packets that contain the WAP request bits are ACKed or after being transmitted the maximum number of times.

HTTP Traffic Model Packet delay is defined as the time elapsed between the instant an HTTP request (main page request or embedded object request) is passed to the transmitter physical layer buffer at the mobile station and the instant all physical layer packets that contain the HTTP request bits are received at a base station.

If HARQ operation is implemented, the HTTP request is considered "received" when all physical layer packets that contain the HTTP request bits are ACKed or after being transmitted the maximum number of times.

Gaming Traffic Model Packet delay is defined as the time elapsed between the instant a gaming packet is passed to the transmitter physical layer buffer at the mobile station and the instant all physical layer packets that contain the gaming packet bits are received at a base station.

If HARQ operation is implemented, a gaming packet is considered "received" when all physical layer packets that contain the gaming packet bits are ACKed or after being transmitted the maximum number of times.

Average TCP ACK delay per user is calculated as the average of all TCP ACK delays during the simulation.

TCP ACK delay is defined as the time elapsed between the instant an TCP ACK is generated at the mobile station and the instant the TCP ACK is received at the base station.

If HARQ operation is implemented, the TCP ACK is considered "received" when all physical layer packets that contain the TCP ACK bits are ACKed or after being transmitted the maximum number of times.

Applicable only to HTTP users.

APPENDIX N: LINK PREDICTION METHODOLOGY FOR UPLINK SYSTEM SIMULATIONS

Figure N-1 below outlines the procedure with supporting notation in Table N-1.

Inputs:
Packet-related information: $N,\ R_{eff},\ E_i,$ modulation.
Channel-related information: Channel model, $\alpha_{i,p},\ \sigma^2_{i,p}$.
SNR of channel estimator : $\Gamma_{i,p}$.
$(p = 1,\ldots,P,$ and $i = 1,\ldots,n)$

↓

Link error prediction algorithm
Output: $\Sigma_{E,eff}$
use Eqn. 0.1 or 0.2 or 0.3

↓

Probability of the packet being in error equals
$$f_{MC}\left(\Sigma_{E,eff} - \gamma_P(R_{eff}) - \gamma_D(\text{Channel model}) + \gamma_{NG}(\text{Channel model})\right),$$
or, equivalently,
$$f_C\left(\Sigma_{E,eff} + \gamma_{NG}(\text{Channel model}),\ R_{eff},\ \text{Channel model}\right)$$

Figure N-1: Outline of Equivalent SNR Method.

The slots spanned by the transmission include H-ARQ retransmissions, if any, and, therefore, need not be contiguous. The output of the link error prediction algorithm (see Figure N-1) is the effective Eb/N0, $\Sigma_{E,eff}$, for the transmission[41]. This is calculated *analytically* using Eqn. N-1 for BPSK and N-2 or N-3, for QPSK.

[41] A block transmission, as referred to here, may consist of one or more H-ARQ retransmissions.

1 **Table N-1. Notations used.**

Variable	**Description**
R_{eff}	Effective code rate $= \frac{N}{N + C_{eff}}$, where C_{eff} is the total number of unique parity bits – excluding systematic bits – transmitted so far[42].
$G_{i,p}$	Complex channel gain on path *p* in slot *i*, which equals $\alpha_{i,p}e^{j\theta_{i,p}}$, where $\alpha_{i,p}$ is the magnitude of the channel gain, and $\theta_{i,p}$ is the channel phase.
$\hat{G}_{i,p}$	Receiver's estimate of the channel gain $G_{i,p}$, which will also be written as $\hat{G}_{i,p} = \hat{\alpha}_{i,p}e^{j\hat{\theta}_{i,p}}$.
$\gamma_P(R_{eff})$	Puncturing penalty for a code of effective rate R_{eff}.
γ_D(Channel model)	Doppler penalty for the given channel model.
γ_{NG}(Channel model)	Adjustment term, which will be called the *Non-Gaussian gain*. This is to account for the fact that the noise introduced due to demodulation using imperfect channel estimates in non-Gaussian.
$f_{MC}(x)$	FER for the mother code, when Eb/N0 equals *x*, on an AWGN channel.
$f_C(x, R_{eff}, \text{Channel model})$	FER for a code of effective rate R_{eff} when the short term Eb/N0 equals *x* on the given channel.
N	Number of information bits.
n	Total number of slots spanned by the transmission (need not be contiguous).
S	Number of modulation symbols transmitted in each slot.
E_i	Transmit energy per modulation symbol in slot *i*.
P	Total number of resolvable paths of the channel.
$\alpha_{i,p}$	Magnitude of the channel gain on path *p* in slot *i*.

42 Note that the effective code rate, by definition, is no smaller than the mother code rate – denoted R_{MC} – and, in particular, remains unchanged in the case of Chase combining. In the case where incremental redundancy is used, the formula for effective rate may need some modification to account for the following case: When the subpacket code rates are very high, and the received SNRs of the subpackets are significantly different, then the link performance can be worse than that due to a transmission of effective code rate $N/(N + C_{eff})$.

Variable	**Description**
$\sigma_{i,p}^2$	Interference plus thermal noise power on path *p* in slot *i*.
$\Lambda_{i,p}$	Equals $\alpha_{i,p}^2 E_i / \sigma_{i,p}^2$. Therefore, this is the received modulation symbol Es/N0 on path *p* in slot *i* of the traffic channel before demodulation. This equals the SNR of the demodulated symbol only when the channel estimates are perfect.
$\Gamma_{i,p}$	"SNR of the channel estimator." See Section 3.1
N_c	Number of chips per slot
$\Omega_{i,p}^2$	Variance of the complex additive noise corrupting each pilot chip. Variance of the real and imaginary parts are each $\Omega_{i,p}^2/2$

Equations (N.1) – (N.3) are all for the case of pilot weighted combining.

Effective Eb/N0 for BPSK Modulation: In this case, $\Sigma_{E,eff}$ is given by

$$\Sigma_{E,eff} = \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}\Lambda_{i,p}\sigma_{i,p}^2\right)^2}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{\Lambda_{i,p}\sigma_{i,p}^4}{\alpha_{i,p}^2\Gamma_{i,p}}\left(\alpha_{i,p}^2\Gamma_{i,p}+\Lambda_{i,p}+1\right)\right]}, \qquad \text{(N-1)}$$
$$= \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}E_i\alpha_{i,p}^2\right)^2}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{E_i\sigma_{i,p}^2}{\Gamma_{i,p}}\left(\alpha_{i,p}^2\Gamma_{i,p}+\Lambda_{i,p}+1\right)\right]}$$

Effective Eb/N0 for QPSK Modulation with I and Q Branches on Different Walsh Codes: The effective Eb/N0, $\Sigma_{E,eff}$, in this case equals

$$\Sigma_{E,eff} = \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}\Lambda_{i,p}\sigma_{i,p}^2\right)^2}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{\Lambda_{i,p}\sigma_{i,p}^4}{\alpha_{i,p}^2\Gamma_{i,p}}\left(\alpha_{i,p}^2\Gamma_{i,p}+\frac{\Lambda_{i,p}}{2}+1\right)\right]} \qquad \text{(N-2)}$$
$$= \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}E_i\alpha_{i,p}^2\right)^2}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{E_i\sigma_{i,p}^2}{\Gamma_{i,p}}\left(\alpha_{i,p}^2\Gamma_{i,p}+\frac{\Lambda_{i,p}}{2}+1\right)\right]}.$$

Effective Eb/N0 for QPSK Modulation with I and Q Branches on the Same Walsh Code: The effective Eb/N0, $\Sigma_{E,eff}$, in this case equals

$$\Sigma_{E,eff} = \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}\Lambda_{i,p}\sigma_{i,p}^{2}\right)^{2}}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{\Lambda_{i,p}\sigma_{i,p}^{4}}{\alpha_{i,p}^{2}\Gamma_{i,p}}\left(\alpha_{i,p}^{2}\Gamma_{i,p}+\Lambda_{i,p}+1\right)\right]} = \frac{S}{N}\frac{\left(\sum_{i=1}^{n}\sum_{p=1}^{P}E_{i}\alpha_{i,p}^{2}\right)^{2}}{\sum_{i=1}^{n}\sum_{p=1}^{P}\left[\frac{E_{i}\sigma_{i,p}^{2}}{\Gamma_{i,p}}\left(\alpha_{i,p}^{2}\Gamma_{i,p}+\Lambda_{i,p}+1\right)\right]}. \quad \text{(N-3)}$$

N.1 Definition of Required Terms

N.1.1 Channel Estimation SNR, $\Gamma_{i,p}$

The receiver's estimate $\hat{G}_{i,p}$ of the channel gain $G_{i,p}$ on path *p* in slot *i* can, in general, be written as

$$\hat{G}_{i,p} = G_{i,p} + \eta_{i,p} \quad \text{(N-4)}$$

where $\eta_{i,p} \sim \mathrm{C}(0,\ 1/\Gamma_{i,p})$. Clearly the variance of $\eta_{i,p}$, denoted $1/\Gamma_{i,p}$, determines the "quality" of the receiver's estimate of the channel gain, which leads to the interpretation that $\Gamma_{i,p}$ is the SNR of the channel estimator. The value of $\Gamma_{i,p}$ depends on the pilot transmission power, the interference plus thermal noise on the pilot channel, and the channel estimator being used. The SNR of specific channel estimators are now provided.

For a channel estimator based on one-slot rectangular filter averaging of the pilot symbols

$$\Gamma_{i,p} = E_i^P N_C / \Omega_{i,p}^2, \quad \text{(N-5)}$$

where E_i^P is the transmit pilot energy per pilot chip, and N_C is the number of chips in the slot and $\Omega_{i,p}^2$ is the variance of the complex additive interference and noise for each pilot chip.

In the case of channel estimation based on two-slot rectangular filter averaging, the channel estimates in adjacent slots become correlated. For the purpose of illustration, consider channel estimation based on averaging two successive slots. The channel estimate for slot *i* is based on averaging the pilot signal from slots (*i-1*) and *i*. Suppose transmission takes place in slots *i* and (*i+1*). There are two sources of correlation in the channel estimates. Firstly, the channel estimates in the two slots both depend on the channel gain in slot *i*. Secondly, both channel estimates depend on the additive noise on the pilot in slot *i*. Two simple approaches are outlined in [28]. One approach captures the correlation due to the channel gains but not the correlation due to the additive noise.

For the case where the center tap of the rectangular FIR filter is at the center of slot $(i-1)$, so that the channel estimate is based on the pilot signal received in slot *i*, $(i-1)$ and $(i+1)$, an equivalent expression for the channel estimation SNR is computed as

$$\Gamma_{i,p} = \frac{4}{\frac{1}{4}\left|\delta_{i,p} - \delta_{i+1,p}\right|^2 + \frac{1}{N_c}\left[\frac{\Omega_{i,p}^2}{E_i^p} + \frac{1}{2}\left(\frac{\Omega_{i-1,p}^2}{E_{i-1}^p} + \frac{\Omega_{i+1,p}^2}{E_{i+1}^p}\right)\right]}, \tag{N-6}$$

where $\delta_{i,p} = G_{i-1,p} - G_{i,p}$ equals the change in channel condition from slot $(i-1)$ to slot i on path p. The effective Eb/N0 is calculated as before using equation (N-1) or (N-2) or (N-3). In the second approach, the correlation in channel gains and noise are captured and an exact expression for the effective Eb/N0 is calculated (see [28]). Results indicate that the first approach, which is simpler, is accurate and therefore the second approach is not described here.

N.1.2 Non-Gaussian Penalty, γ_{NG}

When the channel estimates are very poor – as is the case at very low pilot SNRs –the FER vs effective Eb/N0 curve and the FER curve with perfect demodulation show a small relative shift. This is because the noise introduced by imperfect demodulation is not Gaussian. Therefore, in certain cases, a small adjustment term – called the non-Gaussian gain – may be necessary. Refer to Appendix D in [28] for a more detailed discussion.

N.1.3 Reference Curves

Once the effective Eb/N0 is computed, the probability of error for the transmission or FER, can be obtained in two alternative ways. In the first method, the probability of error for the transmission or FER is simply $f_{MC}\left(\Sigma_{E,eff} - \gamma_P(R_{eff}) - \gamma_D(\text{Channel model}) + \gamma_{NG}(\text{Channel model})\right)$ i.e., the FER is obtained using the lookup curve f_{MC} after adjusting the effective Eb/N0, $\Sigma_{E,eff}$, by applying the Doppler penalty, the puncturing penalty, and the non-Gaussian gain in a manner similar to the methods in Appendix F and [26] Let $f_R(x)$ be the FER of a turbo code of effective rate R on an AWGN channel, when Eb/N0 equals x and the modulation is QPSK. The *puncturing penalty* for the code $\gamma_P(R)$ is simply the additional Eb/N0 required for the code to achieve an FER of 0.01, when compared with the mother code. Mathematically, $\gamma_P(R)$ is such that $f_{MC}(E_{MC}^{1\%}) = f_R\left(E_{MC}^{1\%} + \gamma_P(R)\right) = 0.01$, where $E_{MC}^{1\%}$ is the Eb/N0 required to achieve an FER of 1% for the mother code on the AWGN channel. The *Doppler penalty* captures the channel variability during the transmission as compared to an AWGN channel where the channel gain would remain unchanged during the transmission and is obtained in a manner similar to the puncturing penalty. An accurate method for characterizing this penalty due to channel variations would allow the use of a single AWGN reference curve for all cases, while still capturing the effect of noisy pilot using the ESM method.

In the second method, short-term FER curves for the given channel model and effective code rate are used to obtain the FER. Here, the puncturing and Doppler penalties are not applied, but the non-Gaussian penalty may still be applicable.

The method of generating the short-term curves depends on whether the traffic channel is power controlled or not. The short term FER curve for a given channel model $f_C(x,\ R_{eff},\ \text{Channel model})$ is obtained from link level simulations, assuming perfect channel estimation on each resolvable path of the channel. The goal of this is to characterize the probability of error for a packet transmission over the given channel model when the received Eb/N0 equals x. Of course, since the power of the received signal fluctuates over of the course of the transmission, $f_C(x,\ R_{eff},\ \text{Channel model}) \neq f_{R_{eff}}(x)$, in general. Since the variation in the received power depends on whether or not the traffic channel is power controlled, the method of generating these short-term curves is different for the two cases. When the traffic channel is not power controlled, the power of the received signal fluctuates in accordance with the corresponding channel model. In this case, the transmit power is kept constant over the course of the transmission, so that $f_C(x,\ R_{eff},\ \text{Channel model})$ is simply the FER when the Eb/N0 for the packet transmission equals x. By contrast, when the traffic channel is power controlled, the transmit power varies over the course of the transmission, so that the fluctuation in the power of the received signal is a result of the effect of channel variation *and* the transmit power variation due to power control. Note that since the channel estimation is assumed to be perfect, the degradation due to imperfect channel estimation is not captured in these short-term curves.

With combining of multiple Hybrid-ARQ transmissions, the effect of diversity due to the multiple transmissions would need to be captured in the reference curves[43]. The most precise method would be to have reference curves parameterized based on channel model, code rate, receive antenna diversity order and number of transmissions. However, to keep the number of reference curves reasonable, some simplifications are possible.

a) At low speeds and two receive antennas, a single short-term curve may suffice. For example, for Channel Model A (Pedestrian A) with two receive antennas, a single reference curve per code rate is sufficient and *multiple reference curves based on number of transmissions is not needed*.

b) At medium speeds, if the overall "diversity order" is high, (e.g., large number of multipaths in the channel models and/or the use of receive diversity), then again a single curve may be sufficiently accurate in predicting performance. For example, for Channel Model B (Pedestrian B) and C (Vehicular A) with two receive antennas, a single curve per code rate is sufficient and *multiple reference curves based on number of transmissions is not needed*.

c) Furthermore, if the diversity order is low and the channel variation is high, then some simplifications can still be used to minimize the number of reference curves. For example, the reference curve for the first transmission with two receive antennas is the same as that for the second transmission with only one receive antenna.

[43] Note that the diversity obtained from multiple transmissions reduces the extent of channel variations.

APPENDIX O: REVERSE LINK HYBRID ARQ: LINK ERROR PREDICTION METHODOLOGY BASED ON CONVEX METRIC

A link error prediction (LEP) method that can predict, by using single AWGN reference curve, the performance of different fading channels, the performance of multiple re-transmissions with power imbalances and the performance of different frame durations is described in this text. The method uses the Equivalent SNR Method (ESM) [3] to calculate the effects of a weak pilot signal. The main idea of this method is using a new SNR metric based on Shannon's channel capacity formula to reflect the performance loss due to channel gain variations. The new metric can predict the performance at different fading channels, and the diversity gain due to multiple transmissions.

O.1 Convex Metric based on Channel Capacity Formula

The quasi-static method (QSM) [1] obtains the average frame E_b / N_t by averaging E_b / N_t per slot and by looking up the corresponding frame FER from a reference curve. However, having the same average frame E_b / N_t, the channel with less variation shows better performance than the channel with more variation. Consequently, AWGN channels and slowly varying channels have better error performance compared to fast varying channels. Since the existing QSM does not take this second order statistics into account, a new performance metric, which is based on Shannon's channel capacity formula, is proposed. Consider a deterministic fading channel where we know the channel response. Piece-wise AWGN channels can then approximate the fading channel. Then the channel capacity of the frame can be calculated as

$$C \approx \frac{1}{M} \cdot \sum_{m=1}^{M} C_m \tag{O.1-1}$$

where C is the channel capacity, M is the number of segments that have approximately constant channel response, and C_m is the channel capacity of m-th segment. If we use Gaussian signaling, C_m is expressed by

$$C_m = \log\left(1 + \frac{S_m}{N_m}\right) \tag{O.1-2}$$

where S_m and N_m are the channel energy and AWGN noise variance of m-th segment, respectively. By combining equation (O.1-1) and (O.1-2), the channel capacity is expressed by

$$\begin{aligned} C &= \frac{1}{M} \cdot \sum_{m=1}^{M} \log\left(1 + \frac{S_m}{N_m}\right) \\ &= \frac{1}{M} \cdot \log \prod_{m=1}^{M} \left(1 + \frac{S_m}{N_m}\right) \end{aligned} \tag{O.1-3}$$

As we can see from equation (O.1-3), C is maximized if all (S_m / N_m) have the same value. If the variance of (S_m / N_m) increases, the value of C decreases. This formula can explain the performance degradation due to increased mobile speed, and diversity gain. Diversity gain is obtained by making the channel variations smaller after combining multiple frames. The effective SNR, E_s / N_t, of a frame has one to one function relationships with channel capacity C, and it is calculated from equation (O.1-2) as

$$C = \log_2\left(1+\frac{E_s}{N_t}\right) \rightarrow \frac{E_s}{N_t} = 2^C - 1 \qquad \text{(O.1-4)}$$

In reality, the capacity formula in equation (O.1-4) may not be directly applied to different EP sizes and channel code rates. The size of a segment in our simulation is one PCG (1.25 ms). In case of high speed mobile, channel gain within a segment is not constant and channel variations within a segment are not properly penalized. This can be avoided by making the segment size small enough so that the channel gain is constant over a segment. Furthermore, non-ideal channel interleaver causes additional performance degradation at channels with high gain variations. After all, practical systems do not exactly follow the capacity formula, and we therefore need an adjustment factor in the equation. The modified metric is expressed as

$$C = \log_2\left(1+Q\cdot\frac{E_s}{N_t}\right) \rightarrow \frac{E_s}{N_t} = \frac{2^C - 1}{Q} \qquad \text{(O.1-5)}$$

where Q is a positive real number. A higher value of Q means more convexity of the capacity formula and more penalties for channel gain variations. Thus Q can be considered as a penalty factor.

If we use binary signaling, the mutual information formula in equation (O.1-2) needs to be changed. Let $N_o / 2$ be the noise variance of an AWGN channel. Let E be the transmitted symbol energy, i.e., if s is SNR in dBs then $E_s / N_t = 10^{s/10}$. Assume we transmit a channel bit, X, with signal to noise ratio s dB through the AWGN channel and on the other end of the channel we receive a Gaussian random variable, Y. Let Z be the LLR (log likelihood ratio) value of Y, that is

$$z = \ln\left[\frac{\Pr\left[Y \mid X = +\sqrt{E}\right]}{\Pr\left[Y \mid X = -\sqrt{E}\right]}\right]. \qquad \text{(O.1-6)}$$

Then the variance of Z is $\sigma^2 = 8\cdot 10^{s/10}$ and the mean of Z equals $\sigma^2 / 2$. Using this and the definition of the mutual information, we obtain, after some computations, the following expression

$$I(X;Y) = J(\sigma) = 1 - \int_{-\infty}^{\infty} \frac{1}{\sqrt{2\pi}\sigma} e^{-(v-\sigma^2/2)^2/2\sigma^2} \cdot \log_2\left(1+e^{-v}\right) dv.. \qquad \text{(O.1-7)}$$

Thus there is a one- to- one correspondence between the mutual information and the signal to noise ratio. There is an easy way to compute the function $J(x)$ by using the following approximation

$$J(x) \approx \begin{cases} a_1 x^3 + b_1 x^2 + c_1 x, & \text{if } x \le 1.6363 \\ 1 - \exp(a_2 x^3 + b_2 x^2 + c_2 x + d_2) & \text{if } 1.6363 \le x \le \infty \end{cases} \qquad \text{(O.1-8)}$$

where $a_1 = -0.0421061, b_1 = 0.209252$ and $c_1 = -0.00640081$ for the first approximation, and where $a_2 = 0.00181491, b_2 = -0.142675, c_2 = -0.0822054$ and $d_2 = 0.0549608$ for the second approximation.

In a similar fashion, one can determine the inverse of $J(x)$ by using the following approximation

$$J^{-1}(y) \approx \begin{cases} a_3 y^2 + b_3 y + c_3\sqrt{y}, & \text{if } 0 \le y \le 0.3646 \\ a_4 \ln(b_4(y-1)) + c_4 y & \text{if } 0.3646 \le y \le 1 \end{cases} \qquad (\text{O.1-9})$$

where $a_3 = 1.09542, b_3 = 0.214217$ and $c_1 = 2.33727$ for the first approximation, and where $a_4 = -0.706692, b_4 = -0.386013$ and $c_4 = 1.75017$ for the second approximation.

We can express the above equation as

$$C_B = f\left(\frac{E_s}{N_t}\right) \qquad (\text{O.1-10})$$

where C_B means binary capacity. The function *f( )* is a one-to-one function. Like equation (O.1-5), we have an adjustment factor *Q*. The modified formula is

$$C_B = f\left(Q \cdot \frac{E_s}{N_t}\right) \rightarrow \frac{E_s}{N_t} = f^{-1}(C_B) \cdot \frac{1}{Q} \qquad (\text{O.1-11})$$

The channel capacity of QPSK modulation is derived in a similar fashion and is approximated by

$$C_Q = 2 \cdot f\left(Q \cdot \frac{E_s}{2 \cdot N_t}\right) \qquad (\text{O.1-12})$$

Approximate channel capacities for BPSK and QPSK signalling are plotted in Figure O.1.

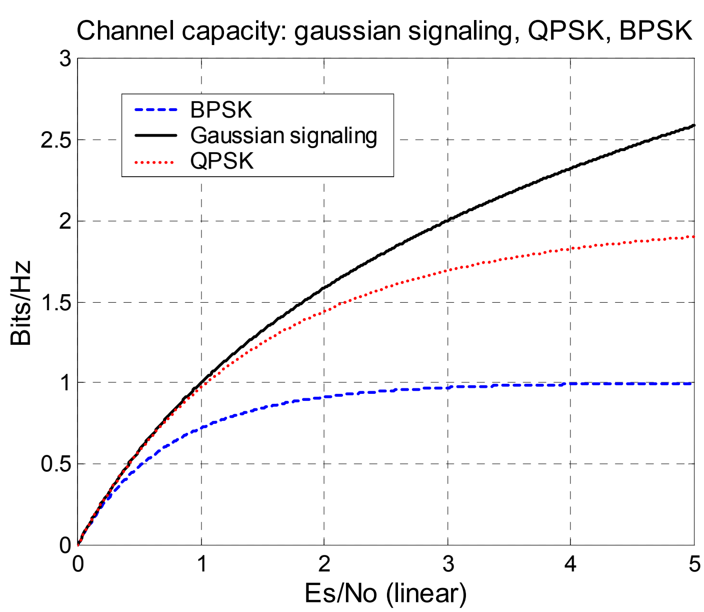


Figure O-1. Approximate channel capacities at BPSK and QPSK signaling (Gaussian signaling case is plotted as a reference).

The key point of using equation (O.1-5) or (O.1-11) is that we have metric that can account the diversity gain or the performance loss due to channel gain variations. And we may use one AWGN reference curve for all fading channels and multiple re-transmissions of H-ARQ packets. However, it does not cover the effect of weak pilot at reverse link. There was another contribution (Equivalent SNR Method: ESM) from Lucent Technologies that provides the formula to calculate the effective E_s/N_t after distortions due to weak pilot [3]. By using ESM, we may not need multiple reference curves for different pilot SNR's. As a result, only one reference AWGN reference curve is necessary for wide ranges of pilot SNR, channel variations and multiple re-transmissions. The enhanced methodology that combines ESM and convex metric is named as *"'ESM based on Convex Metric" (ECM)*, and is described in the next section.

O.2 Equivalent SNR Method based on Convex Metric (ECM)

This scheme combines ESM [3] with the convex metric based on capacity formula. The detailed procedures are explained in the following.

O.2.1 Overview of the Procedure

Assume that *M* is the number of slots, S is number of modulation symbols in each slot, and N is number of information bits. We have all other variables (like channel response, transmission power, noise variances per slot, etc.) available. The ESM method combines all these information to produce effective E_b/N_t per frame and compare it to reference curve (that is obtained on specific fading channel with perfect pilot information).

ECM uses the same formula as ESM, but calculates the effective E_s/N_t per slot. Thus we have the following *M* effective E_s/N_t values per slot.

$$\left(\frac{E_s}{N_t}\right)_1, \left(\frac{E_s}{N_t}\right)_2, \left(\frac{E_s}{N_t}\right)_3, \cdots, \left(\frac{E_s}{N_t}\right)_M \quad \text{(O.2.1-1)}$$

Then the channel capacity of each slot C_m is (assume we use the formula in equation (O.1-5). If we use BPSK or QPSK modulation, we use the formulae in equation (O.1-11) or (O.1-12), respectively)

$$C_m = \log_2\left(1 + Q \cdot \left(\frac{E_s}{N_t}\right)_m\right) \quad \text{(O.2.1-2)}$$

Then we average C_m so that the final capacity C is (from equation (O.1-1)).

$$C = \frac{1}{M} \cdot \sum_{m=1}^{M} C_m = \log_2\left(1 + Q \cdot \left(\frac{E_s}{N_t}\right)_F\right) \quad \text{(O-3)}$$

Then the new effective $(E_s/N_t)_F$ can be obtained as

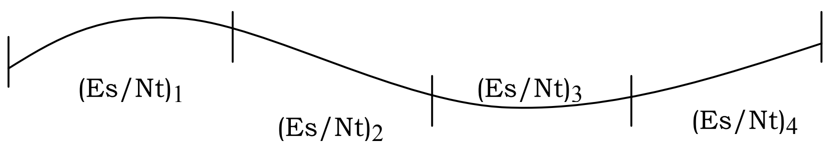
$$\left(\frac{Es}{Nt}\right)_F = \frac{2^C - 1}{Q} \quad \text{(O.2.1-4)}$$

Let $(E_b/N_t)_F$, is final effective Eb/Nt, then $(E_b/N_t)_F$ can be obtained as

$$\left(\frac{E_b}{N_t}\right)_F = N_s \cdot \left(\frac{E_s}{N_t}\right)_F \tag{O.2.1-5}$$

where N_s is the number of modulation symbols per information bit. Then we can obtain the FER by applying $(E_b/N_t)_F$ to an AWGN reference curve. This procedure is illustrated in Figure O.2.

Step 1. Calculate the equivalent $(Es/Nt)_m$ per m-th slot that has weak pilot effects.



Step 2. Calculate the channel capacity C of the frame.

$$C = \frac{1}{4} \cdot \sum_{m=1}^{4} C_m \text{, where } C_m = \log\left(1 + Q \cdot \left(\frac{E_s}{N_t}\right)_m\right)$$

Step 3. Calculate the effective (Es/Nt) of the frame, and obtain (Eb/Nt)

$$\frac{E_s}{N_t} = \frac{2^C - 1}{Q}$$

Step 4. Obtain the FER by applying (Eb/Nt) to an AWGN reference curve

Figure O-2. Overview of ECM procedure (in step 2, we assume Gaussian signaling. If BPSK or QPSK are used, channel capacity formula in equation (O.1-11) or (O.1-12) has to be used).

O.2.2 Detailed Procedure

The procedure is described with a specific example for better understanding. For convenience, all the following notations are defined in Table N-1 in [3].

Let us consider 153.6 kbps R-SCH on channel model A (3 km/h and single path fading). There are two receiver antennas (P=2), and 16 slots per frame (n=16). The number of information bits in an encoder packet is 3072 per frame (N=3072), and the number of modulation symbols per slot is 768 (S=768). Let us assume that we have a fixed traffic-to-pilot ratio (TPR), and let the energy of the transmitted pilot energy per modulation symbol be 1. Then the transmitted energy per modulation symbol in slot i, E_i, is TPR. Actually, the transmitted power is not constant over the frame because of power control. However, we assume that power control is part of the fading channel gain variations, and $\alpha_{i,p}$ includes the power control effects. We can have values of $\alpha_{i,p}$ in the system level simulation. The noise variance $\sigma_{i,p}^2$ includes the thermal noise and multi-path interference. The variance of thermal noise can be calculated from the pilot SNR set point. In this example, there are two chips per modulation symbol. Thus, the pilot energy per chip is 0.5. If the pilot set point per antenna is –20 dB, then the thermal noise variance per chip per antenna is 50. When we have more than one path per antenna, we add the effect of multi-path interference to $\sigma_{i,p}^2$. Even though the multi-path interference does not have Gaussian statistics, we model it as Gaussian random variables. The SNR of the channel estimator $\Gamma_{i,p}$ is derived in section N.1.1 [3].

There is another simple modeling of pilot estimation SNR, and it is described in the following. Given Doppler frequency f_d, the coherent accumulation loss $\Omega(f_d)$ is expressed as

$$\Omega(f_d) = \left(\frac{\sin(\pi \cdot f_d \cdot T)}{\pi \cdot f_d \cdot T} \right)^2 \qquad \text{(O.2.2-1)}$$

Where T is the integration time duration. T is 0.0025 in our simulation (2 PCG). Doppler frequency f_d is

$$f_d = \frac{v \cdot f_C}{v_C} \cdot \cos\theta \qquad \text{(O.2.2-2)}$$

Where v is mobile speed, v_C is speed of light, f_C is carrier frequency, θ is angle of arrival. If we assume θ has uniform distribution, the mean loss $E[\Omega]$ can be calculated per mobile speed. The calculated value of $E[\Omega]$ is 0.623 for 120 Km/h mobile, and 0.969 for 30 Km/h mobile. Thus the channel estimation SNR is 3072·0.623·(pilot SNR per chip) = 1914·(pilot SNR per chip) for 120 Km/h, and 2977·(pilot SNR per chip) for 30 Km/h mobiles. For channel model A and B, we may not need to put penalty on. $\Gamma_{i,p}$.

Then we apply the equation (N.1 in [3]) to calculate effective Es/Nt per slot. In this case the effective E_s/N_t of i-th slot can be calculated as

$$\left(\frac{E_s}{N_t} \right)_i = \frac{\left(\sum_{p=1}^{P} E_i \cdot \alpha_{i,p}^2 \right)^2}{\sum_{p=1}^{P} \frac{E_i \sigma_{i,p}^2}{\Gamma_{i,p}} \cdot \left(\alpha_{i,p}^2 \Gamma_{i,p} + \Lambda_{i,p} + 1 \right)} \qquad \text{(O.2.2-3)}$$

For the convenience of notation, we define the signal component A_i and noise component N_i on i-th slot as

$$A_i \equiv \sum_{p=1}^{P} E_i \cdot \alpha_{i,p}^2$$
$$N_i \equiv \sum_{p=1}^{P} \left[\frac{E_i \cdot \sigma_{i,p}^2}{\Gamma_{i,p}} \cdot \left(\sigma_{i,p}^2 \Gamma_{i,p} + \Lambda_{i,p} + 1 \right) \right] \qquad \text{(O.2.2-4)}$$

Then effective E_s/N_t per i-th slot is calculated as

$$\left(\frac{E_s}{N_t}\right)_i = \frac{A_i^2}{N_i} \qquad \text{(O.2.2-5)}$$

Then we follow the procedure described in section 3.1.

O.2.3 Combining Procedure for H-ARQ

Now, we discuss the combining procedure for H-ARQ (Chase combining and IR). First we consider Chase combining case. Lets define $A_{i,k}$ and $N_{i,k}$ as the signal component and noise component of *k*-th transmission on *i*-th slot. Then, A_i and N_i after Chase combining are calculated as

$$A_i = \sum_{k=1}^{K} A_{i,k}\ ,\quad N_i = \sum_{k=1}^{K} N_{i,k} \qquad \text{(O.2.3-1)}$$

where *K* is total number of transmissions. Then we apply equation (O.2.2-5) for calculating effective E_s/N_t of the *i*-th slot.

Consider the case of IR.[44] Suppose there are total *K* transmissions and each transmission has h(k) number of slots (where *k* = 1,2,…,*K*), and there is no overlapping of transmitted symbols. Then, we apply ECM method as if there is a single transmission of L_K slots where

$$L_K = \sum_{k=1}^{K} h(k)\,. \qquad \text{(O.2.3-2)}$$

The reference channel code rate is the effective code rate after all *K* transmissions. If there are overlapping symbols, we apply the Chase combining metric for that overlapping portion. Simulations and the range of Q values are described in the next section.

[44] Simulation results based on two transmission IR shows that we can have same Q values if the received power set-point difference of both transmissions is less than 3 dB. If the power difference is larger than 3 dB, adjustments for Q values are required.

APPENDIX P: PILOT SINR ESTIMATION FOR POWER-CONTROL COMMAND UPDATE IN LINK-LEVEL SIMULATIONS

In link-level simulations, the following method can be applied to estimate pilot Ec/Nt that is used to update the power-control command. The corresponding procedure for system-level power-control update is specified in section 3.2.3.

The pilot decovered symbols may be modeled as

$$y_{P,l,i} = N_p A_{c,p,l,i} + n_{p,l,i} = N_p \sqrt{E_{c,p,l,i}}\, e^{j\phi_{l,i}} + n_{p,l,i} \tag{P-1}$$

where N_P is the length of the pilot cover code; $A_{c,p,l,i}$, $E_{c,p,l,i}$ and $\phi_{l,i}$ denote the complex chip amplitude, energy and phase associated with the i-th decovered pilot symbol for the l-th RAKE finger output; $n_{p,l,i}$ denotes the corresponding interference which can be modeled as a Gaussian random variable with zero mean and a variance of $N_p N_{t,l}$, where $N_{t,l}$ denotes the interference variance per chip for the l-th RAKE finger output. Assuming that channel conditions remain approximately constant over $N_s = K\ N_p$ chips (i.e. the Ec/Nt averaging length specified in appendix J), the estimates of the complex amplitude $\hat{A}_{c,p,l}$, noise variance $\hat{N}_{t,l}$ over the interval of symbols [n, n+ K) can be obtained as follows:

$$\hat{A}_{c,p,l} = \frac{1}{N_s}\sum_{i=n}^{n+K-1} y_{p,l,i} \ , \qquad \hat{N}_{t,l} = \frac{1}{N_s}\sum_{i=n}^{n+K-1}\left| y_{p,l,i} - N_p\, \hat{A}_{c,p,l}\right|^2 \quad . \tag{P-2}$$

From this, the per-path, per-chip pilot SINR estimate $\left(\frac{E_{c,p}}{N_t}\right)_{est}$ used for power control may be obtained as

$$\left(\frac{E_{c,p}}{N_t}\right)_{est} = \sum_{l=1}^{L} \frac{\left|\hat{A}_{c,p,l}\right|^2}{\hat{N}_{t,l}} \tag{P-3}$$

This expression is consistent with specifications in section 3.2.3.

APPENDIX Q: 1XEV-DV REVERSE LINK SIMULATION AND SCHEDULER PROCEDURES

Q.1.1 Mobile station Requirements and Procedures

IS-2000 Release C-capable mobile stations (MS) should at least support the simultaneous operation of the following channels:

1. Reverse Fundamental Channel (R-FCH): When a voice-only MS has an active voice-call, it is carried on the R-FCH. For data-only MS, R-FCH carries signaling and data. R-FCH channel frame size, coding, modulation and interleaving are specified in [29]. For calibration purposes, we restrict the R-FCH use only Radio Configuration (RC) 3 and the 20-ms frame format.
2. Reverse Supplemental Channel (R-SCH): The MS supports one R-SCH for packet data transmissions. The R-SCH uses rates specified by RC3 in [29]. For the purpose of calibration, we restrict the use of 9.6-kbps, 20-ms convolutional coded frame format and (19.2, 38.4, 76.8, 153.6, 307.2) kbps turbo-coded, 20-ms frame format.

The following procedures are followed by IS-2000 Release C MSs:

1. Multiple Channel Adjustment Gain: When the R-FCH and the R-SCH are simultaneously active, multiple channel gain table adjustment is performed to maintain correct transmission power of the R-FCH. The traffic to pilot (T/P) ratios for all channel rate are also specified in the table 2.1.2.2.3.4-1 as Nominal Attribute Gain values [29]. The Pilot Reference Levels used for multi-channel gain adjustment are specified in Section 3 [29].
2. Discontinuous Transmission and Variable Supplemental Adjustment Gain: The MS may be assigned an R-SCH rate by a scheduler during each scheduling period. When the MS is not assigned an R-SCH rate, it will not transmit anything on the R-SCH. If the MS is assigned to transmit on the R-SCH, but it does not have any data or sufficient power to transmit at the assigned rate, it disables transmission (DTX) on the R-SCH. If the system allows it, the MS may be transmitting on the R-SCH at a rate lower than the assigned one autonomously. This variable-rate R-SCH operation has to be accompanied by the variable rate SCH gain adjustment as specified in [29]. R-FCH T/P is adjusted assuming the received pilot SNR is high enough to support the assigned rate on R-SCH.
3. Overhead transmission of R-CQICH: A data-only MS transmits extra power continuously at a fixed CQICH-to-pilot (C/P) ratio with multi-channel gain adjustment performed to maintain correct transmission power of the R-CQICH. (C/P) value is different for MS in soft-handoff from those not in soft handoff and is given in Section 3.
4. Closed-loop Power Control (PC) command: MS receives one PC command per PCG (at a rate of 800Hz) from all base stations (BSs) in the MS's Active Set. Pilot power is updated by +-1 dB based on "Or-of-Downs" rule, after respectively combining of the PC commands from co-located BSs (sectors in a given cell).
5. Rate request is done with one of two methods. Both methods are to be simulated.
    - Method **a (5 ms)**: Supplemental Channel Request Mini Message (SCRMM) on a 5-ms R-FCH: Signaling overhead due to the SCRMM should be modeled. This is

the only signaling overhead modeled on RL on R-FCH. Each SCRMM transmission is 24 bits (or 48 bits with the physical layer frame overhead in each 5-ms FCH frame at 9.6 kbps).

The MS can send the SCRMM in any periodic interval of 5 ms. If a 5-ms SCRMM needs to be transmitted, the MS interrupts its transmission of the current 20-ms R-FCH frame, and instead sends a 5-ms frame on the R-FCH. After the 5-ms frame is sent, any remaining time in the 20-ms period on the R-FCH is not transmitted. The discontinued transmission of the 20-ms R-FCH is re-established at the start of next 20-ms frame.

The frame error on the 5-ms R-FCH frames should be simulated. If a 5-ms R-FCH frame is in error, the SCRMM carried on this frame is considered lost. No LAC layer retransmission is simulated.

- Method **b (20 ms)**: Supplemental Channel Request Message (SCRM) on a 20-ms R-FCH: Signaling overhead due to the SCRM should be modeled. This is the only signaling overhead message on the R-FCH that is modeled on the RL. One 20-ms 9.6-kbps R-FCH frame transmission could be used to model the SCRM overhead[45]. No data is transmitted on the R-FCH when the SCRM is sent.

The frame error on the 20-ms R-FCH frames should be simulated. If a 20-ms R-FCH frame is in error, the SCRM carried on this frame is considered lost. No LAC layer retransmission is simulated."

The following information shall be reported by the MS to the BS on each SCRM/SCRMM transmission:

- Maximum Requested Rate: It is the maximum data rate a MS is capable of transmitting at the current channel conditions leaving headroom for fast channel variations:

$$R_{\max}(power) = \arg\max_{R} \left\{ \begin{array}{l} R: \quad Pref(R) * NormAvPiTx(PCG_i) * \\ (1 + (T/P)_R + ((T/P)_{9.6k} + C/P)\left(\dfrac{Pref(9.6k)}{Pref(R)}\right)) \\ \le Tx(max)/Headroom\_Req \end{array} \right\}$$

(Q.1.1-1)

$$NormAvPiTx(PCG_i) = \alpha_{Headroom} \frac{TxPiPwr(PCG_i)}{Pref(Rassigned)} + (1 - \alpha_{Headroom}) \times NormAvPiTx(PCG_{i-1})$$

, (Q.1.1-2)

where *Pref(R)* is the "Pilot Reference Level" value specified in the Attribute Gain Table in [29], $TxPiPwr(PCG_i)$ is the actual transmit pilot power after power

[45] With LAC layer overhead included, minimum SCRM payload is 144 bits. This can be transmitted over one 20-ms 9.6-kbps channel.

constraints on the MS side are applied in case of power outage. $NormAvPiTx(PCG_i)$ is the normalized average transmit pilot power.

- Queue Information: For simulation purposes, we can assume that perfect queue information (at the time of transmission of SCRMM) shall be reported.

6. MS receives grant information by one of the two following methods:

- Method **a**: ESCAMM from BS on 5-ms F-DCCH with rate assignment for specified scheduling duration. Method **a** is used for grant whenever method **a** is used for requesting.
- Method **b**: ESCAM from BS on F-PDCH with rate assignment for specified scheduling duration. Method **b** for grant is used whenever method **b** is used for requesting.

A 1% frame error rate is assumed for the 5-ms F-DCCH frames. The ESCAM carried on F-PDCH is considered error free due to HARQ retransmission on F-PDCH.

The assignment delays are different depending on which method is used for rate grant.

During the scheduled duration, the following procedures are performed:

- At the beginning of each 20-ms R-FCH frame, the MS transmits data at the rate of 9600 bps if it has some data in its buffer. Otherwise, the MS sends a null R-FCH frame at a rate of 1500 bps.
- The MS transmits at the assigned R-SCH rate in a given 20-ms period if the MS has more data than can be carried on the R-FCH and if the MS has decided that it would have sufficient power to transmit at the assigned rate (keeping headroom for channel variations). Otherwise, there is no transmission on the R-SCH during the frame. The MS decides that it has sufficient power to transmit on the R-SCH at the assigned rate *R* in a given 20-ms period *Encode_Delay* before the beginning of that 20-ms period if the following equation is satisfied:

$$Pref(R) * NormAvPiTx(PCG_i)\left[1+(T/P)_R+((T/P)_{R_{FCH}}+(C/P))\left(\frac{Pref(R_{FCH})}{Pref(R)}\right)\right] < \frac{Tx(max)}{Headroom\_Tx}$$

(Q.1.1-3)

Even if the MS interrupts the 20 R-FCH frame transmission to transmit a 5 ms SCRMM transmission on R-FCH, it uses T/P ratio corresponding to a 9.6kbps transmission on a 20 ms frame (3.75 dB) to determine whether it should DTX on the R-SCH. The DTX determination is done once every frame, Encode_Delay PCGs before the R-SCH transmission. If the MS disables transmission on the R-SCH, it transmits at the following power:

$$TxPwr(PCG_i) = PiTxPwr(PCG_i)\left[1+((T/P)_{R_{FCH}}+(C/P))\left(\frac{Pref(R_{FCH})}{Pref(R)}\right)\right] \quad \text{(Q.1.1-4)}$$

A MS encodes the transmission frame *Encode_Delay* before the actual transmission.

- If there is less available data than what can be transmitted at the assigned rate, the frame is padded with 0's and transmitted at the assigned rate. Only the actual data is counted towards system throughput.

7. Regardless of the grant, a data-only MS transmits at the rate of 9.6 kbps on R-FCH if

   - There is some data in the MS's buffer (as determined in Step 5 above), or,
   - There is SCRM transmission on R-FCH (see Step 4 above).

   When there is no data in the MS's buffer and when no SCRM transmission is due, the MS transmits a null 20-ms frame at 1.5 kbps on the R-FCH for effective outer-loop PC.

Q.1.2 Base Station Requirements and Procedures

The BS is required to perform the following essential functions:

1. Decoding of R-FCH/R-SCH: When there are multiple traffic channels transmitted by the MS simultaneously, Eb/Nt from other channels should be removed from the total received Eb/Nt before the FER table-lookup is performed. Let the total propagation loss at i^{th} PCG be denoted by PL. The total received power is given by:

$$RxPwr(PCG_i) = PL * PiTxPwr(PCG_i)\left[1 + (T/P)_{Rtx}\left(\frac{Pref(R_{tx})}{Pref(R)}\right) + ((T/P)_{R_{FCH}} + (C/P))\left(\frac{Pref(R_{FCH})}{Pref(R)}\right)\right]$$

(Q.1.2-1)

where R is the assigned R-SCH rate and $R_{tx} = \{0, R\}$ is the transmitted rate on R-SCH. In order to reduce the simulation complexity and the number of link-level curves, the following decoding method shall be used:

- Decoding of R-FCH: Derive the R-FCH and consider the following received Eb/Nt for R-FCH decoding:

$$RxFCHPwr(PCG_i) = PL * PiTxPwr(PCG_i)\left(\frac{Pref(R_{FCH})}{Pref(R)}\right)\left[1 + (T/P)_{R_{FCH}}\right] \qquad \text{(Q.1.2-2)}$$

  Note that the FCH Eb/Nt considered is the same as received. Eb/Nt computed using the above $RxFCHPwr(PCG_i)$ can be used directly to read the erasure probability from the link-level curve corresponding to $(T/P)_{R_{FCH}}$. Also note that the above methodology is conservative since better pilot SNR will slightly improve FCH decoding performance.

- Decoding of R-SCH: Since the MS either transmits at the assigned rate or DTX its transmission, the following received Eb/Nt for R-SCH decoding is used when MS transmits on that channel:

$$RxSCHPwr(PCG_i) = PL * PiTxPwr(PCG_i)\lfloor 1 + (T/P)_{R_{tx}} \rfloor \quad \text{(Q.1.2-3)}$$

  Eb/Nt computed using the above $RxSCHPwr(PCG_i)$ can be used directly to read erasure probability from the link-level curve corresponding to $(T/P)_{R_{tx}}$.

2. Power-control: Power control in a CDMA system is essential to maintain the desired quality of service (QoS). In IS-2000, the RL pilot channel (R-PICH) of each MS is closed-loop power controlled to a desired threshold. At the BS, this threshold, called power control set point, is compared against the received Ecp/Nt to generate power control command (closed-loop PC), where Ecp is the pilot channel energy per chip. To achieve the desired QoS on the traffic channel, the threshold at the BS is changed with erasures on the traffic channel, and has to be adjusted when the data rate changes. Set point corrections occurs due to:

- Outer-loop power control: The power control set point is corrected based on erasures of the R-FCH. A data-only MS while transmitting data and signaling, sends a frame at 9.6 kbps or an null frame at 1.5 kbps on the R-FCH. Outer loop power control does not depend on the R-SCH erasure performance.

  A delay of 20 ms is used before the power control set-point correction is applied. So for a frame decode at time t (corresponding to a frame boundary of 20 ms), the corresponding set-point correction is applied at t+20. In case of a 5 ms SCRMM transmission interrupting 20 ms R-FCH transmission, the power control set-point correction is still applied at 20 ms frame decode boundary. A 5 ms R-FCH erasure detection affects the outer-loop correction in the same way as a 20 ms R-FCH erasure detection.

- Rate Transitions: Different data rate on the R-SCH requires different optimal set point of the reverse pilot channel. When data rate changes on the R-SCH, the BS changes the MS's received Ecp/Nt by the Pilot Reference Levels (Pref(R)) difference between the current and the next R-SCH data rate. The Pilot Reference Level for a given data rate R, Pref(R), is specified in the Gain Table in C.S0002-C. Since the closed-loop power control brings the received pilot Ecp/Nt to the set point, the BS adjusts the outer loop set point according to the next assigned R-SCH data rate:

$$\Delta = Pref(Rnew) - Pref(Rold) \quad \text{(Q.1.2-4)}$$

  Set point adjustment is done $\lceil \Delta \rceil$ PCGs in advance of the new R-SCH data rate if $R_{new} > R_{old}$. Otherwise, this adjustment occurs at the R-SCH frame boundary. The pilot power thus ramps up or down to the correct level approximately in 1 dB step sizes of the closed loop (Figure Q-1).

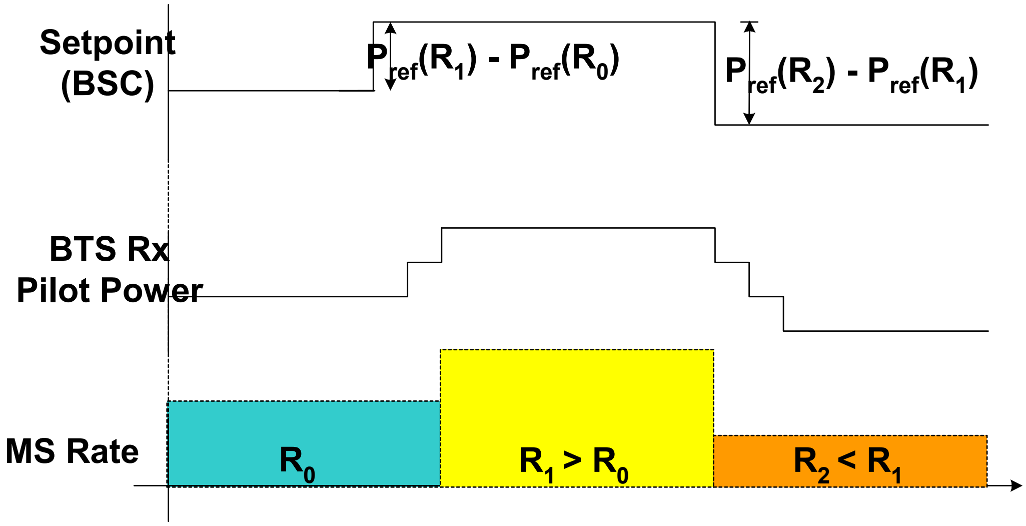


Figure Q-1: Set point adjustment due to rate transitions on R-SCH

Q.1.3 Scheduler Requirements and Procedures

The following assumptions are used for the scheduler and various parameters associated with scheduling:

1. Centralized Scheduling: The scheduler is co-located with the BSC, and is responsible for simultaneous scheduling of MSs across multiple cells. For system simulation with 19 cells, one scheduler for simultaneous scheduling of all 19 cells shall be used.

2. Synchronous Scheduling: All R-SCH data rate transmissions are time aligned. All data rate assignments are for the duration of one scheduling period, which is time aligned for all the MSs in the system. The scheduling duration is denoted *SCH_PRD*. The scheduling timeline is discussed in 0.

3. Voice and Autonomous R-SCH transmissions: Before allocating capacity to transmissions on R-SCH through rate assignments, the scheduler looks at the pending rate requests from the MSs and discounts for voice and autonomous transmissions in a given cell. Details of the scheduler are in 0.

4. Rate Request Delay: The uplink request delay associated with rate requesting via SCRM/SCRMM is denoted as *D_RL(request)*. It is the delay from the time the request is sent to when it is available to the scheduler. *D_RL(request)* includes delay segments for over-the-air transmission of the request, decode time of the request at the cells, and backhaul delay from the cells to the BSC, and is modeled as a uniformly distributed random variable (see Appendix R).

5. Rate Assignment Delay: The downlink assignment delay associated with rate assignment via ESCAM/ESCAMM is denoted as *D_FL(assign)*. It is the time between the moment the rate decision is made and the time the MS receiving the resultant assignment. *D_FL(assign)* includes backhaul delay from the scheduler to the cells, over-

the-air transmission time of the assignment (based on method chosen), and its decode time at the MS and is modeled as a random variable as described in Appendix R.

6. Available Ecp/Nt Measurement: The Ecp/Nt measurement used in the scheduler shall be that available at the last frame boundary before action time minus 31 PCG's and 68 PCG's for Method a and Method b, respectively. See Figure Q-2 below.

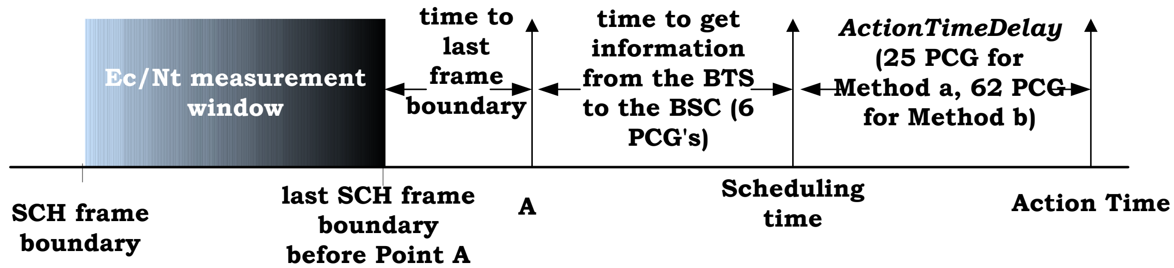


Figure Q-2 Scheduling Delay Timing

Q.1.3.1 Scheduling, Rate Assignment and Transmission Timeline

Given the assumed synchronous scheduling, most events related to request, grant and transmission are periodic with period *SCH_PRD*.

Figure Q-3 illustrates the timing diagram of a rate request, scheduling and a rate allocation.

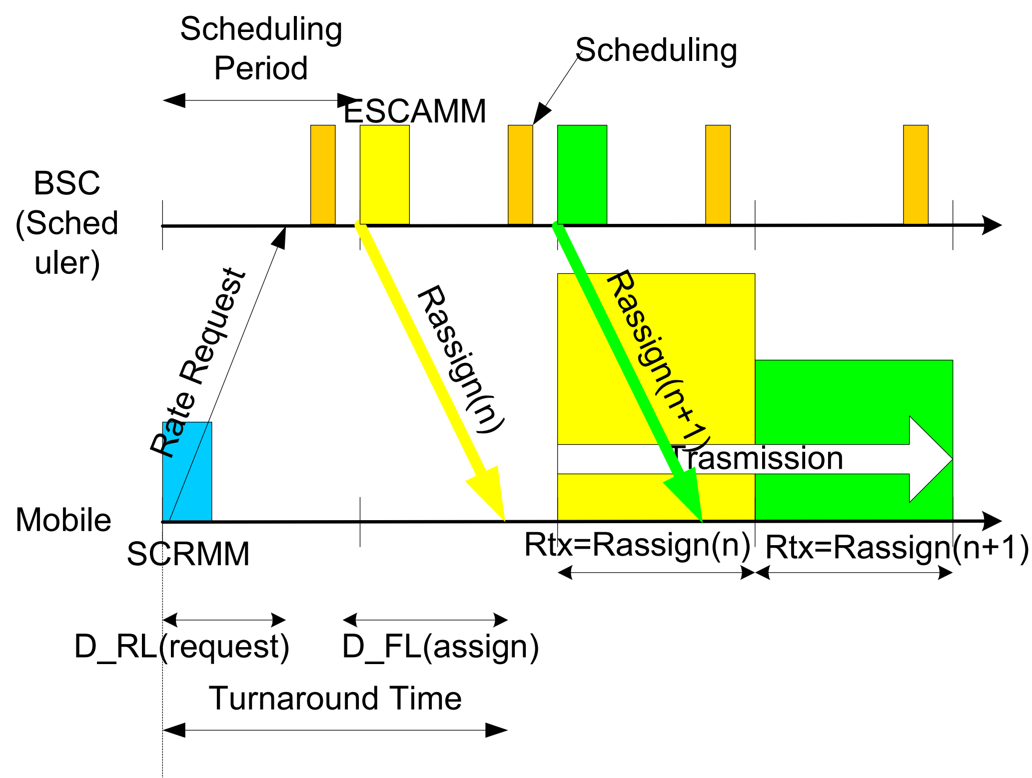


Figure Q-3: Parameters associated in mobile station scheduling on RL

The following characterizes the simulation timeline:

- Scheduling Timing: The scheduler operates once every scheduling period. If the first scheduling decision is performed at t_i, then scheduler operates at $t_i,\ t_i + SCH\_PRD,\ t_i + 2SCH\_PRD...$
- Scheduled Rate Transmissions: Given that the MSs have to be notified of the scheduling decisions with sufficient lead-time, a scheduling decision has to be reached at Action Time of the ESCAM/ESCAMM message minus a fixed delay, *ActionTimeDelay*. The values of *ActionTimeDelay* for Methods a and b are given in Section 0 to ensure that most MSs receive the ESCAM/ESCAMM messages with high probability.

If the ESCAM/ESCAMM message is received after (Action Time – 2 PCG), it is considered lost.

- MS R-SCH Rate Requests: R-SCH rate requests are triggered as described below:

Before the beginning of each SCRM/SCRMM frame encode boundary, the MS checks if either of the following three conditions are satisfied:

1. New data arrives and data in the MS's buffer exceeds a certain buffer depth (*BUF_DEPTH*), and the MS has sufficient power to transmit at a non-zero rate; OR

2. If the last SCRM/SCRMM was sent at time τ_i, and the current time is greater than or equal to $\tau_i + SCH\_PRD$, and if the MS has data in its buffer that exceeds the *BUF_DEPTH*, and the MS has sufficient power to transmit at a non-zero rate; OR
3. If the last SCRM/SCRMM was sent at time τ_i, and the current time is greater than or equal to $\tau_i + SCH\_PRD$, and if the current assigned rate at the MS side based on received ESCAMM/ESCAM is non-zero (irrespective of the fact that the MS may not have data or power to request a non-zero rate). "Current assigned rate" is the assigned rate applicable for the current rate transmission. If no ESCAM is received for the current scheduled duration, then the assigned rate is considered 0. The rate assigned in the ESCAM/ESCAMM message with Action Time at some later time takes effect after the Action Time.

If either of the above three conditions are satisfied, the MS sends a SCRMM/SCRM rate request. A SCRM/SCRMM request made at τ_i is made available to the scheduler after a random delay at $\tau_i + D\_RL(request)$. To initialize the first SCRMM transmissions of the various MSs, last SCRMM/SCRM sent time τ_i is selected from an uniformly generated random variable that uniformly picks one frame in the first scheduling period.

Q.1.3.2 Scheduler Description and Procedures

There is one centralized scheduler for the entire system under consideration (19 cells or 57 sectors). The scheduler maintains a list of all MSs in the system and BSs in each MS's Active Set. Associated with each MS, the scheduler stores estimate of MS's queue size ($\hat{Q}$) and maximum scheduled rate ($R_{max}(s)$).

- The queue size estimate $\hat{Q}$ is updated after any of the following events happen:

1. A SCRMM/SCRM is received: SCRMM/SCRM is received after a delay of D_RL(request). $\hat{Q}$ is updated to:

$$\hat{Q} = Queue\,Size\,reported\,in\,SCRMM \qquad \text{(Q.1.3.2-1)}$$

   If the SCRMM/SCRM is lost, the scheduler uses the previous (and the latest) information it has.

2. After each R-FCH and R-SCH frame decoding:

$$\hat{Q} = \hat{Q} - Data_{tx}(FCH) + Data_{tx}(SCH) \qquad \text{(Q.1.3.2-2)}$$

   where $Data_{tx}(FCH)$ and $Data_{tx}(SCH)$ is the data transmitted in the last R-FCH and R-SCH frame respectively after discounting the physical layer overhead and RLP layer zero bits padding. Since RLP retransmissions are not modeled, data transmitted in the last frames is discounted even if the frame is decoded in error.

3. At the scheduling instant t_i, scheduler estimates the maximum scheduled rate for the MS:

$$\hat{Q}(f)=\hat{Q}-(R_{assigned}+9600)\times\lceil ActionTimeDelay/20\rceil\bullet 20\,\mathrm{ms}$$
$$+((PL\_FCH\_OHD+SCH_{Assigned}*PL\_SCH\_OHD)\times(\lceil ActionTimeDelay/20\rceil)$$

(Q.1.3.2-3)

$$R_{\max}(s)=\min\left\{\begin{array}{l} R_{\max}(power), \\ \underset{R\le 307.2kbps}{\arg\max_R}\{R\mid \hat{Q}(f)\ge((R+9600)\times 20\mathrm{ms}-PL\_FCH\_OHD \\ -PL\_SCH\_OHD)\times(SCH\_PRD/20\mathrm{ms})\ \} \end{array}\right\} \quad (Q.1.3.2-4)$$

where $SCH_{Assigned}$ is an indicator function for the current scheduling period,

$$SCH_{Assigned}=\begin{cases}1 & \text{if } R_{assigned}>0\\ 0 & \text{if } R_{assigned}=0\end{cases} \quad (Q.1.3.2-5)$$

$R_{assigned}$ is the rate assigned on the R-SCH during the current scheduling period and MS is supposed to transmit on the R-SCH until the *ActionTime* of the next assignment. $R_{\max}(power)$ is the maximum rate that the MS can support given its power limit. This max rate is reported in the last received SCRM/SCRMM message.

- Capacity Computation:

The sector capacity at the j^{th} sector is estimated from the measured MSs' *Sinr*s. The *Sinr* is the average pilot-weighted combined *Sinr* per antenna. The combining per power-control group (PCG) is the pilot-weighted combining over multiple fingers and different antennas of the sector of interest. The combining is not over different sectors in case of softer-handoff MS. The averaging is over the duration of a 20-ms frame. The following formula is used for estimating *Load* contribution to a sector antenna [30]:

$$Load_j=\sum_{j\in ActiveSet(i)}\frac{Sinr_j(R_i,E[R_{FCH}])}{1+Sinr_j(R_i,E[R_{FCH}])} \quad (Q.1.3.2-6)$$

where $Sinr_j(R_i,E[R_{FCH}])$ is the estimated *Sinr* if the MS is assigned a rate R_i on R-SCH and $E[R_{FCH}]$ is the expected rate of transmission on the R-FCH. For data-only MSs with autonomous transmission at a maximum rate of 9.6kbps, the maximum autonomous data rate can be used for the expected value. Let the measured pilot *Sinr* (frame average pilot *Sinr* averaged over two antennas) be $(E_{cp}/N_t)_j$, while it is assigned a rate of $R_{assign}(SCH)$ on the R-SCH. Then,

$$Sinr_j(R_i,R_{FCH})=\frac{Pref(R_i)}{Pref(R_{assign}(SCH))}(E_{cp}/N_t)_j\left[1+(T/P)_{R_i}+((T/P)_{R_{FCH}}+(C/P))\left(\frac{Pref(R_{FCH})}{Pref(R_i)}\right)\right]$$

(Q.1.3.2-7)

For voice-only MSs, the following equation is used to estimate the average received Sinr:

$$Sinr_j(0, E[R_{FCH}(\upsilon)]) = \frac{(E_{cp}/N_t)_j}{Pref(R_{assign}(SCH))} \times \left[1 + \begin{pmatrix} (T/P)_{9.6k} P(9.6k) + \\ (T/P)_{4.8k} P(4.8k) + \\ (T/P)_{2.7k} P(2.7k) + \\ ((T/P)_{1.5k} P(1.5k) + \\ (C/P) \end{pmatrix} Pref(R_{FCH}^{\max} = 9.6k) \right]$$

(Q.1.3.2-8)

- Scheduling Algorithm:

The scheduling algorithm has two fundamental characteristics: a) greedy filling for maximum capacity utilization and increasing TDM gain, b) prioritization of MS rate requests such that MSs which were scheduled in the last scheduling period are last in the queue. This resembles Round-Robin scheduler while allowing multiple transmissions for full capacity utilization.

Initialization: The MS rate requests are prioritized. Associated with each MS is a priority count PRIORITY. PRIORITY of a MS is initialized to 0 in the beginning of the simulation. When a new MS enters the system with sector *j* as the primary sector, its PRIORITY is set equal to the $\min\{\text{PRIORITY}_i, \quad \forall i \text{ such that MS}_i \text{ has sector } j \text{ as the primary sector}\}$

1. Let the *Load* constraint be $Load_j \le \max Load$, such that the rise-over-thermal overshoot above certain threshold. For the calibration purposes, max *Load* value of 0.45 will be used by the scheduler. The capacity consumed due to pilot transmissions and transmissions on fundamental channels (due to voice or data) is computed and the available capacity is computed as

$$Cav(j) = \max Load - \sum_{j \in ActiveSet} \frac{Sinr_j(0, E[R_{FCH}])}{1 + Sinr_j(0, E[R_{FCH}])} \qquad \text{(Q.1.3.2-9)}$$

   where max *Load* is the maximum *Load* for rise-over-thermal outage criteria is satisfied.

   MS rate requests are prioritized in decreasing order of their PRIORITY. So MSs with highest PRIORITY are at the top of the queue. When multiple MSs with identical PRIORITY values are at the top of the queue, the scheduler makes a equally-likely random choice among these MSs.

2. Set k=1,
3. The data-only MS at the *k*th position in the queue is assigned the rate R_k given by

$$R_k = \min\left\{ R_{\max}^k(s), \arg\max_R \left[\begin{array}{l} R \mid Cav(j) - \dfrac{Sinr_j(R, E[R_{FCH}])}{1+Sinr_j(R, E[R_{FCH}])} \\ + \dfrac{Sinr_j(0, E[R_{FCH}])}{1+Sinr_j(0, E[R_{FCH}])} \geq 0; \forall j \in ActiveSet(k) \end{array} \right] \right\} \quad \text{(Q.1.3.2-10)}$$

The available capacity is updated to:

$$Cav(j) = Cav(j) - \frac{Sinr_j(R_k, E[R_{FCH}])}{1+Sinr_j(R_k, E[R_{FCH}])} + \frac{Sinr_j(0, E[R_{FCH}])}{1+Sinr_j(0, E[R_{FCH}])}; \quad \forall j \in ActiveSet(k)$$

(Q.1.3.2-11)

4. If $R_{\max}^k(s) > 0$ and $R_k = 0$, increment PRIORITY of the MS

   Otherwise, do not change PRIORITY of the MS

5. k = k+1; if k < total number of MSs in the list, Go to Step 3, otherwise, stop.

Q.2 Baseline specific simulation parameters

Parameter	Value	Comments
R-FCH frame format	20 ms, RC 3 (9.6, 4.8, 2.7, 1.5) kbps	Specified in [29] Voice-only MS uses all four rates Data-only MS uses only 1.5 kbps and 9.6 kbps
R-SCH frame format	20 ms, RC 3 (9.6 k convolutional) (19.2, 38.4, 76.8, 153.6, 307.2) kbps turbo	Specified in [29]
SCRMM Overhead	One 5-ms R-FCH transmission	24 bits of upper layer payload
SCRM Overhead	One 20-ms R-FCH transmission	Minimum of 144 bits of upper layer payload
Headroom_Req	5 dB	Conservative rate request Keeps power headroom for long-term channel variation Reduces DTX on R-SCH
Headroom_Tx	2 dB	Reduces probability of power outage during the duration of R-SCH transmission
Average Tx Power Filter Coefficient α_{Headroom}	1/16	Normalized Average transmit pilot power is computed as filtered version over several PCGs

Parameter	Value	Comments
Encode_Delay	2.5 ms	Delay between encoding of transmission packet and actual transmission
SCRMM Delay (Method a) *D_RL(request)*	Mean 23.625 ms Std Dev. 6.85 ms Uniform [11.75, 35.5] ms	Modeled as a random variable in Appendix R
SCRM Delay (Method b) *D_RL(request)*	Mean 38.625 ms Std Dev. 6.85 ms Uniform [26.75, 50.5]	Modeled as a random variable in Appendix R
ESCAMM Delay (Method a) *D_FL(assign)*	Mean 22.375 ms Std dev. 3.25 ms Uniform [16.75, 28]	Modeled as a random variable in Appendix R
ESCAM Delay (transmission on F-PDCH) *D_FL(assign)*	X + Y X: Uniform [11.75, 24.25] Y: LogNormal with mean and std. Dev. as specified in Table R.5, Appendix R.	Modeled as a random variable in Appendix R
ActionTimeDelay (Method a)	31.25 ms	Based on the ESCAMM delay, including the 2 PCG MS encoding delay
ActionTimeDelay (Method b)	77.5 ms	Based on the ESCAM delay on F-PDCH at the primary sector Geometry of –5 dB. This includes the 2 PCG MS encoding delay
SCH_PRD	200 ms	Scheduling Period sufficiently long to reduce overhead while small enough for effective scheduling. The assigned rate for a given MS does not have to be over the whole 200 ms.
BUF_DEPTH	(19200 X *SCH_PRD*) = 3840 bits	Minimum buffer size for SCRMM/SCRM to be sent
PL_FCH_OHD	24 bits	24 bits per frame is the physical layer overhead on the R-FCH
PL_SCH_OHD	24 bits	24 bits per frame is the physical layer overhead on the R-SCH

Parameter	**Value**	**Comments**
$\alpha_{Geometry}$	0.05	Constant used in the computation of geometry for F-PDCH delay in Appendix R
CQICH-to-(normalized) pilot ratio (C/P)	-5.4 dB (Active Set includes sectors from single cell) -4.3 dB (if Active Set includes sectors from more than one cell)	Overhead transmission on RL due to CQICH is modeled as a continuous transmission with fixed C/P ratio. For a MS not in SHO, average C/P value corresponding to one full CQICH burst and 16 differential bursts is used. For the MS in SHO, average CQICH value corresponding to two full bursts and 14 differential bursts is used
Pref(Data Rate)	Pilot Reference Level [dB]	Pilot Reference Levels are determined such that an average FER of 1% is observed across different channel models based on Evaluation Methodology approved by WG3
9.6kbps	0.0	
19.2kbps (turbo)	1.25	
38.4kbps (turbo)	2.5	
76.8kbps (turbo)	3.75	
153.6kbps (turbo)	5.625	
307.2kbps (turbo)	8.25	

APPENDIX R: MODELING OF D_RL(REQUEST) AND D_FL(ASSIGN)

The four tables below give the minimum expected delays and worst-case delays with SCRMM/SCRM transmission for request and ESCAMM/ESCAM transmissions for assignment.

Table R-1: D_RL(request) delay for Method a

	Minimum Expected delay (ms)	Cumulative Expected Delay (ms)	Reasonable Worst Case Delay (ms)	Cumulative Reasonable Worst Case Delay (ms)
MS processing request message	1	1	1	1
Request message transmission time	5	6	5	6
Decoding time at BTS	2	8	12.5	18.5
Transmission time to BSC	2.75	10.75	8	26.5
BSC processing time	0.5	11.25	2.5	29
BSC to scheduler	0.5	11.75	0.5	29.5
Potential inter-BSC handoff	0	11.75	6	35.5

Request message framing time (time when the new data arrives in the system to when the request is actually sent) is included in the table as "Request message transmission time".

Assuming uniformly distributed backhaul delay between BTS and BSC, D_RL(request) with SCRMM transmission and based on above values can be modeled as uniformly distributed random variable between [11.75, 35.5] ms.

In calculating the transmission time to the BSC, we assumed that the backhaul bandwidth on the forward link and the reverse link are identical. In order to support 2.4 Mbps forward link, the backhaul would need to be at least 2 x 1.544 Mbps. If various sectors are aggregated into the same backhaul, then the rate will be even greater. If an ATM cell consisting of 53 octets is used to transmit the uplink request, then the transmission time is B/R+D/c, where B is the number of bits to be transmitted, R is the transmission rate, D is the distance, and c is the velocity of transmission in the medium. Assuming 10 km distance, D, and a propagation velocity in the medium of one-half the speed of light, then the transmission delay is about 0.275 ms. The actual estimate needs to take into account queuing delay. If the link utilization is 0.9 (a high utilization), then the mean transmission delay will be about 2.75 ms.

Table R-2: D_RL(request) delay for Method b

	Minimum Expected delay	Cumulative Expected Delay	Reasonable Worst Case Delay	Cumulative Reasonable Worst Case Delay
MS processing request message	1	1	1	1
Request message transmission time	20	21	20	21
Demodulation time at BTS	2	23	12.5	33.5
Transmission time to BSC	2.75	25.75	8	41.5
BSC processing time	0.5	26.25	2.5	44
BSC to scheduler	0.5	26.75	0.5	44.5
Potential inter-BSC hand-off	0	26.75	6	50.5

Assuming uniformly distributed backhaul delay between BTS and BSC, D_RL(request) with SCRMM transmission and based on above values can be modeled as uniformly distributed random variable between [26.75, 50.5] ms.

Table R-3: D_FL(assign) delay for Method a

	Minimum Expected delay	Cumulative Expected Delay	Reasonable Worst Case Delay	Cumulative Reasonable Worst Case Delay
Scheduling time	2	2	2	2
Scheduler to BSC	0.5	2.5	0.5	2.5
Transmission time to BTS	5	7.5	10	12.5
BTS processing time	3	10.5	3	15.5
Delay to transmission slot	0	10.5	5	20.5
Transmission time	5	15.5	5	25.5
Grant processing time in MS	1.25	16.75	2.5	28

Assuming uniformly distributed backhaul delay and delay to transmission slot at BTS, D_FL(assign) with ESCAMM transmission and based on above values can be modeled as uniformly distributed random variable between [16.75, 28] ms. Please note that it does not include the delay when the R-SCH transmission is framed by the mobile to when R-SCH starts transmission at the R-SCH frame boundary. This latter delay is naturally modeled by the simulator.

A uniformly distributed random variable between [u1, u2] has a mean of (u1+u2)/2 and variance of $(u2 - u1)^2/12$.

Modeling downlink assignment delay of ESCAM transmission on PDCH requires modeling of backhaul delays and PDCH scheduling delays. Backhaul delays as above are modeled uniform. Assuming no scheduling delays, following table lists the expected minimum delay and worst-case delay.

Table R-4: D_FL(assign) delay for Method b (excluding F-PDCH scheduling delay)

	Minimum Expected delay	Cumulative Expected Delay	Reasonable Worst Case Delay	Cumulative Reasonable Worst Case Delay
Scheduling time	2	2	2	2
Scheduler to BSC	0.5	2.5	0.5	2.5
Transmission time to BTS	5	7.5	15	17.5
BTS processing time	3	10.5	3	20.5
Delay to transmission slot	0	10.5	1.25	21.75
Transmission time	0	10.5	0	21.75
Grant processing time in MS	1.25	11.75	2.5	24.25

To model transmission time or the scheduling delay on the F-PDCH, we did the following experiment. Assuming four users are simultaneously scheduled every 200 ms, 4 ESCAM packets are produced every 200 ms. Other users are added with normal traffic to generate full load on forward link. Signaling messages are given priority over data. A probability density function (PDF) of three MSs at different geometries is shown in Figure R.1. For the cases plotted here, we have

Mean (ms)	Std Dev (ms)
16.9	16.4
19.7	18.1
42.4	90.2

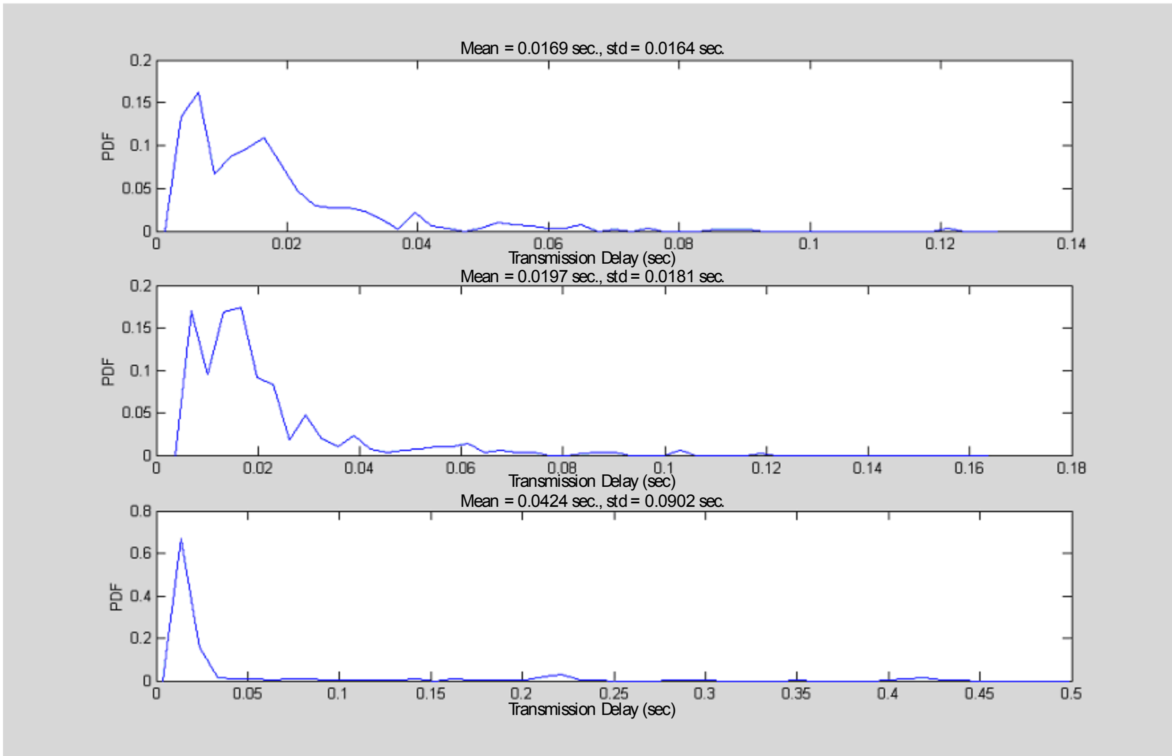


Figure R-1: PDF of FL transmission delays of ESCAM on F-PDCH

To retain the simplicity of modeling, we recommend modeling the delays as lognormal with same mean and standard deviation.

Figure R-2 shows some of the sample points of observed scheduling delays.

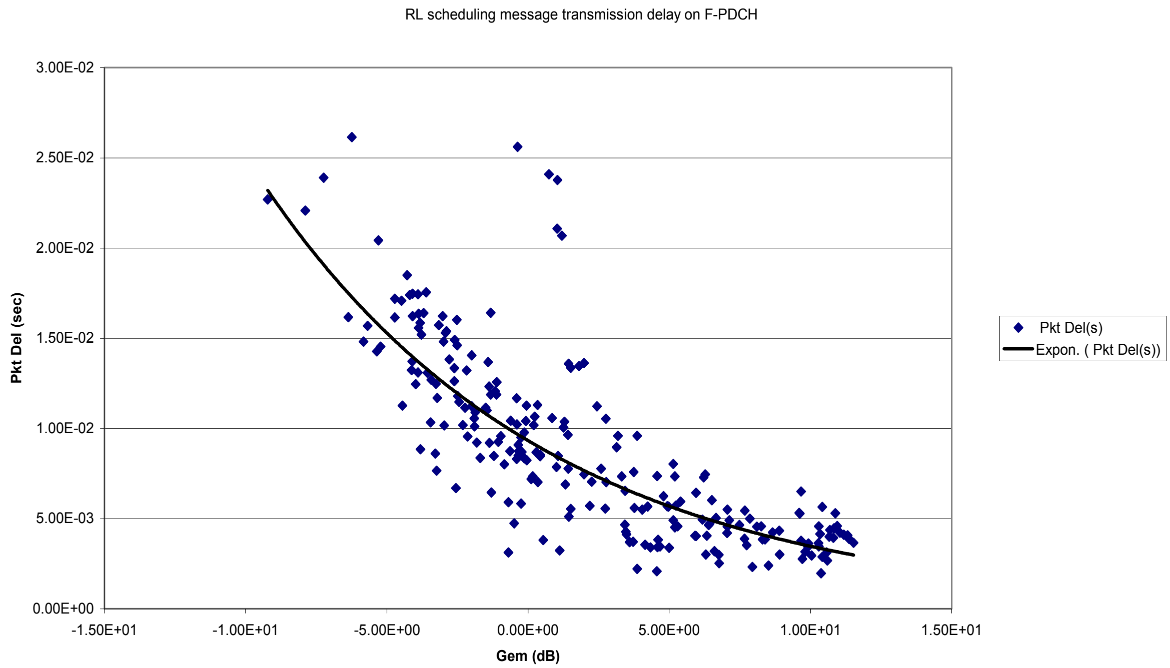


Figure R-2: ESCAM delays on F-PDCH

Since the signaling message is small to fit in any EP size and have priority over the data transmission, downlink transmission time is pretty much independent of channel model as shown in Figure R-2. We can approximate the curve above by interpolating between a few points, for example, based on Table R-5:

Table R-5: Reference table for mean transmission times vs Geometry

Geometry (dB)	Mean Transmission Time (ms)
-9	22
0	10
10	3

To summarize, *D_FL(assign)* = ***x*** + ***y***. ***x*** is uniform [11.75, 24.25]. ***y*** is lognormal with mean and standard deviation obtained by interpolating or extrapolation using the values given in Table R-5 and,

- Geometry = 1 / ((Ioc+No)/Ior + α_{Geometry})
- For geometry computation, all BTSs are assumed to be transmitting at full power (20 W)
- Ior is the total energy per chip received from the strongest sector assumed to be max transmit power (20 Watts)
- Ioc is the sum of total energy per chip from all other sectors, each assumed to use max transmit power (20 Watts)

- No is the thermal noise spectral density