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Test Scenarios and Calibration Cases Issue 2

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Abstract:

This report provides the baseline assumption related to environment, deployment, system design, and algorithms for the three test scenarios "base coverage urban", "microcellular", and "indoor" in which the WINNER system is exemplified and evaluated. Associated key assessment criteria are defined.

Keyword list:

baseline design, test scenarios, simulation assumptions, deployment, assessment criteria

Disclaimer:

Executive Summary

The WINNER radio interface provides a versatile concept and architecture, able to support a wide range of deployment and application scenarios. This is exemplified by three concrete design examples and the associated performance evaluations. Three detailed **test scenarios** have been specified**,** each focussing on a different environment, deployment, physical layer mode, parameterisation, and highlighting different key aspects for future radio interfaces:

- The **base coverage urban** test scenario is an urban macro-cellular deployment using the FDD physical layer mode, a carrier frequency of 3.7 GHz / 3.95 GHz, and 2 x 50 MHz bandwidth. Self-contained ubiquitous coverage for populated areas including the full range of mobility is the focal point of investigations here.
- The **microcellular** test scenario is an urban micro-cellular scenario using the TDD physical layer mode and 100 MHz bandwidth at 3.95 GHz, addressing higher user densities with lower mobility and highlighting adaptivity and flexibility of the WINNER system.
- The **indoor** test scenario investigates in-home and hotspot scenarios, such as hotels, shopping centres, small offices, etc., using the TDD physical layer mode and 100 MHz bandwidth at 5 GHz. Here investigations are centred on questions of self-organisation and self-adaptation.

Important enablers for performance evaluation in these test scenarios are **calibration**, **comparability**, and **reliability management** of the simulations. These topics are addressed in WINNER by several means. This reports aims at the specification of common **baseline simulation assumptions** regarding environment, deployment, system design, and algorithms. It also defines the key assessment criteria and associated measurement procedures. This is of utmost importance especially for system-level investigations, where a direct calibration is infeasible given the complexity of these simulations and one needs to resort to relative performance assessment.

The **baseline system implementation** defined in this report reflects the current status (or a status that is achievable in near-term) of the main simulation tools within the WINNER consortium. It is a minimal configuration which does neither correspond to any particular future WINNER implementation, nor will it provide performance results that can in all cases be indicative for WINNER. Enhancements and/or other options of the features described in this document are also under investigation and evaluation in the project, e.g. as part of the so-called **reference design** [WIN2D6131]. The baseline system design forms a reliable basis for relative comparisons and assessment of the added benefit of new features. It also allows gradual refinement and question-oriented configuration of simulation tools, i.e. to perform a dedicated investigation, the actual simulator might use parts of the most advanced reference design – in particular in the main area of investigation – and simply use baseline design assumptions in other areas. In that way the level of detail and realism in simulations is adapted to the investigated question. Only such a flexible and target-oriented approach can keep the required effort within feasible limits and at the same time provide the required broadness and reliability of investigation. Furthermore the transition from baseline to reference design provides a natural guideline for permanent improvement of simulator capabilities conducted by the WINNER partners.

This deliverable is the successor of [WIN2D6131] and contains the latest updates of the simulation assumption for the test scenarios, of the associated baseline design assumptions, and of the assessment criteria definition. Important updates have occurred with respect to the basic OFDM parameters and dimensioning (increased guard interval and changed chunk dimensions in the TDD mode), the baseline coding scheme (use of the block low density parity check code as baseline coding scheme), the multiple access scheme for non-frequency adaptive transmissions (introduction of B-IFDMA and B-EFDMA), and the baseline spatial processing schemes (simplified assumptions for microcellular and indoor test scenario. Also, new aspects are included and more details are provided, in particular with respect to relaying (basic deployment scenarios and parameters, basic resource partitioning and timing), segmentation (definition of RTU and FEC block sizes), link adaptation (new baseline modulation and coding scheme, adaptive coding and modulation algorithms), and HARQ (detailed specification of protocol and timing).

This report provides the common framework for performance evaluation in WINNER in 2007. Results from many expert discussions throughout the project have been consolidated in order to obtain parameters, assumptions, and algorithms that represent existing or near-term achievable simulator capabilities and at the same time allow meaningful evaluations of the major questions in WINNER system concept and design.

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List of Acronyms and Abbreviations

ACM	Adaptive Coding and Modulation
ARQ	Automatic Repeat Request
BCH	Broadcast (Transport) Channel
B-EFDMA	Block Equidistant Frequency Division Multiple Access
BER	Bit Error Rate
B-IFDMA	Block Interleaved Frequency Division Multiple Access
B-LDPCC	Block Low-Density Parity-Check Code
BLER	Block Error Rate
BPSK	Binary Phase Shift Keying
BS	Base Station
CDF	Cumulative Distribution Function
CE	Channel Estimation
CDF	Cumulative Density Function
CSI	Channel State Information
DAC	Direct Access Channel
DBTC	Dual-Binary Turbo Code
EIRP	Effective Isotropic Radiated Power
FDD	Frequency Division Duplex
FEC	Forward Error Correction (Coding)
FER	Frame Error Rate
FRP	Flexible Re-use Partitioning
GMC	Generalised Multi-Carrier
GoB	Grid of Beams
GOP	Group of Pictures
HARQ	Hybrid Automatic Repeat Request
ICE	Iterative Channel Estimation
IMM	Instant Messaging for Multimedia
IP	Internet Protocol
IPP	Interrupted Poisson Process
IRP	Interrupted Renewal Process
ISD	Inter-Site Distance
LDC	Linear Dispersion Codes
LDPCC	Low-Density Parity-Check Code
LLR	Log-likelihood Ratio
LOS	Line-of-Sight
MAC	Medium Access Control
MCS	Modulation and Coding Scheme
MI	Mutual information (link-to-system level interface, [BA+05])
MIMO	Multiple Input Multiple Output
ML	Maximum Likelihood
MM	Multimedia
MMSE	Minimum Mean Square Error
MPEG	Motion Pictures Expert Group
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PCCC	Parallel Concatenated Convolutional Code (Turbo Code)
PDF	Probability Density Function
PER	Packet Error Rate
PLM	Physical Layer Mode
P2P	Peer-to-Peer

QAM	Quadrature Amplitude Modulation
QP	Quantisation Parameters
RAC	Random Access Channel
RAP	Radio Access Point
RAT	Radio Access Technology
RBD	Regularised Block Diagonalisation
REC	Relay-Enhanced Cell.
RN	Relay Node
RoHC	Robust Header Compression
RRM	Radio Resource Management.
RS	Resource Scheduling
RTU	Retransmisison Unit
SAW	Stop And Wait (HARQ protocol)
SDMA	Spatial Division Multiple Access
SINR	Signal to Interference and Noise Ratio
SISO	Single-Input Single-Output
SMMSE	Successive Minimum Mean Square Error precoding
SNR	Signal to Noise Ratio
SU	Single User
SVD	Singular Value Decomposition
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
UT	User Terminal
VoIP	Voice over IP (Internet Protocol)
VT	Video Telephony

List of Mathematical Symbols

D_f	pilot symbol spacing in frequency direction
D_t	pilot symbol spacing in time direction
Δf	subcarrier distance
$\Delta\gamma$	SINR degradation
ΔR	pilot overhead
f_c	carrier frequency
G	estimator gain for channel estimation
K	segment size
R	code rate
σ^2	noise variance
S_p	pilot boost
T_{av}	average throughput
τ_{av}	average delay
T_G	Guard interval/cyclic prefix duration
T_N	OFDM symbol duration (without guard)/single carrier block length
λ	wavelength
β	pulse roll-off factor
$\theta_{3\mathrm{dB}}$	3dB beamwidth of antenna element pattern

1. Introduction

While the WINNER radio interface provides a versatile concept and architecture, able to support a wide range of deployment and application scenarios, the benefits of WINNER are being exemplified by means of three concrete design examples and the associated performance evaluations. Three detailed **test scenarios have been specified,** each focussing on a different environment, deployment, physical layer mode, parameterisation, and highlighting different key aspects for future radio interfaces:

- The **base coverage urban** test scenario is an urban macro-cellular deployment using the FDD physical layer mode, a carrier frequency of 3.7 GHz / 3.95 GHz, and 2 x 50 MHz bandwidth. Self-contained ubiquitous coverage for populated areas including the full range of mobility is the focal point of investigations here.
- The **microcellular** test scenario is an urban micro-cellular scenario using the TDD physical layer mode and 100 MHz bandwidth at 3.95 GHz, addressing higher user densities with lower mobility and highlighting adaptivity and flexibility of the WINNER system.
- The **indoor** test scenario investigates in-home and hotspot scenario, such as hotels, shopping centres, small offices, etc., using the TDD physical layer mode and 100 MHz bandwidth at 5 GHz. Here investigations are centred on questions of self-organisation and self-adaptation.

Important enablers for performance evaluation in these test scenarios are **calibration**, **comparability**, and **reliability management** of the simulations. These topics are addressed in WINNER by the following means:

- Direct calibration of link-level results and use of a common set of link-level curves in system-level simulators,
- Unified methodologies, i.e. modelling of the link-to-system interface throughout the project [WIN1D27, BAS+05],
- Use of common software for channel modelling [WIN2D111],
- Relative performance assessment on system-level (since direct calibration of different system-level simulators is infeasible given the complexity of these simulators),
- Adaptation of the level of detail in modelling of specific functionalities according to the particular requirements of the actual investigation,
- Specification of common simulation assumptions regarding environment, deployment, function design, and algorithms,
- Definition of key assessment criteria and the associated measurement procedure.

The need for relative performance assessments at system-level, as well as the approach to iteratively improve WINNER system design, requires the definition of a baseline design for the three test scenarios. Such a **baseline system implementation** reflects the current status of the main simulation tools within the WINNER consortium. It is a minimal configuration, which neither corresponds to any particular future WINNER implementation, nor will it provide performance results that can in all cases be indicative for WINNER. However, the baseline system design forms a reliable basis for relative comparisons and assessment of the added benefit of new features. This allows mitigating the highly complex and – given the limited efforts - infeasible calibration of system-level simulators throughout the project. It therefore enables efficient iterative improvement of the WINNER system design.

In contrast to the baseline system implementation, the **reference system design** is the current working assumption of a sensible and high-performance design for the particular test scenario. It includes the synopsis of the most advanced solutions and is based on the latest results of investigations as well as on purely conceptual work. The reference system design therefore goes in sum far beyond the capabilities of individual simulators but represents the target to which the simulators need to be developed.

As mentioned above a question-oriented approach to simulations is adopted, i.e. to investigate particular issues, the actual simulator might use parts of the reference design – in particular in the main area of investigation and associated functions that are sensitive to performance and reliability of the results – and simply use baseline design assumptions in other areas. In that way the level of detail and realism in simulations is adapted to the investigated question. Only such a flexible and target-oriented approach can keep the required effort within feasible limits and at the same time provide the required broadness and reliability of investigation. Furthermore the transition from baseline to reference design provides a natural guideline for permanent improvement of simulator capabilities conducted by the WINNER partners.

This document focuses on the baseline system implementation in order to ensure integrity of the assumptions throughout the project and provide binding guidelines for simulations in 2007. It is the successor of [WIN2D6131] and contains the latest updates of the simulation assumption for the three test scenarios, of the associated baseline design assumptions, and of the assessment criteria definition. It contains also further baseline design details in particular related to relaying, multiple access, link adaptation, and user plane packet processing. References regarding important methodologies, available software, and practical details are provided in order to provide a compendium for efficient implementation of the required tools for simulation and performance evaluation.

To facilitate easy access to information and efficient work, this deliverable is designed as a self-contained document, i.e. all relevant information of [WIN2D6131] is included in this deliverable even in case when no change has occurred meanwhile. However, for conciseness, references to other WINNER deliverables are used for detailed information and rationale behind decisions, wherever appropriate. This document is therefore organised as follows: A description of the environment- and deployment-specific assumptions and parameters is given in Chapter 2, along with information on the channel modelling. Baseline system design, including basic OFDM/GMC parameters, frame and super-frame layout, and baseline algorithms and configurations of system functions are explained in Chapter 3. Chapter 4 defines the key assessment criteria that will be used for evaluations.

2. Description of Test Scenarios

2.1 Environment-specific Parameters

The environment-specific parameters remain mostly as in [WIN2D6131] with some slight adaptation due to the additional consideration of indoor user terminals (UTs) in the base coverage urban and microcellular test scenario.

Whereas no specific topographic details are taken into account in the base coverage urban case, a two-dimensional regular grid of streets and buildings, the so-called Manhattan grid, is considered in the microcellular case. 121 building blocks of 200 m x 200 m size are separated by streets of 30 m width. The indoor test scenario consists of one floor (height 3 m) of a building, containing a regular grid of 40 room (10m x 10m) and two corridors (100 m x 5 m). In each case the number of users is normally a variable parameter. If for particular investigations user densities are required they can be found in [WIN2D6112] for different applications.

In the base coverage urban scenario users are distributed uniformly over the entire area. For the outdoor-to-indoor simulations a penetration loss is simply imposed on all users by the choice of the appropriate channel model (see Section 2.3). In the microcellular scenario users are uniformly distributed in the streets in the outdoor-to-outdoor simulations, whereas they are uniformly distributed in the buildings for the outdoor-to-indoor simulations. For the indoor test scenario 90% of the users are uniformly distributed in rooms and the remaining 10% are uniformly distributed in corridors.

Note, that the definition of the Manhattan grid and the indoor environment follow the definition of [UMTS30.03] to facilitate comparisons. Further details on deployment-specific assumptions and illustrations of the considered environments are given in Section 2.2.

The user mobility models for different simulator classes[1] remain unchanged compared to [WIN2D6131] and are explained in Table 2.1.

Baseline simulations for class III simulators will focus on full queue traffic model. However, to obtain the necessary delay statistics packet models described in Appendix A shall be used. As dynamic aspects are in the focus of class I and II simulators, these shall use traffic models given in Appendix A. The traffic model to be used depends on the actual focus of investigation (e.g. throughput vs. latency aspects), however, as a general guideline, also relevance of the corresponding traffic type to the expected future system load shall be considered. In that respect, HTTP traffic (see Appendix A.1), followed by (highly) interactive traffic classes, such as VoIP and gaming, should be used first. If a traffic mix is investigated, the assumptions shall be based on [WIN2D6112].

2.2 Deployment-specific Parameters

Table 2.2 summarises assumptions about the deployment-specific parameters and assumptions. Apart from general assumptions, parameterisation of base station, user terminal, and relay nodes can be distinguished and are described in the following. In contrast to [WIN2D6131] this section provides details on deployment and parameters of relay-enhanced cells (REC) to be used in the different test scenarios.

2.2.1 General

The FDD physical layer mode (PLM) is used in the base coverage urban test scenario, whereas the other test scenarios focus on TDD. In order to provide proof-of-concept under challenging assumptions all test scenarios use relatively high carrier frequencies, around 4 GHz for base coverage urban and microcellular test scenarios, and 5 GHz for indoor. To facilitate comparisons all test scenarios use a total bandwidth of 100 MHz. These general parameters, along with details on base station, relay node, and user terminal configuration are summarised in Table 2.2 and further explained below.

[1] A definition and description of the simulator classes is given in [WIN2D6131]: class I: protocol level simulators, class II: dynamic system-level simulators, class III: quasi-dynamic system-level simulators, class IV: link-level simulator.

Table 2.1: Environment-specific parameters

	Base Coverage Urban	**Microcellular**	**Indoor**
Environment characteristics	Two-dimensional without topographic details	Two-dimensional regular grid of buildings ("Manhattan grid")	One floor of a building with regular grid of rooms, walls and corridors,
		Number of building blocks: 11 x 11 Building block size: 200 m x 200 m Street width: 30 m	Number of rooms: 40 Rooms size: 10 m x 10 m x 3 m Number of corridors: 2 Corridor size: 100 m x 5 m x 3 m
User distribution model (at simulation start-up)	• Number of users is a variable parameter • All users are uniformly distributed in the entire *area*	• Number of users is a variable parameter • All users are uniformly distributed in the *streets* (outdoor UT simulations), or • All users are uniformly distributed in the *buildings* (indoor UT simulations)	• Number of users is a variable parameter • 90% of users are uniformly distributed in *rooms* and 10% of users are uniformly distributed in *corridors*
User mobility model (class III and IV)	• Fixed and identical speed $\|v\|$ of all UTs • $\|v\| \in \{3, 50, 120$ km/h$\}$ $\angle v = \theta_v \sim U(0^o, 360^o)$	• Fixed and identical speed $\|v\|$ of all UTs • $\|v\| \in \{3, 50$ km/h$\}$ $\angle v$: UTs only move along the streets they are in. Direction is random and both directions are equally probable	• Fixed and identical speed $\|v\|$ of all UTs • $\|v\| \in \{0, 5$ km/h$\}$ $\angle v = \theta_v \sim U(0^o, 360^o)$
User mobility model (class I and II)	See [WIN1D72] (Typical urban C2)	See [WIN1D72] (Typical Urban B1)	See [WIN1D72] (Indoor)
User traffic model (class III)	Single traffic flow per user; Full queue per user or Traffic models specified in Appendix A (traffic type dependent on focus of investigation)		
User traffic model (class I and II)	Traffic models specified in Appendix A (traffic type dependent on focus of investigation)		

2.2.2 Base Station

The parameters for the base station are the same as in [WIN2D631], but with two exceptions. The first thing is that details for REC layout have been added, e.g. inter-site distances. The second deviation from [WIN2D6131] is that sectorisation has been introduced in the microcellular scenario, which allows the use of directional antennas and Grid of Beams (GoB) solutions. The idea is to have two sectors, each with main direction in the street canyon but with 180 degrees coverage in order to provide also indoor coverage.

It is important to understand that the back-to-front ratio of the sector antenna elements in the base coverag urban and microcellular test case is 20 dB, i.e. a maximum of 20 dB SINR can be achieved if resources are re-used.

The values of the deployment parameters, i.e. location/antenna height, transmit powers, inter-site distances, etc. are chosen as typical values in these kind of deployments. Four antennas per sector are assumed in the base coverage urban and microcellular scenarios, while 8 antennas are proposed for the indoor base station, all with λ/2 antenna element spacing.

Table 2.2: Deployment-specific parameters

<table>
<tr><th></th><th></th><th>Base Coverage Urban</th><th>Microcellular</th><th>Indoor</th></tr>
<tr><td rowspan="4">general</td><td>duplexing (asymmetry)</td><td>FDD (1:1)</td><td>TDD (1:1)</td><td>TDD (1:1)</td></tr>
<tr><td>carrier frequency f_c</td><td>3.95 GHz DL / 3.7 GHz UL</td><td>3.95 GHz</td><td>5.0 GHz</td></tr>
<tr><td>channel bandwidth</td><td>2 x 50 MHz</td><td>100 MHz</td><td>100 MHz</td></tr>
<tr><td>Deployment (see Figures 3.1 – 3.3)</td><td>cellular, hexagonal layout</td><td>cellular, Manhattan grid lay-out [UMTS 30.03]</td><td>isolated site[2], regular room layout [UMTS 30.03]</td></tr>
<tr><td rowspan="10">base station</td><td>location/height</td><td>Above rooftop, 25 m</td><td>Below rooftop, 10 m</td><td>3 m</td></tr>
<tr><td>max. transmit power per sector</td><td>46 dBm = 39.81 W</td><td>37 dBm = 5.012 W</td><td>21 dBm = 125.9 mW</td></tr>
<tr><td>inter-site distance (only BS layout)</td><td>1 km</td><td>follows from Figure 2.3 and Table 2.1</td><td>N / A</td></tr>
<tr><td>number of sectors per BS</td><td>3</td><td>2</td><td>4 arrays operated as remote radio heads</td></tr>
<tr><td>number of antennas per sector</td><td>4</td><td>4</td><td>8 elements per array</td></tr>
<tr><td>antenna configuration (per sector)</td><td>Linear array
| | | |</td><td>Cross polarised linear array
X X</td><td>Cross polarised linear array
X X X X</td></tr>
<tr><td>antenna element spacing</td><td colspan="3">$0.5\lambda = 0.5c/f_c$ (f_c =DL carrier frequency, c=speed of light)</td></tr>
<tr><td rowspan="2">azimuth antenna element pattern</td><td colspan="3">$$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3dB}}\right)^2, A_m\right] \text{[dB]}$$</td></tr>
<tr><td colspan="2">A_m= 20, θ_{3dB}= 70°
(A_m= 23, θ_{3dB}= 35° for 6 sector site)</td><td>A_m= 12, θ_{3dB}= 70°</td></tr>
<tr><td>elevation antenna gain</td><td colspan="3">14 dBi</td></tr>
<tr><td></td><td>receiver noise figure</td><td colspan="3">5 dB</td></tr>
<tr><td rowspan="7">user terminal</td><td>height</td><td colspan="3">1.5 m</td></tr>
<tr><td>transmit power</td><td colspan="2">24 dBm = 251.2 mW</td><td>21 dBm = 125.9 mW</td></tr>
<tr><td>number of antennas</td><td colspan="3">2</td></tr>
<tr><td>antenna configuration</td><td colspan="3">dual cross polarised antennas: X</td></tr>
<tr><td>azimuth antenna element pattern</td><td colspan="3">$A(\theta) = 0\,\text{dB}$</td></tr>
<tr><td>elevation antenna gain</td><td colspan="3">0 dBi</td></tr>
<tr><td>receiver noise figure</td><td colspan="3">7 dB</td></tr>
</table>

[2] The indoor test case considers isolated cell for link level simulation but consider deployment based on a couple of cells for radio resource management strategies (e.g. to evaluate coordination mechanisms between BSs).

<table>
<tr><th></th><th></th><th>Base Coverage Urban</th><th>Microcellular</th><th>Indoor</th></tr>
<tr><td rowspan="9">relay node</td><td>location/height</td><td>Below rooftop, 5 m</td><td>Below rooftop, 10 m</td><td rowspan="9">N / A[3]</td></tr>
<tr><td>max. transmit power per sector</td><td>37 dBm = 5 W</td><td>30 dBm = 1 W</td></tr>
<tr><td>number of sectors per RN</td><td>1</td><td>1</td></tr>
<tr><td>number of antennas per sector</td><td>1</td><td>1</td></tr>
<tr><td>antenna configuration (per sector)</td><td colspan="2">single antenna with omnidirectional pattern</td></tr>
<tr><td>antenna element spacing</td><td>N/A</td><td>N/A</td></tr>
<tr><td>azimuth antenna element pattern</td><td>omnidirectional</td><td>omnidirectional</td></tr>
<tr><td>elevation antenna gain</td><td>9 dBi</td><td>7 dBi</td></tr>
<tr><td>receiver noise figure</td><td colspan="2">5 dB</td></tr>
</table>

2.2.3 User Terminal

The user terminal parameters remain unchanged compared to [WIN2D6131]. Dual-polarised transmit and receive antennas are assumed using 24 dBm transmit power in the base coverage urban and microcellular test scenario, and 21 dBm indoors. An ideal omnidirectional antenna characteristic is assumed and the noise figure of 7 dB accounts for cheap mass-market devices.

2.2.4 Relay Node

Relay nodes have different constraints related to deployment, size, and cost compared to base stations. To reflect these constraints, RN use a lower maximum transmit power, a lower number of sectors and antennas. The requirement to have small RN suited for e.g. lamppost mounting, makes it impossible to use the same large vertical antenna aperture (panel antennas) as they are used at BSs. However, an antenna aperture of 10 cm x 4 cm, showing omnidirectional antenna pattern and providing elevation gain of 9 dBi seems feasible using three radiating elements. Therefore for the RN in the base coverage urban test scenario a single antenna with such characteristics is assumed. For the microcellular test case a single omnidirectional antenna with 7 dBi is assumed (the lower elevation gain accounts for the requirement of a larger beamwidth in elevation). These estimations are based on currently available antenna elements [Kat06]. It has been assumed that identical gains can be obtained when scaling the aperture size with the carrier wavelength. The above antenna configurations are considered cheap and feasible for lamppost mounting and are therefore proposed as basic assumptions.

Note, in sum the effective isotropic radiated power (EIRP) of a RN is reduced by 14 dB in the base coverage urban and microcellular test case. Under the requirement of the same QoS at the cell border (approximated initially by the same average SINR in the link budget) the relay cells will have significantly reduced cell ranges compared to the BS.

2.2.5 Network Layout

In the base coverage urban case, no specific topographical details are taken into account. Base stations and relays are placed in a regular grid, following hexagonal layout. A basic hexagon layout for a deployment consisting only of base stations is shown in Figure 2.1, where also basic geometry (antenna boresight, cell range, and inter-site distance ISD) is defined.

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[3] No detailed configuration for relaying in indoor is included in baseline assumption. In case investigations for indoor scenario including relays are performed, it is suggested to use configuration that allow small and cheap RNs, e.g. maximum transmit power (21 dBm), single sector and 2 antenna elements. Mounting at the ceiling (3 m height) can be assumed.

In the base coverage urban deployment option including relays, those are placed along the antenna bore-sight of each sector at 2/3 of the cell hexagon diameter, i.e. 444 m from the BS, cf. Figure 2.2. The antenna array orientation of the relay note is such that it has the same bore-sight direction as the serving BS sector. Each antenna element at the RN has omnidirectional antenna pattern.

Please note, that for the REC deployment, the cell shape will deviate from the hexagon form and the actual shape of the relay cells will depend on the interference situation, i.e. the details of resource partitioning and re-use. It is important to understand that the placement of radio access points (RAPs) does not consider propagation conditions, like shadowing. Users are distributed uniformly over the whole area.

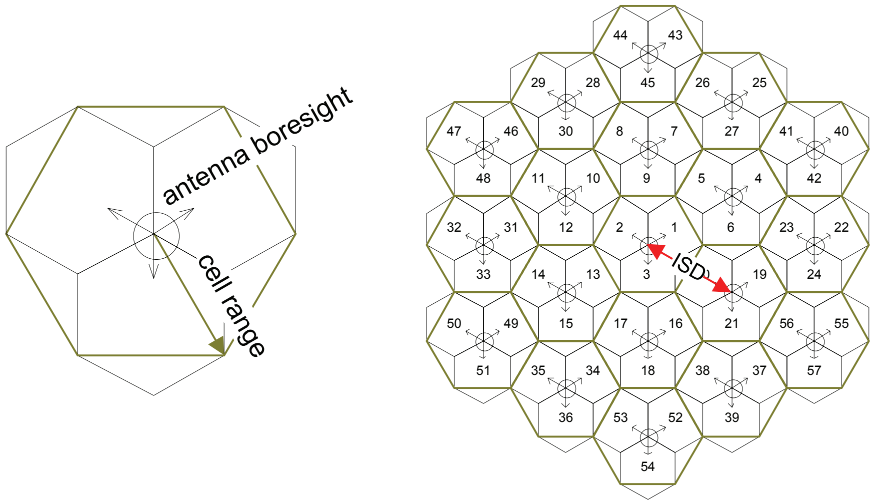

Figure 2.1: Sketch of base coverage urban cell layout without relay nodes

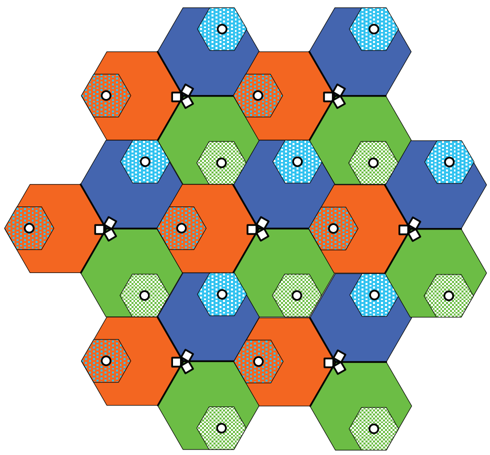

Figure 2.2: Sketch of base coverage urban cell layout with relay nodes for coverage extension

In the microcellular test case, a two-dimensional regular grid of streets and buildings is considered, the so-called Manhattan grid (Figure 2.3). Base stations are placed in the middle of the streets and in the middle between two cross-roads. Two sectors are formed with array bore-sight along the street direction, but with 180° coverage each. The corresponding relay-enhanced cell deployment is shown in Figure 2.4.

The indoor scenario consists of one floor (height 3 m) of a building containing two corridors of 5 m x 100 m and 40 rooms of 10 m x 10 m, as depicted in Figure 2.5. To highlight innovative deployment concepts a remote radio head deployment is investigated here. Four antenna arrays containing each 8 antennas (further details as in Table 2.2), and placed in the middle of the corridor at 25 m and 75 m (with respect to the left side of the building). The antenna array orientation is rotated by 45° as shown in Figure 2.5. It is assumed that all antenna arrays are operated by one central BS.

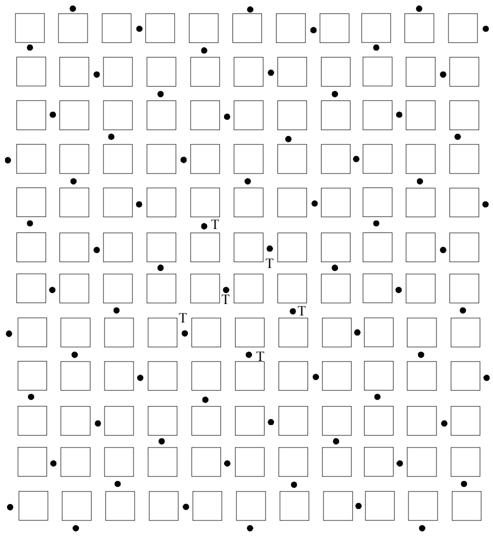

Figure 2.3: Sketch of microcellular cell layout without relay nodes

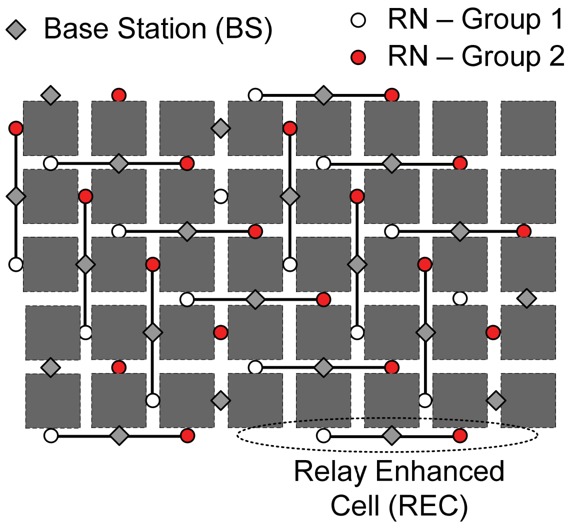

Figure 2.4: Sketch of microcellular cell layout with relay nodes

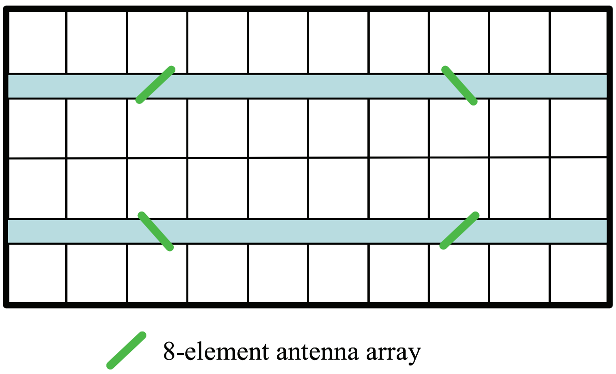

Figure 2.5: Sketch of indoor environment (one floor)

2.3 Channel Modelling

The channel model parameters to be used in the test scenarios are provided in Table 2.3. The channel model nomenclature follows [WIN2D111], where the latest status of the WINNER channel models is captured. Different channel models are used for the links between BS and UT, BS and RN, as well as RN and UT. Note, that in contrast to previous assumptions, all BS-RN links are now assumed to be line-of-sight (LOS).

Table 2.3: Channel modelling parameters

<table>
<tr><th colspan="2"></th><th>Base Coverage Urban</th><th>Microcellular</th><th>Indoor</th></tr>
<tr><td colspan="2">channel model BS↔outdoor UT</td><td>C2/(C1)</td><td>B1</td><td>N / A</td></tr>
<tr><td colspan="2">channel model BS↔indoor UT</td><td>B4</td><td>B4</td><td>A1</td></tr>
<tr><td colspan="2">channel model BS↔RN</td><td>C1[4]</td><td>B5c</td><td>A1LOS.</td></tr>
<tr><td colspan="2">channel model RN↔outdoor UT</td><td>B1 NLOS / (B1 LOS)</td><td>B1</td><td>N / A</td></tr>
<tr><td colspan="2">channel model RN↔indoor UT</td><td>B4</td><td>B4</td><td>A1</td></tr>
<tr><td colspan="2">percentage of indoor users served by outdoor RAP</td><td>70%</td><td>30%</td><td>N / A</td></tr>
<tr><td colspan="2">Path loss models</td><td colspan="3">see [WIN2D111]</td></tr>
<tr><td rowspan="3">Correlation of large scale par</td><td>between sectors of different sites</td><td colspan="3">No</td></tr>
<tr><td>between sectors of same sites</td><td colspan="3">Full, i.e. use identical large scale parameters for all sectors</td></tr>
<tr><td>between UTs of same site</td><td colspan="3">Distance dependent, see [WIN2D111]</td></tr>
<tr><td colspan="2">Correlation of small scale parameters between sectors of same site</td><td colspan="3">Partly full, i.e. use identical small scale parameters except the sub path-phases which are redrawn randomly.</td></tr>
<tr><td colspan="2">noise power spectral density</td><td colspan="3">-174 dBm/Hz</td></tr>
</table>

In the base coverage urban test case, C2 and B1 NLOS channel models should be used for the UT links from BS and RN, respectively. Both are based on NLOS conditions and therefore provide a challenging assumption. To evaluate the sensitivity of results to this assumption, C1 and B1 LOS channel models can be used in optional simulations. In this case, both channel models assume LOS and therefore two extreme cases are covered. As the B1 channel model was designed for use with Manhattan grid simulators, it requires two separate distances d_1 from the RAP to the corner, and d_2 from the corner to the UT. For use in the base coverage urban test case the total distance between RAP and UT d shall be related to these parameters by

$$d = d_1 + d_2\ ;\quad d_1 = d_2 = \frac{d}{2}\ . \qquad (2.1)$$

Serving indoor users by outdoor RAPs is of major importance for the base coverage urban and microcellular test case. Therefore different channel models are defined depending on whether the UT is located outdoor or indoors. Two sets of simulations shall be performed for these test cases, one containing

[4] C1 shall be used as current working assumption. Discussion and potential refinement of the BS-RN link channel model is currently ongoing

only outdoor users and the other containing only indoor users. It is understood that this separation of outdoor- and indoor user simulations does not capture all effects that might be encountered in a mixed and dynamic scenario. However, this approach allows to investigate performance in both extreme cases as well as to draw initial conclusions for overall performance based on a weighted averaging of both results in these test scenarios. Initially this weighting will be based on the percentage of indoor users given also in Table 2.3.

Depending on the topic under investigation, different requirements on the simulation set-up exists. While in some cases it might be important to average over as many uncorrelated user drops as possible, other simulations (e.g. packet delay statistics) might (also) require sufficiently long time evolutions for each individual random placement of a user population in order to obtain sufficient statistics.

3. Baseline System Design

For the test scenarios described in Chapter 2, baseline assumptions for system design have been developed. Major changes with respect to [WIN2D6131] are mentioned explicitly and a short rationale and references for further reading are given. Important updates have occurred with respect to the basic OFDM parameters and chunk dimensions, the baseline channel coding, the multiple access scheme for non-frequency adaptive transmissions, and the baseline spatial processing schemes. Such parameter changes are highlighted by red colour in the tables and figures. Also, new aspects are included and more details are provided, in particular with respect to relaying, segmentation, link adaptation, and HARQ.

3.1 Basic OFDM Parameters and Frame Dimensions

3.1.1 OFDM Parameters and Chunk Definition

With respect to [WIN2D6131] the basic OFDM parameters, outlined in Table 3.1, have been adapted, with modified values indicated by boldface red entries. In particular, the spectral guard bands have been reduced to 10% of the total bandwidth based on recent investigations on complexity and feasibility to implement the associated filters and in order to align with assumptions in other OFDM-based radio access systems. Therefore the number of used subcarriers, the signal bandwidth, and consequently the number of chunks per frame (see Table 3.2) are increased in both physical layer modes. Furthermore, the guard interval T_G in the TDD mode has been increased from 1.28 µs to 2.00 µs (Table 3.1) in order to reduce potential interference from neighbouring BS due to street-canyon effects in cities. The total frame duration is kept constant (and identical to the FDD frame duration), since the duplex guard time has been reduced accordingly to 2 x 8.4 µs, cf. Table 3.2.

Table 3.1: OFDM/GMC parameters

	Base Coverage Urban	Microcellular	Indoor
Subcarrier distance Δf	39062.5 Hz	48828.125 Hz	
Useful symbol duration T_N	25.6 µs	20.48 µs	
Guard interval T_G	3.2 µs	**2.00** µs	
Total symbol duration	28.8 µs	**22.48** µs	
used subcarriers	**[-576:576]** subcarrier 0 unused	**[-920:920]** subcarrier 0 unused	
Signal bandwidth	**2 x 45 MHz**	**89.84 MHz**	
System bandwidth	2 x 50 MHz	100.0 MHz	
FFT bandwidth, sampling rate	80.0 MHz	100.0 MHz	

In the FDD physical layer mode, an optional OFDM parameter set can be used in rural coverage investigations and in investigations where the extent of the cyclic prefix could constrain the performance (inter-cell interference coordination, multicasting that uses inter-cell macro-diversity transmission). This FDD parameter set provides a double-length OFDM guard interval by using a 4096 point FFT with 80 MHz sampling rate. Its OFDM symbol period is 51.20 µs and the OFDM symbol guard time is 6.4 µs.

The chunk size for the FDD mode also remains unchanged compared to [WIN2D6131]. However, the following changes are introduced in the TDD mode:

- The number of subcarriers per chunk is halved from 16 to 8 in order to adapt to the frequency selectivity encountered in metropolitan outdoor scenarios, such as the microcellular test case.
- Due to the new multiple access schemes for non-frequency adaptive transmission (B-IFDMA, B-EFDMA, see Section 3.2.2) and in order to keep the total chunk size in the same range as in the

FDD mode, the TDD chunk extends over the whole half-frame, i.e. over 15 OFDM symbols for 1:1 asymmetry, yielding a total of 120 symbols.

The new chunk dimensions are depicted in Figure 3.1, with modifications indicated by boldface red labels.

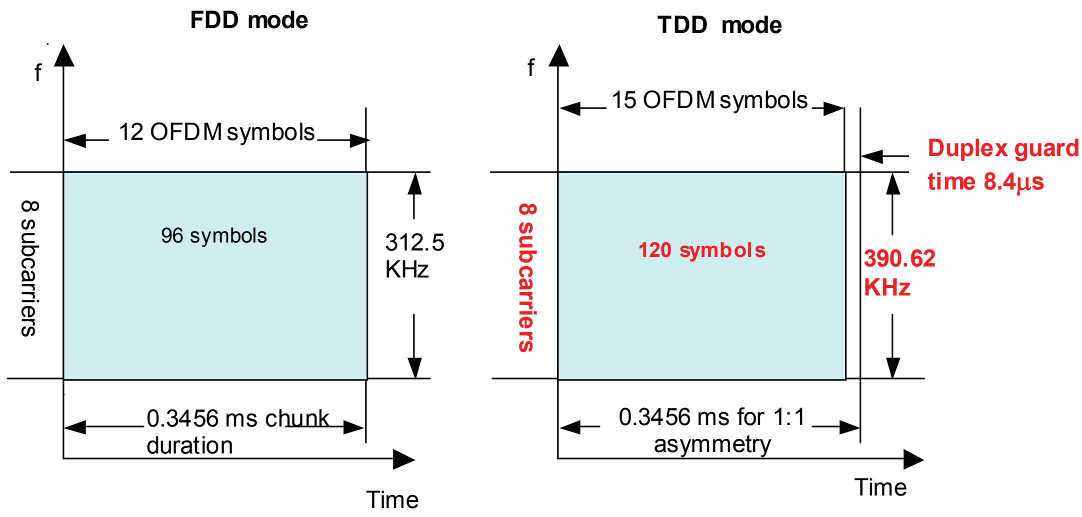

Figure 3.1: Summary of chunk sizes in the two physical layer modes

3.1.2 Frame Design, Pilot and Control Overhead

Pilot and control channel design is still a subject under study in WINNER. Initial assumptions are briefly summarised in the following. On the downlink, common pilots, dedicated pilots, or combinations thereof may be used. Dedicated pilots offer maximum flexibility, to support various spatial precoding approaches, in combination with adaptive multiple access schemes. They are moreover applicable to the downlink as well as to the uplink. Common pilots, on the other hand, have the advantage that interpolation over adjacent chunks is possible. Furthermore, since common pilots do not experience user-specific spatial processing, the unweighted channel matrix over the full band can be estimated. Unfortunately, common pilots are inefficient when the number of antennas is much larger than the number of spatial streams. Also, common pilots are insufficient when adaptive beamforming is employed. The current WINNER baseline design does not allow the use of common and dedicated pilots simultaneously, although it might turn out that this could become necessary. The trade-off and possible combinations between common and dedicated pilots are for further studies.

3.1.2.1 Downlink Dedicated Pilots

- *Frequency-adaptive transmission in the FDD mode*: 4 pilot time-frequency symbols per chunk layer, placed in a rectangular pattern. A frequency spacing $D_f = 6$ and a spacing in time of $D_t = 9$ or 10 that places the pilots near the corners of chunks and in neighbouring positions in different spatial layers/beams, is assumed for frequency adaptive transmission[5]. The maximum speed considered in the test scenarios is 120 km/h[6]. The pilot pattern for a system with 4 spatial layers is shown in Figure 3.2. Pilots are orthogonally separated in time and frequency, i.e. when a pilot is transmitted on a spatial stream, all other streams transmit zeros. With the given parameters up to 4 spatial layers can be supported per chunk.

[5] Having four pilot time-frequency symbols per chunk layer, rectangular spaced in both time and frequency, enables the use of linear interpolation of the time and frequency variation of the channel within each chunk, based only on the pilot symbols that are present within a chunk. These estimates can be used as initial values for iterative channel estimation schemes that use the payload symbols. When utilising also pilots from neighbouring chunks, higher order (Wiener) interpolated channel estimates can be calculated. However, please note that chunk layers that are allocated to non-frequency adaptive transmission use a different, block-based pilot pattern than the chunk layers that are earmarked for frequency-adaptive transmission.

[6] For velocities exceeding 150 km/h an additional pair of pilot symbols should be assumed.

- *Frequency-adaptive transmission in the TDD mode*: With the modified chunk dimensioning of Figure 3.1, (with increased chunk duration) it is practical to use four pilots per chunk in a rectangular pattern also in the TDD mode. The spacing in the time direction should be adjusted to the used uplink:downlink asymmetry ratio. Owing to the similar chunk parameters, the pilot pattern for the TDD mode is similar to the FDD pilots, with the exception that the pilot spacing in time should be variable. With a link asymmetry ratio of 1:1, a pilot spacing of D_f = 6 and D_t = 12 in frequency and time allows to support up to 8 spatial layers.

- *Non-frequency adaptive downlink transmission, FDD and TDD modes*: When using the B-EFDMA scheme for non-frequency downlinks (Section 3.2.2), one pilot symbol is included within each ***block,*** if possible located near the centre of the block. The assumed block size is 4 subcarriers by 3 OFDM symbols, cf. Figure 3.7. The blocks can be assumed flat in time and frequency. A larger number of pilots (one per 4 x 3 block per layer instead of four pilots per chunk layer) are thus required for the non-frequency adaptive transmission, as compared to the frequency-adaptive transmission. With 8 blocks per chunk in FDD and 10 blocks/chunk in TDD, the pilot overhead becomes 8/96 and 10/120, respectively.

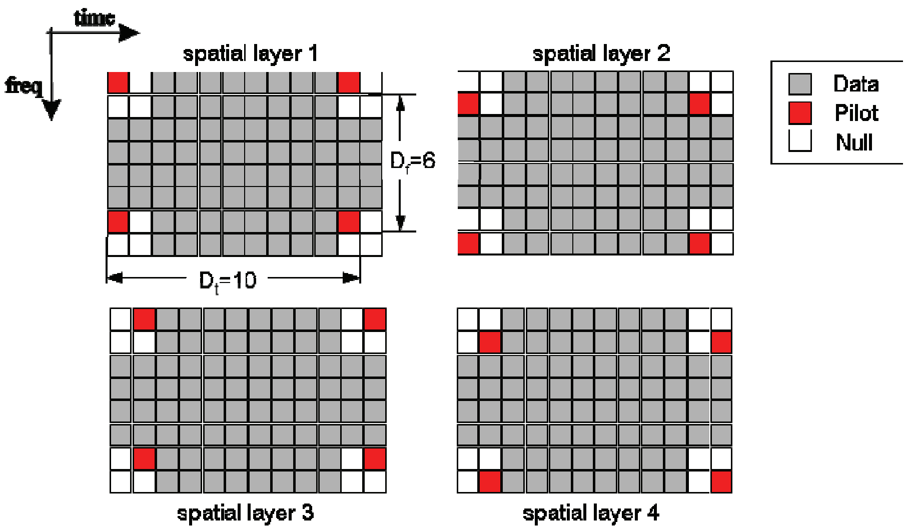

Figure 3.2: Dedicated pilot allocation for 4 spatial streams in FDD mode

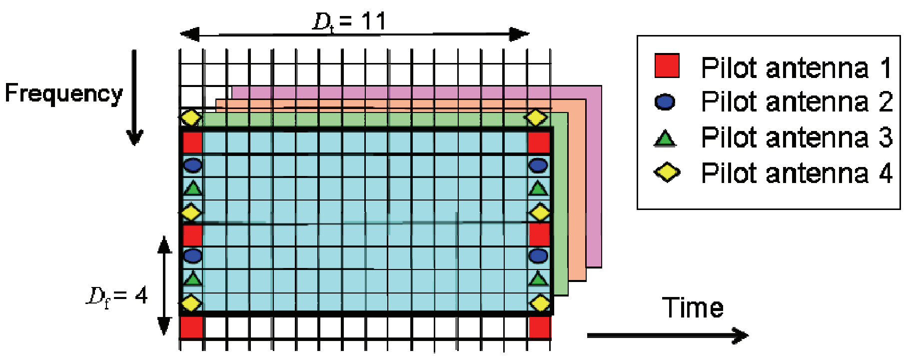

Figure 3.3: Common pilot allocation for 4 spatial streams in FDD mode

3.1.2.2 Downlink Common Pilots

On the downlink common pilots per antenna may be used. According to [WIN1D210] pilot spacings of D_f = 4 and D_t = 11 in frequency and time for FDD mode were identified. For TDD mode pilot spacings become D_f = 8 and D_t = 14. There, the spacing in the time direction should be adjusted to the utilised uplink:downlink asymmetry ratio. This translates to 4 and 2 pilots per chunk per antenna in FDD and

TDD mode, respectively. Pilots from multiple transmit antennas are orthogonally separated in frequency, as illustrated in Figure 3.3 for the FDD mode with 4 transmit antennas. Since common pilots are not subject to user specific processing, interpolation in frequency is possible, and edge effects are not as problematic as for dedicated pilots.

3.1.2.3 Control Overhead and Frame Parameters

The reference design for frequency-adaptive transmission, assumes a downlink control overhead of 6 bits per allocated downlink chunk layer, and 6 bits per allocated uplink chunk layer. If we assume initially QPSK, R=1/2 coding, (appropriate for frequency adaptive transmission at SINR 5 dB or higher, see section 5.2.1 of [WIN2D461]), then 12 control symbols are required for controlling a downlink chunk and a subsequent uplink chunk. The physical placement of these control symbols is not specified. They can be placed either within the downlink chunks that are used for frequency-adaptive transmission (in-chunk control signalling, as exemplified in Section 3.1 of [WIN1D24]) or at separate positions within the slot, preferably using non-frequency adaptive transmission.

The corresponding downlink overhead for non-frequency adaptive transmission is not yet quantified, and it depends on e.g. the minimum assumed SINR. Although less information is required for non-frequency adaptive transmission (identical MCS for all chunks of one FEC block), it requires stronger channel coding, to reliably reach users with SINR < 5 dB. As an initial value we assume 50% more control symbols to be required, i.e. 18 control symbols.

The frame parameters and downlink overhead assumptions are summarised in Table 3.2.

Table 3.2: Frame parameters

		Base Coverage Urban	**Microcellular**	**Indoor**
Overall frame length		0.6912 ms		
Number of OFDM symbols per frame		24	30	
Chunk layer dimension in symbols x subcarriers		12 x 8 =96	**15 x 8 = 120**	
Downlink	Dedicated pilot + control symbols per chunk layer in frequency adaptive transmission	**4 + 12 = 16**	**4 + 12 = 16**	
	Dedicated pilot + control symbols per chunk layer in non-frequency adaptive transmission	**8 + 18 = 26**	**10 + 18 = 28**	
Number of chunks per frame in time and frequency direction		**2 x 144**	**2 x 230**	
Duplex guard time		0 µs	**2 x 8.4 µs**	

The uplink pilot and control overhead is for further study. The pilot patterns for dedicated pilots are equivalent to the previously described downlink case. Channel state information (CSI) feedback reporting overhead should be below 4 bits per chunk layer in control loops for FDD frequency adaptive downlinks (up to 16 vehicular users per competition band), see Section 3.1 of [WIN1D24] and Section B.5.3 of [WIN1D210].

After final figures for in-chunk pilot and control overhead are obtained, the chunk dimensioning might further be adapted in order to match the chunk payload size to the basic granularity of the used channel codes (currently 48 bits for the B-LDPCC described in Section 3.2.5).

3.1.3 Superframe Parameters

The overall super-frame design remains as in [WIN2D6131][7], with some slight adaptations in the assumed length of the different parts in the pre-amble, i.e. the synchronisation slots are each reduced by one symbol, whereas the number of OFDM symbols in the RAC slot has been increased, as it acts also as timing misalignment guard for the initial UL synch symbol transmitted by a terminal. The general layout of a super-frame is given in Figure 3.4, the details of the pre-amble follow in Table 3.3. For more details and background on the basic OFDM and frame dimensioning the reader is referred to [WIN1D210, WIN2D6131].

In the TDD Physical layer mode, a *frame* comprises an UL *slot* followed by a DL slot. In the FDD mode, half-duplex terminals are assigned to one of two groups. Group 1 transmits in the downlink in the first slot of the frame and in the uplink in the latter slot. Group 2 transmits/receives in the opposite way. ([WIN1D210], section C.1.2). Full-duplex FDD terminals are also supported in BS-to-UT transmissions.

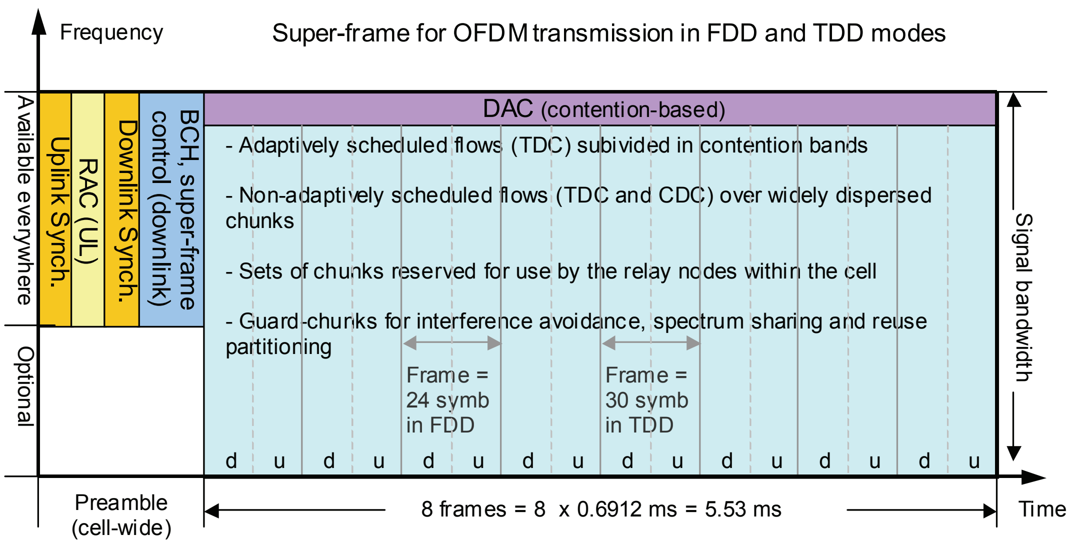

Figure 3.4: Sketch of super-frame and frame structure (FDD and TDD), for TDD: d: downlink, u: uplink

The super-frame is designed to have the same length in the FDD and the TDD physical layer mode, despite different OFDM symbol durations, to facilitate inter-mode cooperation:

The preamble duration is **0.360 ms**, the payload duration **5.5296 ms**, total super-frame: **5.8896** ms.

In the baseline assumption, the preamble occupies the same signal bandwidth as the payload part. In simulations with variable/flexible spectrum use, the preamble shall occupy spectral bands that are available always and everywhere.

Ideal network synchronisation of BS and RNs is the baseline assumption in the base coverage urban and the microcellular scenario. The super-frames of all BS and RNs are synchronised, i.e. begin at the same time).

In FDD, the uplink band contains the UL synchronisation and the RAC part of the preamble, while the remaining preamble duration is empty. The downlink band contains the DL synchronisation and the BCH slot.

The RAC slot duration and the guard times of the preamble differ in the FDD and the TDD physical layer mode. This is for three reasons:

- The preamble durations are thereby made equal in the two modes.
- The guard interval is in FDD required only to guard against RAC packet time misalignments. In TDD, it is needed also as a duplex guard interval for the UL-DL switching, and therefore needs to be larger.

[7] Although the necessity of having a time-multiplexed pre-amble section is currently in discussion in WINNER.

- The RAC time-slot is in TDD of interest not only for initial access by UTs but also potentially for BS-to-BS over-the air communication. The potential for this use increases by increasing its length, which is in TDD set to 6 OFDM symbols.

The re-use 7 factor for the BCH OFDM symbols guards the broadcast channel against inter-cell interference from the other simultaneous BCH transmissions from RNs of the same cell and BSs and RNs in other cells. The exact partitioning of the frequency re-use 7 pattern between base stations and relay nodes is not yet specified.

Table 3.3: Super-frame parameters

		Base Coverage Urban (FDD)	**Microcellular (TDD)**	**Indoor (TDD)**
Payload duration = DAC duration		8 frames = 5.5296 ms		
Preamble	pre-amble duration	**0.360 ms**		
	UL Synch	**2 OFDM symbols**	**2 OFDM symbols**	
	RAC (UL)	**3 OFDM symbols**	**6 OFDM symbols**	
	Guard time	14.4 µs	22.8 µs	
	DL Synch	**3 OFDM symbols**	**3 OFDM symbols**	
	BCH (DL)	4 OFDM symbols	4 OFDM symbols	
	BCH Subcarrier re-use factor	7		

3.2 Basic Configuration of Functions

A summary of the basic assumptions and algorithms used for the main functionalities is provided in Table 3.4 in order to have a quick overview. Further information and details for each area are provided in the subsequent sections.

Table 3.4: Baseline system configuration

PHY Entity		**Base Coverage Urban**	**Microcellular**	**Indoor**
Network Synchronisation		ideal synchronisation of BS		N / A
Flow State Control		all flows in active state (unless flow state control is subject of investigation)		
Intercell Resource Partitioning		Frequency re-use 1 in entire cell, i.e. no resource partitioning[8]		N / A
Resource Partitioning within a REC		**see Section 3.2.1**		N / A
Multiple access	Downlink	Frequency-adaptive RS: chunk based OFDMA-TDMA, optionally SDMA non-frequency adaptive RS: **B-EFDMA**, optionally SDMA		
	Uplink	Frequency adaptive RS: chunk based OFDMA-TDMA, optionally SDMA non-frequency adaptive RS: **B-IFDMA**, optionally SDMA		
Resource Scheduling (RS)		centralised at BS, using Score-based scheduler for frequency-adaptive mode and Round Robin scheduler for non-frequency adaptive mode;		
Constraint Processing		no constraints		
Spatial Scheme Control		fixed spatial scheme per flow		
Spatial processing	BS → UT link	GoB using short term or long term channel knowledge for beam selection	**GoB using short term or long term channel knowledge for beam selection**	**Single user SVD MIMO**
	UT → BS link	OSTBC	OSTBC	OSTBC
	BS → RN link	**GoB**	**GoB**	**N / A**
	RN → BS link	**N / A**		
	RN → UT link			
	UT → RN link	**OSTBC**	**OSTBC**	
	UT → UT link	-	-	Closed loop/LDC
Coding		**B-LDPCC, mother code rate 1/2** or convolutional codes (memory 8, R=1/2, 1/3) for FEC blocks containing less than 200 information bits, e.g. control signalling see section 3.2.5		
Interleaving		random (basic requirement for L2S interface)		

[8] Re-use 1 is used as a simple baseline assumption although it will lead to pessimistic results. Advanced resource partitioning schemes (e.g. adaptations of [Hal83, Ste03] to the test scenarios) are under currently study in the project.

<table>
<tr><th colspan="2">PHY Entity</th><th>Base Coverage Urban</th><th>Microcellular</th><th>Indoor</th></tr>
<tr><td colspan="2">Modulation</td><td colspan="3">M-QAM with Gray mapping, M=2, 4, 16, 64</td></tr>
<tr><td colspan="2">Tx power control</td><td colspan="3">DL: Tx power control might be applied to individual flows but the overall transmit power should be regarded as constant unless specific investigations w.r.t. interference avoidance techniques are carried out
UL: Perfect (signals of all simultaneously scheduled UT in the uplink arrive with same average power at BS)</td></tr>
<tr><td colspan="2">Segmentation</td><td colspan="3">segmentation into FEC blocks of 360 or 1200 information bits (including MAC header)</td></tr>
<tr><td colspan="2">Retransmission Unit</td><td colspan="3">1 RTU (+retransmissions) scheduled per user and slot</td></tr>
<tr><td colspan="2">FEC Block</td><td colspan="3">≥ 1 FEC block scheduled per user and slot</td></tr>
<tr><td rowspan="5">Link Adaptation</td><td>baseline modulation and coding scheme</td><td colspan="3">see Section 3.2.5</td></tr>
<tr><td>MCS calculation for frequency adaptive transmission</td><td colspan="3">identical code rate but adaptive modulation within one FEC block</td></tr>
<tr><td>MCS for non-frequency adaptive transmission</td><td colspan="3">MCS determined based on average SINR of FEC block</td></tr>
<tr><td>Delay</td><td colspan="3">see Section 3.2.7</td></tr>
<tr><td>Error</td><td colspan="3">Ideal</td></tr>
<tr><td rowspan="5">HARQ</td><td>Protocol</td><td colspan="3">N-Channel Stop-And-Wait, with one channel per slot per flow (at most one RTU from one flow transmitted per slot)</td></tr>
<tr><td>Combining</td><td colspan="3">Chase Combining</td></tr>
<tr><td>max. number of retransmissions</td><td colspan="3">0, 4, 10, ∞
(depending on delay requirements of service)</td></tr>
<tr><td>Delay</td><td colspan="3">delay of ACK/NACK: 1 frame
delay of retransmission: 2 frames, i.e.
a retransmission can occur earliest in the 2rd frame (j+2) after the previous transmission in frame j (section 3.2.7)</td></tr>
<tr><td>Error</td><td colspan="3">Ideal</td></tr>
<tr><td colspan="2">Receiver processing</td><td colspan="3">MMSE based equalisation (if required at all; at UT use MRC without interference rejection)
ML based soft output demodulation (if equaliser is present, apply ML criterion to signal at equaliser output)
perfect knowledge of SINR when computing LLR’s</td></tr>
</table>

3.2.1 Baseline Modelling and Resource Partitioning for Relaying

The baseline assumptions for RNs in WINNER is that they will be implemented as decode and forward nodes which are optimised for two hops. Thus the RN is taking benefit from communicating with the BS using a higher MCS than used for serving most of its UTs. The RNs are seen as fixed nodes (no mobility). One RN is connected to only one BS.

In the baseline assumptions the RN should transmit its own preamble (incl. BCH), whereby further implementations are under discussion (see [WIN2D351]). The RN is acting towards its UTs like a BS. In the resource partitioning the RN will get assigned resources for its exclusive use to serve its UTs.

3.2.1.1 Baseline resource partitioning

The baseline concept for resource partitioning is static and based on the assumption of a re-use one network. In TDD the RN is active as serving node (like a BS) every 2nd MAC frame as shown also in Figure 3.5.

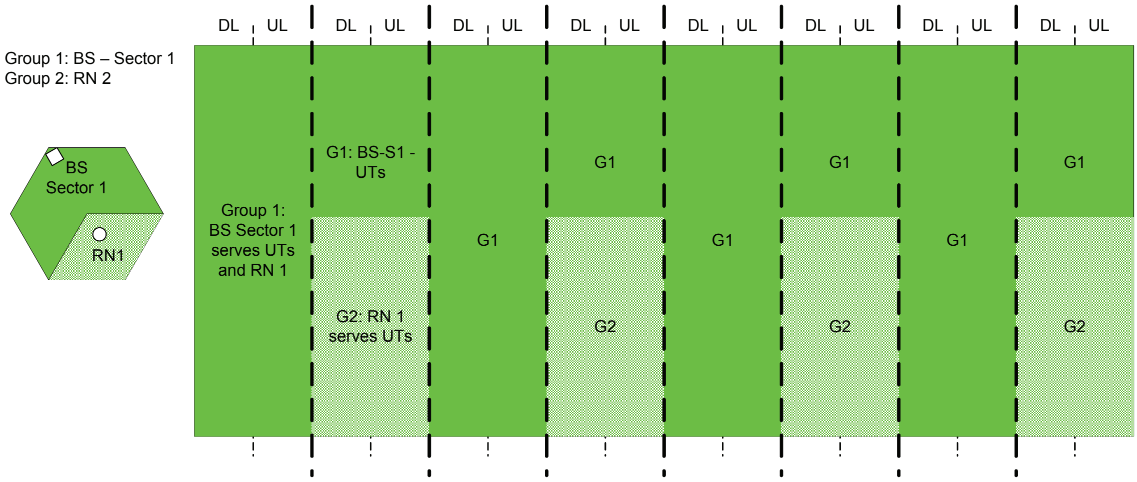

Figure 3.5: Exemplary resource partitioning between RN and BS resources in TDD mode

In FDD, receiving (Rx) and transmitting (Tx) phases of the RN can be distinguished as shown in Figure 3.6. The fat boxes highlight the resources where the RN is serving its UTs either in UL or in DL.

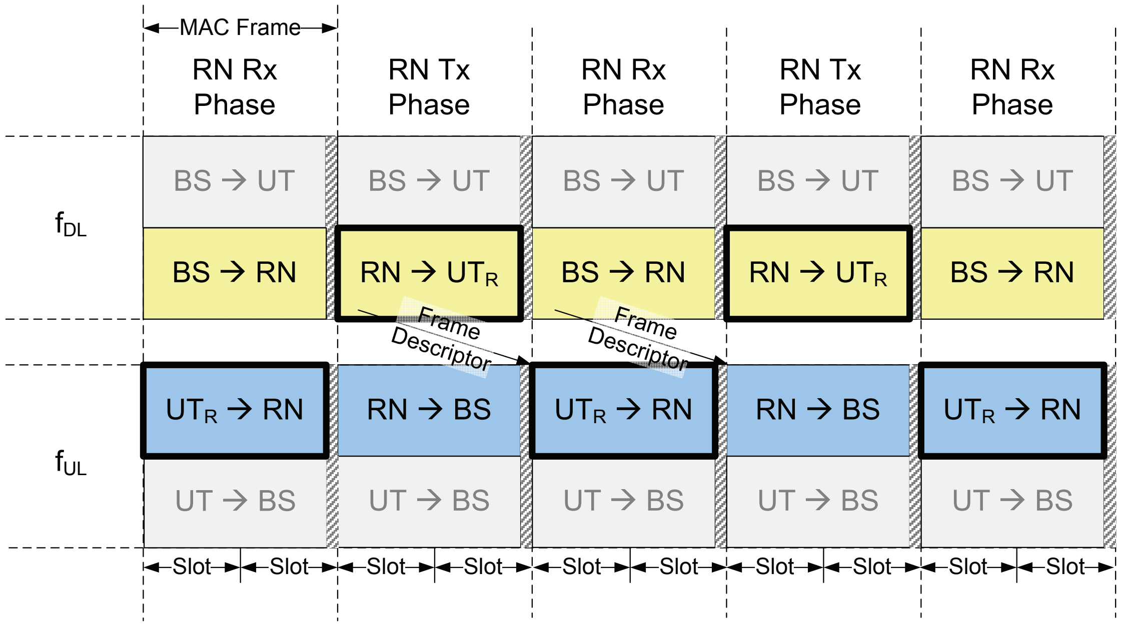

Figure 3.6: RN roles in FDD mode

The amount of resources that has to be dedicated to the BS-UT, BS-RN and RN-UT links respectively is influenced by a number of factors which are briefly discussed in the following:

- The ratio of the area of the relay subcell to the area of the sector that is being served via 1 hop (in fact, the important figure is the number of users in the different subcells of the REC, but we assume a uniform user distribution and hence a number of users that is proportional to the subcell area), we assume the subcell area served by the RN to be one third of the sector area in both the base coverage urban and the microcellular test scenario.

- The throughput (and hence the resource demand) of the relay link in comparison to the average throughput in the relay subcell. For the baseline assumptions the relay link throughput is assumed to be twice as high as the average throughput in the relay subcell for both the base coverage urban and the microcellular test scenario.
- The ratio of the average throughput per user in the one-hop area of the cell versus the average throughput per user achieved on the second hop (i.e. in the relay subcell), for simplicity these figures are assumed as being equal in both base coverage urban and the microcellular test scenario.

The amount of resources partitioned for the use in one sector of a sectorised BS is deployment-specific, whereas the distribution of these resources for the relaying case depends on the factors outlined above. As a result of the above assumptions, the following partitioning ratios for the resources available in a sector are suggested as baseline assumptions.

Table 3.5: Resource partitioning between RN and BS for the base coverage urban and microcellular test case

Link	Portion of Resources
BS-UT	4 / 7
BS-RN	1 / 7
RN-UT	2 / 7

For sake of simplicity the effects of MIMO have been neglected in the above resource partitioning estimation. The capacity required for the relay link (RN-BS link) should be calculated using the parameters and provided in Table 2.2 and Table 2.3. For the baseline assumption, the relay link needs not be modelled explicitly, i.e. the link quality and the required resources to serve the users can be estimated based on long-term average SINR evaluations.

3.2.2 Multiple Access

3.2.2.1 Frequency-adaptive transmission

The multiple access scheme for frequency-adaptive uplink and downlink remains unchanged from WINNER phase 1: Chunk-based **TDMA/OFDMA** is used in both FDD and TDD modes. The same scheme is used for both uplinks and downlinks.

Chunk-based TDMA/OFDMA means that flows are mapped onto individual chunk layers. The mapping is exclusive within the cells, i.e. each chunk layer carries data from only one flow. Individual link adaptation is used within each chunk layer, based on predicted SINR that will be perceived within that particular chunk layer for that particular user, at the time instance when the transmission will occur. One set of link adaptation parameters is used within the whole chunk.

Fast control loops for FDD and TDD enable reliable prediction up to vehicular speeds [WIN1D210]. The uplink control is based on a request for transmission in frame *j-2*. If granted, the transmission is scheduled and prepared during frame *j-1* and is then performed in the uplink slot of frame *j* (see Section 3.2.7, and Section C.1.5 of [WIN1D210]). The downlink transmission in frame *j*, is prepared by using a small amount of downlink control signalling during frame *j-1*. The main downlink control signalling then follows during frame *j*. It is performed simultaneously with the payload transmission to reduce delays.

Note that strong FEC coding will span multiple chunks with individually calculated link adaptation parameters, as described in Section 3.2.5.

3.2.2.2 Non-frequency adaptive transmission

New baseline designs of the multiple access schemes for non-frequency adaptive uplinks and downlinks are introduced. They are called Block Interleaved Frequency Division Multiple Access (B-IFDMA) and Block Equidistant Frequency Division Multiple Access (B-EFDMA) respectively, see [WIN2D461] for further details. The resource allocation for B-IFDMA and B-EFDMA are the same and is illustrated in Figure 3.7 below. The difference between the schemes is that in B-IFDMA a common DFT precoding step is performed over the allocated blocks.

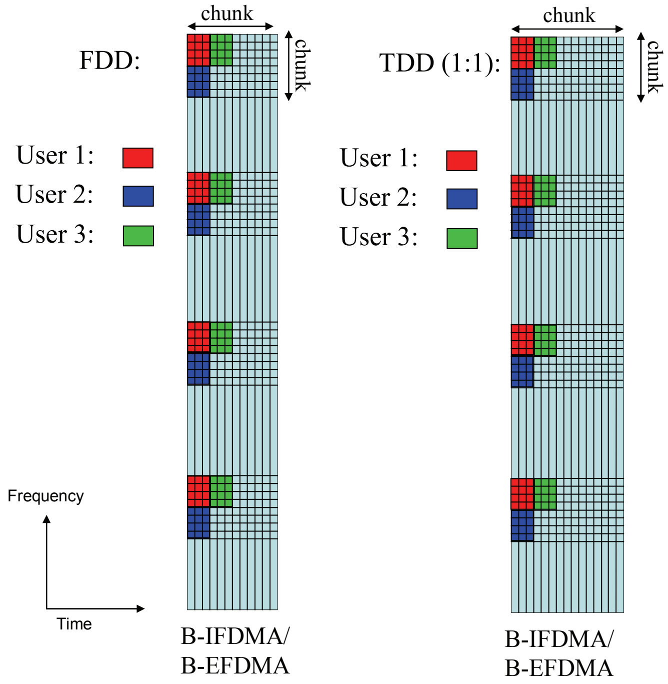

Figure 3.7: Illustration of B-IFDMA and B-EFDMA resource allocation in FDD and TDD.

These schemes aim to maximise frequency diversity, to enable micro-sleep within chunks, keep low addressing overhead and to simultaneously enable low envelope variations of transmitted uplink signals. The similarity of the uplinks and downlinks further simplify the system.

Baseline assumption for parameterisation of both B-IFDMA and B-EFDMA is: the basic block size is 4 subcarriers x 3 OFDM symbols in both FDD and TDD. This block size fits the chunk size in both FDD and the updated TDD chunk size defined in Section 3.1. One common link adaptation is used for all the allocated resources in a chunk layer for the duration of the chunk, and the CQI is based on moving average SINR, averaged over the small-scale (fast) fading of the channel.

For simulations taking resource allocation overhead into account, assumption on basic resource allocation unit is needed. To guarantee full frequency diversity, every 4th chunk in the frequency direction is used, i.e. 1.25 MHz separation of blocks. To maintain reasonable overhead, 16 blocks with equidistant frequency separation are assumed to form one basic resource allocation unit. This corresponds to the size of two physical chunks and spans 20 MHz channel bandwidth. For studies on control signalling, smaller resource allocation units could be assumed.

3.2.3 Resource Scheduling

In the baseline assumptions, simple resource scheduling algorithms are used. For frequency-adaptive transmission a proportional-fair scheduling strategy shall be used, such as the score-based scheduler outlined in [Bon04]. A basic Round Robin scheduler is utilised for non-frequency adaptive transmissions. In both cases a minimum delay of one frame between arrival of a packet in the buffer and its transmission is assumed. Also the CSI/CQI information used for the scheduling decision shall be outdated by a minimum of 1 frame. The resource scheduling shall also prioritise retransmissions, i.e. new initial transmission can only be scheduled when no retransmission is pending.

3.2.4 Spatial Processing

In the baseline system design, it is proposed to implement simple spatial processing schemes that still can capture most of the gain. For the downlink in the base coverage urban scenario this means a fixed Grid of Beams (GoB), which only requires a limited amount of feedback in order to select beam. Also for the microcellular scenario it is suggested to implement GoB, while in the indoor scenario single-user MIMO based on Singular Value Decomposition (SVD) is proposed. The suggested schemes for microcellular and indoor scenarios differ from the ones proposed in [WIN2D6131] which were SMMSE MU MIMO precoding and RBD, respectively. The reason for this is the required simulator implementation effort and complexity of the latter schemes, wherefore somewhat simpler schemes are proposed now in order to ease the simulator implementation.

For the uplink it is still proposed to use an Orthogonal Space Time Block Code (OSTBC), e.g. Alamouti's orthogonal design [Ala98].

In order to keep cost and size of relay antennas low, only a single antenna element is used at relay nodes. Consequently no spatial processing is applied in the baseline assumptions for relay nodes.

3.2.5 Coding, Modulation, and Link Adaptation

In the baseline design a rate-compatible punctured block low-density parity check code (BLDPCC) of mother code rate 1/2 is used for the transmission of information data. Code rates of 2/3 and 3/4 are obtained by puncturing, and combined with different modulation alphabets (BPSK, QPSK, 16-QAM, and 64-QAM). When plotting the throughput versus SINR for these combinations of modulation and code rate, some of the combinations become obsolete, since they don't contribute to the hull curve. Thus a baseline modulation and coding scheme (MCS) for adaptive coding and modulation (ACM) consists of the following combinations:

Table 3.6: Baseline modulation and coding scheme for adaptive modulation and coding

MCS	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**
Mod.	BPSK		QPSK			16-QAM			64-QAM	
R	1/2	2/3	1/2	2/3	3/4	1/2	2/3	3/4	2/3	3/4

The corresponding hull curves for the B-LDPCC (FEC block size of 2304 bits) and using 10% BLER as switching criterion is shown for FDD and TDD in Figure 3.8 and Figure 3.9, respectively. This is based on the initial transmissions, i.e. HARQ retransmissions are not included. Overhead includes in-chunk pilots and control symbols, as well as the super-frame pre-amble

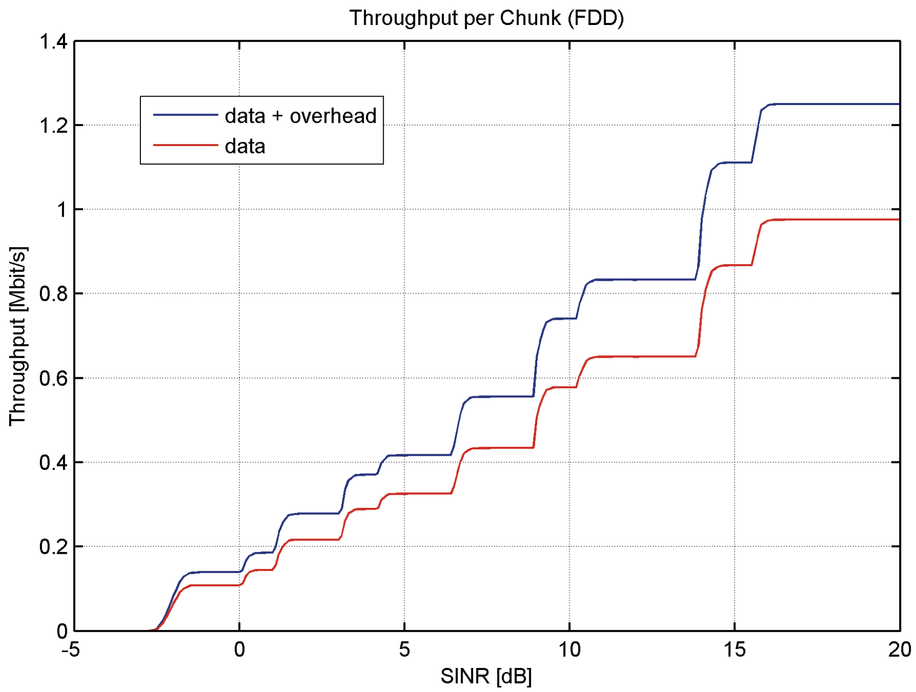

Figure 3.8: Throughput per chunk versus SINR for the baseline MCS (FDD PLM, frequency-adaptive transmisison).

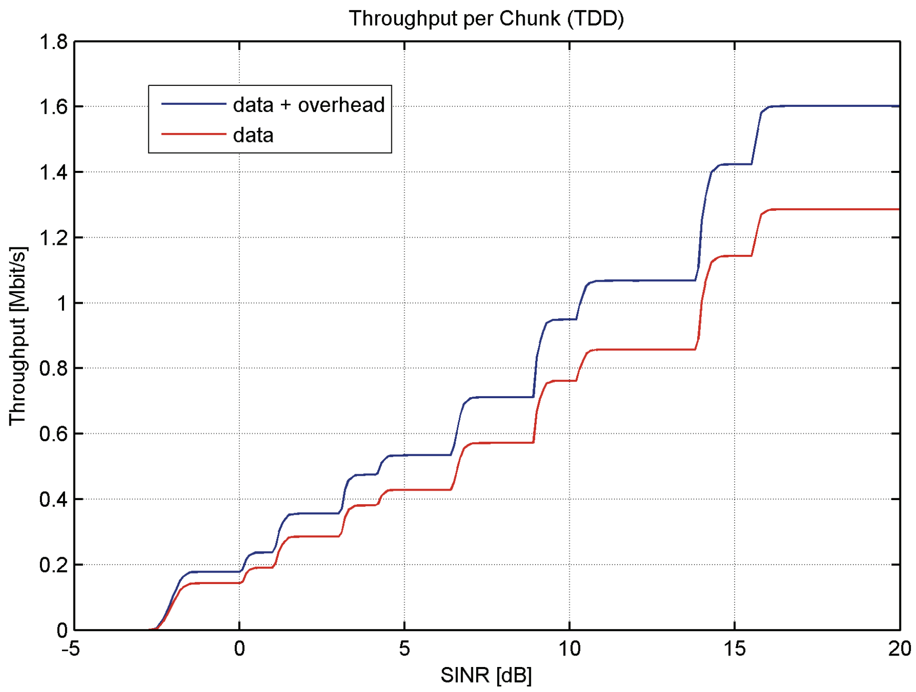

Figure 3.9: Throughput per chunk versus SINR for the baseline MCS (TDD PLM, frequency-adaptive transmission).

The baseline MCS as defined above is pragmatically based on the currently available link-level curves for the B-LDPCC. Future updates should be based on equidistant separation of the MCS along the SINR axis in the AWGN channel (e.g. for the 10% BLER point). Also higher MCS (e.g. 64-QAM, $R = 1$, or 256-QAM) might be required for certain high-quality links, especially in the indoor test scenario.

Convolutional codes are considered as baseline coding scheme and for FEC blocks containing 200 information bits or less, e.g. control signalling. Two memory eight codes [Fre98], one with mother rate *R=1/2* and one with mother-rate *R=1/3*, are presented together with puncturing patterns in Table 3.7 and Table 3.8. For very short word lengths, *circular- or tail-biting encoding* is necessary to avoid the rate loss of termination. The initial state for the tail-biting encoder is given by the last bits of the information block.

Table 3.7: Rate-compatible punctured convolutional code with mother-code rate R=1/2

generator polynom	rate	puncturing pattern
G_1=561 G_2=753	R=1/2	$\begin{bmatrix}11111111\\11111111\end{bmatrix}$
	R=2/3	$\begin{bmatrix}11011111\\10110110\end{bmatrix}$
	R=4/5	$\begin{bmatrix}11011111\\00110100\end{bmatrix}$
	R=8/9	$\begin{bmatrix}11011111\\00100100\end{bmatrix}$

Table 3.8: Rate-compatible punctured convolutional code with mother-code rate R=1/3

generator polynom	rate	puncturing pattern
G_1=575 G_2=623 G_3=727	R=1/3	$\begin{bmatrix} 11111111 \\ 11111111 \\ 11111111 \end{bmatrix}$
	R=1/2	$\begin{bmatrix} 11111111 \\ 10001111 \\ 01110000 \end{bmatrix}$
	R=2/3	$\begin{bmatrix} 11011101 \\ 10000111 \\ 00110000 \end{bmatrix}$
	R=4/5	$\begin{bmatrix} 11011001 \\ 10000110 \\ 00110000 \end{bmatrix}$
	R=8/9	$\begin{bmatrix} 11001001 \\ 10000110 \\ 00110000 \end{bmatrix}$

3.2.6 User plane packet processing

The proposed reference design for frequency-adaptive and non-frequency adaptive transmission is described in [WIN2D461]. In the following, a simplified baseline design is characterised, which aims at easing the implementation in simulators while at the same time maintaining the core features of the reference design. In order to reduce implementation effort, unified baseline assumptions apply for both types of transmission unless otherwise indicated.

Simplifying deviations from the reference design include in particular the constraint that *exactly one* RTU is sent per user and slot and the restriction of the modulation and coding / puncturing to a small set of combinations, denoted as baseline modulation and coding scheme (MCS) as described in Section 3.2.5 above. The following steps describe the baseline assumptions for user plane packet processing:

- RLC PDUs arrive to the MAC layer. These packets represent the retransmission units (RTUs) They are segmented into segments of only a few possible sizes. In the baseline design, we use two specific sizes.
    - TCP ACK packets of 40 bytes = 320 bits are appended with a MAC header into **one segment** of K = 360 bits.
    - Larger packets are segmented into segments of K = 1200 bits (including the MAC header.

  We thus have segments of two possible sizes: **360 bits and 1200 bits.** In the implementation example, packets that have other sizes are partitioned into segments of the above sizes, using zero-padding when necessary.

- Each segment forms a FEC block and is encoded with the BLDPC code described in Section 3.2.5.
- For frequency-adaptive transmission:
    - Based on the frequency-adaptive scheduler's decision appropriate chunk layers are assigned to each user.

  - The baseline link adaptation is based on **adaptive modulation per chunk** and an **average code rate per FEC block**, using the baseline MCS. The average code rate is obtained by puncturing of the BLDPC code and mapped to the code rates of the baseline MCS.

- For non-frequency adaptive transmission:

  - Based on the non-frequency adaptive scheduler's decisions appropriate blocks spread over frequency are assigned to each FEC block

  - Based on the average SINR of all chunks belonging to an FEC block the modulation and code rate is selected from the baseline MCS scheme. The average is calculated by the link-to-system interface methodology [BAS+05].

- Depending on the overall capacity of the allocation of one user per slot, an integer number of FEC blocks of each flow are drained from the buffer and mapped onto the slot. For each flow, they comprise part of an RTU or at most one complete RTU. Note that the large segment sizes and the coarse MCS might result in significant padding loss. Further investigations are required to address these problems. Details on any modified assumptions shall be specified along with the simulations.

- No chunk-specific power control is used, i.e. the available transmit power is evenly distributed across all used chunk layers, observing the total power constraint per antenna.

- Downlink control signalling is not explicitly modelled in baseline investigations (unless it is particular target of the investigation), however control overhead (Section 3.1) and delay (Section 3.2.8) is taken into account.

Particular implications of these necessary simplifications include a potential high padding loss, due to large segment sizes and the few combinations in modulation and coding in the baseline MCS. Furthermore it restricts the scheduler and degradations in multi-user scheduling gain can occur depending on the overall configuration.

The downlink control information used for controlling downlinks as well as uplinks has to be transmitted with adequately low error rate at a reasonable coding overhead. The use of frequency-adaptive transmission is in [WIN2D461] suggested to be limited to SINRs above 5 dB for this reason. This SINR-limit is near those reported from [WIN2D341] for the use of spatial multiplexing schemes in preference to spatial diversity schemes. This is a significant insight from the investigations in WINNER II: Frequency adaptive transmission and compatible multi-antenna transmit schemes are useful in roughly the same SINR regions. Earlier investigations (see Section 3.1 of [WIN1D24] and Appendix B.5.2 of [WIN1D210]) have shown that CSI prediction errors put an upper limit on the use of adaptive transmission at vehicular velocities, which depends on the prediction filter used, the required accuracy, the carrier frequency, and the actual SINR. At SINRs above the switching point of $SINR = 5$ dB, this velocity is in the order of 50 km/h [WIN1D210, SFS05].

3.2.7 Basic Timing Assumptions

The air-interface packet delay is defined as the time that elapses between the arrival of a packet in the transmission buffer until correct reception of the corresponding packet at the receiver, see also Section 4.4. The transmission occurs no earlier than the frame following arrival in the transmission buffer. Correct reception at the receiver is finalised one slot before the ACK/NACK feedback is sent.

The timing of the transmission control loops for frequency adaptive and non-frequency adaptive transmission have been outlined in Sections 3.1, 3.2, 4.1 and 4.2 of [WIN2D461]. The target is to attain a very short delay over the air interface. In the transmission control systems, a scheduling computation delay of max. 0.1 ms has been assumed. The computation delay for channel quality or state prediction is likewise assumed to be max. 0.1 ms.

Regarding the delay of ACK/NACK for (link) retransmission, we have to add the delay of decoding. The results of Table B-2 of Appendix B in [WIN1D210] show that a delay of below one clock cycle per decoded bit is attainable, with appropriate parallel implementations of LDPC and DBTC decoders. For example, the assumed FEC block sizes of max. 1200 bits, this corresponds to less than 6 μs when 200 MHz of the total clock cycles are allocated to decoding. We below allow the total *receiver processing* *delay to use up to one slot (345.6 μs).* Decoding of FEC blocks of size up to 1520 bytes should require less than 60 μs, so this provides ample time also for iterative turbo decoding /channel estimation.

Let "UL *i*" and "DL *i*" denote the uplink/downlink slots of frame number *i*, respectively. The timing of the transmission over one hop, as outlined in Sections 3.1, 3.2, 4.1 and 4.2 of [WIN2D461] is summarised with this notation by Table 3.9. In the tables below, it is assumed that in half-duplex FDD as well as TDD, a DL slot precedes an UL slot within the frame. The attainable delays involved in a retransmission (Section 3.2.9) are summarised by Table 3.10 below.

Table 3.9: Transmission and decoding delays, summary from [WIN2D461]

	Transmit request	Predict. update Scheduling, DL control	Trans-mission	Decoding of RTU	1-hop delay incl. decoding (frames)	1-hop delay (ms)
Frequency-adaptive **uplink**	UL *j-2*	UL, DL *j-1,* DL *j*	UL *j*	DL *j+1*	3.0	2.1
Non-frequency adaptive **uplink**	UL *j-1*	DL *j*	UL *j*	DL *j+1*	2.0	1.4
Frequency-adaptive **downlink**		DL *j-1*, DL *j*	DL *j*	UL *j*	2.0	1.4
Non-frequency adaptive **downlink**		*j-1*, DL *j*	DL *j*	UL *j*	2.0	1.4

Table 3.10: Retransmission delays

	FEC block received	ACK/NACK transmission	Retransmission	Retransmission delay (frames)	Retransmission delay (ms)
Downlink	DL *j*	UL *j+1*	DL *j+2*	2.0	1.4
Uplink	UL *j*	DL *j+2*	UL *j+2*	2.0	1.4

The following examples illustrate the range of attainable delays over the air interface.

Example 1. Consider a downlink transmission over two hops (BS-RN-UT) in TDD. The BS-RN relay link transmission is initiated during slot DL *j-1* and executed in slot DL *j*. Decoding is finalised during slot UL *j*. Simultaneously with the decoding, the forwarding/scheduling is prepared for the next hop over the RN-UT link. An ACK/NACK is transmitted over the reverse RN-BS relay link during slot UL *i+1*. Transmission over the RN-BS link is performed during slot DL *i+1*, with decoding completed at the UT during slot UL *i+1*. (We here assume that the forwarding and queuing at the MAC layer within a RN does not induce any extra delay.) If no retransmission is required, the transmission over two hops thus requires in total 6 slots or 3 frames, with total delay 2.1 ms. (In the FDD mode with relay nodes, the timing becomes somewhat different, see Figure 3.6.)

Example 2: A one-hop downlink transmission is performed for a 576 byte TCP-IP packet that is segmented into four RTUs, which each comprises one FEC block of 1200 bits. Let's assume that the TCP-IP packet is received in frame *k*. In the baseline design, at most one RTU can be transmitted per frame, so the blocks are transmitted over subsequent frames *k+1, k+2, k+3, k+4*. Error-free transmission would be completed in frame *k+4*, with the last ACK transmitted in frame *k+5*. Under the worst-case assumptions that the last FEC block is in error, the retransmission starts in frame *k*+6 and the associated ACK is received in frame *k*+7.

If the TCP-IP packet comprises one single RTU block that is segmented into four FEC blocks of 1200 bits, then all four FEC blocks (that belong to the same RTU) can be transmitted within the same frame, if enough transmission resources are available. This would reduce delay for error-free transmission (transmission+decoding) to 2 frames, or 1.4 ms.

3.2.8 HARQ

As baseline assumption a *N*-Channel Stop-and-Wait (SAW) protocol is used. For both, up- and downlink a fixed relationship between the transmission of a packet and the associated ACK/NACK feedback is assumed with respect to time and used physical resources.

In order to align with state-of-the-art in other systems, the timing of the retransmission is the following:

- in downlink the retransmission can happen at any time exceeding or equal to the retransmission delay of τ_{rtr},
- in uplink, the retransmission can happen at any time $k \cdot \tau_{rtr}$, where k is an integer number greater or equal to 1.

Similar assumptions were taken in legacy systems, such as HSPA, LTE [TR25.858, TR25.896, TR25.814] as they provide a reasonable trade-off between flexibility and control overhead.

The retransmission delay τ_{rtr} is decomposed in the following way:

- the estimated decoding delay of the transmission allows still to send the ACK/NACK feedback in the subsequent frame,
- the ACK/NACK feedback can be decoded in the same slot and the retransmission can be scheduled in the following slot.

As a result the retransmission delay is $\tau_{rtr} = 2$ frames, see Table 3.10. For baseline investigations, error-free ACK/NACK feedback is assumed and the feedback needs not be modeled explicitly.

For each physical layer mode, the number of SAW channels *N* should allow continuous scheduling of one user in time. Since the baseline assumption is to transmit only 1 retransmission unit (RTU) per user and slot, there is no need for multiple SAW channels *per slot*.

Chase Combining is used, i.e. any retransmission of a packet will contain exactly the same coded bits as the initial transmission. A RTU can comprise one or several FEC blocks (Section 3.2.6). Retransmissions are given highest priority. Transmission of a new RTU of a flow is not allowed in a slot where a retransmission of the same flow takes place. No restriction related to the physical resources used in retransmissions need to be taken. However, the following simplifying assumptions may be taken. As the baseline assumption is transmission of *complete* FEC blocks, the allocated resources at time of retransmission must accommodate at least one complete FEC block of the RTU. This assumption obviously poses some constraints on the scheduler and it might lead to a reduced efficiency.

Depending on the actual investigations and the simulator class, further simplifications might apply and the rationale for doing so needs to be explained.

For example, an initial guess on performance without explicit modeling of HARQ can be obtained for low average BLER ($BLER_{av}$) by assuming that successful transmission would have happened after one retransmission. In this case, the average throughput T_{av} can be approximated by:

$$T_{av} = T_{MCS} \cdot (1 - BLER_{av}) + \frac{T_{MCS}}{2} \cdot BLER_{av} \ , \tag{3.1}$$

where T_{MCS} is the maximum throughput obtained by the applied modulation and coding scheme. Under the same assumptions, the average delay can be approximated by:

$$\tau_{av} = 2 \text{ frames} (1 - BLER_{av}) + 4 \text{ frames} \cdot BLER_{av} \ , \tag{3.2}$$

Note, that such simplifications are not applicable for high initial BLER, caused either on purpose since the scheduler wants to use time diversity due to HARQ or by imperfections due to measurement and feedback errors, or delay.

3.3 Modelling of Imperfections

In the baseline design, calibration for spatial processing and measurements of signal, interference and noise shall be assumed ideal. Also channel estimation (CE) is assumed ideal for first investigations. Optionally, channel estimation errors can be considered by a simple but accurate error model which has been developed within WINNER [WIN1D23, AuC07]. This model allows expressing the channel estimation error as SINR degradation or as an additional noise term according to the signal model shown in Figure 3.10.

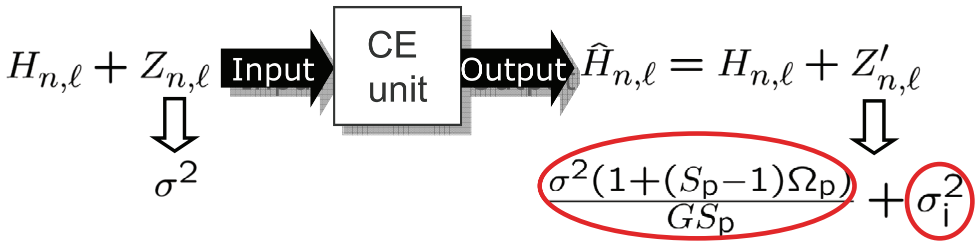

Figure 3.10 Signal model for channel estimation error modelling

The performance of an estimator can be approximated by modelling the channel estimation error ε_l as a modified Gaussian noise, where

- the noise variance of the input process σ^2 is reduced by the estimator gain G and by a factor $S_p/(1+(S_p-1)\Omega_p)$ which depends on the pilot boost S_p and the pilot overhead Ω_p, and
- an additional residual interpolation error σ_i^2 is introduced.

The estimator gain G is defined as:

$$G = 1/(\mathbf{w}^H \mathbf{w})$$

where the $M_f M_t \times 1$ column vector $\mathbf{w}$ represents an arbitrary linear estimator, e.g., FIR filter, with M_f and M_t denoting the number of pilots in frequency and time, respectively. Furthermore, the pilot boost S_p is the factor obtained by dividing the pilot power by the data power, whereas the pilot overhead Ω_p is the factor obtained by dividing the number of pilot symbols per chunk with the total number of symbols (data plus pilots) per chunk.

The *SINR* degradation $\Delta\gamma$ can thus be described by [AuC07]:

$$\Delta\gamma = (1+(S_p-1)\Omega_p)\left(1+\frac{1}{GS_p}\right)+\sigma_i^2\gamma\,. \tag{3.3}$$

As baseline assumptions for techniques operating on common pilots, the following assumptions can be made:

- Ω_p baseline assumptions are discussed in Section 3.1
- G = 3 dB for pilot-aided channel estimation (PACE)[9],
- σ_i^2 = - 40 dB, and
- S_p = 0 dB, i.e. no pilot boost.

For spatial processing this Gaussian error model can be regarded as worst-case due to possible correlations in the error processes. As a rule of thumb, between 1 dB and 2 dB degradation in *SINR* due to channel estimation errors typically result for techniques based on common pilots. For those techniques that rely on dedicated pilots interpolation errors become dominant and detailed investigations are required.

Ideal synchronisation is the baseline assumption. For dedicated investigations constant offset in time and frequency might be used or an additional noise power due to synchronisation errors might be used. The parameters for these error models are for further study.

A summary on modelling of transmitter and receiver imperfection is given in Table 3.11.

[9] In some previous investigations estimator gains of 13 dB have been used, which are obtainable using iterative channel estimations (ICE) in a regime with negligible decision feedback error, such as high SINR and low UT velocity. This might be a realistic assumptions for spatial processing techniques used in the indoor test scenario.

Table 3.11: Modelling of potential Tx/Rx imperfections

PHY Entity	**Base Coverage Urban**	**Microcellular**	**Indoor**
Calibration for spatial processing	Ideal		
Channel estimation accuracy	Ideal first, optionally the Channel Estimation Error model described above can be used		
Link Synchronisation accuracy	Ideal first, Insert constant offsets [(µs)in time and frequency (ppm]] or Additional variable noise power N'= r*N, r~G(0,σ), G=Gaussian distribution or r constant		
Measurement of Signal Power S	Ideal		
Measurement of Interference I	Ideal		
Measurement of Noise N	Ideal		

4. Assessment Criteria

This section presents a set of criteria that are to be used within WINNER for the purpose of system performance assessment. The descriptions in this section are focused on the refinement of the corresponding descriptions in [WIN2D6131] as well as the technical interpretations of the relevant requirements in [WIN2D6111].

The system performance, expressed in terms of the assessment criteria described in this section, may be sensitive to the overall design and assumptions related to environment and deployment. For instance, the performance of a MIMO scheme may depend on the resource management/scheduling strategy used. Accordingly, when analysing the performance of a specific strategy, the other assumptions should be well described and the impact of changing these other strategies/assumptions should be investigated as far as possible.

Future modifications of the design assumptions might affect the system performance in many areas. However, it can not be predicted easily how much such a modification will affect the system even if only a minor change is performed, e.g., a change of a few system parameters, change/replacement of some algorithms/functional modules, and so on. Thus, in case any modification/change is going to be proposed, the motivation needs to be justified with a proof that all the performance requirements are going to be still met and/or the performance can be even improved.

4.1 Bit error rate

The bit error rate (BER) is one of the most commonly used measures in the evaluation of digital communication systems. It characterises the robustness of a communication system to the noise and/or interference which the system might encounter while transmitting and receiving information data.

In case the performance of the modulation, channel estimation, and synchronisation schemes is concerned, the BER is to be measured as raw performance at the output of a demodulator. In order to include the performance of the channel coding scheme, it is to be measured at the output of a channel decoder.

The BER may concern the Class IV simulator only. As it may not be sufficient enough in characterizing the performance of a packet data system, it is often preferred to consider the frame error rate (FER) which is to be described subsequently.

4.2 Frame error rate

Although the frame error rate (FER) generally refers to the ratio of the number of correctly received packets to that of total transmitted ones, it can be subdivided according to at what specific point it is to be measured on the receiver chain:

- Codeword Error Rate (CWER) to be measured at the output of a channel decoder.
- Block Error Rate (BLER) also to be measured at the output of a channel decoder. In case a block is composed of a single codeword, it will be the same as the CWER. In case of HARQ, a residual BLER is to be looked after, for which an error will be checked after the decoding of all the retransmissions of the same packet.
- Frame Error Rate (FER) in which a frame represents the information block protected by cyclic redundancy check (CRC) in the RLC layer. In the WINNER design, the FER is equivalent to the CWER and the BLER as well.
- IP Packet Error Rate (IPER) to characterise the error probability of an entire IP packet.

In the sense that both throughput and data rate are to be measured while taking PHY and MAC overhead into account, the FER and IPER seem more adequate than the CWER and BLER.

4.3 User throughput

The average user throughput is defined on a link for a given user as the ratio of correctly received information bits to the simulation run time that would have elapsed in a real system. Both PHY and MAC overhead needs to be taken into account while the user throughput is being measured, for which all the information on those overheads is given in Section 3.1 of this document.

In case of a packet call, the user packet call throughput or active session throughput is to be used, where the simulation run time for each packet call is defined as the duration from when the first packet of a call enters the transmission queue to when the last packet of a call is received successfully.

In order to investigate the efficiency and fairness of a scheduling algorithm, the cumulative distribution function (CDF) of user throughputs normalised by the average user throughput is to be checked at those percentile points specified in [WIN1D72]. The 95%-ile plays an important role also in the definition of the satisfied user criterion for spectral efficiency calculation, see Section 4.7.

4.4 Delay

The end-to-end delay means the delay from the data source somewhere in the network, e.g., WWW server, to the sink, e.g., WWW browser, which IP packets experience while they are transmitted over the system. Among various contributors to the end-to-end delay, the air interface will be the main concern from the WINNER perspective. Focusing on the contribution of the air interface only, the delay shall be measured from the MAC of a transmitter side to that of a receiver side, which includes processing delay, scheduling delay, retransmission delay, relaying delay, propagation delay, and so on. However, it does not include the packet flow establishment time which will be explained later in this section.

As each source of delay will contribute a different fraction to the end-to-end delay, each of them needs to be modelled separately and accordingly. That is, the minimum scheduling delay of one frame mentioned in [WIN2D6131], for example, refers to each execution time of scheduling, and the maximum number of retransmissions in relation to HARQ should also be mentioned with the consideration of the quality of service (QoS) requirement for each service.

The measurement of the user throughput, data rate, and system capacity need to be done while the delay constraint being met. End-to-end delay criteria are specified in Table 3.2 of [WIN2D6112] and the CDF of packet delay and/or packet call delay is to be checked at a 95 percentile point, where the packet delay and the packet call delay are defined as follows:

- Packet delay is the time interval from when the packet enters the transmission queue to when the packet is received successfully. If a packet is not successfully delivered by the end of a simulation run, none of the information bits of the packet shall be counted.
- Packet call delay is the time interval from when the first packet of a packet call enters the transmission queue to when the last packet of the packet call is received successfully. If a packet call is not successfully delivered by the end of a simulation run, the packet call shall not be counted in the performance statistics.

4.5 Data rates

Both PHY and MAC overhead needs to be taken into account while the data rate is being measured, for which all the information on those overheads is given in Table 3.2 in Section 3.1 of this document.

As for the support of average session data rate of up to 50 Mbps in [WIN2D6111], the averaging with respect to the user population shall include only those terminals that can support 50 Mbps or higher and for which the services generate a sufficient traffic load. Assessment assumption shall include practical amount of interference from neighbouring cells and might assume a single user in the serving cell excluding disadvantageous area within a cell such as the cell edge. Here, the cell edge is defined as the point at which the CDF of normalised user throughputs is 95 percentile.

The consistent and ubiquitous data rate of 5 Mbps [WIN2D6111] shall also be met only in case a terminal can support 5 Mbps or higher and for which the services generate a sufficient traffic load.

4.6 Cell throughput

The cell throughput is defined as the ratio of the aggregate number of correctly received information bits in a cell to the total simulation run time that would have elapsed in a real system. It is equivalent to the sum of user throughputs in a cell.

In relation to macro diversity, in case a packet has been successfully received and has been transmitted over n links, each of the links may be attributed $1/n$ of the total transmitted information bits of the packet. However, this will be definitely dependent on how the macro diversity is to be supported specifically in connection with the air resource allocation and usage. The point here is that the same packet should not be counted multiple times when a terminal receives the same packet from multiple base stations.

4.7 Spectral efficiency

Based on the definition in [WIN2D6111], the spectral efficiency needs to be measured while the satisfied-user criterion is being met, i.e., an average active session throughput of 2 Mbps or higher needs to be guaranteed for 95 percentile of users in the downlink, and an average active session throughput of 1.3Mbps or higher needs to be guaranteed for 95 percentile of users in the uplink. Thus, it can also give a rough estimate of system capacity in terms of the number of users who can be served with an average active session throughput of 2Mbps or higher.

In order to reduce the scenarios under which the spectral efficiency is going to be measured, it might be measured for two extreme cases, e.g., all outdoor users and all indoor users for the base coverage urban and microcellular test case (refer to Section 2.3 of this document) and then a simple interpolation of those two figures might give us an estimate for a scenario in which there exist both indoor and outdoor users simultaneously.

According to the requirements in [WIN2D6111], the following spectral efficiencies are expected to be obtained in the different assessment scenarios:

- 2-3 bps/Hz/site for the downlink and 2/3 thereof for the uplink in wide area deployments, e.g. the base coverage urban test scenario in an operation point that meets the satisfied-user criterion,
- 2-5 bps/Hz/site for the downlink and 2/3 thereof for the uplink in metropolitan deployments, e.g. the microcellular test case in an operation point that meets the satisfied-user criterion,
- 10 bps/Hz/site for the downlink and 2/3 thereof for the uplink in isolated sites, e.g., the indoor test case, in an operation point that meets the satisfied-user criterion.

4.8 Packet Flow Establishment Time

Each packet stream is denoted as a flow and is identified and transmitted individually according to its QoS requirements. For example, a real time service and a file transfer service may be mapped to different flows, thus allowing the scheduler/service level controller to grant a privilege of staying within the delay constraint to the real time service. Therefore, each flow requires an individual ID and separate queue.

When the first packet of a new flow arrives at the upper layer of an air interface system, the flow parameter will be generated and the flow context will be established at a BS, a user terminal, and each relay node between them. The time from the arrival of the first packet of a new flow until the flow is established in the system and then the packet is received and passed to the layer above the air interface system on the peer side is defined as packet flow establishment time. This includes the transition of the user terminal from the idle state to the active state in case no other flow is transmitted. The packet flow establishment time may concern the class I simulator mainly.

4.9 Radio resource management related criteria

When assessing the overall performance of a system, the end-user perception of the service plays a key role. Apart from data rate and user throughput, the session availability and continuity should also be assessed as a part of the global assessment. Both are driven by the radio resource management algorithms such as call admission control, load control, and handover management, and related criteria are defined as follows:

Session rejection rate is the ratio of the refused new sessions to the total number of incoming sessions, where the refusal means that the first packet of a new session has not been completely transmitted within 30 ms of its arrival at the transmission queue, for example, in case of a packet voice according to Section 3.1.2 of [WIN1D14].

Session drop rate is the ratio of the number of sessions that must be dropped for any reason to the total number of ongoing sessions, where the drop will occur if the packet delay and/or packet call delay criteria of 100 ms can not be met, for example, in case of a simple telephony according to Table 3.2 of [WIN2D6112].

4.10 Complexity, costs, power consumption

Assessment criteria of the technology choices shall include complexity, costs, power consumption, reliability, form factor criteria in order to be able to propose trade-offs when engineering the system.

Typical trade-off could be the performance gain in terms of system capacity, for example, vs. the complexity of algorithms.

The cost is a multi-dimensional criterion as it needs to cover both CAPEX and OPEX. From the perspective of CAPEX analysis, the base station density to provide a given coverage with a given quality in a given area shall be assessed. In addition, the impact of introducing relay stations shall be assessed and the backhaul capacity required for the base station shall be assessed as well. From the perspective of OPEX analysis, aspects like computational complexity of the algorithms and power consumption shall be assessed. These cost aspects have already been considered in the definition process of the test scenarios, e.g., in the basic parameterisation of relay nodes, and in the algorithm proposals for the baseline and the reference design.

4.11 Performance Metrics

The assessment results investigated through each class of simulator shall be presented in terms of the following performance metrics, but not necessarily limited to the metrics listed in Table 4.1.

Table 4.1: Performance Metric of different simulator classes

	Performance Metrics
Class I Protocol Level Simulator	1. Investigation results with respect to protocol-related aspects, e.g., packet flow establishment time
Class II (Dynamic) System Level Simulator	1. Session rejection rate vs. the number of incoming sessions 2. Session drop rate vs. the number of ongoing sessions 3. Investigation results with respect to handover-related aspects
Class III (Quasi-Static) System Level Simulator	1. 1 to 2 from the Class IV Link Level Simulation 2. A scattering plot of the link SINR vs. the distance from the serving cell/sector to users' locations, which is observed in one of the sectors of the centre cell or just in the centre cell if no sectorisation has been introduced 3. A pdf of the link SINR observed in one of the sectors of the centre cell or just in the centre cell if no sectorisation has been introduced, where the SINR in 2 and 3 includes path loss, shadowing, and sectorisation 4. Cell throughput and spectral efficiency vs. the number of users 5. Investigation results with respect to the support of average session data rate of up to 50Mbps, and the support of a consistent and ubiquitous data rate of 5Mbps 6. Average user throughput (or average user packet call throughput) along with a scattering plot and a histogram of user throughput (or user packet call throughput) vs. the distance 7. Average packet delay per sector/cell along with a scattering plot and a histogram of user packet delay vs. the distance 8. Average packet call delay per sector/cell along with a scattering plot and a histogram of average user packet call delay vs. the distance, where the distance mentioned in 6 through 8 refers to the distance from the serving cell/sector to users' locations 9. A scattering plot of user throughput vs. average packet delay 10. A scattering plot of user packet call throughput vs. average packet call delay
Class IV Link Level Simulator	1. All link level results for both traffic and control channels, which are often presented in terms of error performance (e.g., BER, CWER, BLER, FER, IPER) vs. SINR 2. The performance of any estimator/predictor implemented in a simulator along with its details, e.g., channel estimator, synchronisation unit, etc.

5. Conclusion

This deliverable is the successor of [WIN2D6131] and contains the latest updates of the simulation assumption for the three WINNER II test scenarios, of the associated baseline design assumptions, and of the assessment criteria definition. Important updates have occurred with respect to:

- the basic OFDM parameters and dimensioning: (increased guard interval and changed chunk dimensions in the TDD mode),
- the baseline coding scheme (use of the block low density parity check code as baseline coding scheme),
- the multiple access scheme for non-frequency adaptive transmissions (introduction of B-IFDMA and B-EFDMA), and
- the parametrisation of the channel estimation error model.

Also, new aspects are included and more details are provided, in particular with respect to

- relaying (definition of basic deployment scenarios and parameters, basic resource partitioning and timing),
- segmentation (definition of RTU and FEC block sizes),
- link adaptation (new baseline modulation and coding scheme, adaptive coding and modulation algorithms), and
- basic timing of control loops (definition of processing and protocol delays)
- HARQ (detailed specification of assumptions related to protocol and timing).

This report contains now a single source of information for mandatory assumptions used for performance evaluation in WINNER during 2007. Results from many expert discussions throughout the project have been consolidated in order to obtain parameters, assumptions, and algorithms that represent existing or near-term achievable simulator capabilities and at the same time allow meaningful evaluations of the major questions in WINNER system concept and design.

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Appendix A. Traffic Models

In this appendix traffic models in the scope of WINNER are presented which are important for the expected traffic in WINNER deployments and at the same time are already investigated to an extent where detailed traffic models can be derived.

The most relevant statistical parameters, like session arrival rate, session duration, packet call size, time between packets calls, etc., that maybe used as inputs for simulators, will also be presented here, for each traffic model.

Apart from full queue traffic modelling, HTTP traffic shall be investigated first, followed by (highly) interactive traffic classes, such as VoIP and gaming.

A.1 Internet applications

Internet and multimedia traffic can be characterised by frequent transitions between ON and OFF periods, active and inactive states. The ON period corresponds to the file-downloading period and the OFF period corresponds to the user reading time.

In a circuit-switched network, the dedicated bandwidth is wasted during the OFF period. However, the packet-switched technology allows higher data transmission rates and uses the bandwidth only during ON periods.

A.1.1 Web browsing

The number of web services and the amount of information that can be found in the web is constantly growing. This happens for many reasons: first, because web is almost suitable for any kind of service or application which is based on text and graphics; second, HTTP is adequate to transfer types and file sizes; third, and maybe the most important, web has become a kind of universal interface: the simple and user friendly “look & feel” has contributed to the spreading of the relevant services. Within this context, it is important to understand how web traffic is composed.

The term web traffic comprises all HTTP traffic generated during a session with a typical web browser. However, the way that a web page is downloaded differs from browser to browser and between different HTTP versions.

In order to understand the different methods for data transmissions, the basic method and the web page structure must be explained. A typical web page consists of ASCII text i.e. the HTML code. This part of the page is referred as main object. In HTML code, images or other objects, like Java scripts may be embedded with a reference to an external file, which can be or not in the same server, (see Figure A.1).

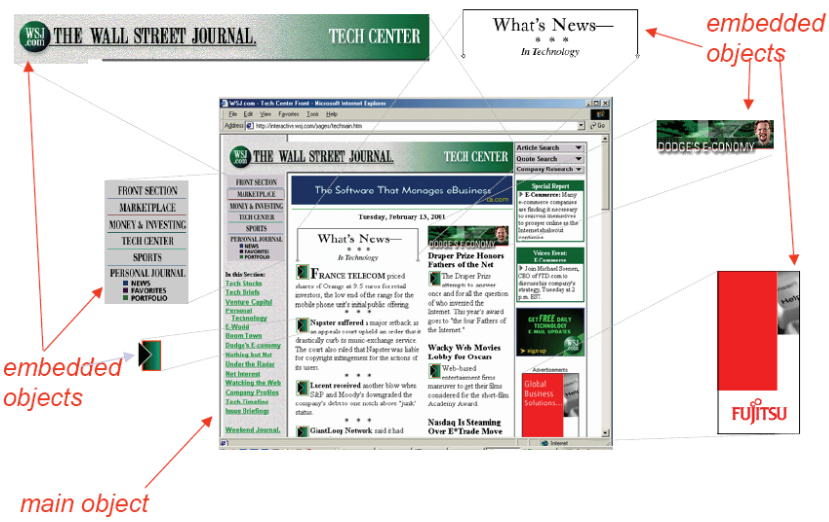

Figure A.1: Typical structure of a WWW page.

Web browsing is the most dominant application for broadband system and it has been extensively investigated. In Figure A.2 a typical web browsing session is presented: the ON and OFF period periods are the result of human interaction, where the packet call represents the user's request for information and the reading-time is the time that user needs to digest the web page.

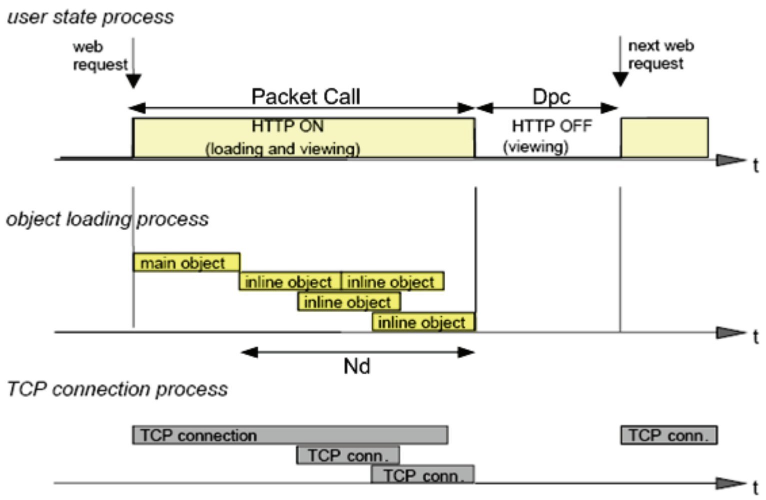

Figure A.2: Packet trace for a typical web browsing session.

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, sides of the page, stylised buttons, Java scripts, etc. The retrieval of the initial page and each of the constituent objects is represented by ON period within the packet call. Next, the user will read the information downloaded by the web browser, OFF period.

The parameters for the web browsing traffic are the following:

- SM: Size of the main object in a page
- SE: Size of an embedded object in a page
- Nd: Number of embedded objects in a page
- Dpc: Reading time
- Tp: Parsing time for the main page

Table A.1 and Table A.2 indicate the relevant parameters for both down and uplinks, respectively.

Table A.1: HTTP Traffic Model Parameters [IEEE80216].

Component	Distribution	Parameters	PDF
Main object size (SM)	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 (SE)	Truncated Lognormal	Mean = 7758 bytes Std. dev. = 126168 bytes Minimum = 50 bytes Maximum = 2 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 2.36, \mu = 6.17$

Component	Distribution	Parameters	PDF
Number of embedded objects per page (Nd)	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 Nd
Reading time (Dpc)	Exponential	Mean = 30 s	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.033$
Parsing time (Tp)	Exponential	Mean = 0.13 s	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 7.69$

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.

Table A.2: HTTP Traffic model parameters for uplink [3GPP2WG31].

Component	Distribution	Parameters	PDF
Main object size (SM)	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 (SE)	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 (Nd)	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 Nd If x > max, then discard and re-generate a new value for x

Component	Distribution	Parameters	PDF
Reading time (Dpc)	Exponential	Mean = 30 s	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 0.033$
Initial reading time (Dipc)	Uniform	Range [0, 10] s	$f_x = \frac{1}{b-a}, \quad a \le x \le b$ $a = 0, b = 10$
Parsing time (Tp)	Exponential	Mean = 0.13 s	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 7.69$

A.1.2 E-mail

In the e-mail traffic model, the message is downloaded from the mail server to UT during the ON period: The length of the ON period depends on the message size and the instantaneous throughput available for the user. The OFF period is the time taken by the user to read the message.

The ON period is characterised by a Weibull distribution and the OFF period is characterised by a Pareto distribution.

Assuming that user will take about 2 to 3 min reading an e-mail message, it is reasonable to assume that k_e = 30 – 60 s, OFF period. Regarding e-mail size, it is true to say that it depends of the attachment size, because the e-mail message is based ASCII characters, resulting in e-mails with few kbytes (without the attachments). The parameters to modulate an e-mail session are shown in Table A.3.

Table A.3: Wireless packet data traffic model for E-mail.

Period	Distribution	Formula	Parameters
Packet arrival	Poisson	$P(m_e = n) = \frac{(P_e \lambda_e T_e)^n}{n!} e^{-p_e \lambda_e T_e}$	
ON	Weibull	$F_e(x_e) = \begin{cases} 1 - e^{-e^{k_1} x_e^{c_1}} \\ 1 - e^{-e^{k2} x_e^{c2}} \end{cases}$	C_1 = 1.2 – 3.2 (Mean = 2.04), C_2 = 0.31 – 0.46 (Mean = 0.37) k_1=14.0 – 21.0 (Mean = 17.64), k_2 = 2.8 – 3.4
OFF	Pareto	$\Gamma_e(t_e) = 1 - \left(\frac{k_e}{t_e}\right)^{\alpha_e}$	k_e = 30 – 60 s, α_e = 0.5 - 1.5
E-mail attachment upload file size	Truncated Lognormal	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 2.0899, \quad \mu = 0.9385$	Min = 0.5 kbytes Max = 500 kbytes If the value generated according to the lognormal PDF is larger than Max or smaller than Min, then discard it and regenerate a new value. The resulting truncated lognormal distribution has a mean = 19.5 kbytes and standard deviation = 46.7 kbytes

A.1.3 Instant Messaging for Multimedia (IMM)

Real-time multimedia messages will be exchanged between users, using a multimedia (MM) server. This data source is similar to web browsing model, presented in section A.1.1, but with potential higher average in file size due the nature of content. A multimedia traffic is characterised by heavy tailed distribution patterns. Each session is modelled as a WWW application, consisting of a sequence of packet calls corresponding to file downloads.

It is considered that messages will not have a minimum size, and therefore the Weibull distribution is used to model packet call size, see Table A.4 [JF05].

Table A.4: IMM traffic model parameters.

Entity	Random variable	Parameter
Session arrival [h^{-1}]	Exponential	Mean = 0.15
Session duration [min]	Exponential	Mean = 15
Packet call size [KB]	Weibull	α = 1, β = 640 (Mean = 640 KB)
Inactive time distribution [s]	Pareto	α = 1.5, k = 30 s (Mean = 90 s)

A.2 Voice over IP (VoIP)

Voice over Internet Protocol (VoIP) has emerged as a significant enabling technology and the adoption of industry standards has accelerated its deployment. VoIP technology is generating wide interest across several markets [Am05]. There is now growing interest in delivering VoIP services over a range of wireless technologies, including 3G, WLAN, WiMAX and systems beyond 3G.

VoIP application requires timely packet delivery with low latency, jitter and packet loss values. Three parameters emerge as the primary factors affecting voice quality within networks that offer VoIP technologies: clarify, end-to-end delay and echo. To support interactive voice application on an IP network we must be able to control four QoS categories: bandwidth, latency, jitter and packet loss [KK01].

A.2.1 Source files for VoIP model

One source file is specified for VoIP which is used by each UT with a unique starting offset in the file for each UT.

The audio file was generated based on the Markov Service Option (MSO) model IS-871, but with some alterations. Modelling of 1/8th frame rate blanking is achieved by only transmitting the first 1/8th rate frame of each silence interval (as a silence indicator), and then one out of every 12 consecutive 1/8th rate frames (see [3GPP2DTX] for a description of this approach to model blanking). Assuming a 4-byte robust header compression (RoHC) overhead for each IP packet and this is included in the size of each VoIP packet in the source file [3GPP2WG31].

A.2.2 VoIP delay jitter model

VoIP delay jitter model is applied for the generation of source files for the forward link VoIP simulation. Laplacian distribution with $\alpha = 0$ and $\beta = 5.11$ ms is used to model VoIP delay jitter.

$$F(X) = \frac{1}{2}\exp\left(-\frac{|X-\alpha|}{\beta}\right), \text{ for } X \le \alpha \text{ , and}$$

$$F(X) = 1 - \frac{1}{2}\exp\left(-\frac{|X-\alpha|}{\beta}\right), \text{ for } X > \alpha \text{ .}$$

For a voice frame generated at T, the corresponding VoIP packet arrives at BS equals T + τ, where τ ~ L(0, 5.11ms) with limit $-80ms < \tau < 80ms$. The VoIP source file with delay jitter applied is available as [3GPP2WG32].

A.2.3 Simulation Specifics

In this section some specifics of the VoIP traffic models, related to 3GPP2 and based on those specific parameters, are repeated for information. Adaptation to WINNER assumptions and design parameters is required.

The VoIP simulation is to be run for 60k slots, with 10k warm-up slots. Units of the simulation are specified in DO slots (i.e. 5/3 msec). Information used for packet scheduling and dropping is to be what is actually available in the specified design. Any packet may be dropped on both FL and RL at any point in the simulation, but statistics on dropped and lost packets are collected for each UT [3GPP2WG31].

When presenting simulation results, parameters configurable in the system should be summarised, such as MAC and QoS parameters.

Each link (FL and RL) is simulated separately. The simulation flow is as follows [3GPP2WG31]:

1. Drop a number of users (K) per sector.
2. For each user perform server selection and redrop the user to another location if either FL/RL server selection is unsuccessful (i.e. the user is on either FL/RL coverage outage).
3. For each user (e.g. the k-th user), find the delay that corresponds to the 98-th percentile of the user's packet delay CDF - this is denoted by D_1.
4. Store all users' D_1 values and plot the CDF.
5. Find the largest K that has the 95-th percentile of the CDF less than the delay criterion D_0 by increasing K (go to step 1).
6. Report the capacity as the number of users K_0 that satisfied the step 5.

The values of D_0 to be used are:

D_0_FL: 50 ms, 70 ms

D_0_RL: 50 ms, 70 ms

To avoid hunting for the exact integer value of K, it is assumed that K takes on values 10*n* where *n* is integer.

A.3 Video Telephony (VT)

Video telephony is full-duplex, real-time audio-visual communication between or among end users. Video telephony is the ultimate friends-and-family plan. It connects people face to face, over any distance, to share milestones and precious moments.

In this age of e-mail, instant and text messaging, video telephony shares the personal nuances that only come from experiencing face-to-face communications. Inflections, expressions, and other non-verbal cues that are lost in cyberspace are preserved with video telephony, helping reconnect people during life's important moments[10].

The concept of video telephony has been around for more than 50 years, but only recently it has come to fruition. The basic technology required to transmit images and sound over the global communications network was feasible, but the infrastructure required to support practical video telephony was inadequate[10].

The primary challenge facing developers of the video telephone is the fact that full-motion, high-resolution video data requires far more bandwidth than audio data. Video telephony is an important but complex service, operators are working hard to promote the flagship service.

A.3.1 Source Files for Video Telephony model

Two source files are specified for VT, one for audio and one for video. The file for audio is the same file as that used for VoIP. For VT each UT has two source flows, one video and one audio, represented by

[10] http://www.physorg.com/news5717.html

these two files. Each UT uses the same pair of source files, but each UT uses a unique starting point offset in each file, as specified in the source configuration file. [3GPP2WG31].

The video file was generated using an H.263 encoder on reference video clips. A source file could be created in the following manner. A set of reference video clips was encoded under a few different reasonable assumptions to create a set of reference encodings. The video clips used were: crossing, doctor, foreman, friends, stunt, walk, zoom (7 of them). Two rate types of encoding were used: fixed rate at 44kbps, and fixed quality with quantisation parameters (QP) fixed to a value of 22. Four encoding modes were used, varying fps and the group-of-pictures (GOP, i.e. the rate of I-frames vs. P-frames): 1) 15fps GOP 45; 2) 15fps, GOP 37; 3) 10fps, GOP 27; 4) 10fps, GOP 23. Altogether, then, there are 7 * 2 * 4 = 56 separate reference encodings. The source file was generated from these 56 encodings by concatenating a completely random set of them (i.e. each subsequent clip is chosen uniformly from the set of 56 clips) [3GPP2WG31].

We could assume a 4-byte RoHC overhead for each IP packet, and this is included in the size of each application packet in the source file. We account for IP fragmentation in the source by assuming a maximum IP packet payload size of 1460 bytes, and adding in the 4-bytes of RoHC overhead per IP packet needed to carry the application packet. The sum of these overheads is included in the application packet size in the source file [3GPP2WG31].

A.3.2 Simulation Specifics

In this section some specifics of the VoIP traffic models, related to 3GPP2 and based on those specific parameters, are repeated for information. Adaptation to WINNER assumptions and design parameters is required.

The Video Telephony simulation is to be run for 60k slots, with 10k warm-up slots. Units of the simulation are specified in DO slots (i.e. 5/3 msec). Information used for packet scheduling and dropping is to be what is actually available in the specified design. Any packet may be dropped on both FL and RL at any point in the simulation, but statistics on dropped and lost packets are collected for each UT [3GPP2WG31].

When presenting simulation results, parameters configurable in the system should be summarised, such as MAC and QoS parameters. [3GPP2WG31].

A.4 Streaming

Streaming applications have been constantly gaining ground in terms of popularity, and this is mainly due to the bandwidth abundance and hardware sophistication the end-user is experiencing, specifically during the last few years. The next two sections provide an overview of the most relevant traffic models for video and audio streaming.

A.4.1 Video Streaming

A simplistic video model is introduced in this section that represents self-similar video traffic with local Hurst parameter ranging from 0.73 to 0.93 which is the case for motion pictures expert group (MPEG) video in a 25fps rate [IEEE802.162]. Each video source in this framework is represented by a superposition of two Interrupted Renewal Processes (IRP). The difference with the basic Interrupted Poisson Process (IPP) is that the sojourn time in both (on-off) states is now Pareto distributed, and each source needs two parameters so that its behaviour is described: one for each Pareto distribution corresponding to these two possible states. The generic video model based on 2IRP is introduced in the next table.

Table A.5: Generic 2IRP video model.

source_i	pkts/time_unit	c1_i	c2_i	avg_pkts
IRP_1	44.95	1.14	1.22	26.74
IRP_2	61.90	1.54	1.28	23.78
Avg.Rate of 2IRP process (pkts/time_unit) =			50.52	

The model can be scaled so as to represent any variable bit rate video and the following table demonstrates such a scaling for the case of a 1.9Mbps MPEG video.

Table A.6: 1.9Mbps 2IRP video model.

source_i	pkts/time_unit	c1_i	c2_i	avg_pkts
IRP_1	1123.80	1.14	1.22	668.49
IRP_2	1547.50	1.54	1.28	594.51
Avg.Rate (pkts/sec) =			1263.00	

A.4.2 Audio Streaming

Recent works [LH01] on modelling of RealAudio which is the most popular format for streaming audio applications have indicated the multi-scale variance of this type of traffic. More specifically, in scales of tens of seconds a single streaming audio flow has a constant rate while in smaller scales it behaves like a bursty ON-OFF source, with the OFF periods appearing in multiples of 1.8 seconds approximately; this bursty behaviour is also noticed in aggregate streaming audio flows. Other important characteristics incorporate the approximately fixed packet size and the strong correlation of flow requests with the time of the day. The main assumptions of the streaming audio model are summarised in Table A.7.

Table A.7: Streaming audio traffic model.

Streaming Audio Parameter	Distribution	Mean
Session Duration (sec)	Pareto (a=1.6)	2400
Session Inter-arrival Times (sec)	Exponential	5.45
Packet Size (bytes)	Deterministic	300/500 (16 and 20kbps respectively)
Bit Rate – compressed audio, low quality (kbps)	Deterministic	20
Bit Rate – compressed audio, FM radio quality (kbps)	Deterministic	32
Bit Rate – compressed audio, high quality (kbps)	Deterministic	128-256
Bit Rate – uncompressed audio, (kbps)	Deterministic	1411

A.5 File Transfer (FTP)

In [3GPP2EV] a straightforward FTP traffic model is proposed which incorporates two main parameters that describe the behaviour of an FTP transaction, namely the D_{pc}, i.e. the reading time (the time between successive downloads of the same user) and the distribution of the file size (S) to be transferred. The model is summarised in Table A.8 for the downlink and in Table A.9 for the uplink.

Table A.8: FTP Traffic model parameters (downlink) [3GPP2EV].

Component	Distribution	Parameters	PDF
File size (S)	Truncated Lognormal	Mean = 2Mbytes Std. Dev. = 0.722 Mbytes Maximum = 5 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \ge 0$ $\sigma = 0.35, \mu = 14.45$

Component	Distribution	Parameters	PDF
Reading time (D_{pc})	Exponential	Mean = 180 s.	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

Table A.9: FTP Traffic model parameters (uplink) [3GPP2WG31].

Component	Distribution
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

A.6 Interactive Applications

In this section we report on interactive applications that comprise a big part of current and foreseen data exchanges. Firstly we look into the internet based gaming. The popularity of such online entertainment is increasing in a fast pace and there exist already games in the market that are available only to be played online. In this framework game traffic characteristics and game traffic models are studied and presented. The widespread adoption of on-line gaming makes this service one of the most demanding and most complex from the modelling point of view. Finally in this section we report also on tele-presence, tele-surgery and e-learning services and their traffic characteristics.

A.6.1 Internet Gaming

A.6.1.1 Online gaming Quality of Service (QoS) requirement

In order to evaluate the impact of gaming data delay or gaming data loss to the quality of service of providing network gaming over wireless systems, it is necessary to define the QoS metrics for the game traffic model. For car racing games, an average round trip time of 100ms is suggested. Based on the work in [Fa02] and a subjective quality assessment [SERZ02], an average round trip time of 139ms would provide sufficient game quality for first person shooter games like Counter Strike® or Quake®.

Assuming 50ms average network delay and 30ms average downlink delay, the average delay for the uplink wireless air interface will be 59ms. It is observed [FA02] that players experience serious degradation of game playability with a round trip delay of 200 – 225ms. Impact on performance is perceived starting at 75ms one-way end-to-end latency [3GPPR2] which has been set as the preferred limit in [3GPPSA1]. To maintain the playability, a maximum delay of 145ms is applied to all data transfers, i.e., gaming data is dropped if it is not delivered after 145ms. There are very few statistics available for the tolerance of network/mobile gaming to data loss, partly because there is no clear threshold of data loss rate beyond which the game becomes unplayable. The playability of games decreases as the data loss rate increases. While real-time strategy games seem to tolerate only 1% packet loss, massive multiplayer online role playing games may allow up to 10% packet loss [3GPPR2]. In 3GPP a preferred packet loss <3% and a limit of 5% have been agreed [3GPPSA1].

The topic of game traffic modelling is relatively new and few publications exist on this issue.

Nevertheless some important first work exists on source models of network game traffic [Bo00] and the validation of the proposed traffic model [Fa02].

It is deemed valuable to report here below the associated state of art that may represent the basis for further analyses and for gaining a perspective of the possible future evolution of the recent successful gaming applications and of the associated traffic models.

A.6.1.2 Game traffic characteristics

Among network games, action games are the most popular and within this genre the most popular game is Counter-Strike® followed by Quake®. A network game model for Counter-Strike® is proposed in Network Game Traffic Modelling [Fa02], which is an evolved model based on the network game model for Quake® proposed in Source Models of Network Game Traffic [Bo00]. The game communication model of both games follows the client/server approach and uses UDP packets for the exchange of small update information to maintain fairness of the game and player synchronisation. The server sends game state information to each client where packets are read and processed. Clients synchronise the server game state with their local game state, process player commands, and return update packets with the players' movement and status information. Game traffic has been monitored and registered over a LAN with 50 total participants for overall 36 hours. More precisely, several matches with 8 to 30 active players have been observed with the matches lasting from 30 to 90 minutes each (6.5 hours gaming in total). Details about such traffic observations and associated characteristics can be found in [Bo00].

Network game traffic generates a significant share of today's Internet traffic. In [MC00] it is reported that 3-4% of all packets in a backbone could be associated with only 6 popular games. A high market potential, increasing usage as well as sharp real time requirements make this kind of traffic interesting for Internet service providers and manufacturers. In order to profit from the high popularity of online gaming, networks are enhanced for gamers, i.e. components and protocols are optimised for game traffic. Although there are other popular online games emerging with more focus on strategy or role playing, first person shooters are still the most popular multiplayer games found in the Internet and they impose the hardest real time requirements on the network.

A.6.1.3 Game Traffic Model

[Fa02] provides a simple traffic model for fast action multiplayer games. Although multiplayer game traffic shows strong correlations due to a shared game state it has been shown in section "Traffic Characteristics" that the variance is small, i.e. these dependencies only lead to slight traffic changes. Thus, the game traffic can be modelled by independent traffic streams from each client to the server and a burst traffic stream from the server to the clients. Therefore the approach assumed in [Fa02] is:

(1) Clients behave independent of each other,

(2) Server traffic per client is independent of the number of clients and

(3) Client traffic is independent of the corresponding server traffic.

Based on the scope of the evaluation the modelled traffic only reflects active game phases without interruptions due to change of scenario or game options. During game interruptions client and server traffic may pause for a short time after which larger update packets are transferred to synchronise all clients. Note, that this traffic is not time critical. Those dynamics are out of the scope of this work and have to be modelled on a higher level if desired. The game traffic model proposed consists of only two independent modules, the client traffic model and the server traffic model with a burst size equal to the number of clients participating in the simulated traffic. For a mathematical description of the distribution functions for inter-arrival time or packet size it is necessary to find a function of similar shape and fit its parameters to the empirical data. In [Bo00, Fa02] the Extreme Value distribution has been identified to fit best.

The Extreme Value distribution is given by the following expressions:

$$f(X) = \frac{1}{b} e^{-\frac{X-a}{b}} e^{-e^{-\frac{X-a}{b}}}$$

$$F^c(X) = e^{-e^{-\frac{X-a}{b}}}$$

b > 0

Server – Model

The inter-arrival time for the server denotes the burst inter-arrival time. Within a burst a packet is sent to every client as soon as possible. Packet sizes are generated independently for each destination. Table A.10 shows traffic characteristics of the observed data as well as the suggested distribution [Fa02]. For games with a small number of players it has been found that inter-arrival times of server bursts show four clear peaks comparable to client inter-arrival times, i.e. at 50 ms, 55 ms, 60 ms and 65 ms instead of a continuous distribution function as obtained for matches with many players. It has been assume that this behaviour is caused by the server nearing its performance limit in games with many clients.

Client-Model

As the distribution functions of client packet inter-arrival times is characterised by one to three peaks a multimodal distribution is suggested. Significant peaks are identified at 34 ms, 42 ms, 50 ms and 60 ms. As most observed clients show their peak at 42 ms it has been suggested a deterministic distribution for this inter-arrival time (see Table A.10, [FA02]).

Table A.10: Counter Strike traffic characteristics and suggested approximation.

	Server (per Client)		Client	
	Characteristics	Approximation	Characteristics	Approximation
(Burst) Inter-arrival time	peak = 55 ms mean = 62 ms coeff. of variation = 0.5	Extreme (a=55,b=6)	mean = 41.7 ms coeff. of variation = 0.24	Deterministic (40 ms)
Packet Size	mean = 127 Bytes coeff. of variation = 0.74	Extreme (a=120,b=36)	mean = 82 Bytes coeff. of variation = 0.123	Extreme (a=80,b=5.7)

A.6.1.4 Usage of Game Traffic Model

The simplicity of the presented model allows to use it either to simulate traffic on a link to and from a subset of clients as well as traffic to and from the server communicating with all active clients. The number of active clients as well as session durations have to be set for the duration of the simulation or must be described on a higher model level[11], e.g. using the results of [HB01]. The game traffic model is not suited to provide background traffic for evaluations of other traffic flows. Its use is clearly in the evaluation of quality of service (QoS) aspects of networks in respect to games. In order to asses the impact of packet delay or packet loss experienced in a simulation, it is necessary to define QoS metrics for gaming applications. Today's games can cope with an enormous lag (ping, round trip time) and loss. These applications are thought to be used over the Internet with a typical round trip time of 50 to 150 ms. If analogue modems are used, each use introduces an additional latency of 30 to 40 ms, i.e. an additional 120 to 160 ms to the round trip time for a dial-up player. Ping times frequently show 300 ms and more. Consideration of loss and lag are an essential part of the game design. Game designers try to optimise for 200 to 250 ms round trip time and provide robustness for larger lag. This is achieved by client-side prediction of the game state, i.e. movement of objects and other players [SERZ02] [HB01]. By combining movement with inertia or reducing maximum velocity of objects prediction is even more effective. Such considerations result in very robust games tolerating lag up to one second and loss up to 40%. However, these values should not be taken as criteria for good or bad QoS since acceptable game play requires far better performance. Ping times of 50 ms or 150 ms make a huge difference. In [AG01] an evaluation of player effectiveness over that players ping time shows that players with lower ping times score significantly better than others. Based on [TR02] we find that a ping below 50 ms is associated with excellent game play. A ping below 100 ms is good and above that, playability decreases noticeably. Ping

[11] http://www.acm.org/sigs/sigmm/MM2001/ep/henderson/

times above 150 ms are often reported to be intolerable but many players claim to have no problems with ping times around 200 ms. An evaluation on "Half Life" reported in [TR02] shows that players who experience high ping times of over 225 ms do not quit and look for a faster server but stay and continue to play with this high lag. It has been assumed that those players use 56k modems and do not expect to get a better connection elsewhere. The study reveals that many gamers (40%) play with a high lag of over 225 ms despite of the decreased playability. The impact of packet loss is rarely discussed as it is experienced as lag as well. However, a high ping time without packet loss is preferable to a small ping time with packet loss of around 10%.

A.6.1.5 Future Mobile Gaming Applications QoS metrics

According to the above discussion it is expected that future wireless mobile systems can assure high quality as for lag (ping, round trip time). Currently understanding is that 50 ms lag is considered excellent quality while 100 ms lag is considered good quality: future wireless system networks should be able to achieve such lag range (50 – 100 ms) target.

While on the issue of the impact on QoS of the lag some understanding has been gained trough previous works, there are very few statistics available for the tolerance of network/mobile gaming to data loss, partly because there is no clear threshold of data loss rate beyond which the game becomes unplayable. Apart form the obvious consideration that the playability of games decreases as the data loss rate increases, there is the need for collection of statistics and users feedback on this issue. It is expected that the increase in complexity of games may lead to a further need of data loss control and low data loss may become a key parameter of the mobile games QoS metrics.