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Proposal of the best Suited Deployment Concepts for the identified Scenarios and related RAN Protocols

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Abstract: D3.5 presents relaying as an integral part of the WINNER system concept. The relay based deployment concept allows deploying the WINNER broadband radio interface cost efficiently. Suitable assumptions have been made on the technical solutions as guideline for the definition of protocol requirements. The relay is presented as part of the overall WINNER system concept in the WINNER logical nodes architecture. Further the logical nodes architecture and the relation between functions in system layers and specific logical nodes is shown. Protocols and functions are presented that are necessary to implement relaying in the WINNER system concept. Therefore the protocols of different layers namely the radio resource control layer, the radio link control layer and the medium access control layer have been investigated in more detail, also under consideration of the physical layer aspects. In addition to that two add-on solutions to increase the efficiency of relay based technologies have been investigated and are described. These add-ons are cooperative relaying and mobile relays.

Keyword list: WINNER, relay based deployment concepts, multi-hop, fixed relay nodes, deployment concepts, relaying, logical nodes, protocol architecture, radio resource management (RRM), radio resource control (RRC), medium access control (MAC), mobile relays, cooperative relaying, resource partitioning, flexible protocol architecture, multi mode

Executive Summary

This deliverable presents the WINNER system concept from the deployment concept point of view with a strong focus on relay based deployment concepts. The document is addressing the layered system architecture from the protocol perspective. The same layered system architecture was also taken up by WP7 from the functional point of view. The protocols needed to allow for efficient relaying are presented layer by layer bringing up requirements and showing particular solutions.

The work on relay based deployment concept for a mobile communication system is strongly related to the system engineering work. The placement and need of function required in the relay as well as the addressing of what information and signalling has to be relayed is reflected in the system architecture including a mapping of system layers on the specific WINNER logical nodes as represented in the WINNER logical nodes architecture.

Further the flexible WINNER reference protocol architecture is presented. The reference architecture enables the necessary adaptivity of the WINNER air interface protocols to the wide range of envisaged scenarios. A use case is presented to demonstrate the different functional units and their mode dependencies in the BS/UT and in the relay node.

It is further shown that the protocol layers have to be designed appropriately to allow for efficient relaying. One of the major issues for a relay based system are the radio resource management functions which can be found in the radio resource control (RRC) layer as well as in the medium access (MAC) layer. It is shown how the radio resource partitioning is communicated in between the different nodes directly involved in the radio access, which are the user terminal (UT), the base station (BS) and the relay node (RN). As solution a three level approach is presented for the resource partitioning, where the partitioning information is broadcasted to the UTs by their serving radio access point (RAP), which can be either a BS or a RN. The MAC plays an important role for the envisaged relay based system as the forwarding of data is assumed to take place on MAC level. Thereby the quality of service (QoS) based shaping of the data is performed by the service level controller, which is placed in the BS for the DL and only necessary for the transmission of uplink user data. The RLC layer taking care of the reliable data transfer needs dedicated relaying solutions for to allow reliable end-to-end data transfer, as further detailed in the document including some simulation results in the annex.

In addition, final research results on cooperative relaying which has been investigated under consideration of different candidate concepts are presented. Cooperative Relaying might serve as an add-on technology for a relay based system like WINNER to further improve the performance through the exploitation of spatial diversity, which is an inherent feature of relay-based systems.

Also the application of mobile relays has been investigated in order to estimate their usability for the WINNER concept. It has been shown that the main focus in this field should be put on moving networks due to their economic impact.

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1 Introduction

This deliverable is the final deliverable of WP3 and therefore the last in a series of 5 deliverables of WP3. In the first two deliverables [WIND31] and [WIND32] different relay based deployment concept have been studied showing the benefit of relay based deployment concepts, e.g., using fixed relays to cover otherwise shadowed areas. In [WIND31] system level performance results have been presented to show that throughput and capacity in a system using fixed relays, e.g. in a Manhattan like scenario will be appropriate to allow cost efficient broadband radio coverage. The results shown in [WIND31] and [WIND32] were based on existing systems like IEEE 802.16. In addition to the relaying concepts the work on protocols has been started by introducing a first version of the logical nodes architecture and the multi mode reference model in [WIND31] which have been continuously further developed as one of the essential parts of the WINNER system concept. In [WIND34] a first approach has been made to integrate the relaying concept into the WINNER system concept by matching different relaying concepts to different WINNER scenarios. As continuation of the integration of relays into the WINNER system concept this deliverable presents a full view on how a system and related protocols have to be designed to allow for inherent relaying support.

The need for innovative relay based deployment concepts as inherent part of the WINNER system concept is motivated by the limited range of broadband radio interface as studied by WINNER due to high attenuation at carrier frequencies beyond 3.4 GHz, a limited transmission power (EIRP) owing to regulatory constraints and unfavourable radio propagation conditions, e.g., in densely populated areas. Conventional cellular radio network deployment concepts would require a very high density of base stations to achieve sufficient radio coverage there. As a consequence, the system deployment cost in terms of Capital Expenditure (CAPEX) and Operational Expenditure (OPEX) for broadband radio will increase dramatically, resulting in a high cost per bit transmitted.

It is well known that an increased data rate (for a given power and carrier frequency) leads to a reduced radio range and that the available data rate decreases with increased distance from a base station (BS) as illustrated in Figure 1-1. In general, the service quality in terms of data rate, delay, outage probability, etc. seen by the user does not depend on its location in a cell.

Assuming a constant number of users per area element in a cell, the number of users increases with the distance d from the BS following a square law. It appears reasonable that the requirements on 4G radio systems in terms of capacity, delay, user-experienced data rate and deployment cost cannot be met using conventional cellular deployment concepts. Instead, a novel disruptive deployment concept is urgently needed.

To meet the goal of low cost radio network deployment for both, short-range and ubiquitous (wide-area) coverage, fixed layer-2 relay node based deployment concepts appear to be the most promising technology. Relay nodes don't need a wired (fibre) backbone access reducing deployment costs (CAPEX and OPEX) and introduce a high flexibility in relay positioning, allowing a fast network rollout and adaptive traffic capacity engineering. Relays may also be used to provide indoor coverage from outdoor BSs.

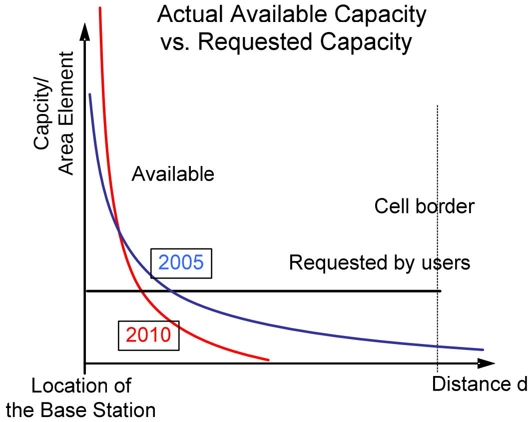


Figure 1-1: Facts in the available capacity vs. distance from a base station compared to the requested capacity [WaWiSc06]

In the following, different types of deployment concepts are discussed starting with a section on the potentials of deployment concepts based on relays as part of the fixed infrastructure, followed by a section explaining some further relay based deployment concept (see also [WaWiSc06]).

The layer-2 relays as considered by WINNER in an infrastructure based cellular deployment are at least temporarily fixed in location. In the following they are being denoted Fixed Relay Nodes (FRN), although they could also be movable in order to, e.g., temporarily increase the capacity in a certain service area, e.g. for the duration of an exhibition. In the following different application scenarios will be shown that have different characteristics and also different impact on the WINNER protocols.

1.1 Relays to extend the service range of a BS (service area size optimisation)

FRNs introduced to a cell (to become a Relay Enhanced Cell - REC) may be used to enlarge the coverage area of the BS as shown in Figure 1-2. If the FRN is placed outside the coverage area of the BS, antenna gain is needed to connect BS and FRN. The higher the antenna gain on the BS-FRN link is, the larger is the capacity of the FRN.

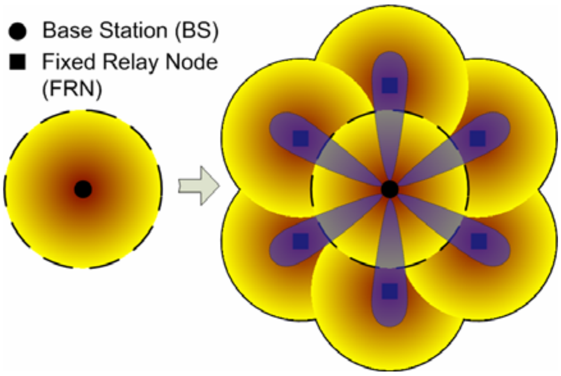


Figure 1-2: Left: Conventional cell; Right: Relay Enhanced Cell (REC) using layer-2 Relay Nodes (RN) to enlarge the cell area [WaWiSc06]

1.2 Relays for Optimised Cell Capacity and Minimum Transmit Power

FRNs may be used in order to increase the capacity at outbound cell regions as shown in Figure 1-3. In both scenarios shown in Figure 1-3 the capacity per area element in the REC scenario approximates the requested capacity better than possible with a conventional (single-hop) cell. For a cellular radio deployment the channel re-use distance is minimised when receive antenna gain instead of transmit antenna gain is used.

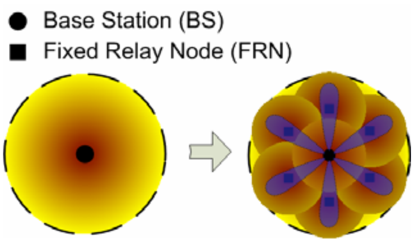


Figure 1-3: Left: Single BS cell; Right: REC with RN to increase the capacity at the cell border and balance the capacity per area element [WaWiSc06]

The solution presented in Figure 1-3 can also be used to minimise the transmission power needed by user terminal (UT), BS and FRN. It is referred to as *Power Minimising* concept. In this concept the UTs benefit from the reduced energy consumption, whilst the reduced output power at BS and FRN leads to reduced HW cost.

1.3 Relays to cover otherwise shadowed areas

A capability not available from any other deployment concept is that a FRN can be used to serve areas otherwise shadowed from the BS as shown in Figure 1-4.

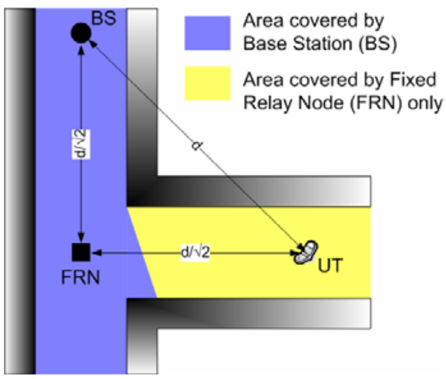


Figure 1-4: Relay Node to cover otherwise shadowed areas

1.4 Structure of D3.5

This deliverable aims at the description of the WINNER system concept from the deployment concept perspective. The idea of relaying has already been taken up by the first description of the WINNER system concept in [WIND76] in form of the logical nodes architecture which is a direct outcome of WP3 (see [WIND31] and [WIND32]. In [WIND210] as final deliverable of WP2 the work of [WIND76] has been further detailed with respect to the air interface elaborating on the interworking between the layer with a clear focus on the PHY and MAC functionalities. For this deliverable the further development of the system concept will focus on clear layered architecture as well as a first outline of the protocols required by the WINNER system concept will be given.

In the next Chapter a set of definition as used commonly in WINNER (see also [WIND76]) is introduced followed by Chapter 3 which describes the WINNER vision and the idea of modes in the WINNER system concept. This should allow the reader to understand the work on the WINNER system concept as provided in the latter chapters. Further the WINNER test scenarios are briefly listed in Chapter 3 which have been used to map the WINNER deployment concept on different scenarios.

A set of more technical working assumption was required in order to get a good idea on what requirement the protocols have to serve. Therefore some basic system aspects have been defined and are presented in Chapter 4. These system aspects provide assumptions on how a final WINNER system could technically look like, i.e., what does the system concept has to support. These assumptions have been made based on the current research results, such as from WP2 on the one hand and on the assumption that the final WINNER system concept should be as simple as possible. The identified system aspects should not be seen as final system concept, but as clear working assumption and reference case for further studies.

Chapter 5 shows the further development of the layered architecture as outlined in [WIND76]. It should be mentioned that the provided architecture is not a new architecture, but the further development of the work which has been started with the description of a layered service or functional architecture in [WIND76].

In Chapter 6 is providing more detail with respect to the RAN protocols of the different layers of the WINNER system concept. Thereby no final protocol specification is presented but protocol requirements and challenges are shown, while at the same time solutions for some particular protocols are proposed. The more detailed specification of the protocols will be left for Phase II of WINNER.

In Chapter 7 the final results on cooperative relaying are presented. The idea of cooperative relaying has been consequently further developed through the deliverables [WIND31], [WIND32] and [WIND34].

Chapter 8 gives a final overview on the Mobile Relay concepts (Type I/II/III) that have been investigated through the previous deliverables [WIND31], [WIND32] and [WIND34]. Some final remarks and comments are included and final selection takes place with regards to the MR-based concept that will be taken for further research within the Phase II of WINNER.

The conclusions in Chapter 9 provide a matching of deployment concept in terms of relaying and mapping of logical to physical nodes for the different main WINNER test scenarios.

Finally the Annex provides more details about some protocol solutions as well as a number of system level simulation results that show the potential of relay based deployment concepts.

2 Definitions

2.1 General

Mode	**-**	Specific combinations of algorithm assignations or ranges of algorithm assignations may be referred to as "Modes". The two main considered physical layer modes (PLM) are based on, and denoted, FDD and TDD. A System mode is a PL mode combined with MAC and RLC assignations.
Radio Access Technology	**RAT**	The radio access technology (RAT) is the air interface that is used to allow the link between User Terminal and Base Station or Relay Node of the RAN. This includes also multi-hop/relaying elements. The WINNER RAT can be derived into several Physical Layer Modes.
Deployment Concept		The term “Deployment Concept” describes network element types and their functions (i.e. logical network elements), (a) how these network element types are linked in a network topology, (b) how logical network elements are mapped onto physical network elements and (c) where physical network elements are deployed according to the radio propagation scenarios for which the deployment concept is applicable.

2.2 Physical Network Elements

Physical Network Element		A physical network element denotes a physically existing device in the RAN that incorporates certain functionality, thereby representing one or possibly even more logical network nodes.
(Physical) Base station	**BS**	A stationary physical network element serving relay nodes or user terminals via its radio access capabilities. Base stations are interconnected with network elements belonging to the RAN. A physical base station contains one or more base station logical nodes.
User terminal	**UT**	A physical network element used by the end user to access a service or set of services.
(Physical) Relay node	**RN**	A physical network element serving other RN or UT in a given geographical area via its radio access capabilities. It is wirelessly connected to a base station, another relay node and/or a user terminal and forwards data packets between these network elements.
Heterogeneous Relay Node		A heterogeneous relay node is a relay node that uses different radio access technologies (or different modes of the same RAT) using common or different sets of transmission resources (e.g. RF channels) for its links (BS-RN, RN-RN, RN-UT). The radio access technologies that a heterogeneous relay incorporates can be different modes of the same RAT (i.e. in the WINNER context), one WINNER RAT-mode and another (possibly legacy) RAT, or two (legacy) RATs, where the latter case is not in the WINNER scope of research.
Homogeneous Relay Node		A homogeneous relay node is a relay node that uses the same radio access technology and mode in a common set of transmission resources (e.g. RF channels) for its entire links (BS-RN, RN-RN, RN-UT).
(Physical) Radio access point	**RAP**	A physical network element in the radio access network responsible for radio transmission and reception to or from the user terminal via its radio access capabilities. A RAP can be either a relay node or a base station.

Access System		The access system is used to connect the WINNER user terminals to the base station either directly or via relay nodes. The elements of the access system are the WINNER base stations and the WINNER relay nodes.
Feeder System		The feeder system is the transport system used to feed the base stations. The distinctive characteristic compared to the access system is that WINNER users shouldn't connect to this network directly. The transmission technology used by the feeder system could be wireless or wired and is irrelevant and transparent for the final user.
Site	**-**	A site is defined as the physical co-location of base station hardware serving a set of antennas. Users may be connected to a site either directly or through relay nodes

2.3 Logical Nodes

Logical Node	**LN**	A Logical Node is defined by the service (or group of services) it provides towards other nodes and the service (or group of services) it requires from other nodes. Identical Logical Nodes terminate an identical set of protocols and provide/require the same group of services. One physical element can comprise one or several LNs."
Base Station Logical Node	**BS_{LN}**	A logical node terminating the transport network layer protocols on the network side as well as the radio protocols on the UT and RN side. It contains a single MAC entity corresponding to a single cell, and it manages the logical relay nodes connected to it.
Relay Node Logical Node	**RN_{LN}**	A logical network node with relaying capabilities that is wirelessly connected to a BS_{LN}, UT_{LN} or another RN_{LN}. Like the BS_{LN} it terminates the radio protocols (MAC and PHY) on the UT side as well as on the BS side and, in case of more than two hops, also on the RN side. The RN_{LN} does not terminate the transport network layer protocols. It contains a single MAC entity corresponding to a single cell.
User Terminal Logical Node	**UT_{LN}**	A logical node comprising all functionality necessary for it to communicate directly with another UT_{LN} or the RAP.
Radio Access Network Gateway	**$RANG_{LN}$**	A logical node terminating the RLC-UP protocols.
Access Control Server	**ACS_{LN}**	A logical network node that controls the access to the radio interface resources. It terminates Control Plane protocols of the RLC.
Access Router Logical Node	**AR_{LN}**	A logical IP layer node that performs the tasks attributed to an Access Router as defined in relevant IETF specifications. In the WINNER architecture the AR_{LN} contains all functionalities of the IP Convergence Layer (CL).
Cooperative RRM	**CoopRRM**	The CoopRRM will be responsible for the decision making process of the cooperation mechanisms (handover, admission control and QoS management) and is foreseen to be physically located outside of the involved RANs.
Radio Access Network	**RAN**	The WINNER RAN comprises $BS_{LN,}$ $RN_{LN,}$ $RANG_{LN,}$ $ACS_{LN,}$ AR_{LN}

2.4 Links, Flows, Cells and Handovers

Link	**-**	A link is a radio connection between two physical network elements of the WINNER access system. It subdivides into relay link between radio access points and the user link between the

		user terminal and the radio access point.
Flow	-	A flow is a packet stream from one source to one or several destinations, classified by QoS requirements, source and destination(s)
Cell	-	A cell is defined by the geographical coverage area of its broadcast channel. A cell uses a single PLM on a particular carrier frequency.
Base station serving area or Relay enhanced cell	**REC**	The geographical area covered by the broadcast channels of cells whose resources are managed by a single base-station and its connected relay nodes.
Multi-Homing (Multi – RAN Transmission)	-	Multi-homing means that a UT is associated to more than one RAN simultaneously.
Multi-Mode-Transmission	-	Multi-mode-Transmission means that a UT is connected by more than one link to different cells of one WINNER RAN. These cells use either different WINNER PLM or the same mode at different carrier frequencies.
Handover	**HO**	A Handover is a change in the set of links between a RAP and a UT. This includes a hard "switch" from one cell to another, moving into and out of a multi-mode-transmission and a changing of the links used for multi-mode-transmission
Intra-system HO	-	Intra-system HO is a handover between two different radio cells within the same system, with the same or different radio mode. It subdivides further into Inter-mode HO, Inter-cell HO and Inter-frequency HO. The term horizontal handover is equivalent to intra-system handover.
Inter-mode HO	-	Inter-mode is a intra-system-handover between WINNER cells operating in different system modes (FDD, TDD and P2P).
Intra-mode HO	-	Intra-mode is a intra-system-handover between WINNER cells operating in the same system mode (FDD, TDD and P2P).
Inter-cell HO	-	Inter-cell HO is a intra-mode-handover between WINNER cells operating in the same system mode at the same frequency.
Inter-frequency HO	-	Inter-frequency HO is an intra-mode handover between WINNER cells operating in the same system mode but at different frequencies.
Inter-system HO	-	Inter-system handover: An inter-system handover is a handover between two different radio systems e.g. WINNER <->WLAN, UMTS <->GSM. Two subcategories are distinguished: inter-system handover of radio networks belonging to the same operator and inter-system handover of radio networks belonging to different operators. The term vertical handover is equivalent to inter-system handover.

2.5 Transport Channels

Transport channels		Transport channels have in the WINNER system been defined as the User Plane interface between the RLC and the MAC.
Broadcast Channel	**BCH**	For control information to all terminals within a cell.
Random Access Channel	**RAC**	Contention based random access channel, for initial access to master device
Direct Access Channel	**DAC**	Contention based direct access channel
Common Data Channel	**CDC**	Scheduled transport channel for point-to-multipoint communication
Targeted Data	**TDC**	Scheduled transport channel for point-to-point communication

Channel		
Protocol Data Unit	**PDU**	Output from a protocol layer
Service Data Unit	**SDU**	Input to a (protocol) layer. A packet in a transport channel is a MAC SDU and a RLC PDU

2.6 MAC and PHY-specific Terms

Service level controller	**SLC**	Service level controller in RLC User Plane
Resource scheduler	**RS**	MAC User Plane. Controls the resource mapping onto PHY channels
Service level control buffer	**SLCB**	Flow queuing in RLC layer for scheduled flows. The SLCB contains MAC SDUs.
Resource scheduling buffer	**RSB**	Per-flow queuing in MAC layer for scheduled flows. Each RSB contains one queue per active flow and one RS controls it. A RSB contains coded segments of MAC SDUs, denoted FEC blocks
Cyclic Redundancy Check	**CRC**	Code sequence added to re-transmission units
MAC Re-transmission unit	**RTU**	Retransmitted individually by link ARQ for scheduled flows and DAC. Formed by (a segment of) a MAC SDU +CRC code+ segment number
FEC block		Coded transmission block with whole or part of an RTU as payload. Content of RSB.
Adaptive resource scheduling	**ARS**	Uses channel quality or state info. at the transmitter
Non-frequency-adaptive resource scheduling	**NRS**	
Chunk		Basic resource unit on radio channel. A time-frequency resource consisting of nsub adjacent subcarriers and nsymb consecutive OFDM symbols with chunk duration Tchunk
Chunk layer		Chunk within one spatial channel (layer). There are Qc layers in the cell.
Generalised Multicarrier Transmission	**GMC**	Has OFDM and frequency-domain based serial modulation as special cases. See [WIND21] and [WIND23] .
Slot		Time interval for uplink or downlink transmission in half-duplex FDD and in TDD.
Frame		Time-frequency-spatial resource unit. The frame duration in time covers one uplink slot and one downlink slot in half duplex FDD and TDD transmission.
Superframe	**SF**	Time-frequency-spatial unit on the physical channel. Contains resources for all transport channels and control signalling, and includes main synchronization pilot symbols. Consists of preamble followed by a number of frames.
Resource mapping		Mapping of FEC blocks onto SF preamble and chunk layers.

2.7 Spectrum related Terms

Coexistence		The concurrent operation of different services or RANs in the same or in adjacent frequency bands without causing degradation to any service, with emphasis on the indicated limitations in terms of, e.g., frequency separation, physical separation, and transmission powers.

Sharing	The use of a same frequency band by different RANs or services, either with coordination or possibly without any coordination between the systems, with emphasis on the spectrum access schemes and methods.
Dedicated Spectrum	Spectrum is available for a single deployment of the WINNER based RAN (e.g. similar to current GSM bands).
Single system shared spectrum	Spectrum is available for WINNER only, but multiple independent deployments are possible in the same bands (e.g. similar to current DECT bands).
Open shared spectrum	Spectrum can be used by WINNER (one or more deployments) and also other systems (e.g. similar to ISM bands).

3 The WINNER Context and Requirements

This chapter should inform about the boundary conditions that had to be taken into account when defining the deployment concepts and about the initial assumptions that led to the presented solutions

3.1 WINNER Vision

The goal of WINNER is to define a ubiquitous radio access system concept, capable of providing the connectivity required to enable the long-term vision of the “Wireless World”. This vision has at its heart the idea of user centricity – new technologies are not introduced just because they exist, but because they address the users' needs and desires.

In order to address the goal of WINNER within this vision, the following general assumptions form the guiding principles towards a WINNER system concept:

- WINNER will develop a single new ubiquitous radio access system concept whose parameters can be scalable or adapted to a comprehensive range of mobile communication scenarios from short-range to wide-area.
- The ubiquitous radio access system concept will provide terrestrial communications, but not including BAN and PAN elements.
- The ubiquitous radio access system concept will be self-contained, allowing WINNER to target the chosen requirements without the need for interworking with other systems.
- Where other systems are available (including BAN/PAN, as well as, for example, evolved 3G and WLAN), cooperation, interworking and infrastructure reuse may be used for mutual benefit. The WINNER concept will fit into a multi access structure allowing an “Always Best Connected” solution.
- First deployment expected at the earliest in 2010, widespread from 2015.
- The WINNER RAN should provide significant benefit to users, manufacturers, providers and any potential actors compared to alternative technologies, such as evolved 3G or WLAN systems. Examples of benefit might include cost, performance, and ease of use or ubiquity of service availability.
- Requirements will be further developed which relate to the expected stakeholder experiences. E.g. from the end user perspective, continuous & ubiquitous link throughput, delay and negotiable quality of service.
- WINNER develops a single Radio Access Network (RAN)
- The WINNER RAN is further based on one WINNER Radio Access Technology (RAT). The WINNER RAT may provide different modes to come up with a flexible solution for different scenarios and propagation conditions.
- The ubiquitous radio access system concept will be scalable in terms of service requirements, capacity-per-area-unit and stepwise increasing complexity and related performance.

Details about WINNER requirements can be found in [WIND71].

3.2 Modes

Modes are used in the project as a synonym for adaptivity of the system to different application scenarios, radio environments, spectrum bands, etc. In [WIND71] the concept of a mode was introduced and defined in the following way (bold added):

“A goal of the WINNER project is to develop one RAT which can be adapted to a wide range of situations and environments, e.g. ranges, mobility, user densities. The adaptation of the RAT might require different parameterisations or use of different algorithms. **Certain combinations of parameter or algorithm assignations or ranges of parameter or algorithm assignations may be referred to as “Modes”**.

A more detailed understanding and further definition has been made in [WIND76] and can be summarised as:

- A minimum set of modes should be defined in order to meet the WINNER requirements

- The multi-mode protocol reference architecture provides the framework
- A **System Mode** is the combination of a PLM and a MAC.
- A **Physical Layer Mode (PLM)** can be defined where there is a significant impact (discontinuity in adaptation) of PHY functionality on the air interface concept.
- At the present time only one such functionality leads to different PLMs – duplex. Therefore only 2 PLMs are considered.
- All other PHY parameters are assumed to be adaptable within the ranges considered in the project (e.g. carrier frequency, bandwidth, multiple access, coding, underlying modulation, modulation alphabet).
- MAC
- Current thinking is that different MACs are needed for:
- FDD/TDD "cellular"
- P2P
- The RLC and MAC design takes into account horizontal sharing with coordination and vertical sharing.
- Node/device (i.e. BS, RN, UT) capability will take into account both modes (system and PLM) and other adaptive parameters.
- Please note that there is currently no reference or definition of "wide area" or "short range" modes. Similarly there will not be device "classes" of "wide area" or "short range"

3.3 Basic deployment scenarios

Table 3-1: WINNER High-level deployment scenarios as defined by [WIND71]

	Name	Coverage	#	Propagation Conditions	Mobility	Traffic Density (Indicative)
Scenario A	In and around building	Localised and non-ubiquitous coverage	A.1	Indoor	0-5 km/h	[High]
			A.2	Indoor to outdoor		
Scenario B	Hot Spot/Area	Area wide but non-ubiquitous coverage	B.1	Typical Urban	0-70 km/h	[High]
			B.2	Bad Urban		
			B.3	Indoor	0-5 km/h	
			B.4	Outdoor to Indoor		
			B.5	LOS – Stationary	0 km/h	[High]
Scenario C	Metropolitan	Ubiquitous coverage	C.1	Suburban	0-70 km/h	[Medium]
			C.2	Typical Urban		[Medium]/[High]
			C.3	Bad urban		
			C.4	Outdoor to Indoor		[Low]-[High]
			C.5	LOS – Stationary	0 km/h	[Low]/[Medium]
Scenario D	Rural	Ubiquitous coverage	D.1	Rural	0-200 km/h	[Low]
			D.2	LOS – Moving Networks	0-300 km/h	[High]

4 Initial Deployment Concept assumptions

A wide variety of concepts for network deployments, with different maturity status, has been proposed in the deliverables D3.1 – 3.4 [WIND31]- [WIND34]. Some concepts focused on "traditional" approaches while some proposals were more "fancy" with regard to their presented concept. However mostly different assumptions and requirements have been selected by partner for their deployment concepts proposals. This section will focus on the description of the agreed assumptions for the deployment and implicitly define the initial requirements for the underlying protocols that will be outlined in more detail in Chapter 6 and will be further specified in the next phase of WINNER.

The assumptions provided in this chapter are working assumptions that serve as basis for the work on the system architecture and the protocols as presented in Chapter 5 and Chapter 6. They should in no case be seen as the final system assumptions. Most of them have been chosen following the rule "Keep it simple". Thus a change of the assumptions would in most cases mean to add complexity to the system and should be compared to the results as found for the basic system aspects as outlined in this chapter.

4.1 General assumptions

The WINNER Deployment Concept (DC) will utilize relay nodes to enable deployment improvements with respect traditional solutions in scenarios around 5GHz, where coverage problems due to shadow fading and limited range can be expected. These *relay-enhanced cells* (REC) offer the advantage of increased coverage flexibility, potential for indoor coverage and a more flexible radio resource management.

The presence of relays implies additional requirements on the system. RNs need to be integrated in the RAN architecture including the realisation of the necessary radio resource partitioning between the BS and its connected RNs. A relay based system might benefit from relay specific functions, like, e.g. a intra REC handover, that allows to exploit the already existing knowledge about the flow which is to handover within on REC.

The DC will be designed to allow a flexible number of hops in order to have a decent flexibility in coverage strategies, link budget and resource partitioning mechanisms depending on the scenarios. In practice the actual number of hops (for a certain target delay) will likely be limited, due to performance limitations and system complexity increase. More than two hops are likely to result in more complex routing and forwarding schemes, increased protocol complexity, additional overhead and potentially more end-to-end delay. Therefore the basic DC will be optimized for two hops in a first step (without ruling out the possibility to allow more than two hops).

The cellular MAC design within WP3 has focused on the more challenging case, where RNs and BS use the same physical layer mode and share spectral resources but a different operation on the RAP-RAP and the RAP-UT connections (heterogeneous relaying) enabling optimization for stationary and mobile use can be envisaged as well. The same mode on UL and DL links is assumed in any case.

UTs will need to support only a limited set of modes, mainly due to complexity reasons.

4.2 Topology assumptions

An important aspect of the DC is the set of possible topologies for the interconnection of the Network Elements UT, RN, and BS. It should noted that the term BS is here referred to the "Physical Base Station" Node, that could comprise several logical nodes (BS, RANG, ACS, AR – see also section 3), e.g., to allow for a stand-alone deployment of a single BS (which could also feed some RNs).

UT and BS can be connected to each other directly or by the means of one or more RNs. From the user viewpoint, the RN should be "transparent", i.e., the user should perceive the service without even knowing whether the UT is connected to a BS or to a RN. The UT however, in its working conditions, must obviously know the kind of supported connection, in order to correctly support handover (BS to BS, or BS to RN, or RN to RN); therefore, the RN is seen by the UT as an individual node, i.e. like a BS.

Each RN can be physically connected to one BS at most, so that inter REC meshing is not part of the basic deployment concept. This has been agreed since an increase of complexity could be implied due to the required coordination of the resource assignment to different BS (coordination across BSs), the required coordination signalling and the need of two transceivers when nodes operate on different resources. In sight of possible DC evolutions, some advantages of inter-REC meshing like the ability to handle BS failure, improved mobility support and load balancing could be taken into account in the future.

Each RN has one path towards the BS, so that intra REC meshing is not part of the basic deployment concept because it increases coordination signalling requires the support of multiple connections at the RN and enhanced routing functionalities. In sight of possible DC evolutions, some advantages of intra-REC meshing could be taken into account in the future. The ability to increases the network reliability (e.g. by establishing redundant paths), allows for load balancing (in particular greater resilience against congestion) and might improve QoS aspects.

In the basic DC it has been assumed that multiple flows from/to the same UT will not follow different paths through the network for simplicity reasons. The possibility to follow different paths increases complexity and the number of states especially for routing. However the potential gains of such a solution, will be evaluated in the future, when a clearer view on the system is available. An exception which will be evaluated as well is the case of a multi-mode terminal where the two (or more modes) are handled independently, so that all the flows related to one mode must follow a given path, while the flows related to the other mode can follow one different path.

Peer-to-peer communication is part of the DC but interference to the rest of the network needs to be carefully observed. It should be noted that peer-to-peer has not been thoroughly studied so far and the term "peer-to-peer" is here referred to a single hop terminal to terminal direct communication supported (i.e., established, released) by a stationary node (BS or RN), according to the direct mode of HiperLAN2. Some open issues for peer-to-peer communication remain, like the business model for the operators, the resource question (in the same spectrum as the cellular mode or not) and the control of the peer-to-peer connection (centrally controlled by a fixed and operator controlled network element or decentrally UT controlled) are open issues which will be further investigated.

The feeder system (i.e., the transport network which is connecting of the BS to the backhaul network) is not part of the DC, even if it can be itself realized through a wireless technology. The feeder connection can be implemented either through wireless or through wireline solutions. In case of a wireless feeder, the overall system will consist of the cascade of different wireless links (feeder + WINNER Multi-Hop), where the feeder link is not to be interpreted as a further "WINNER hop". Therefore, the WINNER DC will not foresee a dedicated mode for this use (wireless feeder), although not excluding that a wireless feeder can be adopted, based on existing technologies.

4.3 Relay assumptions

All relays are fixed or temporarily fixed for the time of operation. Except for the moving networks case, no mobile relays will be assumed within the basic WINNER DC due to the currently unresolved issues with regards to this concept. This applies to mobile UT acting as relays as well, where at this stage the advantages of this concept cannot provide a considerable incremental gain compared to the currently unresolved issues of complexity, battery consumption and security. Relaying via user terminal is considered for the "fixed UTs" case only, where power supply is available and individual user equipment may be used for relaying external traffic. In this case, a bonus could be envisioned for the owner, e.g. free traffic, but the related "business scenario" is critical and its identification not yet completed. For more information on the mobile relay based DCs, see Chapter 8 which includes are more detailed discussion of mobile relays. Due to economic reasons priority should be given to the concept of moving networks (Type I concept) e.g. on publics transport like busses or trains.

Out of the other concepts presented in Chapter 8, UTs acting as mobile relays (Type III) is regarded as very promising case, but due to the currently unresolved problems like battery consumption, security and complexity, it will not be taken forward within WINNER. It has to be seen whether the progress in terms of power saving, security, complexity the next 5-10 years is sufficient enough that UTs acting as mobile relays (Type III) will have the chance to be further investigated as part of a mobile communication system. Dedicated Mobile relays (Type II) are not regarded as promising technology due to the large number of issues that need to be resolved.

RNs can be movable (i.e. the RN is temporary stationary) and with self-configurable capability even if it requires more expensive network deployment. Movable RNs can be temporarily set up (e.g. for fast network roll-out), can be used on ad hoc basis i.e. when needed, in multiple instance positions, so the network can be easily extended. This allows reusability in multiple locations and the possibility to shift the resources of low loaded cells to areas where sporadically there is a high traffic demand (e.g. football matches). The self configuration capability of RNs is optional as it increases node complexity (also for the BS that can contain self configuration capabilities too) and is not always required, but allows an easy network deployment and increases flexibility of network deployment. The protocols shall be designed so that self configuration is supported.

It has been assumed that ***RNs operate in "decode and forward" manner***, avoiding noise amplification and guaranteeing variable rate protocols. RNs without intelligence denoted as repeaters in 2G/3G systems, (i.e., operating as "amplify and forward") have been rejected from the basic assumption of the WINNER DC. If "decode and forward" relays are widely deployed for coverage or capacity extension as envisaged in the current DC, their amount will likely be much larger than the traditional repeaters which currently cover highly shadowed areas like subway stations. Because of cost savings when producing larger amounts of the same device no major difference in cost is expected between both relay types. Therefore and because repeaters do not allow heterogeneous relaying and increase the error probability and the noise amplification problem they are currently not considered in the DC. This assumption has of course be verified and re-evaluated if the above assumptions change.

RNs are assumed to work with one transceiver only in the TDD mode, unless the performance gains show that the increased costs and complexity for a second transceiver are justifiable. In case of FDD the RN is likely to work with two transceivers in the full duplex mode like the BS.

Cooperative relaying (and the related macro-diversity techniques) is not taken into account, at this stage, for the basic WINNER system. However, these advanced techniques have been considered in the WINNER landscape, and will be further investigated, in order to better clarify issues related to cost/complexity versus benefits (see also Chapter 7)

No definite choice about the "physical" deployment of RAPs will be made within WINNER as this is strongly scenario dependent. In order to cope with this assumption it should be able to build RNs sufficiently small to enable flexible deployment and sufficiently cheap to compete with the fixed line or legacy feeder link backbone connection. The exact cost of a RN (as compared to a BS) however is an open issue that needs to be further investigated.

The possibility privately owned RNs as part of an operator driven WINNER network is an aspect depending on business case scenarios which should be further investigated.

4.4 Resource sharing assumptions

In the WINNER relay scenarios, different kinds of sharing are possible in principle, based on different variables (time, frequency, space, or combinations), providing several degrees of freedom in the overall system design and optimization.

About the "spatial" dimension, spatial processing is seen as an integrated part of the WINNER air interface. Therefore, multiple antennas at the WINNER nodes are foreseen in principle. The more typical scenario will be characterized by multiple antennas at BS and RN, while the coexistence of multiple antenna and single antenna terminals will probably take place (single antenna terminals will be supported too by WINNER, in order not to limit possible applications and business opportunities).

The radio resource partitioning shall be flexible between BSs and RNs and no pre-allocation of radio resource shall be assumed. Hence the total time-frequency resources are dynamically partitioned into parts used by the BS (shared parts) and parts used by RNs. This partitioning is computed by the MAC that is implemented at the BS, and is then signalled to all RNs and UTs. This partitioning increases complexity and overhead (due to signalling), but allows flexible resource sharing and increases spectral efficiency. The functions of coordination across nodes, resource partitioning and related signalling are required. The update interval may be adaptive and different update intervals for the resource allocation may be used for intra and inter REC resource allocation.

If RNs are present, some UTs may transmit to/receive from these RNs. The MAC implemented in each RN controls those transmissions, thus, the RNs essentially control separate sub-cells. A complete MAC layer is assumed to be implemented at each BS and also at each RN.

While defining resource partitioning strategies, it should be highlighted that macro-diversity based Soft Handover is not taken into account, at this stage, for the basic WINNER system. Soft Handover and Fast cell selection will be possibly further investigated as a possible enhancement.

4.5 Deployment in WINNER test scenarios

The deployment in the WINNER test scenario depends on the considered scenario. Most of the presented proposals in the deliverables took the scenario B.1 and to some extend C.2 into account. Few proposals where presented for D.1 and D.2. Please see Chapter 8 for the discussion of moving networks in scenario D.2.

The proposals which where presented for scenario B.1 can in principle also be used for ubiquitous coverage as envisaged by scenario C.2, although they do not take a strict hexagonal cell layout into account and may not be the most cost efficient solution. With this respect both scenarios can not be completely separated at this point. Most of the proposals assumed a deployment "below rooftop" with highly shadowed areas which enables capacity improvements if relays are used in most of the cases directional antennas or beamforming between the network nodes has been assumed. Below rooftop deployment comes along with greater spatial separation of RAPs which potentially enables a smaller reuse factor while the connection can be realized with pathloss savings on the LOS paths between the network nodes for the whole transmission. As such it is in general a preferable scenario for relays; however this can only be partially exploited in the WINNER test scenarios (Manhattan grid) due to the long LOS streets, if relays are placed on the street crossings. Below roof top deployment allows and requires smaller network equipment associated with smaller sub-cells and enables more flexible coverage strategies. However, a higher number of RAPs will be required due to range restrictions.

A deployment "Above Rooftop" improves the Line-Of-Sight (LOS) properties of the connections, allows an easy roll-out and through better initial coverage. Moreover existing 2G/3G sites and infrastructure can be reused. As drawbacks, this solution increases inter-RAP interference, increases transmission power and limits deployment possibilities. In most of the cases a directional link (beamforming or directional antennas) has been used for linking the BS and the RN. In some sense this deployment is has similarities to the traditional hexagonal structure and the deployments found today where BS are fed by directional feeder links in an orthogonal resource. Like today this deployment is more suited for the hexagonal based scenarios D.1 and C.2.

4.6 Summary

Based on the discussion given above, the following main assumptions have been made for the basic deployment concept. These assumptions will be used in the following WINNER phases to complete the WINNER system and to design the required protocol concepts. It should be noted however that some aspects might still be missing and further refinement may become necessary as the work continues and the view of the WINNER system becomes clearer. In the light of these assumptions it is expected that more and more partners will take those into account when evaluating and studding their respective proposals and future contributions will be based on these assumptions.

- The DC is based on Relay Nodes
- The nodes should be able to self-configure
- Base Stations may be deployed stand alone
- A feeder system is not considered in the DC
- Peer-to-peer communication is part of the DC, but will not utilize multiple hops.
- The DC will not rely on a resource pre-allocation (e.g. frequency planning).
- BSs and RNs will dynamically share Radio Resources
- Resource partitioning should be flexible from super-frame to super-frame
- Multiple antennas should be assumed in the BS and the RN and optionally in the UTs
- Each RN is exactly connected to one BS and will only use one path for data transfer towards this BS
- Multiple flows from/to the same UT will not follow different paths within the same set of resources. If different resources are used this could be the case.
- The DC will not exploit macro-diversity in the context of soft handover or cooperative relaying.
- The number of hops shall be flexible; however the DC will be optimized for two hops whenever possible
- RNs will decode and forward any user data to be relayed.
- The use of RNs shall be transparent to the user in terms of service experience. RNs do not need to be transparent to a UTs
- A complete MAC layer is assumed to be implemented at each BS and also at each RN
- The control of *downlink flows* resides in the transmitting RN/BS. Most MAC control functions for *uplink transmissions* reside in the receiving RN/BS.
- The RN can be movable (i.e. the RN is temporary stationary)
- The only Mobile Relay DC considered will be that of Type I i.e. that of moving networks.

- No choice about the physical deployment of RN will be made within WINNER
- UL and DL shall use the same mode
- Different modes can be used for the RAP-RAP and the RAP-UT connections
- A UT can only support a limited set of/one modes
- The DC does not rely on the availability of a certain system mode[1]

[1] This represents an assumption by WP3. In the light of ongoing mode discussions within WP7 this could be discussed and revised within WP7 again.

5 WINNER System Architecture

To enable communication between different nodes of the WINNER system WINNER has started to develop layered architecture with layers, entities, service access points, protocols, connections, etc. Within this layered architecture layers serve as a subdivision of the architecture.

A layer consists of protocols which are a set of rules and formats (semantic and syntactic) which determine the communication behaviour of the entities of a layer in the performance of the functions of the respective layer. Note that protocols are not algorithms.

A layer has service access points (SAPs) at which entities of the layer provide their services to entities in the next higher layer.

A service is a capability of a layer and the layers beneath it which is provided at the boundary between the layers to entities at the next higher layer.

An entity is seen as an active element of the layer embodying a set of capabilities defined for the layer.

A function is then part of the activity of an entity.

For information to be exchanged between two or more peer entities of the same layer, an association is established between them in the next-lower layer using the protocol of the next-lower layer.

These are the definitions as introduced by the ITU in the ITU-T recommendation X.200 [ITUTX200].

In addition to the layered architecture as defined in by the ISO/OSI reference model the exchange of control information between two entities of adjacent layers without the establishment of a peer-to-peer communication is seen necessary mainly to allow efficient radio resource control.

Figure 5-1 shows the layered WINNER System architecture which has been further developed from the service specification as presented in [WIND76]. The WINNER system architecture consists of 4 layers namely Radio Resource Control (RRC), Radio Link Control (RLC), Medium Access Control (MAC) and the physical (PHY) layer. The layers provide services to higher layers at well defined service access points (SAPs). The main difference compared to the version provided in [WIND76] is the introduction of the Radio Resource Control (RRC) layer.

For WINNER two types of service access points (SAP) have been defined

1. Conventional SAPs where a lower layer provides services to the higher layer in order to allow the higher layer to communicate with its peer entity.
2. At the Control SAPs a layer provides services to a higher layer which allow the higher layer to request information and to impact the behaviour of the lower layer, e.g. by parameterisation. The services provided here are not related to peer-to-peer communication.

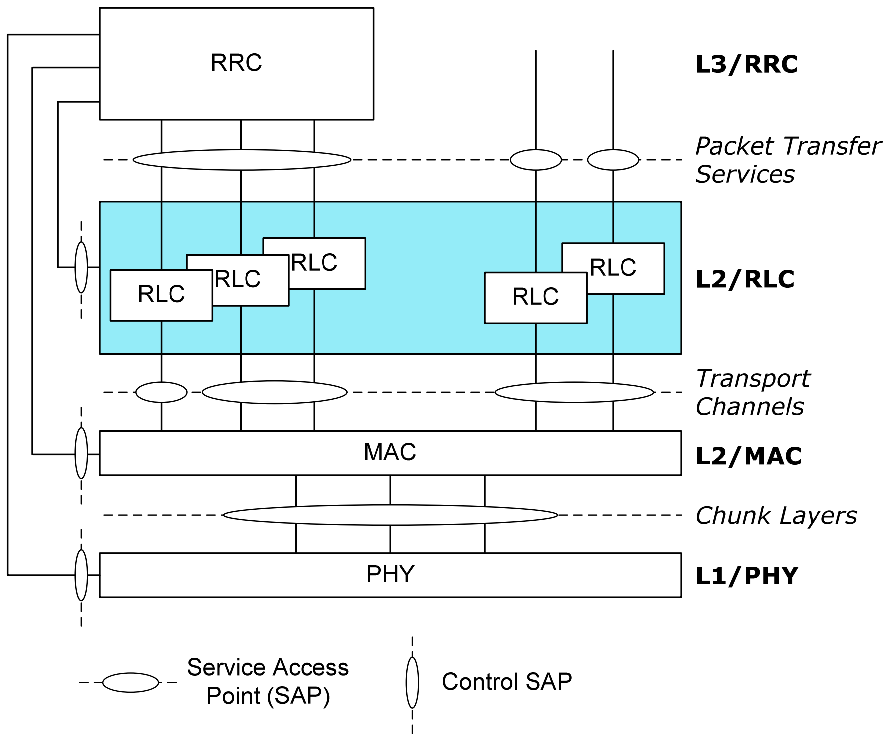


Figure 5-1: WINNER System Architecture

In the following the different layers and their characteristics will be explained briefly in section 5.1. In section 5.2 the WINNER flexible and multi mode capable protocol reference architecture as introduced and described in [WIND31] and [WIND32] is briefly presented in order to show how the WINNER system is able to adapt flexibly to different scenarios needs allowing even multiple modes within one radio communication system. Section 5.3 shows how the different layers are mapped onto logical nodes.

5.1 WINNER System Layers

In [WIND76] a first outline of a layered WINNER architecture has been presented. The architecture was an outcome of the service specification performed to define the interfaces between layers. As the goal of WINNER is to develop a communication system it is essential that peer entities of the same layer exchange information without knowledge about how this information was gathered. To allow for the communication between peer entities the set of rules must be defined and implemented in protocols. These protocols provide the means for peer entities to communicate with each other as well as they define the communication behaviour of each entity.

In order to be able to provide ubiquitous radio access the WINNER system has to comprise cellular radio network architecture. Thereby the RRC layer reflects that WINNER is addressing more than an isolated cell scenario, where only a number of radio links have to be controlled. It reflects the idea that WINNER, like, e.g. 2G or 3G system, is also going for a cellular coverage where allocation of among different access nodes radio resources have to be controlled. To reflect the capability that the WINNER system concept is addressing a radio access network as well as to build the WINNER system consistent and therewith also make easer to understand for people outside of WINNER the RRC layer has been introduced in comparison to [WIND76]. The introduction of the RRC layer does not create a completely different architecture compared to [WIND76]. Many of the RRC functions are taken from the RLC Control Plane (CP) and the IP Convergence (IPC) CP layer as presented in [WIND76] as they are providing functionalities addressing the control of the overall radio resources rather than the control of a single radio link or the IP convergence. The IPC layer is left out in D3.5 as its services and functionalities are seen in the scope of the WWI IP Ambient Networks [AN].

5.1.1 Radio Resource Control

The RRC is a pure control plane layer, which is responsible to control the overall radio resources assigned to the WINNER system. The radio resource can be seen on different levels from the RRC perspective: Control of radio resources

1. between WINNER and other systems

2. between different WINNER systems (operators)

3. between different WINNER modes

4. between the different RAPs of one WINNER system

The RRC takes care of load-, spectrum- and micro-mobility control.

The RRC provides also functions for flow establishment and release and location services.

5.1.2 Radio Link Control

The RLC User Plane provides reliable packet transfer over the air-interface. It also performs confidentiality protection and packet prioritization in order to meet the QoS goals.

5.1.3 Medium Access Control

The User Plane provides the services ‘Radio packet transfer’, i.e. transmission and reception over the radio interface of packets belonging to the transport channels (see Section 6.3 defined as the interface between RLC and MAC layers.

The Control Plane provides the ‘MAC radio resource control’ service, i.e. acceptance and execution of control messages from higher layers that specify required transmission parameters and boundary conditions. Furthermore it implements ‘MAC control feedback’, i.e. messaging that supports the flow control, the QoS control and the spectrum assignment and other functions at the RLC and RRC layers.

5.1.4 PHY

Handles the physical transmission of chunks and measurements and control signalling directly related to the radio interface.

5.2 WINNER Protocol Reference Architecture

5.2.1 Requirement - Adaptability

Among the Requirements for the WINNER radio access system are the ubiquitous, spectrally efficient radio access at high data rates and low delays, embedded into a unified radio access technology. To meet these requirements, an efficient adaptation of the system to different environments and scenarios (such as short-range vs. wide-area, LOS- vs. NLOS propagation etc.) is needed. Such an adaptation in many cases will take place in terms of switching between algorithms of certain air interface functions or changing between (sets of) parameters. In some cases, the adaptation may even involve changing the behaviour of the air interface, e.g. by switching to another duplex scheme. Different duplex schemes are in the WINNER terminology referred to as so-called different “physical layer modes” of operation. These modes represent the highest degree of adaptation and optimization to different scenarios. To limit overall system complexity, the central design paradigm is to keep the number of such options to the minimum needed to ensure efficient operation in all envisaged scenarios.

It is felt by the WP3 partners that a flexible structure of the protocol software and architecture is inevitable to facilitate the following three levels of adaptation:

The parameterization level - requires that individual functions of a protocol can be adapted in their parameters to suit a certain communication environment or -situation. Examples for this parameterization include, but are not limited to:

.1 Window sizes and retransmission timers of Automatic Repeat ReQuest functions
.2 Segment sizes of Segmentation and Reassembly (SAR) functions
.3 Padding Sizes
.4 Cyclic Redundancy Checksum lengths
.5 …

The algorithm level - requires that certain functions can be adapted in their behaviour through the choice of appropriate algorithms, all belonging to the same class of algorithms, i.e., the same kind of functionality. Examples for this level of adaptation include:

.1 The choice of certain scheduling algorithms for Service Level Controller (SLC) or the Resource Scheduler (RS)
.2 The choice of a certain type of ARQ algorithm (e.g. Stop-And-Wait, Go-back-N, Selective-Reject, Selective-Repeat, with and without Block Acknowledgements, etc)
.3 The choice of certain error correction coding/decoding schemes
.4 …

The adaptation on this level is in some cases expected to include the choice of 'No' algorithm at all to disable certain protocol features, e.g. the ARQ or the Segmentation.

The functionality level - requires that functionalities of the protocol layer are replaced by other functionalities or the logical ordering, respectively the interworking of functionalities, be re-arranged, to result in a different behaviour of the entire protocol layer, or if such changes affect more than one protocol layer, to result in different behaviour of the entire protocol stack.

Whether or not the third level of adaptation will be necessary to enable the different physical layer modes of operation of the WINNER air interface will be subject to further study in project phase 2, when the properties of these physical layer modes are better understood and the differences become clearer. If only the logical order of certain function is concerned, an alternative to re-ordering functionality is to foresee two functional blocks of a certain type at different logical positions in the processing chain and enable/disable one or the other as described under "algorithm level". Such a way forward has been drafted by the Cross-Workpackage (XWP) group on RLC issues regarding the different optional combinations of Retransmission and Segmentation Functions.

The common understanding across Work Packages 2 and 3 is hence that the WINNER protocol reference architecture should account for at least the first two levels of adaptation.

5.2.2 Layered and Modular Structure

The basic protocol architecture as proposed in [WIND32] aims at providing a framework which enables all three levels of adaptation mentioned above. The goal is to allow a flexible configuration of the WINNER protocols and the efficient integration of multiple potential WINNER modes in a complementary way, thereby allowing to take maximum benefit of the commonalities between the modes (see [BePaScWa05]). As a consequence, such a reference model requires protocols that conform to the structure given in Figure 5-2. In order to exploit commonalities between different modes of operation, the software of these protocols would (i) ideally follow a modular approach to allow a high degree of reusability and (ii) provide suitable structures and interfaces for the flexible composition of the individual modules.

Another main requirement towards the structure of the WINNER Reference Protocol Architecture is to match the layered service architecture as proposed in [WIND210, WIND76].

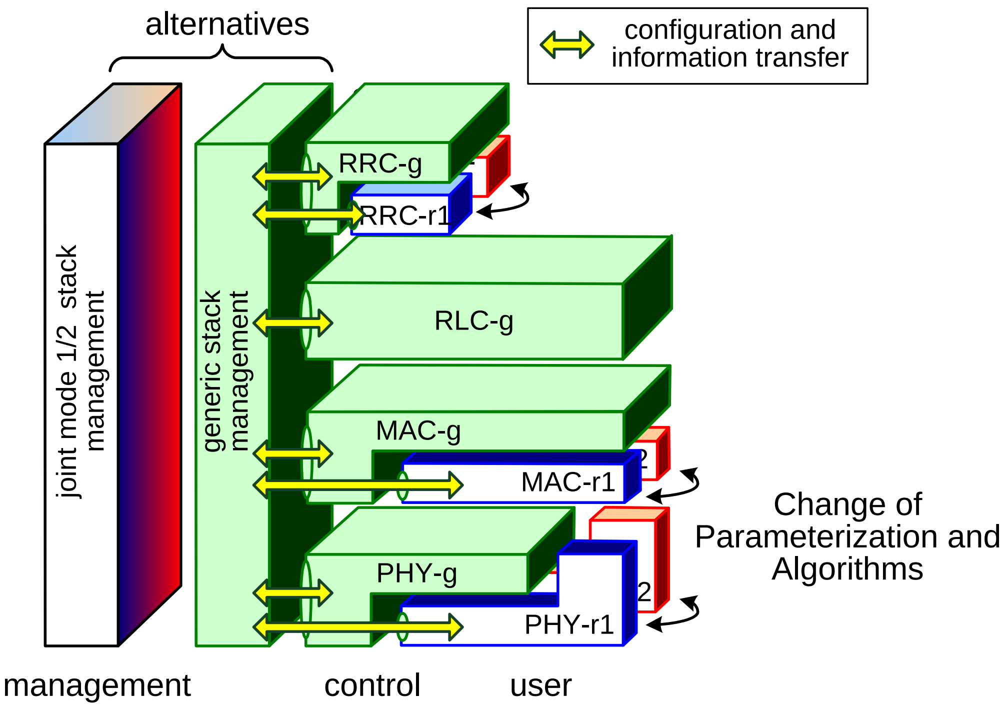


Figure 5-2: Overview of Layered Protocol Architecture and Management Plane Interaction

A complete reference to the envisaged protocol functions/services in the different Service- respectively Protocol-layers can be found in [WIND76]. A mapping of these layers to the logical nodes as defined by WP3 can be found in Section 5.3. A brief overview is given here:

RRC:

RRM-related functions

RLC:

Flow setup, release and E2E addressing
E2E Retransmission Protocol
Flow Control
Higher-level Segmentation

MACg:

Service level controller (QoS Control)
Hop-wise addressing in Multi-Hop setup
Hop-wise retransmission protocol
Segmentation

MACr:

Resource Scheduler (QoS Control)
Spatial Scheme Selection
Duplex Scheme Implementation
Adaptive- / Non-adaptive transfer
Broadcast transfer

To match the requirements about modular composition of the functionality of the protocol layers which result from the reference model in Figure 5-2, the set of functions identified above have been decomposed into a set of so-called Functional Units (FUs) [PaScDeWa05], each having cohesive responsibilities and a unified interface. We further describe the identified functional units and how they are connected to form what is called a Functional Unit Network (FUN). This Functional Unit Network is a representation of the protocol that helps to understand and visualize dependencies between different Functional Units and thus improve protocol design.

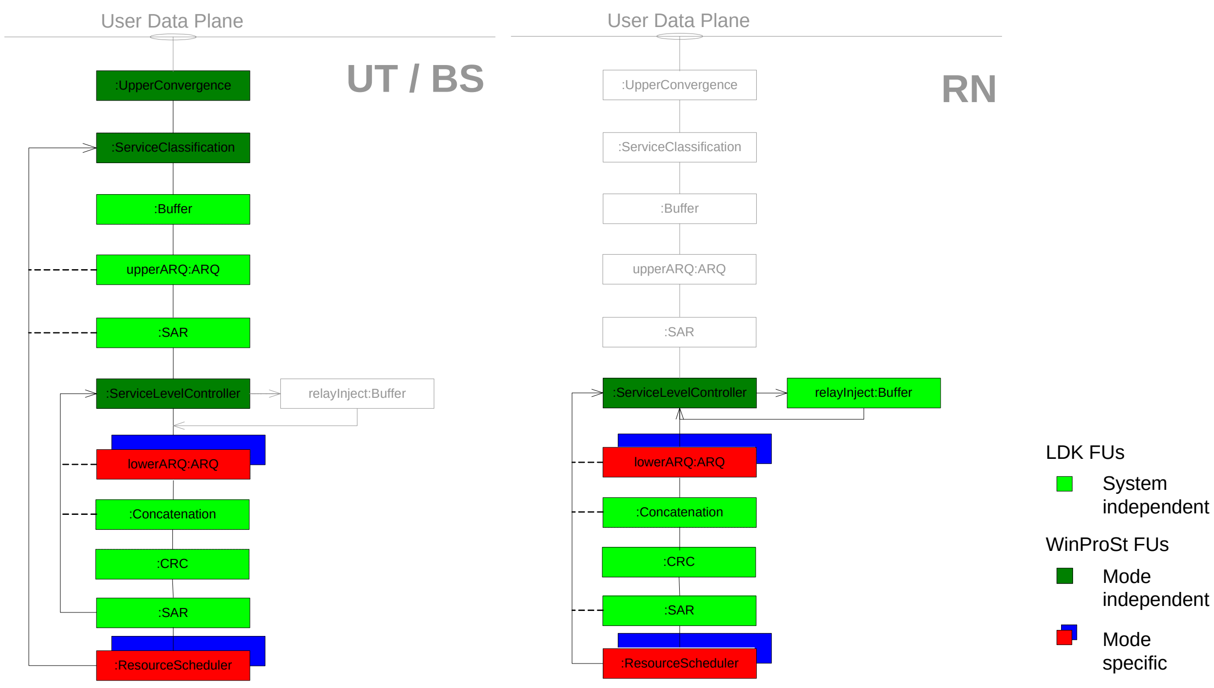


Figure 5-3: Functional Units in BS / UT and RN. Dotted lines describe dependencies between units.

Figure 5-3exemplarily shows how the intended functionality of the Data Link Layer-User Plane as it is currently discussed within WINNER can be composed out of a set of mode independent Functional Units and a small number of mode-specific Functional Units. The used Units can be further subdivided into three different classes:

1. The common, system-independent functions, shown as light green boxes, these can be taken from a toolbox of generic protocol functions that can also be used to implement protocols for other, non-WINNER radio systems, examples are
    a. ARQ units. The figure shows an upperARQ for securing packets end-to-end over multiple radio hops and a lowerARQ that operates on a per-hop basis.
    b. Buffers • Segmentation and Reassembly Units
    c. Helper Units such as a Synchronizer, used for FUNinternal flow control (The Synchronizer function is an implementation detail and exceeds the scope of this article; it is shown here only for the sake of completeness.)
2. The mode-independent, but WINNER-specific functions, shown as dark green boxes • IP Convergence Layer
    a. Service Classification, among others dealing with flow handling and addressing
    b. Relay Inject Buffers for PDU handling at the Relay Nodes
    c. The Service Level Controller (SLC). In the WINNER terminology, this is how the Modeindependent QoS scheduling and per-flow buffering is referred to.This unit also deals with flow addressing.
3. The WINNER-mode-specific functional units, which are shown as red/blue boxes. Example for such functions are
    a. The so-called Resource Scheduler, which performs the actual mapping of data flows onto physical resources. It is therefore a unit that is specific to a certain physical layer mode being employed.
    b. The lowerARQ, which is likely to be a Hybrid ARQ and thus also closely linked to the physical layer mode being used.

Note that the combination of both is referred to as MAC-r (where 'r' stands for radio-specific) in the reference model in Figure 5-2. This also illustrates that common and mode-specific functionality can be arbitrarily located inside the protocol layer, since the functional units referred to as MAC-r do not necessarily have to be directly connected to each other.

The mode-specific parts are the only Functional Units that may be affected by functionality-level adaptation to different WINNER physical layer modes. The mode-independent boxes may undergo a change of their parameter set, but their essential functionality remains the same. The change of the parameters would be the responsibility of the entity that manages the respective protocol layer (see Figure 5-2). Figure 5-3 also shows that different subsets of the Functional Unit Network can be active in different logical nodes. The greyed-out boxes denote the Functional Units that are unused at the BS/UT or RN respectively.

5.2.3 Management Plane

The work related to protocol design in WINNER Phase I has to a large extent focused on the control and user data plane functionalities. An overview about the requirements for the management functionalities associated with the presented new kind of protocol architecture will be given in this section. In addition, we give an overview about the most challenging research topics in the area of protocol management to be investigated in WINNER Phase II.

The protocol management can be differentiated into two different groups of management tasks, Layer Management and Stack Management tasks.

5.2.3.1 Layer Management Tasks

According to a chosen set of parameters and algorithms, the functionality of a protocol layer (i.e., the Functional Unit Network that represents the layer) has to be composed from generic, system-specific and system-Mode-specific Functional Units. Composition in this context means to instantiate, parameterize and interconnect the appropriate set of Functional Units i.e., select behaviour and / or set parameters of individual Functional Units. More advanced tasks include changes of the configuration at runtime, and context transfer. Context transfer involves either the preservation of internal protocol states to be reused after a configuration change or the translation of internal protocol states into corresponding states after a configuration change, if possible.

5.2.3.2 Stack Management Tasks

The stack management is envisaged to handle the overall stack configuration. Triggering configuration inside Layers by instructing Layer Managers accordingly and handling of parameters that have to be jointly set/optimised across different layers are among the tasks of the stack management. In the case of operation of multiple modes, the stack management may also control their operation, i.e., the coexistence of and the switching between modes.

5.2.4 Conclusions / Further research

The presented protocol architecture achieves the high level of adaptability required from the WINNER system concept through composition of functionalities from functional units with cohesive responsibility.

The main benefits of this concept for the system design are:

- Flexible design: possibility to easily investigate different protocol options and logical ordering of functionalities and faster performance evaluation of concurrent design proposals
- Efficient design: through increased re-use of functional units with cohesive functionality.
- Reduced complexity of the design: making dependencies between different functional units explicit helps to (i) understand (and question) their necessity and (ii) maintain the modular design approach.
- Reliable design: better testability of protocol software

The presented concept also opens up potential for an abstract description and with this the possibility of external configuration of the protocol stack and -layers via a formal description language, while the current status is a static management through external configuration of the composition and parameterization only. As indicated above, the layer and stack management tasks will be among the research issues for WINNER phase II. Another focus will be on the integration of Control and User Plane functionalities (already ongoing) in the context of Functional Unit Networks. The degree of commonalities between WINNER modes will have be further investigated, since keeping the number of options to the minimum possible will be one of the key issues to ensure moderate system complexity.

Other interesting -but lower-priority- research topics will be to study whether Functional Unit Networks can spread across different logical nodes and whether parameterization and configuration instructions can be sent to User Terminals or Relay Nodes over the Air.

5.3 The WINNER Logical Nodes architecture

5.3.1 Introduction

The main goal of the logical node architecture model is to assist in grouping functions, between which there may be a need for defining open interfaces. In particular, the logical node architecture needs to support all envisioned deployment scenarios for WINNER (as well as not yet foreseen deployment scenarios) without introducing too many logical nodes and/or interfaces. Note that the list of reference logical nodes presented here is preliminary, and logical nodes may have to be added, removed or combined during the development of the final WINNER architecture.

Definition:

"A Logical Node (LN) is defined by the service (or group of services) it provides towards other nodes (the provided service access points) and the service (or group of services) it requires from other nodes.

Identical Logical Nodes terminate an identical set of protocols and provide/require the same group of services (i.e. identical service access points). One physical node can comprise one or several LNs."

A physical node instead can comprise more than one logical node, e.g. one physical node can comprise a RANG logical node and an ACS logical node (Definition of RANG and ACS see below). Physical nodes can have different logical node configurations in different physical deployment concepts.

While the logical Nodes Architecture is assumed to be the same for all WINNER scenarios the specification of physical nodes can be different for the different scenarios as explained in the Conclusions in Chapter 9.

The WINNER logical nodes architecture is shown in Figure 5-4. It is shown that the LNs RANG, ACS and AR are located inside the network. One RANG or ACS can be connected to several BS.

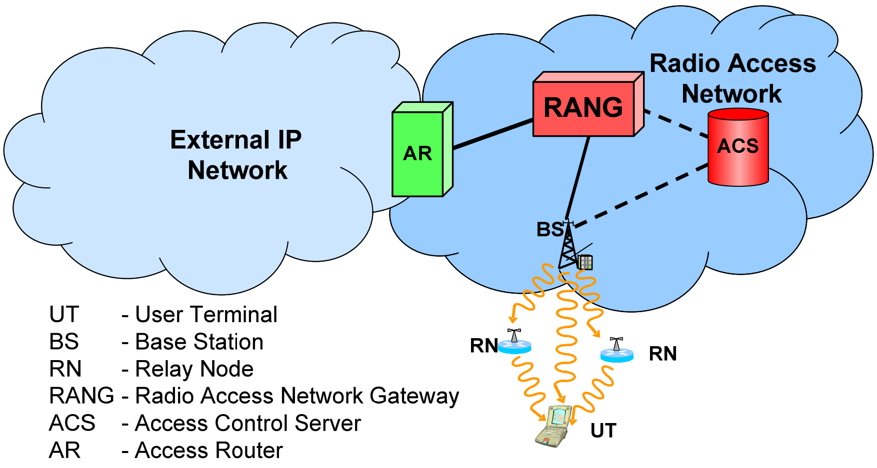


Figure 5-4: WINNER Logical Nodes Architecture

In the following the different logical nodes are defined in more detail.

5.3.2 User Terminal Logical Node

User Terminal Logical Node (UT_{LN}) is a logical node comprising all functionality necessary to communicate directly with another UT or the network, i.e. a BS or a RN.

5.3.3 Base Station Logical Node

Base Station Logical Node (BS_{LN}) is a logical node terminating the transport network layer protocols on the network side as well as mode specific radio protocols on the UT side.

5.3.4 Relay Node Logical Node

Relay Node Logical Node (RN_{LN}) is a logical network node with relaying capabilities that is wirelessly connected to a BS_{LN}, UT_{LN} and/or another RN_{LN}. Like the BS_{LN} it terminates the radio protocols (MAC and PHY) on the BS side as well as on the UT side (i.e. in case of more than two hops also towards subsequent RNs). The RN_{LN} further comprises all necessary functionalities to associate itself to the network (placed in the RLC-CP). To avoid unnecessary signalling between the RN_{LN} and the network the SLC functionalities are required on the uplink. Further the RN_{LN} will comprise the peer entity for the network side resource partitioning (RP) control entity (part of RRC) in order to receive and interpret the RP information coming from the central node. In case of more than two hops resource partitioning functionalities may be required to control the subsequent hops.

Remarks:

Another difference to a BS_{LN} is that it does not terminate the transport network layer protocols. Due to future work it might not be sufficient to classify only one RN_{LN}. The RN_{LN} may need to be further partitioned, e.g. depending on whether it is mobile or not (i.e. classified as a Fixed Relay Node (FRN_{LN}) or Mobile Relay Node (MRN_{LN})) or on what layer it performs the forwarding (e.g. classified as a RN with layer 3 routing capabilities ($RN3_{LN}$)). The number of necessary RN_{LN}s is currently under discussion and left for future work.

5.3.5 Radio Access Network Gateway Logical Node

Radio Access Network Gateway Logical Node ($RANG_{LN}$) is a logical network node terminating the mode independent RLC-UP protocols.

5.3.6 Access Control Server Logical Node

Access Control Server Logical Node (ACS_{LN}) is a logical network node that controls the access to the radio resources. It terminates the control plane protocols of the RLC and RRC.

5.3.7 Access Router Logical Node

Access Router Logical Node (AR_{LN}) is a logical IP layer node that performs the tasks attributed to an Access Router as defined in relevant IETF specifications [IETF]. In the WINNER architecture the AR_{LN} contains all functionalities of the IP Convergence Layer (CL).

5.3.8 WINNER System and Protocol Layer in the Logical Nodes

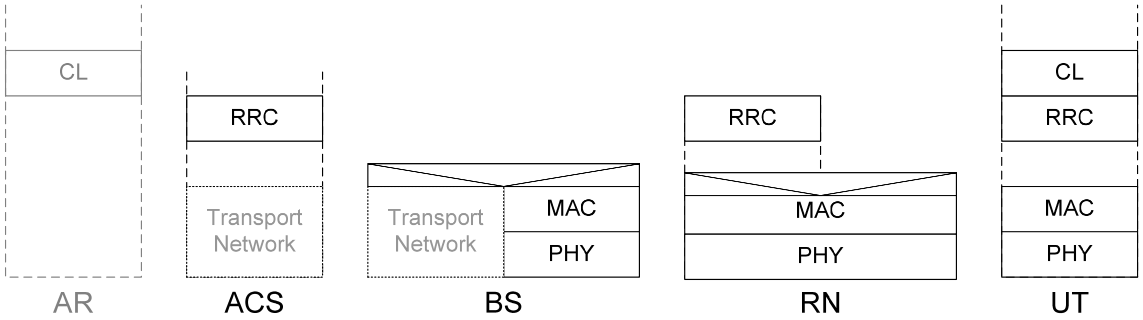


Figure 5-5: Logical Nodes Architecture in the Control Plane

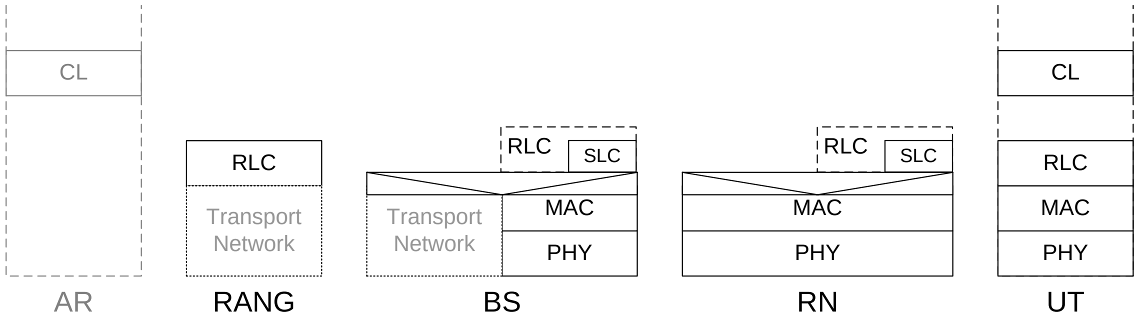


Figure 5-6: Logical Nodes Architecture in the User Plane

6 Protocol Details

6.1 Radio Resource Control

According to WINNER System architecture presented in Chapter 5, RRC is a pure control plane layer which is responsible to control the overall radio resources assigned to the WINNER system. The radio resource can be seen on different levels from the RRC perspective: Control of radio resources between WINNER and other systems, different WINNER operators, WINNER modes and between different RAPs of one WINNER system.

In general, RRC takes care of control aspects and higher layer signalling of Radio Resource Management (RRM) functions. In this section some of the main RRM functions within the scope of WINNER system concept are presented and discussed including Routing, Admission Control (AC), Load Control and Resource Partitioning. Inevitably, each of these functions has interactions with other RRM functions to optimize the overall system performance.

Two basic routing strategies, centralized and distributed, are presented and discussed in Section 6.1.2. In centralized strategy, route computation is performed in a central controller while in the distributed strategy, all network nodes between the source and the destination jointly perform route determination. The final decision on the routing architecture remains to be made based on further investigations and more advanced performance assessments. Also it is not finalized whether routing decisions should be made based on the uplink or the downlink and it remains for further investigation in the WINNER phase II.

Admission Control is another key RRM function to ensure the control of the system load aiming to exploit and maximise the available capacity. It is responsible for making access decisions in response to a user's session request based on a range of criteria including required QoS level, service class priority, bandwidth requirement, coverage, cost, etc. In a multimode WINNER system, AC has to be able to provide different load metrics for each mode and in different directions (uplink and downlink) and also in the adjacent cells for load sharing. In Section 6.1.2, the main basic functional blocks of the policy system model are presented. An interference-throughput policy based algorithm is proposed to make policy-based access control decisions taking place both at network side and at the user terminal based on PHY measurements, network priorities, terminal capabilities, user-defined QOS parameters, costs, etc.

Load Control in WINNER, alongside Admission Control and Routing, is a RRM function to balance the load in the system and avoid congestion. The main aspect of Load Control is to balance the load of each cell with the amount of resources allocated to them, in order to provide a fair geographical distribution of network resources and QoS. As a result, the chance of having uneven load distribution in the network can be minimized and high resource utilization should be achieved. Load control and its components are presented in Section 6.1.3.

Resource Partitioning in WINNER is a bit tricky since there are several control functions involved with radio resources and spectrum management. Each of these functions has different responsibility and fully or partially separate task to perform. The overall network performance in terms of resource utilization and spectral efficiency relies not only on individual performance of those functions but also (more importantly) the interactions between them. It is clear that any poor-coordinated interactions between these functions will deteriorate the overall performance.

Section 6.1.4provides an overview of Resource Partitioning in the context of the WINNER system as a whole, and further exploits the inter-cell resource partitioning. In addition, a flexible reuse partitioning method is presented.

6.1.1 Routing in the WINNER Relay-based Air Interface

6.1.1.1 Introduction

The objective of the routing function in the context of the WINNER systems is to optimise system performance (e.g. throughput). Routing functionality also provides input to, or has interoperations with, other RRM functions such as access-point selection, flow admission, load control and hand-off.

From the architectural perspective, two basic routing strategies are envisioned in relay-based systems: *centralized* and *distributed*. Under the centralized strategy, route computation is performed in a central controller, which normally possesses powerful processing capabilities and has knowledge on global network status. With the distributed strategy, all network nodes between the source and the destination (inclusive) jointly perform route determination. This strategy can function when no central controller is reachable, but its performance is normally limited by the processing capabilities of the network nodes and its knowledge of the network status. *In the WINNER air interface, both the centralised and distributed routing strategies are feasible to implement.*

Here we present both approaches; the final decision on the routing architecture remains to be made based on further investigations and more advanced performance assessments in the WINNER II.

In the following the assumptions as made in Chapter 4 are taken as basis. The air interface technology is a fixed-relay based with the maximum of two hops. Routing is done per user; however per-flow routing can be also considered as an option to facilitate more advanced load balancing and radio resource management schemes. Communication between BS and UTs follows a star topology, i.e., there are no links between FRNs, and each FRN communicates only with one BS (no meshing). We also assume that the uplink and downlink communications follow the same route. Routing decision can be made based on either uplink or downlink. At this point it is not finalized that the routing decisions is made based on the uplink or the downlink. This should be further investigated in the WINNER II. Finally, packet forwarding is performed in the MAC layer.

6.1.1.2 Centralized Routing Architecture

Centralized routing architecture in the WINNER is envisioned as shown in Figure 6-1. The route calculation (routing) resides at the Access Control Server (ACS), which is the logical node that controls the access to the air interfaces radio resources.

For the purpose of packet forwarding, the centralised route information needs to be distributed to the *Forwarding Information Bases (FIBs)* of BSs and RNs in order to facilitate their packet forwarding. It is worth mentioning that UTs are always sources in upstream and destinations in downstream, i.e., they do not need to forward packets. Therefore, no *FIB* is needed in a UT.

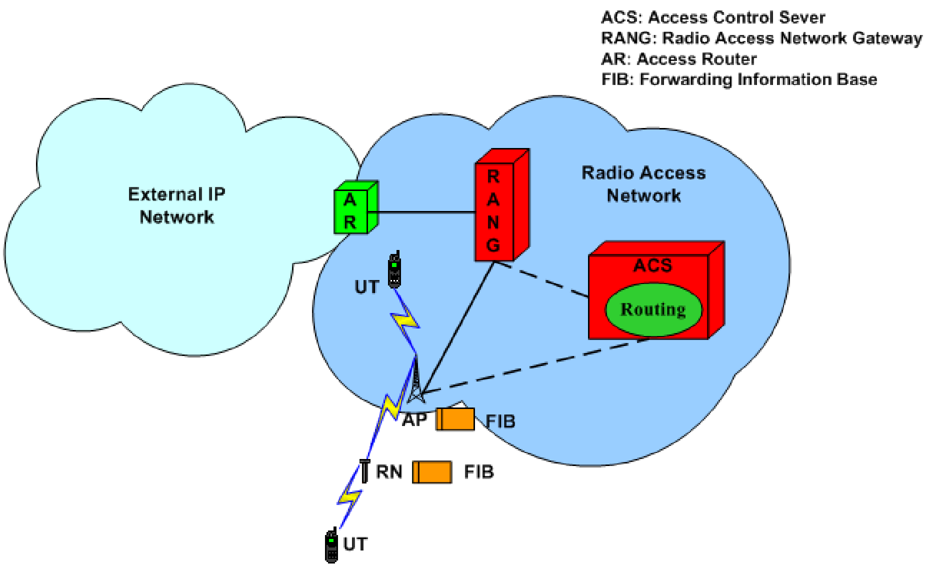


Figure 6-1: Architecture of the centralized routing function in WINNER

6.1.1.2.1 Functional Description

Generally, within the centralised routing functional module, there are three basic functional blocks (Figure 6-2), namely, *Route Discovery*, *Proactive Routing* and *In-use Routes Updating*, as well as a database, called *Candidate Routes Database.*

The *Route Discovery* is responsible for calculating initial candidate routes under the request of *Initial Cell/Mode Selection* during flow establishment. This is basically an initialization to proactive routing function. The *Proactive Routing* can be executed periodically or on demand in order to maintain the overall quality of the candidate routes in the database. It is able to perform cross-optimisation (if necessary) for the routes in the system. The main task of the *In-use Routes Updating* is to update the "in-use" flag of corresponding routes in the database under the requests of other RRM functions, e.g. *Admission Control*, and *Handover*. Each active flow has one in-use route but the *Candidate Routes Database* holds up-to-date candidate routes for all the active flows. Each flow can have multiple candidate routes in the database, and the in-use flag associated with each candidate route indicates whether the route is in use or not.

Measurements and flow statistics from *MAC Control Feedback*, and load information from *Load Control*, provides the required information to the *Route Discovery* and the *Proactive Routing*. The execution time scale of the *Proactive Routing* generally depends on system dynamics (e.g. user mobility, flow traffic volume fluctuations), and normally is fairly slow, e.g. once per several seconds. Moreover, if necessary, an adaptive time scale could be considered for the *Proactive Routing*, i.e., different time scales are adopted for users with different mobility or traffic characteristics.

It is worthy mentioning that the *Route Discovery* and the *Proactive Routing* are only responsible for providing candidate routes of flows. Other RRM functions such as *Admission Control* and *Handover* will eventually decide which of these candidates should be used, and then send requests back to the *In-use Route Updating* to update the in-use flag of corresponding routes in the database.

The candidate routes in the database can be output to other system functions, such as *Initial Cell/Mode Selection, Mobility Monitor,* and *Load Sharing.* The in-use routes in the database should be signalled to BSs and RNs by means of *Forwarding Information Distribution*, to update their *FIBs* for packet forwarding purposes.

Generally, a FIB is a database purely for the purpose of packet forwarding. Each entry of it consists of the minimum amount of information necessary to make a forwarding decision on a particular packet. The typical components within a FIB entry are the address of the destination (or flow ID) and the address of the next hop.

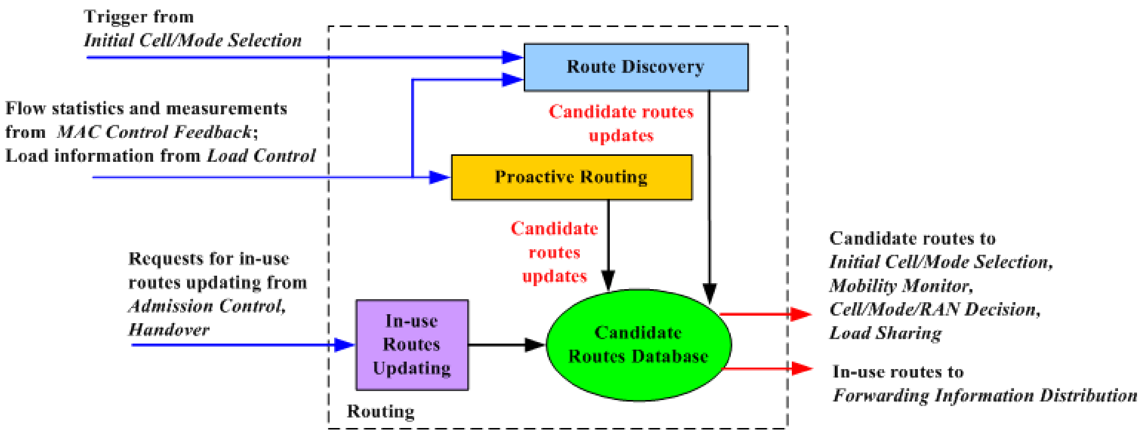


Figure 6-2: The internal structure of the routing functional module

FIBs are maintained by means of *Forwarding Information Distribution* based on the corresponding in-use routes in the centralised *Candidate Routes Database.*

In WINNER phase1, Maximum of two hops are envisioned for a connection, That means for downstream, the BS and RNs are supposed to do the packet forwarding, whereas for upstream, only the RNs act as packet forwarders.

In upstream case, RNs always forward packets to the BS and as such, the upstream *FIB* of each RN essentially has only one fixed entry, and therefore can be omitted. The distribution of the *FIBs* of network nodes is illustrated in the following figure.

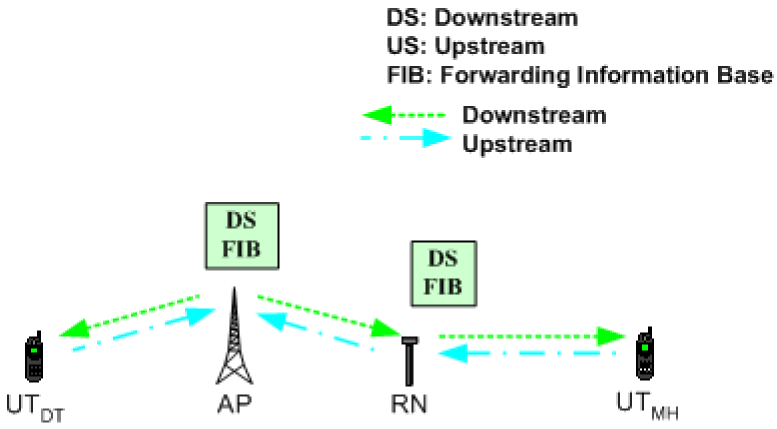


Figure 6-3: The distribution of the *FIBs* of network nodes

6.1.1.2.2 Qualitative Statement on System Complexity

As mentioned earlier, one major advantage of centralised routing strategy is that the cross-optimisation can be performed in order to improve the system performance. As a result, the centralised route calculation algorithm usually involves some degree of algorithm complexity. Nevertheless, thanks to the powerful process capability of the central node, sophisticated routing algorithms are normally feasible to implement. Moreover, heuristic algorithms can also be devised to notably bring down the algorithm complexity with small performance penalty.

The route calculation algorithm needs some input measurements, e.g. channel qualities between users and BSs/RSs, and the loading status of BSs and RSs. Fortunately, they are normally available already in the central node to be used by other RRM functions such as handover, admission control and load sharing. As a result, the centralized routing strategy is not likely to induce much extra complexity due to the input information gathering.

User candidate routes calculated will be stored in the Candidate Routes Database, and the corresponding forwarding information will be distributed to the relevant network nodes once the flow is established. Hence some system complexity will be brought to maintain the Candidate Routes Database. However, this is normally not a problem to the central node, especially based on the consideration that the route updates are happening in a fairly slow time scale.

6.1.1.3 Distributed Routing Architecture

In the distributed routing architecture, the UT makes the decision on the RAP (i.e., BS or FRN), it is going to be connected to, which is implicitly a routing decision. The routing decision is made in UT based on the information which is broadcasted by the relays and base-station in the network coverage area (see section 6.3.3.3). With respect to the logical nodes architecture as shown in Figure 5-4 this would mean that only a respective routing algorithm has to be placed in the UT, which of course is relaying on the information conveyed by each RAP during the broadcast phase.

6.1.1.3.1 Functional Description

FRNs and BSs include *link cost information* in their broadcast channel (BCH). Link cost information for the BS-RN link indicates the radio channel status (e.g., path-loss, interference) as well as the BS or FRN utilization (e.g., their traffic load). For the RAP-UT link, the link cost only indicates the utilizations cost of the RAP as the UT can extract the radio channel status from its measurements.

The UTs listen to the BCH and evaluate the optimal access point. Implicitly, the selection of the optimal server also results in a route to be followed toward the access point into the core network. Note that here, the selection of the optimal route involves radio channel status and traffic load.

Based on the downlink pilot signal measurements, the UT evaluates the cost of the air interface link UT-RAP for received pilots. From BCH the UT also extracts the costs related to the RAPs utilization as well as the cost of the RAP-BS link. By selecting the RAP with the lowest total cost the UT also selects an appropriate or optimal route. Hence, it can be seen that the UT selects not only the *best server*, but actually the *best access route*. The algorithm can be enhanced to consider additional parameters such as the user subscription attributes, etc.

One solution could be that the radio resources in each relay enhanced cell including the BS and the first tier of FRNs are administered from the BS. In this way, for example, if the traffic load increases within the coverage area of a RN, the BS can assign it additional resources at the expense of other RNs in the cluster. The BS would not have visibility into resource assignment/ scheduling within a RN (i.e., between flows / UTs). Similarly, the assignment of resources for competing BSs will be arbitrated by the central ACS entity. The ACS will not have visibility into cluster level resource assignments, which are controlled by the BSs.

6.1.1.3.2 Qualitative Statement on System Complexity

In the above centralized routing architecture, the signalling overhead related to the routing is limited to the cost information broadcasted by BCHs. UTs then make routing decision based on a simple decision making procedure to find the route with the lowest cost. Therefore the routing complexity here is mainly a function of the number of received BCHs. For cases with more than tow hops signalling overhead would be an important factor that should be further investigated. In such cases the decision making procedure would be also more complex.

6.1.1.4 Routing Interaction with Other RRM Functions

In this section we describe the interactions of the routing with other RRM functionalities in the WINNER air interference. To avoid confusion we use the terminology used in Section 6.1.1.2.

6.1.1.4.1 Interaction between routing and admission control

In order to make an appropriate *admission control* decision at the flow establishment, apart from flow QoS parameters, network load status etc., the possible routes for the new coming flow are also necessary. Therefore, prior to the *Admission Control* decision making, *Route Discovery* needs to be performed to find out candidate routes for the flow. If the flow is admitted, the in-use route for the flow is therefore chosen, and hence, *Route Information Updating* will be executed to send requests to the *In-use Route Updating* in order to set the in-use flag of the corresponding route in the *Candidate Routes Database*, and update the *FBIs* of evolved BSs and RNs based on the in-use route.

The interaction between *Routing* and *Admission Control* is shown as follows.

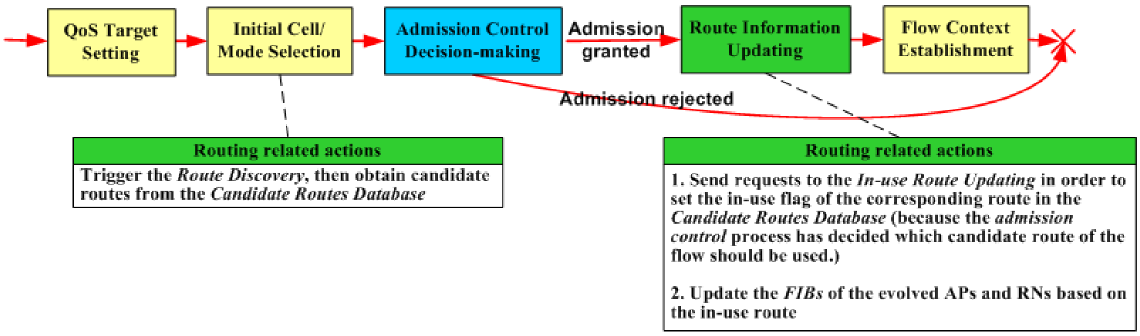


Figure 6-4: Interaction between *Routing* and *Admission Control* at the flow establishment

6.1.1.4.2 Interaction between routing and handover

In WINNER, the areas covered by BS and RN essentially comprise separate cells (i.e. RNs will have their own broadcast channels. For the definition of *cell*, please refer to [WIND31]). The handover in WINNER can therefore take place between BSs and RNs.

As mentioned earlier, the *Proactive Routing* is executed periodically or on demand to maintain the candidate routes in the *Candidate Routes Database*. *Mobility Monitor* keeps monitoring the quality of the candidate routes for individual flows. Once the *Mobility Monitor* finds the in-use route of a flow needs to be updated, it triggers a handover process. If the handover is successful, requests will be sent to the *In-use Route Updating* in order to modify the in-use flag of corresponding routes in the *Candidate Route Database*, and the *FBIs* of evolved BSs and RNs will be updated based on the in-use route. The interaction between *Routing* and *Handover* is illustrated in Figure 6-5.

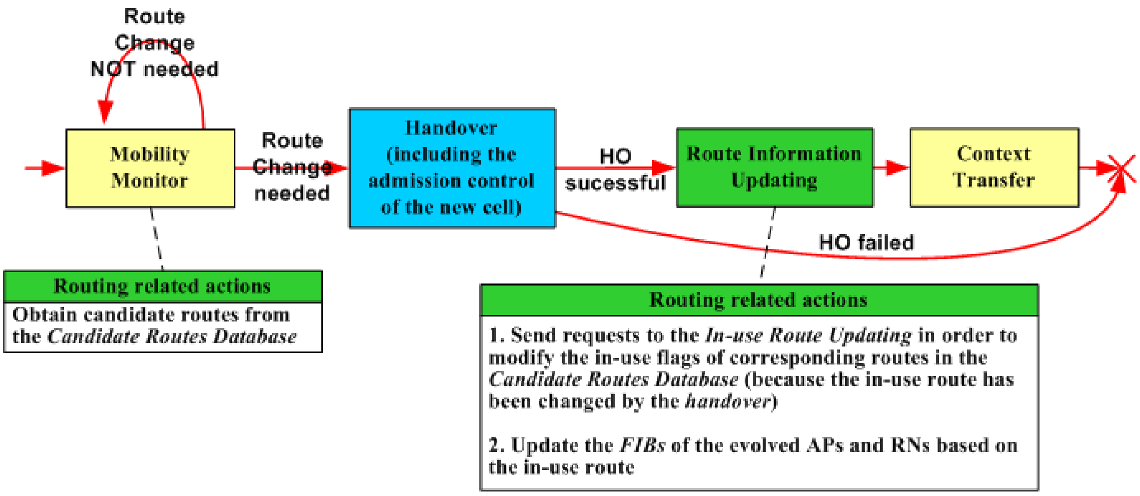


Figure 6-5: Interaction between *Routing* and *Handover*

6.1.1.4.3 Interaction between routing and load sharing

Load Sharing in WINNER is an important congestion avoidance control, and is responsible for the prevention of the system overloading by means of appropriately sharing loads between RNs and BSs. The *Load Sharing* periodically checks the candidate routes of flows. If it finds beneficial for the system (e.g. the overloading of some RNs or BSs can be avoided) to change the route of a flow, it will trigger a handover process. The routing related actions after the handover are the same as those explained in the last section, hence are skipped here.

The interaction between *Routing* and *Load Sharing* is depicted in the following figure.

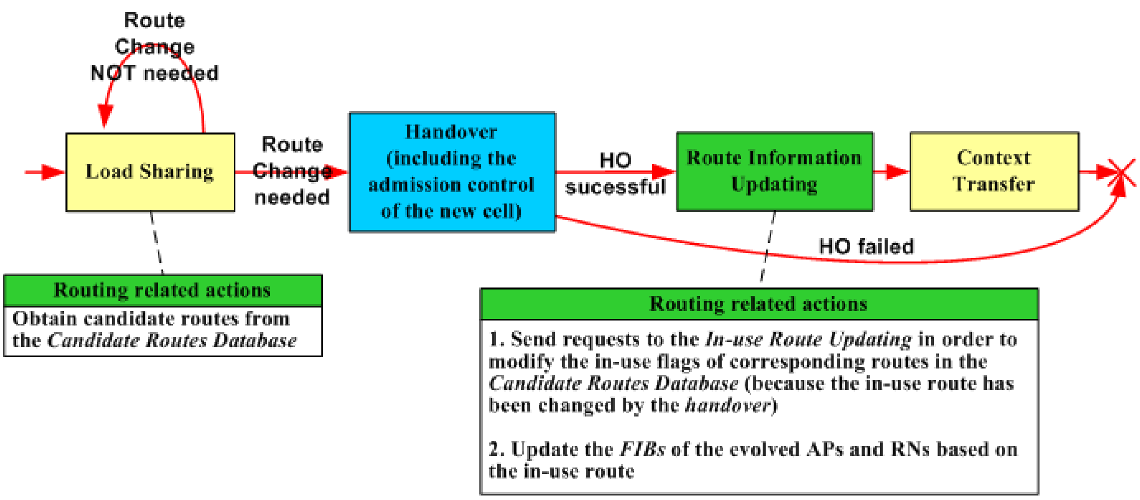


Figure 6-6: Interaction between *Routing* and *Load Sharing*

6.1.1.5 Conclusion

We propose *centralized* and *distributed* routing strategies for the WINNER air interface. Under the centralized strategy which is *network-oriented*, route computation is performed in a central controller, which normally possesses powerful processing capabilities and has knowledge on global network status. In the distributed strategy which is *user-oriented*, all network nodes between the source and the destination (inclusive) involve in performing route determination. This strategy can function when no central controller is reachable, but its performance is normally limited by the processing capabilities of the network nodes and its knowledge of the network status. *In the WINNER air interface, both the centralised and distributed routing strategies are feasible to implement.*

Further technical comparisons between centralized and distributed approaches remain to be done in WINNER II. This comparison would deal with the classical debate on the pros-and-cons of centralized and distributed decision making procedures which involves complexity, signalling overhead, response time to variations in different time-scales etc.. In the case of having at most two hops with no meshing and very slow network dynamics, it is simple to show that given perfect link cost information, the two presented methods results in a unique routing solution.

Since the objective of the routing function in the context of WINNER systems is to optimise system performance (e.g. throughput), instead of to maintain the network connectivity, the centralised strategy is potentially more capable to fulfil this objective. However, the computational complexity and signalling overhead of centralized approach should be taken into consideration. In WINEER phase I, maximum of two hops are envisioned. Therefore, the signalling overhead and delay of the centralised strategy (caused by the information gathering) would be acceptable. In addition, some other RRM functions require lots of measurements gathering, e.g. link qualities, and some of these measurements can be shared with the centralised routing function, and thus further mitigate its problem of information gathering.

Distributed routing approach reduces signalling overhead. Since the routing decision in this approach is made locally, potentially this approach can react with a lower latency in responding to changing conditions in the air interface. The distributed routing however may require a higher level of computational complexity in UTs. Such approach because of its distributed nature may also make implementation of the advanced traffic management functionalities harder than that of in the centralized architecture. The signalling overhead for this approach would be increased for cases with more than two hops and or meshing.

6.1.2 Admission Control

6.1.2.1 Introduction

Admission Control (AC) function belongs to the RRM function group and is identified as the key function to ensure the control of the load of the system aiming to exploit and maximise the available capacity. The admission control algorithm evaluates whether a new session should take place or a current one can be modified. Fundamentally, admission control is responsible for making access decisions in response to a user’s session request based on a range of criteria including required QoS level, service class priority, bandwidth requirement, coverage, cost, etc. From a network point of view, admission control tries to achieve the highest capacity and spectrum efficiency network usage. From the user perspective control admission algorithms may also assist the users giving the chance to control their choices in the session requests taking into account the conditions agreements to the operators in terms of QoS and associated costs by accepting or refusing any conditions related with the admission in a flexible way. Sophisticated admission control schemes are required to maintain QoS agreements and continuous connectivity in an environment where users can dynamically roaming between different wireless technologies or even between modes.

Due to the different nature of the traffic AC has to be able to differentiate the service classes for prioritisation. Real time services (typical delay sensitive services) and non real time services (less delay sensitive and background services) are treated in different ways when a new session is requested to the AC. For non real time services, AC and PS (Packet Scheduling) will cooperate together in order to manage the maximum throughput on the network ensuring the minimum required QoS and security levels. In a multimode system such as WINNER AC has to be able to provide different load metrics for each mode and in different directions (uplink and downlink) and also in the adjacent cells for load sharing. This way different AC can be deployed to each mode TDD and FDD.

In this chapter is presented a interference-throughput policy based algorithms either based on the network side but taking into account the user preferences, priorities and costs.

6.1.2.2 AC algorithms based

6.1.2.2.1 Wide band power based

Most of the current 3G networks such as UMTS AC algorithms are base on wide band power measurements carried out on the air interface in the own cell in both directions uplink and downlink and adjacent cells to control the load and the admittance of a new session or call. The prioritisation of different services types are done taking into account the services attributes (minimum required throughput, delay, etc).

$$P_{RxTotal} \le P_{RxTarget} + P_{RxOffset} \tag{1}$$

6.1.2.2.2 Throughput based

Throughput based AC algorithms are pretty simple by principle. The strategy is to sum the overall aggregated throughput of the different flows on paths and networks elements to control the admittance of the new session request. A new request for a service is admitted if the total load after admittance stays below a pre-defined threshold (2).

$$\eta_{old} + \Delta L \leq \eta_{new}, \tag{2}$$

In this sense AC throughput based are not technology dependant but service QoS have to be guaranteed. The main drawback of the effective throughput based AC lies in the fact that is does not exploit the statistical behaviour of the traffic sources and therefore might not optimise the RRM resources and be efficient. In order to improve the accuracy of the AC dynamic throughput (effective bandwidth) based algorithms are proposed such as LMGF (Logarithmic Moment Generating Functions) which goal is to dynamically determine in advance the effective bandwidth for each AC decision taking into account the number of traffic sources and their probability density function to occur such traffic pattern.

In the effective bandwidth methods the users are grouped into classes and an effective bandwidth value is assigned to each type of user somewhere between the mean and peak requirement. Then the required system capacity is estimated through the effective bandwidth values and the number of the users in each classes. Moreover effective bandwidth values depend on the QoS parameters and on the stochastic behaviour of the users' traffic as well. Effective bandwidth based algorithms are able to adapt continuous and dynamically to different radio environments conditions and provides trade off between decision efficiency and complexity [WIND31].

6.1.2.2.3 Policy based

Typically policy based AC is formed by a set of evaluating rules and instructions which allow the AC to take the final decision at the beginning of a session request. The evaluation of a policy is triggered by an event i.e. session request which results in a policy decision taking into account a set of input parameters (signal strength, network load, user preferences & priorities, velocity, traffic requirements, etc). These inputs parameters are then evaluated at both sides, UT and network side (i.e. ACS) in a IF…Then…Else fashion.

6.1.2.3 AC: policy based system

As state in the last subsection, the policy based AC scheme is responsible for making access decisions to a user's access request based on a range of criteria and input parameters including required QoS level, bandwidth requirement, coverage, cost, etc. From a network perspective, admission control aims to optimise the network resources usage allowing high capacity and spectrum efficiency. This can be achieved through load balancing between WINNER modes, enforcing for mode handover to meet the required QoS constraints. Sophisticated admission control schemes are required to maintain QoS agreements and continuous connectivity in an environment where users might dynamically roaming between WINNER modes. In the context of the admission control the goal is to admit the higher number of users as possible taking into account the system stability and users required QoS based on the several input parameters such as received signal strength, average bit error rate, current QoS level and the required QoS, access network coverage, cost of both current and requested networks, battery power of user terminal, user preferences made available to the provider.

Most of the current AC uses a network perspective based, i.e. mostly based on the PHY measurements disregarding the users priorities preferences, network priorities, commercial issues, etc. For this reason a joint user and network perspective is proposed in the AC.

6.1.2.3.1 IETF concept of policy-based algorithm

In a policy-based network the decisions are taken according to a set of policy rules. Therefore, it enables operators to define a set of strategic rules that will influence the network actions. In IETF, it is specified a framework to set a policy-based control in order to take admission control decisions. It is proposed to define two main elements for a policy-based network: Policy Decision Point (PDP) and Policy Enforcement Point (PEP). PDP brings back the policies from the policy repository and then, using this data, it carries out some policy interpretation. After taking a decision, PDF translates it into a format understandable by the PEPs. Finally PDF forwards the translated decision to all the PEPs. The PEP is an entity where the decisions are enforced. Next figure shows the basic framework; it is in the policy repository that is stored the data used by the PDP to take their decisions.

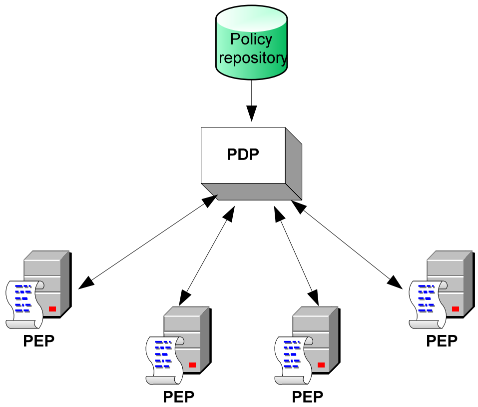


Figure 6-7 Policy algorithm architecture

The basic process of communication starts when PEP receives a request that needs a policy decision. When this happens, PEP sends the respective request to the PDP entity. PDP will look at the policy repository for data in order to take a decision. After having the right decision, PDP returns it to PEP, which will enforce the decision by accepting or denying the request.

Next graphic shows the overall factors that may influence the final PDF decision. As it can be seen, besides the network availability and the services demands, the commercial strategies also play an important role. Based on all of these factors, the PDF will be able to make the right decision related with each user at each time.

6.1.2.3.2 Policy parameters

A policy rule is specified in terms of a number of network and user parameters. These parameters are continuously updated and stored in a policy repository in both the network side and user terminal. The architecture of the policy based management system is further discussed in the section 6.1.2.4. Policy parameters include, coverage (best server), load measurements, C/I, power consumption, commercial conditions, user preferences and priority.

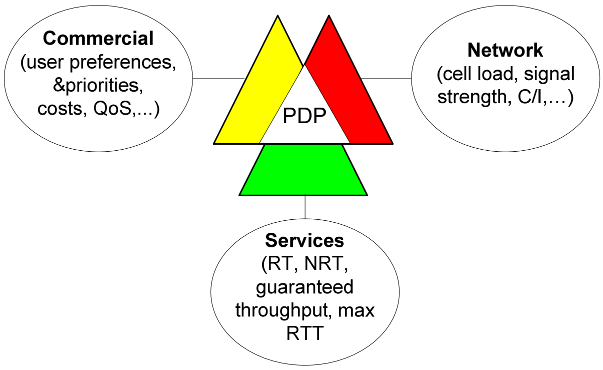


Figure 6-8 Input policy parameters

6.1.2.3.3 Service aspects

When a user intends to setup a session the policy engine on the UT shall evaluate a set of input parameters related with the traffic characteristics (pattern), required QoS and service type such as RT and NRT. After UT making the request, PDP engine should be able to sort out the RAN/mode most suitable according the terminal capabilities and service requested to the PDP at network side.

6.1.2.3.4 Network conditions

Another important factor is the network condition. In order to make a proper RAT selection afterwards a admission request PDF needs to know what is the current status of the network. For that purpose it needs to get information about the status of WINNER system and also in respect the network resources availability. This information is collected periodically updated in the repository through the different RRC messages both at network and at UT.

6.1.2.3.5 *Commercial issues*

Commercial issues shall take into account to decide on the allocation of the network resources, meaning to select the most suitable RAT/mode in which the user should be admitted to.

The commercial strategies should reflect the plans made by the operators to reach their business plans. Several rules are taken into consideration at PDP on these preferences such as cost of the network, time of the day, etc.

6.1.2.4 Policy system model

A policy system architecture is proposed to make policy-based access control decisions and network prioritisation/selection decisions in the network and at the user terminal respectively. Policy system architectures is based on the functional relationship between the point where the outcome of a policy is enforced, e.g. point in a network where admission control is enacted, and the point where the decision on whether policy conditions are satisfied is taken, i.e. the Policy Decision Point (PDP).

For WINNER the PDP for network access control is envisaged to be implemented in the ACS server using an open interface to the enforcement points (PEPs) since the most important information is stored, maintained and updated in the repository database such as cell load or aggregated throughput of different available RAT/modes, L2 buffer load, Rx power level, UT Tx power, location, etc. All this information has to be kept updated in the repository database. The architectural approach below also introduces the possibility to update the policy rules through the network. At UT side the PEP is located at the user terminal, which is typically powerful enough to run a PDP restricted to their capabilities to perform measurements. Adaptive nature of the policy rules within the PDPs is expected to perform load balancing, perform handover across the WINNER and legacy systems allowing an optimisation of the maximum carried traffic over the spectrum available. However the policy rules have to be carefully designed to avoid policy conflicts which could compromise system stability as the network load levels change.

6.1.2.4.1 *UT side*

Policy engine mechanism to assist the user requirements and preferences should be available at the user terminal in order the user express not as a simple set of parameters and values but also clear rules of choice committing the conditions under those ones the request should be carried out. This mechanism requires automatic and adaptive negotiation from/to network (interaction with PDP engine at network side) in terms of user preferences and associated costs, reliability, application delays, application minimum required QoS, etc.

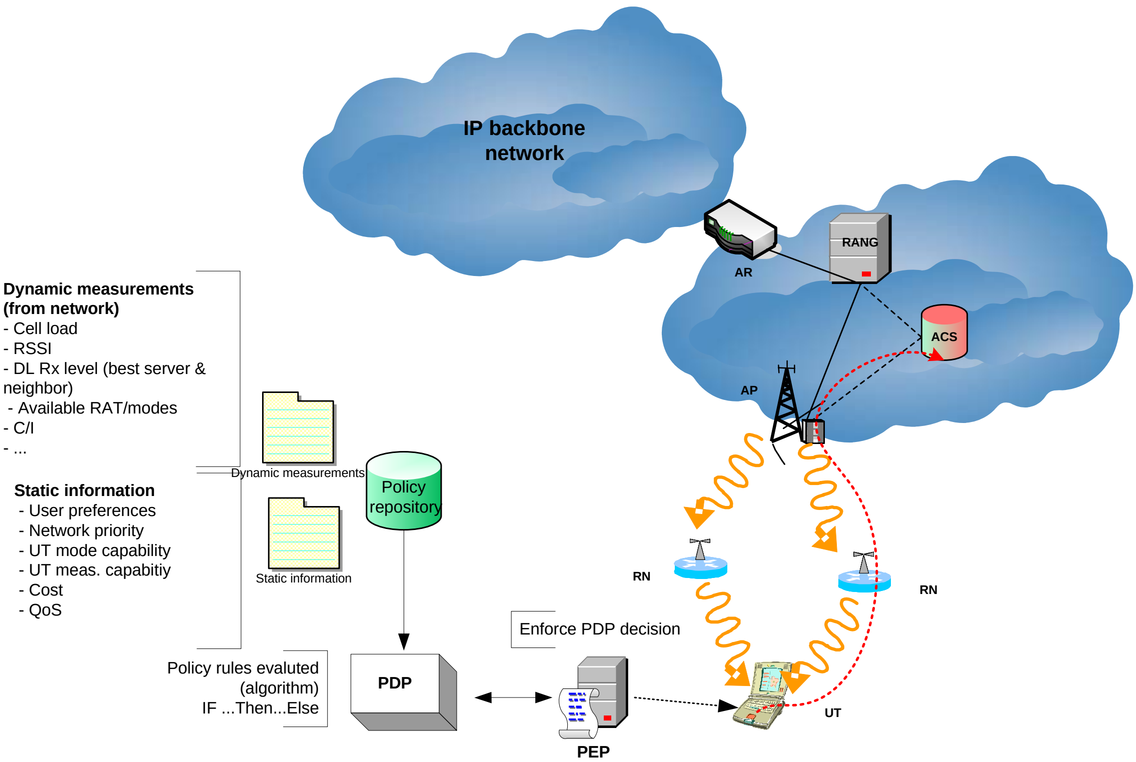


Figure 6-9 Policy architecture: UT side

6.1.2.4.2 Network side

Currently 3G networks are mainly based on the power measurements on the air interface to control the admittance of a new call or session request disregarding sometimes other important input parameters user centred such as QoS required, user preferences, costs, and networks priorities. When a new session request is placed over the network the policy engine at UT checks for the available prioritised list of the RAT/modes and cell load measurements mainly. If the current cell load level is higher enough the policy engine at network side may admit the new request if there are available modes after checking the estimated cell load of the neighbour cells in order to offer the required QoS or otherwise simply deny the request.

In the end and in the context of admission control, the resulting action at PDP will be the admittance of users into the requested network or the force handover either to other mode or legacy systems. The set of the parameters made available to the policy engine comprise for example the received signal strength at the requested BS, average BER, current QoS level and the required QoS, radio coverage, cost of both current and requested networks and user terminal battery power.

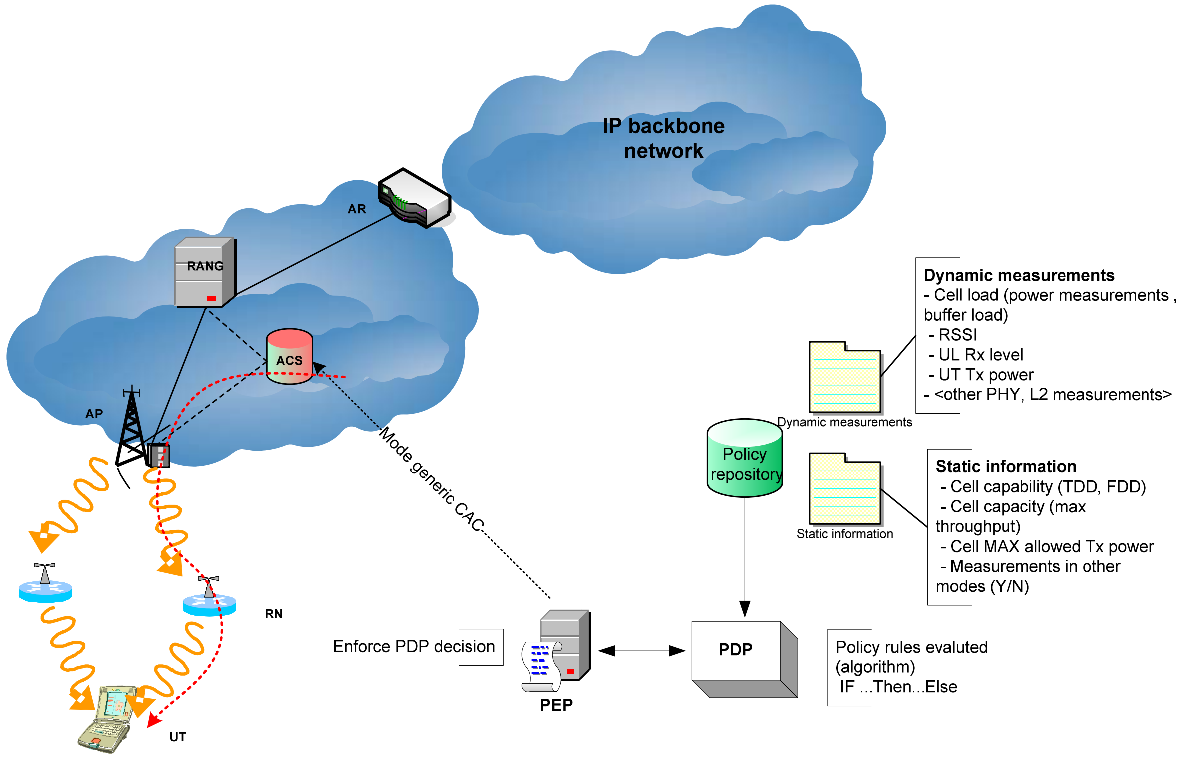


Figure 6-10: AC policy architecture: network side

6.1.3 Load Control

Load Control in WINNER is considered as one of the main functions under Congestion Avoidance Control alongside Admission Control and Proactive Routing as shown in Figure 6-11.

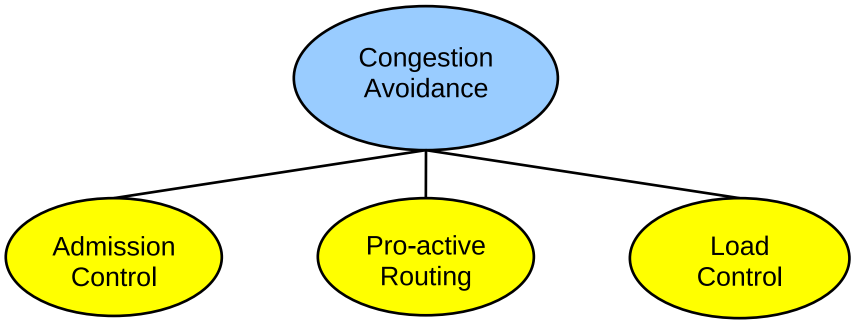


Figure 6-11 Load control as a congestion control mechanism

We can further divide this function based on the load situation: In under-load situations, the main aspect of load control is to balance the load of each cell with the amount of resources allocated to them, in order to provide a fair geographical distribution of network resources and QoS. As a result, the chance of having uneven load distribution in the network can be minimized and high resource utilization should be achieved. Resource partitioning plays a big role here and is considered one of the main actions of the load control in this kind of situations. We will discuss the resource partitioning in detail in 6.1.4.

Over-load situations however could still happen when the amount of total resources available in the network service area is not enough to provide sufficient QoS for existing active users. In this case, load control takes preventive/reactive mechanisms to bring the system back to normal. Figure 6-12 illustrates the procedure of load control: First, the load of each cell is calculated. If the cell load for all the cells remain below a predefined threshold, then it is an under-load case otherwise it is at least an overload situation for one cell. Different actions are taken according to the system load situation. If the system is in under-load situation, load sharing is performed. Load sharing is a set of actions in load control which tries to continuously balance the load of each cell with its allocated resources. On the other hand, if the system is overloaded, preventive/reactive load control is executed to take a set of more drastic actions to bring the overload cells back to normal.

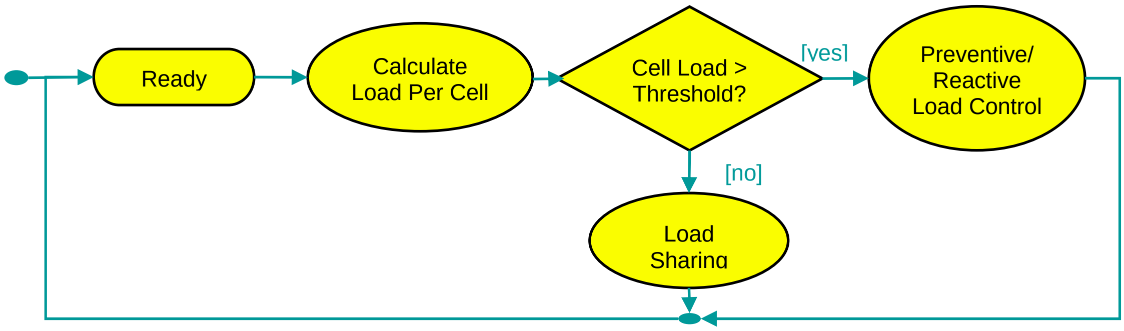


Figure 6-12 the procedure of load control

6.1.3.1 Load Sharing

Within the WINNER system architecture, the existing functions dealing with spectrum utilization, interference management and load balancing are mainly spectrum control (including spectrum assignment) [WIND76], and constraint processing [WIND210].

In the process of resource utilization and load balancing as a whole, the load sharing tries to harmonize the process in such a way to achieve efficient use of spectrum under all sorts of traffic conditions.

The main purpose of spectrum assignment function [WIND63] is to assign and/or partition the available spectral resources between overlapping WINNER RANs in a geographical area. One of the functions for spectrum assignment is to solve in a sense the same problem as traffic balancing between networks by balancing the network resources according to the traffic. The spectral resources assigned to the networks can be portions of spectrum or time.

After resource negotiation between networks took place, a portion of spectral resources in terms of spectrum and/or time will be dedicated to each network for a period of several minutes.

The final resource allocation in each network is done by resource scheduler in both adaptive and non-adaptive manner [WIND210]. It takes place at MAC layer of BS/RN by means of second level of mutual negotiations among cells within same network through constraint processing block. This can be seen as some kind of dynamic resource allocation which has shown good performance in conventional networks under low to medium traffic conditions [WIND33].

It is a well-known fact from conventional networks that as load increases, the performance of dynamic allocation degrades and resource partitioning by some central entity becomes important to achieve high spectral efficiency. The Load sharing tries to address this problem and provide a flexible structure for spectrum utilization in WINNER RANs in which high efficiency could be achieved under any traffic circumstances. In particular, it will fill up the gap in spectral efficiency for medium-high traffic.

Another important point is the time scale of resource allocation modules in WINNER RANs. While long-term spectrum assignment works periodically in the order of several minutes, and short-term spectrum assignment in the order of several tens of seconds, the resource scheduler in MAC has to work much faster (few hundred microseconds) to support adaptive scheduled flows and avoid excessive delays. The load sharing can fill this timing gap as well and hence provide smoother (in time) load balancing by working in the order of a second.

The time scale of load sharing should be slower than that of the spectrum assignment in order to adjust resource utilization according to traffic circumstances and distribution. On the other hand, it should be long enough to avoid unnecessary changes in the resource partitions. This longer time scale can also let the network benefited from all enhanced techniques with shorter time scale, e.g. adaptive scheduling and link adaptation, (H)ARQ, and beam forming. Load sharing only applies after the effects of these techniques have been settled down. Therefore, if the total received power of a cell averaged over the load sharing period (e.g. over one second) exceeds some threshold, it means the resource partition for this particular cell does not fulfil the load demand of the cell. This will trigger the load sharing function to reshuffle the resources and readjust resource partitioning accordingly.

Another way to trigger load sharing is by Service Level Controller (SLC) requests e.g. by SLC buffer tokens, which reflect the needs for extra resources in order to maintain the QoS requirements for existing active users. This is also after the SLC and the resource scheduler fail to adjust available resources within current resource partition.

In the beginning of network operation, the whole resource pool assigned by spectrum control could be shared by all cells (i.e. starting with 100% dynamic resource allocation). Load sharing process can be started in an event-triggered manner when the traffic increases and then perform periodically afterwards. It will resolve the conflicts in the negotiation process for low-medium load and then will replace mutual negotiations by resource partitioning as load increases.

Figure 6-13 shows the overall structure of resource allocation process and interactions between spectrum allocation, constraint processing and load sharing. Although resource partitioning is the major task for the load sharing functionality, it may take other actions such as handover if needed.

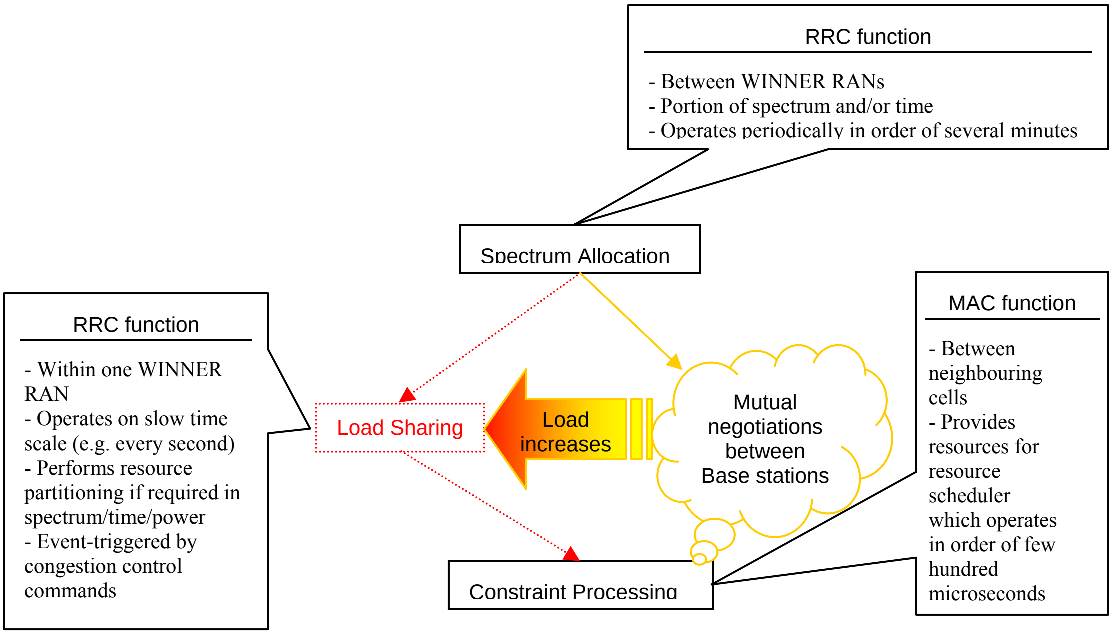


Figure 6-13 Load sharing interactions

6.1.3.2 Preventive/Reactive Load Control

When Load Sharing fails to bring an over-load situation back to normal, another set of actions may be taken by the load control. These are more drastic actions including tolerating QoS, dropping flows, etc. It can also trigger the short-term spectrum assignment [WIND63] to request for extra spectrum from another WINNER operator. Now, there are two strategies here for the short-term spectrum assignment: the first is to use the extra spectrum locally in the overloaded cells, and the second is to add the extra spectrum into the resource pool and perform resource re-partitioning accordingly. There are some issues to be considered for both of these approaches. For instance, local use of extra spectrum is much simpler and does not need any changes in other cells' partitions. On the other hand, the second strategy needs much less extra spectrum, as it reuses the extra spectrum throughout the network and thus has higher resource utilization efficiency.

The other issue to be considered is the possibility of roaming for WINNER users into the coverage of different spectrum partitions or different operators. The WINNER spectrum range is most likely too wide compared to user's hardware spectrum range.

An important issue in the load control is how to set the load threshold appropriately to achieve maximum spectral efficiency. As mentioned earlier, this threshold determines whether to perform load sharing or reactive load control and therefore it is crucial.

Aside from the load control, some other RRM functions, e.g. admission control, handover, and routing, also take different actions based on system load situations. Cell load also influences resource allocation which is a MAC functionality. Therefore it is very important to define the system load in an appropriate and measurable way.

The load in WINNER systems has different perspectives in different layers. For instance, in RLC layer load situation of a cell can be monitored by buffer status of its active users and delay performance of different flows. We can easily find when buffers are getting congested and which part of the network it happens but it is hard to quantify the system load in this way. In MAC layer, average interference level in chunks prior to resource assignment could be an indicator for the load. In a longer time scale, the frequency of re-transmissions is another indication of over-load situation.

Measuring cell load looks more complicated in WINNER systems than legacy counterparts like UMTS because of different resource dimensions we use especially spatial domain. If we assume RoT (Rise over Thermal noise) level to be the definition of the cell load in UMTS, finding the equivalent in WINNER systems remains for further study.

6.1.4 Resource Partitioning

In WINNER systems there are several control functions involved with radio resources and spectrum management. Each of these functions has different responsibility and fully or partially separate task to perform. The overall network performance in terms of resource utilization and spectral efficiency relies not only on individual performance of those functions but also (more importantly) the interactions between them. It is clear that any poor-coordinated interactions between these functions will deteriorate the overall performance.

Nevertheless, all these functions are involved in the resource partitioning process in general, with different granularity and different time scales. To give an overview of resource partitioning in the context of the WINNER system concept, it is useful to categorise these functions based on their operation time scales. We can categorize resource partitioning in long, medium and short time scales as illustrated in Figure 6-14.

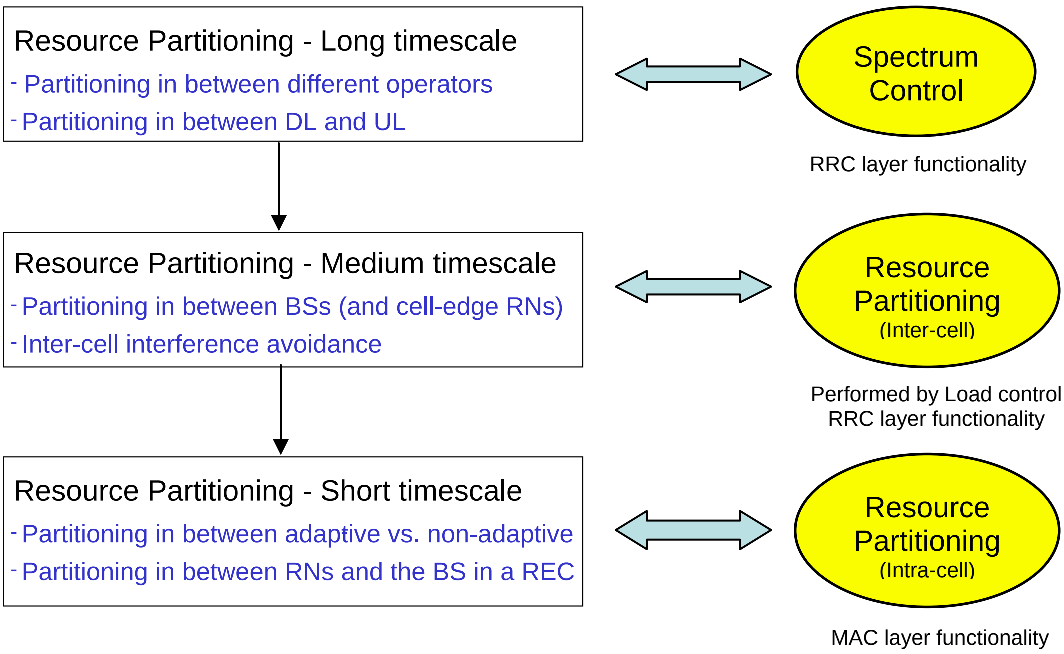


Figure 6-14: Resource partitioning time scales and mapping to control functions

6.1.4.1 Long timescale Resource partitioning

Long timescale resource partitioning is performed by Spectrum Control [WIND63]. Spectrum control is the main RRM function for coordinating flexible spectrum use and sharing between WINNER RANs and also with other radio access systems with different radio access technologies. Spectrum control has two components, spectrum sharing which coordinates the sharing of spectrum with systems using other RAT, and spectrum assignment, providing flexible use of spectrum between multiple WINNER RANs. Spectrum assignment itself is further divided into long-term spectrum assignment providing slowly varying spectrum assignments for large geographical area and short-term spectrum assignment providing short-term, local variations to the large-scale solution. Long-term spectrum assignment partitions the overall resources between different operators and then later on, partitioning will change on very long timescales (in the order of months or even longer). On shorter timescales (possibly several hours) short-term spectrum assignment can adjust the available spectrum to the load situation or change the UL/DL asymmetry. These actions however need to be triggered by load control (see 6.1.3) to avoid unnecessary changes (possibly oscillation) in the spectrum partitioning. Details of spectrum control and its components are provided in [WIND63].

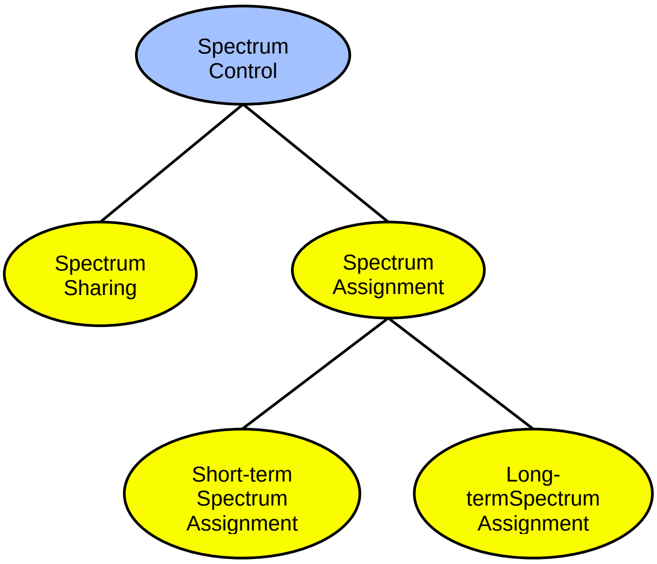


Figure 6-15: Spectrum control and its components

Short-term Spectrum Assignment particularly needs more attention because it is more or less overlapping with load control (medium timescale resource partitioning) both in timescale and functionality. Short-term spectrum assignment provides:

- Short-term, local (cell-specific) deviations and adjustments to the long-term spectrum assignments
- Reacts to local, short term demands, increasing the flexibility in the spectrum assignments
- Triggered by load control, or at the beginning of new spectrum assignment period (to further optimise the result of long-term spectrum assignment)

Since reacting to local load variations by increasing spectrum maybe unnecessary and reduce spectral efficiency, this action must be taken after Load sharing fails to balance the load with available spectrum. Therefore, the best is to trigger Short-term spectrum assignment by preventive/reactive load control, see 6.1.3.

6.1.4.2 Medium timescale resource partitioning

Medium timescale resource partitioning is performed by Load Control. As mentioned in section 6.1.3, load control includes load sharing function whose purpose is to balance the load in each cell with the amount of resources provided by resource partitioning so that the resources are efficiently utilized. Load sharing provides a centralized control on the resource partitioning over the cells (inter-cell partitioning) whereas resource partitioning performed by MAC (short time scale, on super frame basis) can be regarded as intra-cell partitioning.

The aim of inter-cell resource partitioning is to prevent any cell from becoming overloaded as the result of load fluctuation and inter-cell interference, by means of balancing the cell load with its available resources. At the resource assignment instant in Resource Scheduling (MAC layer), a normal load situation is when good CQI chunks for both adaptive and non-adaptive flows are available and will be assigned. In this case, re-transmission level is acceptable showing interference caused by interfering cells is low. An overloading situation can occur in several ways. We can identify the hard-limit overloading as when all chunks within the allocated resource partition are being assigned already. The soft-limit overloading is when there are some chunks not being assigned but interference level in those chunks are very high. Therefore to overcome interference and provide required SINR, high transmission power is required even for low transmission rates, which in turn may cause even further inter-cell interference. For uplink, user transmission power headroom may not be sufficient either to provide the required SINR. As a result, re-transmission level goes high which should have been prevented by an appropriate resource partitioning.

6.1.4.2.1 Flexible reuse partitioning method

Inter-cell resource partitioning can be performed in a variety of schemes. Main partitioning schemes are discussed and compared in [WIND33]. Basically, inter-cell resource partitioning can be categorized as fixed, dynamic and hybrid partitioning. In fixed partitioning schemes, a high degree of separation provides robust interference avoidance. It is suitable for high traffic situation with stable load pattern otherwise has poor resource utilization. Dynamic partitioning, on the other hand, is suitable for low traffic with varying pattern but it has low resource utilization in high traffic situation and also has weakness against inter-cell interference. Hybrid partitioning is the combination of fixed and dynamic partitioning approaches to gain benefit from their advantages.

From cell coverage point of view, Reuse Partitioning is a method which implements different reuse pattern per cell. In principal, it divides cell coverage into zones with different reuse factors.

In the following, we propose a method called flexible reuse partitioning which is suitable for the WINNER system concept. Figure 6-16 illustrates some basic aspects and terms used in this method for the single-hop deployment case.

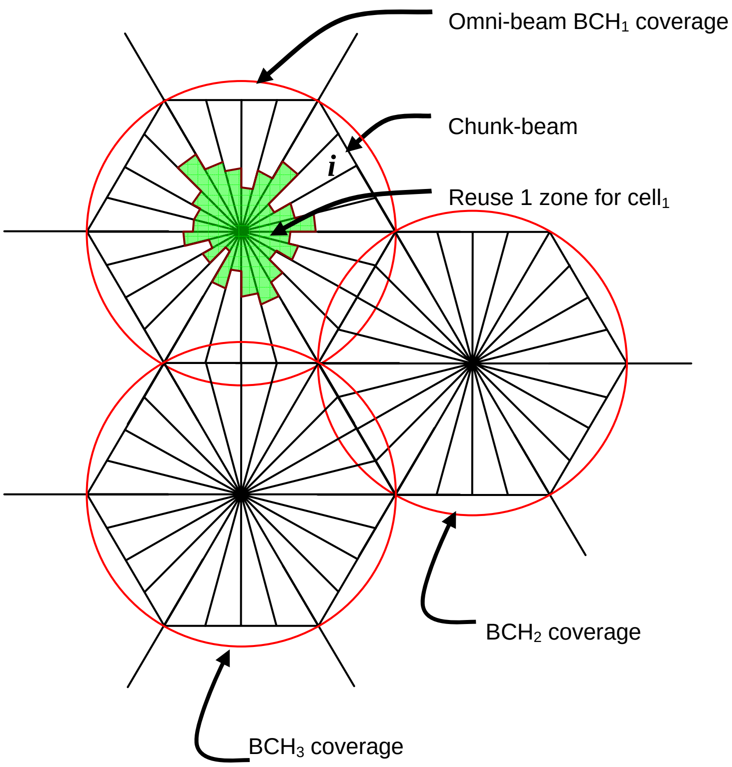


Figure 6-16: BCH coverage, reuse 1 zone and chunk-beams

We assume a cell is recognised by its Broadcast Channel (BCH) coverage area. The BCH is part of the preamble in a MAC super frame and should be broadcasted on an Omni-beam so that all the users could receive and detect the required cell information. The preamble should be broadcasted on resources that are available everywhere in the network coverage area.

A Chunk-beam is the smallest spatial (directional) component of the resource pool in WINNER systems as illustrated in Figure 6-16. Just for simplicity, we have shown chunk-beam as a narrow sector in azimuth direction. It should be noted that the flexible reuse partitioning method presented here does not perform partitioning on the chunk-beam basis since different flows may use the antenna resources in different ways and therefore the partitioning on that level may not be efficient.

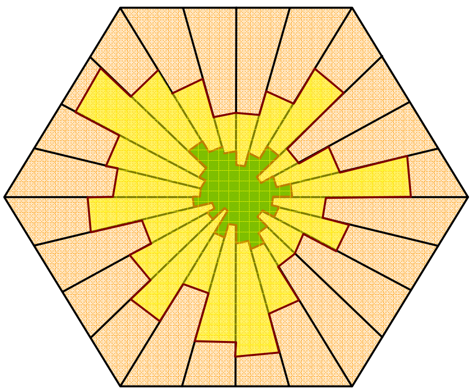
In spatial domain we perform the partitioning in a co-centric manner by partitioning the cell coverage into co-centric zones with different reuse factors as illustrated in Figure 6-17. We define reuse 1 zone as the core of the cell (close to the BS) which will be exempted from resource partitioning restrictions. In ideal case (i.e. complete shielding), reuse 1 zone criterion for cell$_1$ is:

$$\sum_{i=1, i \neq j}^{N} RSS(BCH_i) = 0 \tag{3}$$

where *RSS(BCH$_i$)* denotes the received signal strength of the BCH of cell$_j$ and *N* is the total number of cells. This means reuse 1 zone has to be completely isolated from other parts of the network. Of course it is not a realistic assumption and if such area exists, it would be in the immediate vicinity of the BS. We replace the ideal criterion in (3) with a practical condition:

$$RSS(BCH_1) \Big/ \sum_{i=1, i \neq j}^{N} RSS(BCH_i) \geq Threshold_{\mathrm{Reuse1}} \tag{4}$$

The condition in (4) extends the reuse 1 zone and allows inter-cell interference to some extent based on $Threshold_{Reuse1}$ value. In other words, the amount of intercell interference in this zone can be adjusted by $Threshold_{Reuse1}$. We can further extend this definition to zones with higher reuse factors that gives more protection against inter-cell interference. For instance, reuse 3 avoids co-channel interference from the first tier of neighbouring cells and reuse 7 avoids two tiers of neighbouring cells and provides a high degree of resource separation. A cell with different reuse zones is illustrated in Figure 6-17 where green zone is the area exempted form partitioning (reuse 1), and yellow and orange zones have reuse 3 and reuse 7 partitioning respectively.



Orange zone criterion for cell$_j$: (e.g. Reuse 7)

$$RSS(BCH_1) \Big/ \sum_{i=1, i \neq j}^{N} RSS(BCH_i) \geq Threshold_{\mathrm{Reuse7}}$$

Yellow zone criterion for cell$_j$: (e.g. Reuse 3)

$$RSS(BCH_1) \Big/ \sum_{i=1, i \neq j}^{N} RSS(BCH_i) \geq Threshold_{\mathrm{Reuse3}}$$

Green zone criterion for cell$_j$: (Reuse 1)

$$RSS(BCH_1) \Big/ \sum_{i=1, i \neq j}^{N} RSS(BCH_i) \geq Threshold_{\mathrm{Re\,use\,1}}$$

Figure 6-17: Different reuse zones per cell

In the flexible reuse partitioning method, a central partitioning entity (e.g. located in the ASC) allocates subsets of recourses per each reuse zone to every cell according to the respective reuse pattern in a similar way as if the whole cell was using single reuse factor. Therefore every cell knows its available resource subset per each reuse zone already prior to resource assignment in MAC. Obviously there would be some overlapping between resource subsets in each cell but in fact that has no negative impact on the resource assignment. For instance, in the resource assignment in MAC, we start with assigning the resources to users in orange zone first. The resource subset in orange zone is always smaller than yellow zone and so on because of its bigger reuse factor. When the assignment for users in orange zone is finished, we carry on with the resource assignment of yellow zone users and then we use all the resource pool defined by spectrum assignment (see section 6.1.4.1) left for green zone users. One of the main advantages of this method is that there is no need for re-partitioning during network operation and thus there is no need for signalling the partition subsets either since it is fixed. The flexibility of the method comes from the fact that by changing reuse partitioning thresholds (see Figure 6-17), we move users in between different reuse zones and balance the load with available resource subset per zone accordingly. Reuse partitioning thresholds are automatically adjusted by the load sharing function based on the load in each cell e.g. by continuously monitoring the level of re-transmissions for users in each zone and then signalling to the constraint processing in MAC for the resource assignment. This is actually the only signalling required in this method. It is also possible to perform cell breathing by adjusting the BCH transmit power. In high traffic situation, the load sharing may reduce the BCH power of congested cells if their neighbouring cells are under-loaded. Therefore the amount of recourses allocated to the congested cell will be used by less number of users and the excess load will be passed to the under-loaded neighbouring cells. It can also increase the BCH power for under-loaded cells to balance the load with available resource subsets.

The flexible reuse partitioning can be extended to provide resource subsets to RNs as well. We differentiate between RNs depending on their position in respect to the BS. A Relay-enhanced cell (REC) includes RNs which are well inside the coverage area of the BS as shown in Figure 6-18a. As a result, the inter-cell interference situation in the RN's coverage area is not much different from that of the BS. Therefore resource partitioning is performed by MAC at the base station for those RNs on super frame timescale basis (intra-cell resource partitioning). On the other hand, a relay at the border of the BS's coverage (cell-edge relay, Figure 6-18b) is treated just like a separate cell from resource partitioning point of view.

Nevertheless, in all cases of RN deployments, reuse 1 criterion is applicable i.e. partitioning restrictions are not applied to green zones in Figure 6-18.

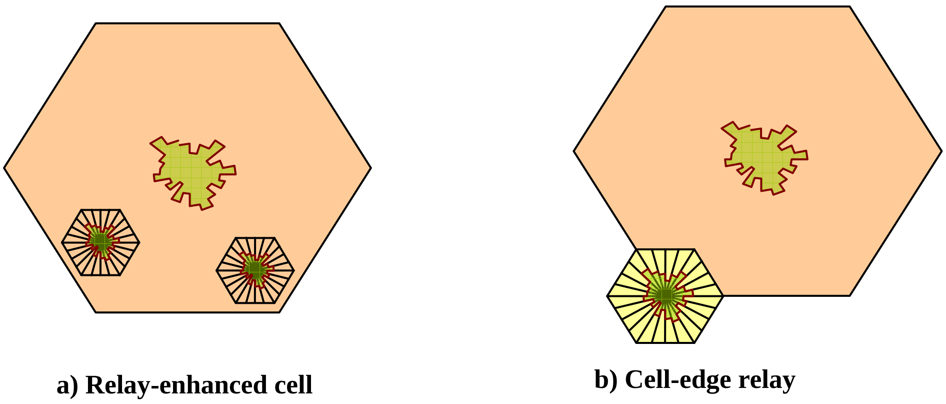


Figure 6-18: Resource partitioning for different RN deployments

6.1.4.3 Short timescale resource partitioning

The finest level of resource partitioning which we refer to as intra-cell partitioning in Figure 6-14 is performed by MAC with granularity of 1 Frame (out of 8 in a super-frame) on the time-axis and 1 chunk on the frequency axis [D2.10]. The resource partitioning in MAC partitions the super-frame into sets used for adaptive, non-frequency adaptive and DAC transmissions, as well as chunks set aside for uses by RNs and BS-to-RN links.

In the following, a simple example of intra-cell resource partitioning is given. Four different periods are defined in the course of two consecutive frames each consisting of an UL and a DL period. The first period is dedicated to BS DL transmissions towards both RNs and UTs, the second period is dedicated to UT UL transmissions, either towards the BS or towards their respective RNs, the third period is dedicated to DL receptions of user terminals from either the BS or the connected RNs, and finally the fourth period is dedicated to UL receptions by the BS from UTs and RNs.

On top of this basic scheme, a more fine-grained partitioning of the resources can be defined as shown in Figure 6-19. During the periods controlled by the RNs (i.e. second and third period mentioned earlier), each RN is assigned a subset of the overall available resources. A subset of the resources could also be assigned to a group of the RNs. This subdivision is expected to be performed in the frequency domain only. Details of intra-cell resource partitioning and constraint processing are provided in [WIND210].

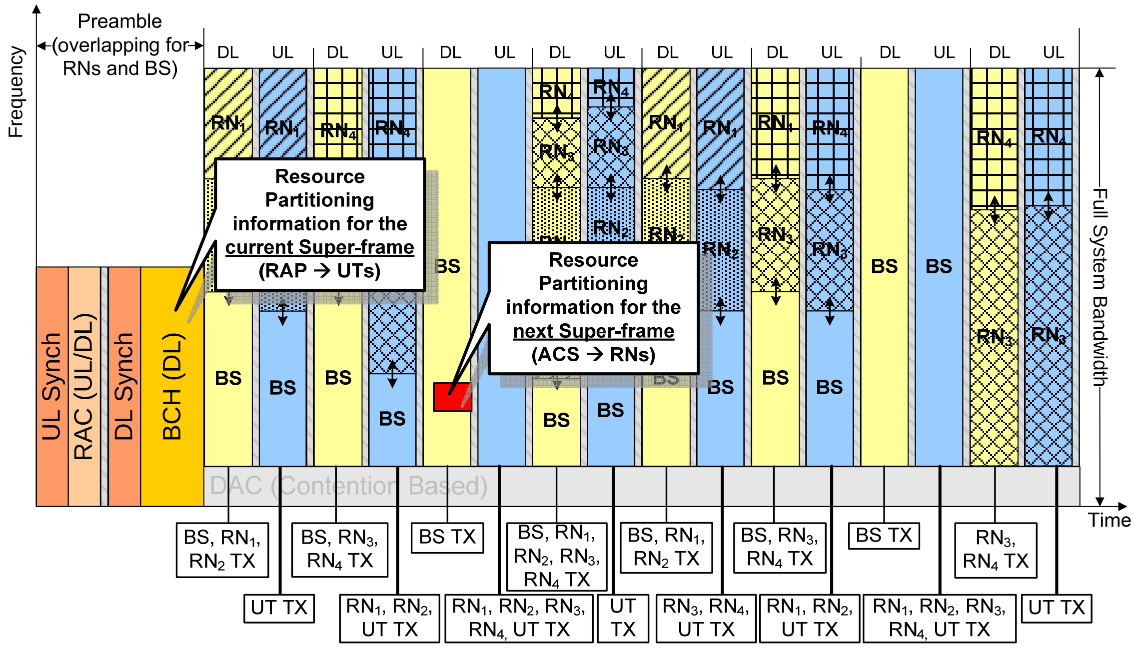


Figure 6-19: An example of resource partitioning between four relay nodes (RN$_x$) and the base station (BS) in a 2-hop relay-enhanced cell (REC) for the TDD MAC super-frame showing the distribution of resource partitioning information by the BCH and as RRC protocol information

6.1.5 Coordination across Base Stations in the WINNER Deployment Concept

It is obvious that WINNER system will have to reuse resources in order to provide high data rate using scarce spectrum resources. Therefore, like any other wireless systems, mutual interference is a crucial issue to be addressed properly in order for the WINNER system to evolve successfully. The concepts and criteria for base station coordination to avoid mutual interference in single-hop cellular, 2-hop, and feeder systems have been discussed in [WIND33]. The idea of base station (BS) coordination and *opportunistic non-orthogonal scheduling* scheme for relay-based systems has been presented in [WIND33] In that scheme, a group of BSs forms *RAP set* (downlink transmissions from each BS become dominant interference for the user terminals (UTs) and relay nodes (RNs) in other BSs in the set) and BS exchanges channel state related information with each other in the set. Interference aware scheduling is performed in a coordinated manner within the RAP set, which aims to maximize the resource utilization based on the available intra RAP set mutual interference information. The idea is that if the interference levels (hence the signal-to-interference-plus-noise-ratios (SINRs)) are predicted and are known to each BS in the set, then by exploiting channel fluctuations the interfered *time-frequency chunks* can be reused in many occasions, which would achieve better aggregate spectral efficiency compared to the case when the chunks are masked proactively.

Although proposed *opportunistic non-orthogonal scheduling* scheme is equally applicable for uplink, for illustration purpose we limit our discussions to downlink transmissions only. In the subsequent sections, we describe the scheme along with feasibility and complexity issues with respect to currently defined WINNER PHY and MAC.

6.1.5.1 Description of the scheme

Figure 6-20 shows the layout of 19 3-sector cell sites, which is assumed as a baseline layout for WINNER simulations by XWP Simulations group [WINXWPSMI]. We consider the downlink of a wide area cellular system that uses half duplex FDD with available frequency spectrum of 40 MHz in the 5 GHz band. We further assume that every sector is equipped with a RN in order to improve overall network coverage/capacity. The downlink transmission uses *orthogonal frequency division multiplexing* (OFDM) and the smallest resource unit is called as chunk as defined in [WIND210]. In this study, we assume that the available 40 MHz spectrum is equally distributed among three sectors as shown in the figure. This gives around 341 subcarriers per sector, of which 277 are utilized and partitioned among different logical channels.

For the simplicity of illustrations, we consider SISO antenna configurations in all associated links. We assume that the UT receive and RN transmit antennas are omnidirectional. The sector antenna is directional (pattern is defined in [WINXWPSMI]) to cover the serving area of the sector. The RN receive antenna is a narrow beamwidth directional antenna pointing towards the associated sector antenna.

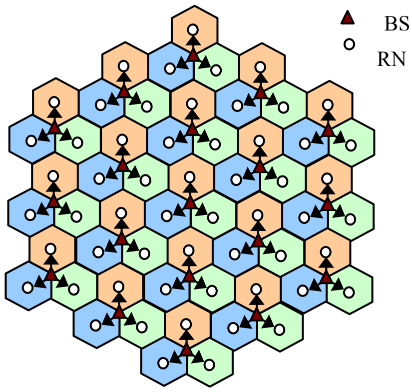


Figure 6-20: Layout of 19 3-sector cell sites

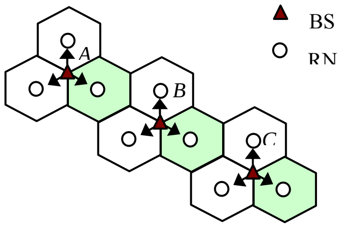


Figure 6-21: The RAP set {A, B, C} considering the shaded sectors

It is intuitive that the downlink transmission from BS is the strongest interferer for the RN because of the high transmission power and directional antenna gains. Furthermore, narrow beamwidth directional receive antenna at the RN makes it very susceptible to dominant interference from neighbouring front facing BSs. For example, as shown in Figure 6-21, downlink transmission from BS *B* and BS *A* would be two most dominant interferers for the RN and UTs in BS *C* (consider the shaded sectors). Similarly, BS *A* and wrap-around BS *C* would be the two most dominant interferers for RN and UTs in BS *B*. For this example, BS set {*A*, *B*, *C*} forms a RAP set. Resources may be allocated dynamically in a slow time-scale in order to increase the network efficiency and flexibility [WINXWPRRM]. In that case, the RAP sets would have to be updated accordingly.

We further consider that RN-UT uses same set of contiguous subcarriers (as *targeted data channel*, TDC) as used by BS-RN or BS-UT partitioned in time domain. Partitioned TDC portion to be used by the RN are reserved by the serving BS and the RN is signalled to schedule its traffic in the reserved portion. Transmissions in the RN-UT links in the RAP set can occur simultaneously with minimal interference as the co-channel sectors are spatially separated and the RNs transmit with low power and omni-directional antennas. No BS in the set is allowed to transmit during RN-UT transmissions.

The scheduling algorithm resides at the *resource scheduler* (RS) in the MAC layer of the BS. It receives channel state information from the RLC layer. The *service level controller* (SLC) cache maintains queues on per-flow basis. The scheduler in the BS selects flow in a round-robin (or depending on the QoS requirements) fashion. The schedulers in cooperation with the *constraint processing* units at the MAC layers find combination of candidate flows that yields the highest predicted aggregate spectral efficiency when transmitted concurrently from the BSs in the RAP set.

The proposed interference aware opportunistic non-orthogonal scheme is summarized as follows.

- UTs predict SNIRs both for the direct link BS-UT and partial link RN-UT. For the case of BS-UT link, we additionally assume that UTs are also capable of predicting the conditional SINRs based on the absence or presence of dominant interferers in the RAP set. For instance, a UT in BS *A* monitors BS-UT link quality for following three cases: i) BS *B* and *C* do not transmit concurrently, ii) BS *B* transmits but *C* does not, and iii) both *B* and *C* transmit.

- RN predicts the conditional SINRs based on the absence or presence of interferers in the set.

- The above channel state information is compressed and fed back to the serving BS.

- Involved BSs in the RAP set exchange this information using wireline connections.
- For the first portion of the frame, scheduler finds combination of BS(s) in the RAP set to be scheduled concurrently at a particular time that yields highest aggregate spectral efficiency.
- All RN-UT transmissions in the set take place during the second portion of the frame.

The scheduler also provides decision on how the packet from a flow will be routed. For instance, scheduler may find that packets from flow 4 and 7 of BS *A* and *C*, respectively, should be scheduled concurrently (note that BS *A* and *C* are using same set of time-frequency resources) for the transmissions; and BS *A* packet is routed directly to the respective UT while BS *C* packet goes through the RN.

6.1.5.2 Feasibility and Complexity Issues

1.) The proposed scheme requires all involved nodes in the RAP set be synchronized. This is feasible through *self-organized synchronization* supported by the super-frame.

2.) To perform link adaptive transmission under the control of resource scheduler, WINNER system will support prediction of the channel quality for the user terminals at up to vehicular speed [WIND24]. This information can be compressed and fed back to the BS with reasonable data rate. Proposed scheduling algorithm needs this information to be exchanged among BSs in the RAP set. This can easily be done as the BSs are expected to communicate via wireline connections.

3.) The quality of BS-RN link does not change rapidly as connected RAPs are fixed. Therefore, channel coherence time for this link could be in the order of tens of super-frames length long and link quality reporting does not have to be frequent. However, mobility of the user terminal demands more frequent reporting for the BS-UT or RN-UT link.

6.2 Radio Link Control (RLC) Layer

An important issue that needs to be taken into account is that the radio interface is solely based on packet transfer. Even though packet transfer is an efficient method for sharing communication resources among multiple users there is also a downside. Usually the underlying network introduces plenty of uncertainties that generally impair information transfer reliability. Scheduling functions, retransmission protocols and (dynamic) routing schemes (together with multi-hopping techniques) are examples of techniques that may change the packet transmission and reception sequence. Similarly, retransmission ambiguities, signalling errors, and unreliable feedback channels may result in spurious retransmissions and residual errors. Out-of-order delivery, duplicates, and lost packets are therefore typical error events that occur in this type of communication systems. A problem arises, since all these events often degrade the upper layer protocol or user perceived communication quality. The responsibility of the RLC protocol is to provide reliable packet transfer services towards the upper layer [WIND76].

6.2.1 Model of the RLC layer

The RLC layer consists of entities that are further composed of sender and receiver sides. The upper layer submits Service Data Units (SDU) to the sender side through a RLC Service Access Points (SAP) and the receiver side delivers SDUs to the upper layer through RLC SAPs. The sender side of the entity transmits RLC Protocol Data Units (PDU) and the receiver side of the entity receives PDUs. It may also perform confidentiality protection such that the transferred PDUs are not made available to unauthorized individuals, entities or processes. A model of two communicating RLC entities is illustrated in Figure 6-22 where SAPs are illustrated with circles and arrows show the direction of data flow.

The problem of reliable information transfer essentially boils down to the questions how the sender and receiver buffers are managed and what the receiver is supposed to do with the data units before they are delivered to the upper layer. As different users may have different requirements on the communication quality, the level of transfer reliability is adjustable, i.e. the RLC layer is configurable with respect to its actions on observed error events. Some users may indeed tolerate certain type of errors more than other users, e.g. some applications (such as voice) and upper layer protocols (such as TCP) tolerate out-of-order delivery to some extent, which is not always the case for network/connection control functions. In other words, all error events should not necessarily always result in similar kind of actions. Similarly, different users may have different requirements with respect to security and therefore confidentiality protection is optional.

The operation of the RLC layer can be exemplified with an implementation that ensures in-sequence delivery of SDUs without duplicates and residual errors. The sender side of the transmitting entity assigns a unique sequence number to incoming SDUs and the resulting PDUs are stored in the retransmission buffer. These PDUs are further ciphered before they are submitted to the lower layer though MAC SAPs. At the receiver side of the receiving entity, these PDUs are deciphered and duplicates, i.e. multiple and identical copies of the same PDU, are erased. The received PDUs are also buffered in the reception buffer for the sake of in-order delivery. The receiver side of the receiving entity reports the status of received PDUs to its own sender side (using 'Send ACKs' in Figure 6-22) that further transmit a status report to the transmitting entity. The receiving side of the transmitting entity extract the status reports from the received data and forward them further to the retransmission buffer (using 'Received ACKs' in Figure 6-22). Acknowledged PDUs are released from the retransmission buffer and the sender side of the transmitting entity retransmits the remaining content of the retransmission buffer.

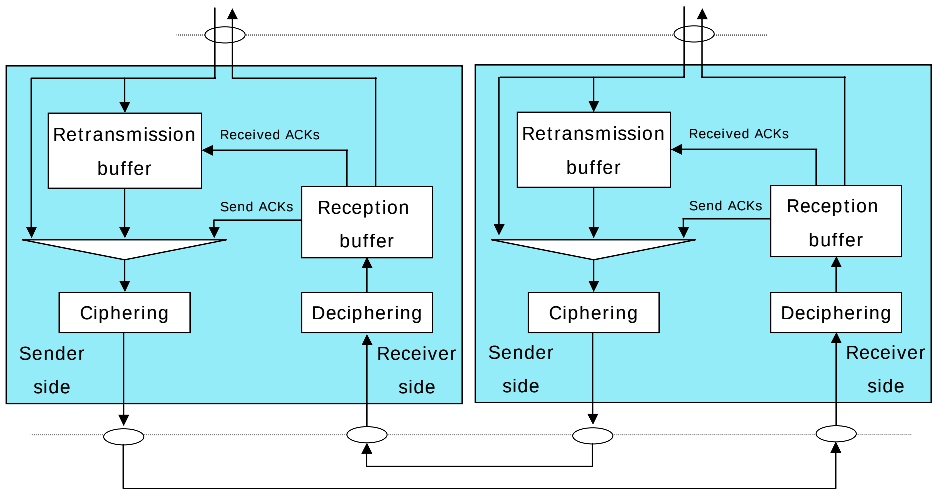


Figure 6-22 RLC model.

6.2.2 ARQ Framework

Transmission in wireless multi-hop (MH) networks should result in the same reliability as in single-hop (SH) wireless networks to guarantee a required residual packet error rate (PER). Traditionally, cellular mobile communication systems comprise only SH wireless transmission. Hence, the wireless end-to-end connection equals the one-hop connection and an ARQ protocol dimensioned for a single-hop is sufficient to achieve a high reliability. This traditional approach is not suitable for MH communication, like it is envisaged in the WINNER air-interface. Specifically, for the efficient support of mobility and at the same time reliability, an end-to-end (e2e) automatic repeat request (ARQ) protocol should be introduced in addition to a single-hop (SH) ARQ protocol. With the introduction of the SH-ARQ protocol, which acknowledges successful data transfer on each hop, the SH reliability and spectral efficiency is increased. The e2e ARQ protocol should be introduced to re-transmit lost packets on the way to the final destination and to guarantee the required e2e reliability.

These issues have been discussed already in [WIND32] and in the current service specification, the RLC layer is responsible for the end-to-end ARQ service, whereas the MAC layer is responsible for the hop-ARQ service. In [WIND32], two possible solutions were suggested for the reliable transfer service, a so-called Layered ARQ approach and a co-optimized so-called Relay-ARQ approach. Initial work towards a comparison of these two solutions is presented in the appendix A.2.2. A full end-to-end ARQ approach is still the current working assumption, but below we also present another possible solution for the retransmission service called Multi-hop ARQ, that terminates the end-to-end ARQ protocol in the RAP closest to the UT. The last hop connecting the UT to the RAN relies on a hop-ARQ service.

6.2.2.1 Layered ARQ for error recovery

The most straightforward solution is to have two layers of error recovery on top of each other. The lower layer provides error recovery per hop, while the upper layer performs error recovery for the complete multi-hop route (edge-to-edge). Such a set-up for a two-hop path is shown in Figure 6-23:.

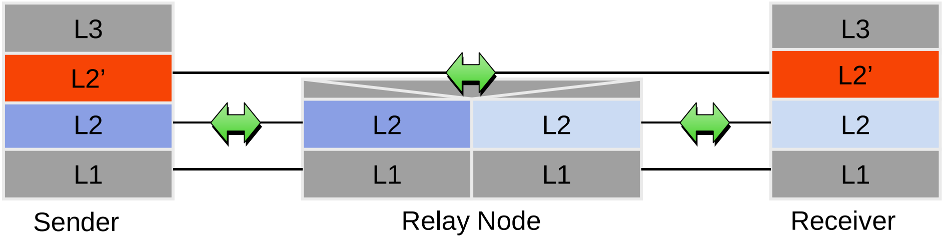


Figure 6-23: Layered ARQ for error recovery

However, such a simple and straightforward approach has some inherent problems: i) ARQ protocol layering may potentially cause harmful interactions.[2] As there are two competing error recovery control loops (ARQ processes) it can happen that missing data is concurrently being retransmitted by the upper and the lower layers; thus wasting radio resources. Configuring the upper layer with a very conservative error recovery procedure can reduce this effect. But this has as consequence that packet loss due to a change of route is recovered very slowly; ii) ARQ protocol layering leads to additional overhead, since both layers need header and/or control information (e.g. two different sequence numbers) and both have to send acknowledgements. These drawbacks are further discussed in [WIND32] Annex I section 8.3.1.1.1.

The problem with competing timers may be alleviated if the two layer's timers are coupled. Nevertheless, this operation is not as straightforward as it first may seem since: i) Packets are out-of-sight of L2' after passing first L2 hop; ii) 3 different ARQ windows will have to be handled/compared; and iii) Out-of Order delivery in relays is problematic for L2' timers. An example outlining the messages exchanged when using the proposed scheme is given in [WIND32] Annex I section 8.3.1.1.2.

6.2.2.2 Relay-ARQ for error recovery

A retransmission protocol overcoming many of the aforementioned drawbacks of layered retransmission protocols is outlined in this section. The basic principle of the proposed scheme (henceforth referred to as Relay-ARQ) is to define a single error recovery protocol that spans over the complete multi-hop route (see Figure 6-24). Instead of a L2 protocol termination in each node, the same protocol state is used for all hops. This may be accomplished by enforcing that the transmission in the chain of links is based on the same L2 PDUs, including the use of the same sequence numbers (i.e. segmentation and reassembly may not be performed on L2). This allows the re-use of state information and the exchange thereof beyond the preceding link.

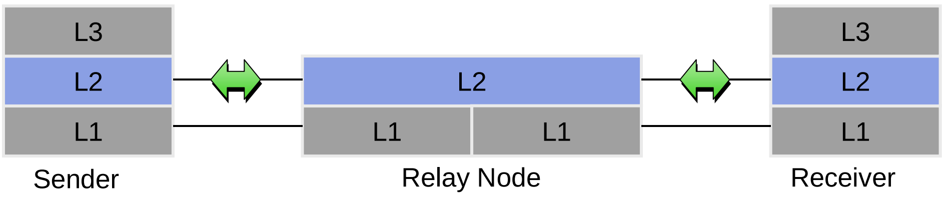


Figure 6-24 Common ARQ layer for complete route.

As in the preceding cases, the retransmission of lost data is separated into a two-stage error recovery process. When an intermediate relay node successfully receives the data, it replies with a feedback report indicating a Relay Acknowledgement (RACK), i.e. instead of binary ARQ feedback, ternary feedback is used to express the status of L2 PDUs. The RACK indicates to the previous node in the multi-hop chain that a data packet has been successfully received at the next hop. Consequently the previous node delegates the retransmission responsibility to the next node. Nevertheless, the transmitted data is not yet deleted from the ARQ buffer. It is only deleted when the data packet has been received at the final receiver and a final acknowledgement (ACK) is sent back, that all nodes along the way (upon receiving this ACK) remove the data from the ARQ window. In this way, the previous relay node may take back the retransmission responsibility in case the next relay node drops out of the connection. If on the other hand a relay node joins the route they build up a soft-state ARQ protocol state for this connection.

[2] The most prominent example is the interaction between TCP and wireless link layer protocols. Another scenario with harmful interactions is layered link layer ARQ protocols, e.g. GPRS with its LLC and RLC protocols.

More details about Relay-ARQ and its potential benefits may be found in [WIND32] Chapter 2.4.2.2.2 and Annex I section 8.3.1.1.3.

6.2.2.3 Multi-hop ARQ

Two coupled ARQ protocols with feedback are introduced for reliable transmission in MH networks. The first ARQ protocol runs as e2e protocol between the BS and the last RN that has a direct connection with the UT. The second ARQ protocol is a SH-ARQ protocol. The SH-ARQ protocol is a conventional ARQ protocol used in wireless systems, e.g., a selective-repeat ARQ or hybrid ARQ protocol. The e2e ARQ protocol is coupled with the SH-ARQ protocol running between the last RN and UT. E.g., if the UT is connected via an RN, the e2e ARQ protocol is terminated for the downlink (DL) transmission, i.e. a transmission from the BS to the UT, at this RN. Respectively, the e2e ARQ is started for the uplink (UL) transmission at this RN and is terminated at the BS.

The protocol stack for the coupled multi-hop ARQ (M-ARQ) protocol is depicted in the Figure 6-25.

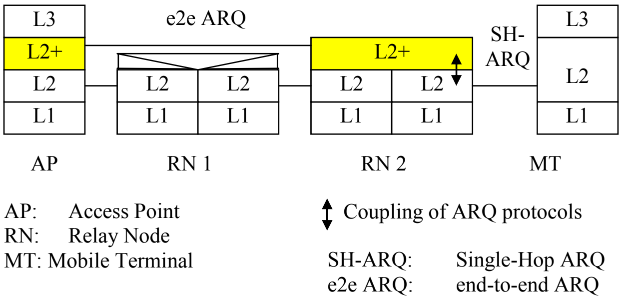


Figure 6-25. Protocol stack of the Multi-hop ARQ protocol

The e2e ARQ protocol is invisible for the UT. The UT recognizes no difference between being connected to an RN or a BS. In the example the e2e ARQ protocol is terminated at the BS on the one side, and at the last RN connected directly with the UT on the other side, which is RN 2. Consequently, the buffers for the e2e ARQ protocol are located at the BS and the RN 2, but not at the UT.

On the UL all correctly received and acknowledged (ACK) packets between the UT and RN 2 are stored in the buffer of the e2e ARQ protocol that resides in the layer L2+. Different to conventional systems where the packet forwarding on the UL is only controlled on the hop to the next entity (RN 1 in Figure 6-25), the new e2e ARQ protocol will take care of correct delivery to the BS. Only packets correctly received at the BS and acknowledged (ACK) will be released at the RN, which is RN 2 in the example in Figure 6-25.

On the DL all packets are stored in the RN before they are transmitted on the last hop to the UT. Only correctly delivered packets on the last hop will be ACK by the RN towards the BS and will be released. For that purpose a coupling of the e2e ARQ and last-hop ARQ protocol is required. The ACK for the packets correctly received at the UT will generate a corresponding ACK for the e2e ARQ protocol, which will be delivered to the BS. The RN and BS in turn will release this packet from their transmission queues. A more simple operation exists if the UT is directly connected to the BS. In this case the e2e ARQ protocol is terminated in the same point, and hence, can be simplified.

More details on the Multi-hop ARQ proposal can be found in the appendix A.2.1.

6.3 Medium Access Control Layer

The MAC system layer design supports and enables several innovative features of the system concept, such as adaptive scheduling and link adaptation, advanced spatial multiplexing/multiple access, fast retransmission also for delay sensitive flows, resource allocation in relay enhanced cells and constraint processing to support spectrum sharing between other networks and operators. In D7.6, the MAC system layer as part of the WINNER system concept is described and in D2.10, the MAC functional design is given. In addition, a tight co-design of the MAC system layer with the physical layer is important for the WINNER system proposal to meet the performance goals of the WINNER concept.

There are three MAC modes identified within the system concept

- **FDD cellular MAC**,
- **TDD cellular MAC,**
- **MAC for peer-to-peer transmission**, using the TDD physical layer mode.

Thus, the combinations of PHY and MAC modes lead to three WINNER system modes. The PHY modes are described in section 6.4. Also refer to this chapter for definitions of physical layer entities not defined in this chapter.

The functional behaviour of the FDD and the TDD MAC are very similar, but it is unlikely that the same super-frame structure can be used for the two modes. The MAC for peer-to-peer has a different design. In the current design, it could be seen as an internal function of the TDD MAC since the resources for the peer-to-peer MAC are controlled by TDD MAC. Thus, it needs a BS to operate. If the peer-to-peer MAC should support peer-to-peer communication also without a RAP in range is for further study. The Peer-to-peer MAC is described in [WIND210].

The MAC design is focused on Relay Enhanced Cells with two-hop relaying using Fixed Relay Nodes (FRNs). Cooperative multi-antenna transmission and distributed antenna systems are supported, as long as all antennas are regarded as encompassing one BS. The control of *downlink flows* resides in the MAC that is implemented in the transmitting RAP (radio access point, i.e. BS or RN). Most MAC control functions for *uplink transmissions* reside in the receiving RAP.

In the following section as basis for the protocol work the MAC functions will be listed as stated in [WIND76] and briefly explained. For further details on the MAC functions and services the interested reader is referred to [WIND76] and [WIND210]. As basis for all MAC communication the structure WINNER MAC super-frame will be introduced in Section 6.3.2. Following the super-frame description some protocol functions will be discussed in general. All the MAC protocol work is currently in the status of discussion and not specification. The section on the MAC protocol highlights challenges of the WINNER MAC protocol in the context of relaying as well as concepts to solve particular problems.

6.3.1 MAC functions

The functional behaviour of the WINNER system concept is described by splitting the functions into control plane functions and user plane functions. In that way, it is possible to build systems with independent scalability of the control functions and the transport of the user data flows, by terminating control plane and user data plane functions in different nodes. This functional grouping is very important for the economy of large networks.[3]

The MAC layer provides the following services to the RLC layer:

- **Radio packet transfer**, i.e. transmission and reception over the radio interface of packets belonging to any of the transport channels defined below.
- **MAC radio resource control**, i.e. acceptance and execution. of control messages from higher layers to the MAC that specify required transmission parameters and boundary conditions.
- **MAC control feedback**, i.e. messaging from the MAC that supports the flow control, the QoS control and the spectrum assignment and other functions at the RLC and higher system layers.

The WINNER *transport channels* are defined as being the interfaces between the RLC protocol layer (in the RLC system layer user plane) and the MAC protocol layer (in the MAC system layer user plane). They define the basic types of radio packet transfer that are provided by the MAC system layer:

- Broadcast channel (**BCH**) for broadcasting system information from RLC and higher layers to all terminals inside the coverage area of the cell,
- Contention based random access channel (**RAC**) for initial access to a master device, and also for BS-to-BS control signalling in TDD systems,
- Contention based direct access channel (**DAC**) for contention-based uplink data transfer,
- Common data channel (**CDC**) for scheduled point-to-multipoint communication,
- Targeted data channel (**TDC**) for scheduled point-to-point communication.
- Targeted control channel (**TCC**) for control-plane generated control messages.

[3] In the presently assumed design, a MAC system layer is assumed present in all BS and RN logical nodes (Section 5.3). A distinction between MAC control plane and user plane is made (see A.3, but it is not a necessary feature of the WINNER MAC layer in the assumed logical node architecture.

6.3.2 WINNER MAC Super-frame

The *super-frame* (SF) is a time-frequency unit that contains pre-specified resources for all transport channels; Figure 6-26 illustrates its preliminary design for the TDD case. More details about the super frame in the FDD scenario are given in Annex A.3.1. The super-frame is designed to

- include self-organized synchronisation of all involved base stations, relay nodes and user-terminals. This enables an improved spectral efficiency in two ways: It makes large guard-bands unnecessary and enables interference-avoidance scheduling between cells and relay nodes with fine granularity in time and frequency
- enable the resource partitioning to work efficiently in conjunction with inter-cell interference avoidance schemes. It is also designed for relay-enhanced cells, so that base stations and a set of relay nodes can share the total spectral resources efficiently.
- enable adaptive resource partitioning. On the super-frame time-scale, the resource partitioning can adapt to the traffic demand to/from different nodes in the REC over different transport channels

As shown in Figure 6-26 the main part of the super-frame is shared by the contention-based direct access channel (DAC) for the peer-to-peer mode and the scheduled data channels CDC and TDC, and their related control signalling. It also contains time-frequency-spatial resources that are not to be used, due to interference avoidance constraints. The DAC channel is provided a constant set of frequencies over the whole super-frame. The preliminary super-frame design comprises a preamble followed by n_f MAC frames. The current proposal uses $n_f = 8$, resulting in super-frames of approximate duration 5.6 ms.

The highlighted MAC frame in Figure 6-26 shows the MAC frame in its current reference parameterisation[4] with 6 chunks per frame in the time direction for the TDD mode. The respective super-frame for the FDD mode is assumed to have two chunks per MAC frame in the time direction. The available number of chunks in the frequency direction could vary with the geographical location and the available spectrum. It is assumed that for both the FDD DL and UL and for TDD, there exist frequency bands that are available everywhere. The preamble is transmitted cell-wide in those commonly available bands.

For further details about the design and the choice of parameters the interested reader is referred to [WIND76] and [WIND210]. The super-frame provides the structure for the MAC protocol.

[4] Please note that the shown structure in terms of chunk numbers chunks size number of frames is not finalised but has been identified as reference for harmonised assessment.

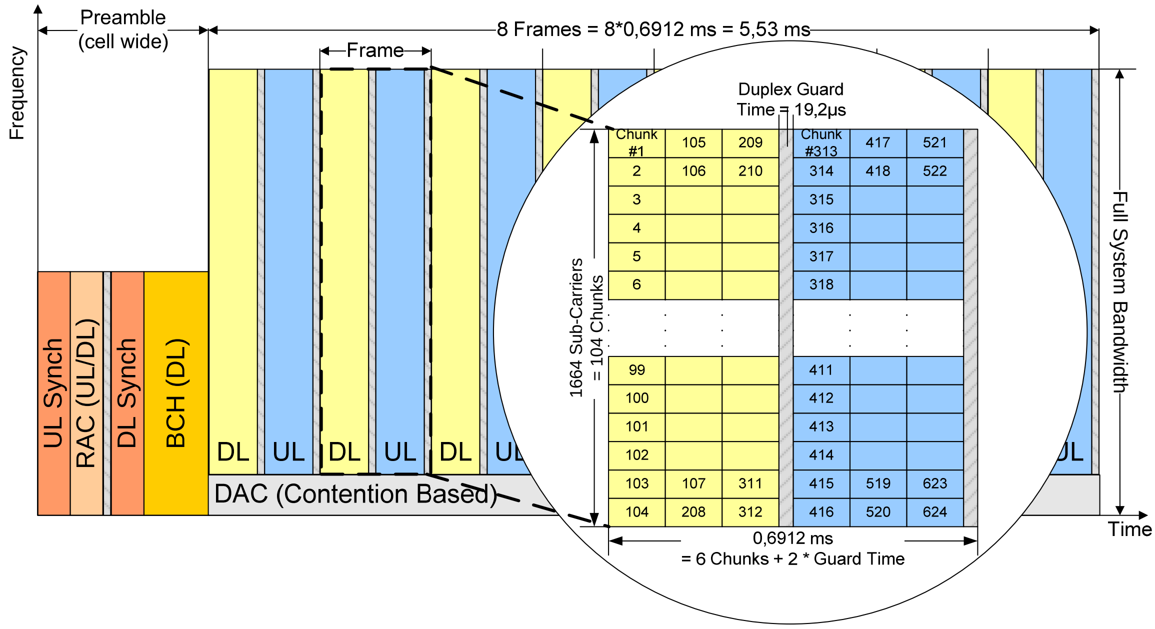


Figure 6-26: WINNER MAC super-frame example consisting of 8 TDD frames with asymmetry 1:1, a preamble and the DAC for contention based peer-to-peer communication. The frame for TDD cellular transmission with its chunk-based substructure has been enlarged.

6.3.3 The Super-frame Preamble

As shown in Figure 6-26, the initial timeslots form the **preamble** of the super-frame. They are located in a part of the band available cell-wide in all cells. The remainder of the super-frame may use other spectral areas that are available at some locations, or to some operators, but not to others, or can be received by some types of UT only. The super-frame has been arranged in a way so that as less guard times as possible are needed.

The WINNER relay node should appear to the UTs as a BS. Therefore it has to provide the same preamble to its UTs. As WINNER assumes a fully synchronized system, all RAPs have to transmit their preamble at the same time. Thus the super-frame preamble is used by several adjacent RAPs in parallel as shown in Figure -6-27. To be able to distinguish between the different RAPs the resources for the preamble have to be separated in frequency. In praxis this means that each RAP is transmitting its synchronization signal on specific sub-carriers. The position of the sub-carriers also denotes on which sub-carrier the RAP sends its BCH as well the sub-carriers where the random access procedure will take place. Consequently a RN is not able to receive a BCH from its BS or any other RAP as it transmits its BCH at the same time.

After the transmission of the preamble the downlink of the first MAC frame is started which consequently cannot be used for inter-RAP communication as the RN does not get the necessary time to switch between sending and receiving.

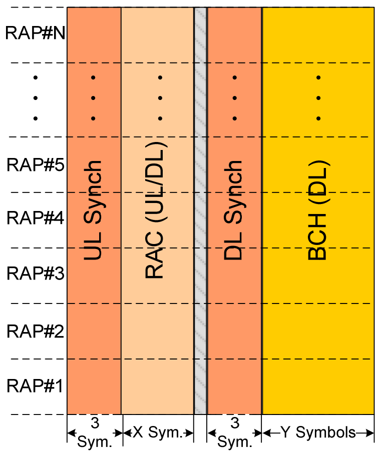


Figure -6-27: Parallel usage of Super-frame Preamble on the example of TDD

The super-frame structure relies on a tight self-organising synchronisation of terminals and network nodes as described in [WIND23] and [WIND210]. The structure of the super-frame, with its two initial sets of synchronization OFDM symbols, had been designed to facilitate self-organized network synchronization This is important since the availability of e.g. GPS timing references at all BSs and RNs is not assumed to be guaranteed.

In the following the parts of the super-frame preamble will be described in more detail.

6.3.3.1 Synchronisation

At the beginning of each super-frame there are **two synchronisation slots**. Self-organising synchronisation of terminals and network nodes, as described in [WIND23] and Appendix B.1.4 of [WIND210] can be used on a super-frame basis by this design. In the **second slot** (DL Synch, see Figure -6-27), each base station/relay node transmits on four OFDM symbols. The first symbol, the *T-pilot*, is used for coarse synchronization. In the remaining three symbols, each BS transmits on two adjacent sub-carriers, with the others set to zero. On reception, they are used for updating the UT synchronizations. The **first slot** (UL Synch, see Figure -6-27)of three OFDM symbols the next super-frame is the *uplink* synchronization slot. Here, all terminals transmit on the two adjacent sub-carriers that were received strongest, i.e. those that were used by the BS/RN closest to them. This part of the iteration is used for self-organizing synchronization of the base stations and relay nodes, which receive uplink synchronization symbols from terminals at other BS, that in turn synchronise to those BS. In the FDD mode, the UL synch. slot is in the UL super-frame and the DL Synch slot is in the DL super-frame

6.3.3.2 The Random Access Channel (RAC)

In-between these synchronization slots a short timeslot over the whole band is reserved for the contention-based **random access channel (RAC)**. This channel enables initial access to a master device. Placing the RAC and its guard time in-between the synchronisation slots gives the RAP sufficient time to process the uplink synch signal and adjust its synchronisation before transmitting the downlink synchronisation signal. The RAC time-slot in TDD systems could also be used for BS-to-BS and RN-to-BS over-the-air control signalling. Of course, the main communication between RN and BS and between RNs will be performed in scheduled channels, while the main BS-to-BS communication is done via the backbone in order to save radio resources.

6.3.3.3 The Broadcast Channel (BCH)

Subsequently, a set of OFDM symbols carries the **downlink preamble control transmission**. It contains the broadcast control channel (BCH) from the RRC layer. It also contains a control message that specifies the overall resource allocation used within this super-frame. Adjacent BS and RN should use orthogonal time-frequency sets within this timeslot for their downlink preamble control transmission, to limit their mutual interference.

In the BCH each RAP broadcasts relevant control information to its UTs, which are

- the cell ID,
- location of resources used by the respective RAP in the coming MAC frames of the MAC super-frame
- partitioning of the UL vs. DL resources.

- In addition it might be necessary to transmit information about the number of hops and the link quality of the next hop towards the BS

Thus after having successfully received the BCH the UT knows that it has received and decoded the correct BCH based on the cell ID. Further the UT knows which of the n_f MAC frames (and sub carriers) it has to receive and decode in the upcoming MAC frames. This information allows the UT to stay idle in the other MAC frames to save battery power.

The additional information on the hop numbers and link quality is necessary to allow the UT to choose the best possible RAP if it can listen to more than one. In some cases it might be more appropriate to connect to the BS (or a RN closer to the BS), even though the SINR of the RN is stronger than the one received from the BS, e.g. due to delay constraints, or due to the fact that the overall throughput over the multi-hop link is less than the one received directly from the BS (or next RN).

The range of the safe reception of the BCH message places an upper bound on the size of the cell. Thereby two types of RNs can be distinguished

1. RNs that have a serving area which is spatially partly or completely orthogonal to the BS serving area, which means at least some of the UTs in their cell cannot hear the BS or any other BCH. This is true for the scenarios depicted in Figure 1-2 (right) and Figure 1-4. This type of relay will be further referred to as ***RN Type I***

2. the RNs which serve UTs that can also hear the BS BCH (see Figure 1-3 (left)) further referred to as ***RN Type II***

In case 1 each RN has to transmit its own BCH to be visible for all UTs in its serving area (cell). Thus the RN controls a separate cell. In case 2 the BS can transmit one BCH for the whole cell, which might contain some additional information of the UTs. The RN could be simpler in this scenario. Only the issue of the cell ID and the hop count must have to be signalled to the UTs in some way! In scenario 2 the RN is able to listen to the first DL sub-frame sent by the BS (or next RN) as the BCH is send by the BS. Please note that RNs of Type I also work in the RN Type II scenario but not vice versa.

6.3.4 Signalling of Resource Partitioning information

The focus of the cellular WINNER MAC design is set on the case where RNs and BS use the same Physical Layer mode and share spectral resources. Relaying via user terminals is currently not considered in the WINNER system concept and therefore also not part of the MAC design.

The total time-frequency resources (chunks) of one super-frame are partitioned into parts used by the BS, shared parts, and parts used by RNs and parts for DAC use. Further the resource partitioning contains information about the UL/DL switching point. The calculation and processing of such resource partitioning is different for RN Type I and RN Type II

RN Type I

This partitioning is computed by the RRC located centrally at the ACS. This resource partitioning information is distributed to the connected BSs and as RRC control message in a MAC frame as shown by Figure 6-19 to the RNs. In order to avoid inconsistent resource partitioning information among the nodes in the REC, the resource partitioning information is time-advanced one super-frame (assumes two-hop relaying). This allows the RN to timely distribute the resource partitioning information to its UTs. The resource partitioning information is broadcasted to the UTs by means of the BCH as explained in section 6.3.3.3 .

RN Type II

For RN Type II the resource partitioning can be performed on REC level by the BS. This allows a resource partitioning on a shorter time scale as the BS can broadcast the resource partitioning to all RNs and UTs at the same time in its BCH.

6.3.5 MAC flow setup function

New uplink and downlink flows are established by the RLC Flow establishment function. It requires the detailed set-up of a flow context over each involved hop. That flow context establishment is executed by the MAC ***flow setup*** function Flow context release is initiated by the RLC and is executed by the MAC ***Flow termination*** function. These two functions require the MAC protocol to transfer the flow and flow context information.

When a new downlink flow is established, it is given a local flow address that is unique within the Relay Enhanced Cell. Its destination UT (or UTs in the case of CDC point-to-multipoint flows) is notified and a resource scheduling buffer queue is initialized [WIND210].

In order to save signalling overhead for further resource allocation next the *RLC flow ID* a REC wide *MAC flow ID* is assigned

6.3.5.1 Downlink:

- The BS MAC receives a flow establishment request (*flow_setup_req*) from the RLC at RANG containing the flow context = UT address, QoS parameter, source ID (ACS) and the *RRC flow ID*
- The MAC assigns a local flow address (or is this already done by higher layers?) (REC-wide) and stores this information in its database and initialises the queue
- BS sends a MAC control signal to the RN including the *MAC flow address*, the *flow context* including the *UT ID* and the *RRC flow address*.
- The RN stores this information in its database and initialises the queue for the RS.
- The RN transmits the flow establishment with including the *MAC flow address* and the flow context to the terminating UT. The RN can use the same or different *MAC flow address* than the *MAC flow address* assigned by the BS.
- The UT processes the MAC flow set up signal by informing its RLC and transmits after positive response a confirmation to the RANG RLC-CP with the *MAC flow ID*

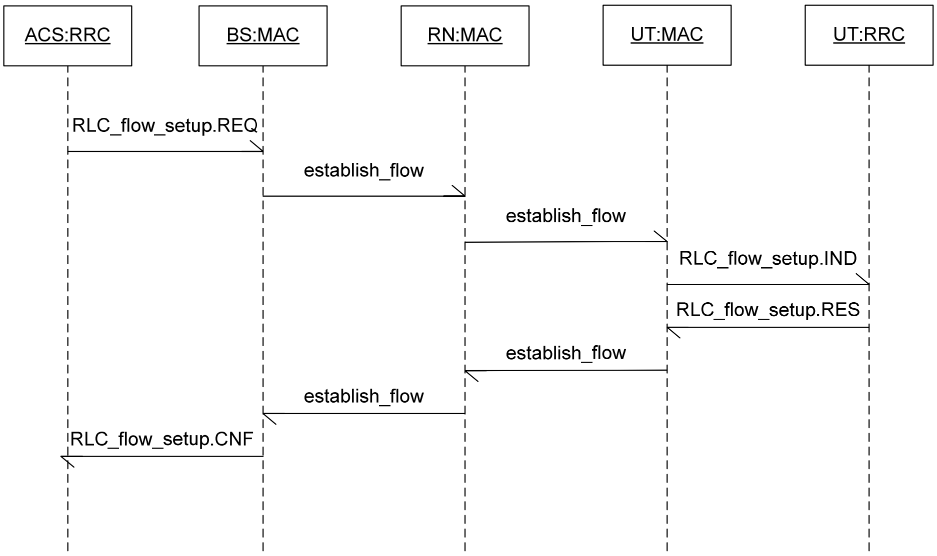


Figure 6-28: Flow establishment signalling

Final status after successful flow set up:

- The BS has full knowledge about the flow and flow context (QoS constraints, flow ID and RN it has to transmit to on the next hop – if any)
- The RN knows the flow address and the flow context in terms of QoS constraints and destination (or optional the next hop)
- The UT knows the flow context, whereby it is not necessary for it to know about the QoS parameter. It is important for the UT to know the *RLC flow address,* which allows a fast handover in case of inter REC handover. The *MAC flow address* is important to identify the flow user data.
- With the receipt of the *RLC_flow_setup.CNF* primitive the end-to-end flow has been established and is ready for use.

To allow the multi-hop flow set-up the BS has to be aware that the UT is assigned to one of the relays in its REC. Therefore it has to initiate a multi-hop flow set up in order to inform the RN about the flow context. In comparison to the single hop where only the UT will be informed about the flow set up, the BS has in addition to the flow ID to transmit the QoS parameters.

6.3.5.2 Uplink:

The RLC sends a flow setup request on the feedback channel to its peer entity in the RANG. The RLC-CP at the RANG then initiates the flow establishment, which means the same procedure as for the DL is performed.

6.3.6 Resource Allocation

6.3.6.1 Resource allocation as the last step of resource partitioning

Within the MAC layer, resource partitioning (time-frequency resources) and spatial scheme control (spatial resources) in a REC is performed per super-frame. The allocation to different transport channels is adjusted, based on the aggregated demand within each transport channel. Guard chunks for interference avoidance scheduling are also defined and re-adjusted, to enable flexible spectrum use between WINNER operators/users and adaptive interference avoidance between neighbouring cells, beams and sectors within a REC.

After receiving the BS-to-RN control message, the Resource partitioning functions of the MAC at each relay node perform the further refinement of the resources assigned to these RNs. A table holding the chunk set allocations for RN exclusive use and RN shared usage is used together with other constraints from upper layers to form the resource allocation within the RN cell. After updating the constraint combining and the spatial scheme pre-selection, the RN MAC updates the resource partitioning:

1. The directional properties of all constraints are first converted to be with respect to directions related to the relay node cell/sector, instead of the BS sector.
2. The ***DAC assignment*** chunks at RNs are assumed to be the same as that used by the BS, to simplify the DAC transmission.
3. ***Assign RN control chunks***.
4. Subtract the so defined DAC, Control chunks and the chunks allocated for the BS-RN links to obtain the remaining assignable chunk set for RN-UT links.
5. ***TDC adaptive assignment*** is performed in the same way as in the base station MAC
6. ***RN-to-UT control message formation*** is similar to the BS-to-UT control message formation by the BS MAC. This message is also transmitted during the SF preamble.

Further details on the proposed algorithms for the MAC Resource Partitioning within an REC can be found in D2.10 [WIND210] Appendix C.1.1 “MAC radio resource control: Resource partitioning and constraint combining” and D2.10 [WIND210] Appendix C1.3” MAC radio resource control: Spatial scheme pre-configuration and selection”.

6.3.6.2 Signalling of Resource Allocation in the non frequency adaptive case

The resource allocation has to be signalled for every MAC frame by every active RAP. Depending on the length of the MAC frame the resource allocation has to be signalled in the MAC frame it is addressing or one MAC frame ahead. The resource allocation send out by each MAC frame will look in the non frequency adaptive case as shown in Figure 6-29. As the correct receipt of the full resource allocation table is important the table entries have to be protected by CRC checksums that are not shown in the current figure.

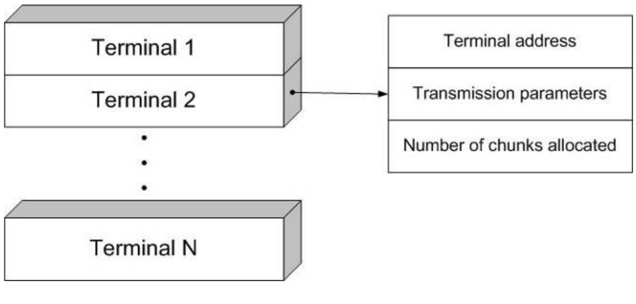


Figure 6-29: Structure of allocation information for the non frequency adaptive chunks

Thereby it is assumed that the resource allocation information for a specific MAC frame will be signalled in the first chunks of the same MAC frame. If the time constraints are too restrictive and the signalling has to be performed one MAC frame in advance, this would mean that a RN is blocked for two frames for communication with the serving BS/RN. On the opposite the communication with serving BS/RN will cost two MAC frames for which the RN to UT communication is idle.

Further investigations are necessary to allow the signalling of the spatial component in the resource allocation table.

6.3.6.3 Signalling of Resource Allocation in the frequency adaptive case

In this scheme, the active flows and therewith UTs for a UL/DL slot in a frame are first identified by listing their addresses. The number of bits reserved for allocation signalling is most likely fixed, which makes it necessary to restrict the maximum number of allowed users in the slot (the active set).

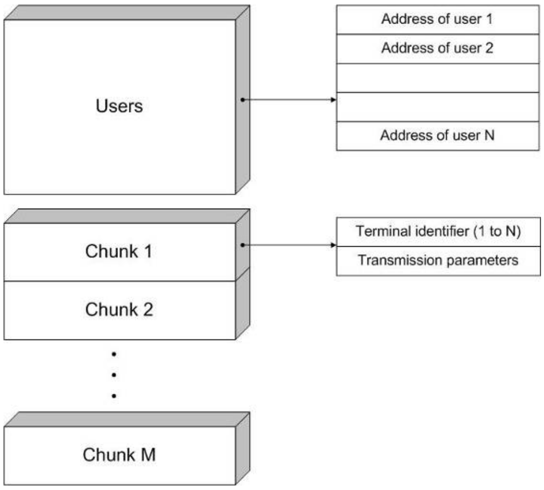


Figure 6-30: Structure of allocation information in Scheme 1

After the active users have been identified, each chunk is assigned to one of the active users. The ordering of the active users in the list mentioned above is used as a temporary user identifier. In addition to the user identifier, transmission parameters are specified for each chunk. The structure of allocation information is illustrated in Figure 6-30. In [WINXWPMAC] it is further that there might be no need to explicitly transmit the address list if the number of active flows is below a maximum limit.

6.3.6.4 Resource allocation for inter RAP communication

In general there are two different possibilities to assign resources to the flows belonging to UTs that are connected to a RN:

a) The flows are treated by the BS in the same manner as all other flows. This means they will be scheduled individually. In this case the RN will act as receiving UT for all this flows and does not have to separate them.

b) The RN and BS will establish one flow and multiplex all flows belonging to the UTs of the RN on this one flow. In this case the individual flows belonging to different UTs will be separated at the RN and then put in the resource scheduler for further transmission. The advantage of heaving only one flow lies in a reduced signalling overhead and a more efficient resource usage. The flow between BS and RN is only known on the MAC level. This single flow has to be well protected as the one failure of receiving, e.g. the control message where to receive it in the DL will lead to an empty DL frame on the next hop and to the need to retransmit the whole DL traffic of one MAC frame for one RN cell. One solution could be to do the inter RAP traffic on fixed resource. It might also ease the work of the resource scheduler as it has to treat a reduced number of flows.

c) As for b) the flows will be multiplexed, but this time on a number of flows set up between the RN and the BS with respect to QoS requirements. The RN has to separate the flows again.

The RN has to listen to the BS or RN it is connected to in the DL phase in order to receive the resource allocation table. This means the RN cannot serve its UTs in this time frame. This is especially critical if the resource allocation tables have to be transmitted one frame in advance as this would mean that the RN would miss two MAC frames in order to receive and transmit data to the further hop (most likely the BS). Therefore it could be beneficial if the system reserves specific resources for the inter RAP communication, which is on the one hand less flexible in terms of adapting to changing traffic demands, but allows on the other hand to assign resources for the inter RAP communication with only very few signalling. Further it would force the system to work on the RN centric resource allocation type b)or c) as explained above.

6.3.7 Forwarding of User Data

For the forwarding of user data the situation for UL and DL have to be distinguished:

6.3.7.1 Forwarding of downlink user data

The forwarding of DL data is an easy task, as the data arriving at the RN is already pre-scheduled by the SLC of the sending node (the BS in the 2-hop case). The only job of the RN is to separate the different flows, e.g. by means of the MAC flow addresses, for the different UTs (flows) and shift the data blocks to the different queues of the RN resource scheduler. The resource scheduler is then further processing the packets based on the flow context known at the RN.

6.3.7.2 Forwarding of UL user data

The UL data to be forwarded by the RN has first to be scheduled by the SLC at the RN. The result of the SLC is signalled to the UT or next RN by the resource allocation. The incoming data is then shifted to the outgoing queues UL queues, which will be drained with respect to the resources assigned by the BS/RN.

6.4 Physical Layer

Currently two different Physical Layer Modes (PLM) can be distinguished in the WINNER concept, defined in detail in [WIND210]. These two PLMs are needed in order to efficiently meet the different requirements of the WINNER deployment scenarios:

- Frequency division duplex (FDD) transmission is performed over paired bands and supports half-duplex FDD terminals.
- Time division duplex (TDD) transmission over unpaired band.

Although any PLM can be configured for any kind of deployment, the FDD mode is primarily intended for the wide-area cellular deployment scenario. For that reason, the currently defined frequency bands are more narrow (2x20 MHz and 2x40 MHz are being studied) than for the TDD mode (100 MHz). The TDD PLM is primarily intended for metropolitan (short-range cellular) deployment, peer-to-peer communication and links to relay nodes. Both PLMs use Generalised Multi-Carrier (GMC), which include Orthogonal Frequency Division Multiplexing with a standard cyclic prefix (CP-OFDM) and serial modulation (block-based single-carrier modulation as well as multi-band single-carrier transmission) as special cases. Serial modulation is interesting in uplinks, where power-constrained users may use single-carrier waveforms within their assigned transmission resources to improve their transmission range.

The basic time-frequency unit for resource partitioning is denoted a *chunk*, and is chosen in such a way that it experiences essentially flat fading in its time-frequency extent. It consists of a rectangular time-frequency area, Figure 6-31 (a), which contains payload symbols and pilot symbols. It may also contain control symbols that are placed within the chunks to minimise feedback delay (in-chunk control signalling). In transmission using multiple antennas, the time-frequency resource defined by the chunk may be re-used by spatial multiplexing. A *chunk layer* represents the spatial dimension, Figure 6-31 (b). The number of offered payload bits per chunk layer depends on the chunk quality, by dynamically adjusting the modulation and coding formats.

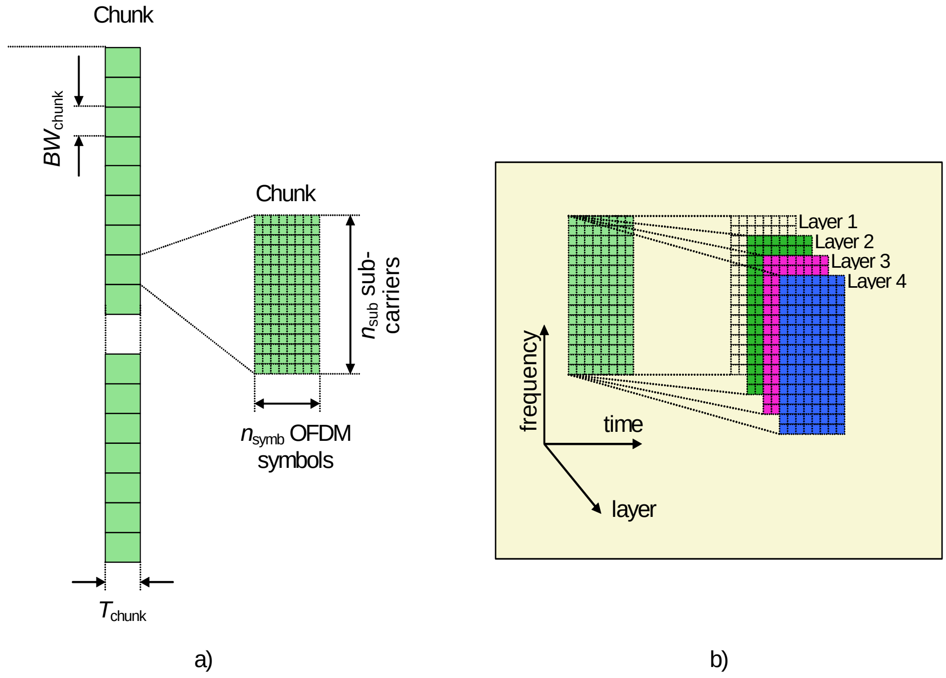


Figure 6-31: a) Multi--carrier Downlink physical channel structure and chunks. b) Chunk layers obtained by spatial re-use.

Chunks are pre-assigned in each radio access point (RAP) for use with different GMC settings, see Figure 6-32 for an example. This setup is dynamically adjustable on a super-frame timescale, based on e.g. flow type, terminal capabilities and channel quality. In addition, the antenna resources are dynamically configured based on a generic processing chain for different spatial processing techniques based on the available chunk layers, see [WIND210] chapter B.2, and they can be used differently for different flows to/from a user terminal.

Chunks can be allocated for either adaptive or non-frequency adaptive transmission. In adaptive transmission, resource allocation and link adaptation relies on accurate channel state information and adapts towards the small-scale fading of the channel. These chunks are envisioned to carry in-chunk pilot and control signals, since the accuracy of the channel state information depends on a fast control loop. Non-frequency adaptive transmissions are used whenever adaptive transmission is infeasible or when the adaptive transmission doesn't make sense.

Different flows may also use different variants of GMC, as depicted in Figure 6-32. For example, in uplinks power-constrained users may use single-carrier waveforms within their assigned transmission resources, while other terminals use CP-OFDM.

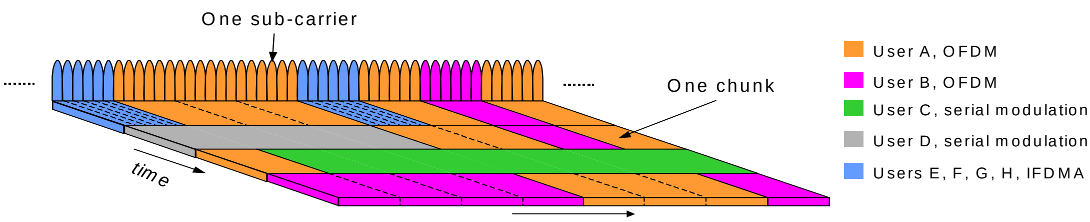


Figure 6-32: Use of different variants of GMC modulation for different uplink users.

A *physical channel* corresponds to a set of chunk layers onto which the resource scheduler in the MAC layer maps data associated with one flow (or sub-flow for certain spatial processing schemes, see D2.10 chapter B.2). These physical channels define the interface between the MAC and the PHY layers.

7 Cooperative Relaying

7.1 Introduction

This section presents the cooperative relaying extension to conventional relaying to exploit spatial diversity in those scenarios where the UT has a sufficiently reliable connection to both RN and BS. In principle, advanced cooperation protocols and algorithms among the relays and BSs can be used to obtain further performance gains at the physical layer (e.g., cooperative diversity, virtual antenna arrays,...) and at the upper layers (e.g., interference avoidance and management, smart scheduling and routing, load balancing, ...). At the first glance using such methods may result in a higher level of system complexity. In this chapter we propose some cooperative techniques for the WINNER air interface and show that they significantly improve system performance however, they only cause a slight increase in the deployment costs.

The WINNER Project has a longer time horizon in comparison to the IEEE 802 Task Groups currently developing the relay-enabled standards. Therefore, it makes most sense for the WINNER Project to have more advanced goals in the context of relaying. Towards that end, this section discusses possible cooperative diversity scenarios in relay-based deployment concepts and points out the flexibility of cooperative diversity concerning future developments and underlying architectures, e.g. Fixed and mobile relaying types in section 7.2 and 7.2, one protocol scheme for single-antenna scenarios in section 7.4, the Cyclic Delay Diversity for MIMO-scenarios in section 7.5.

7.2 Relay-Assisted Cooperative Diversity

The explicit user cooperation diversity schemes discussed in [LaTsWo04][SeErAa03] require that at least two users are in a network and they are willing to cooperate. To sustain this cooperation, an *incentives system* is required for the cooperating partners which could tackle implementation problems. Furthermore, explicit cooperation requires that each cooperating partner detects the data of the other user. Therefore, each user's data have to be protected from "a malicious" partner. This will add another dimension to the *security challenges* of wireless networks. On top of that, the explicit cooperation requires that terminals are modified to prepare them for their new tasks, leading to increase in *terminal complexities*.

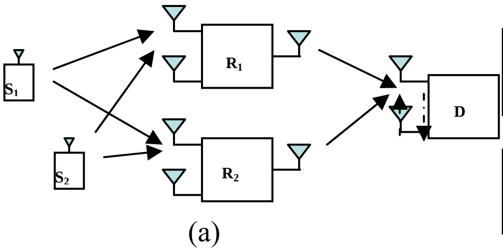
To circumvent the problems of the user-dependent cooperative diversity, we present fixed relay-enabled user cooperation. The scheme, in its simplest form, operates as follows. Two users are engaged in cooperation through two fixed relays. The user terminals do not need to detect each other's data. Not only that the privacy of users is not compromised but sanction/reward for them to cooperate is not required. Additional benefits of the fixed relay-enabled user cooperation include simple user terminals, guarantee of service, ease of routing, and flexibility in system deployment. These are features any service provider would like to maintain an unfettered control. For instance, since the two hops of this new scheme are decoupled, the scheme can easily support heterogeneous relaying allowing two air interfaces in contrast to single air interface used in homogeneous relaying.

7.2.1 System Description

In the proposed scheme the signal processing at the relays can be classified into two types: spatial division multiple access (SDMA) and multiuser detection (MUD) based processing. In the multi-antenna relay architecture the relays can utilize optimum combining scheme to separate users through the SDMA (Figure 1). However, for the single antenna relay architecture, the multiuser detection (MUD) capability will be required at the relays where it is assumed that the users employ some spreading (orthogonal) codes.

In general, after the relays detect the signal in the first hop, they could enter into cooperation using, for example, distributed space-time coding (DSTC) [Alamouti98].

The scheme using SDMA operates in the following manner. Each relay detects the signal of one user by directing a null towards the other. To engage the data of the two users in the distributed space-time coding, the signal of the nulled user is obtained by the other relay node by exploiting the broadcast nature of wireless channels. This exchange of user signals is accomplished in the second time slot when each relay node transmits to destination Figure 7-1 (b).



S_2 (d_2) ----> R_2	R_1 (d_1) ----> D and R_2 R_2 (d_2) ----> D and R_1	R_1 (d_2) ----> D R_2 (d_1) ----> D

(b)

S_1 (d_1) ----> R_1, R_2 S_2 (d_2) ----> R_1, R_2	R_1 (d_1) ----> D R_2 (d_2) ----> D	R_1 (d_2) ----> D R_2 (d_1) ----> D

(c)

Figure 7-1: Fixed relay-enabled user cooperation: Realization I and II: (a). Relay-enabled cooperative diversity scheme, (b). Protocol for realization I: SDMA is used to separate the users at the relay, Protocol for Realization II: MUD is used to detect the cooperating partners

The three-slot protocol combines the two-hop relay network with two (virtual) transmit antenna scheme. The sum of the times allocated to the three slots is made equal to that of the reference scheme for fair comparison. Alternatively, to allow for the same bandwidth with the reference scheme, a higher modulation constellation could be employed in the links.

The second realization uses MUD at each relay to make detections of the signals of the two users. In this realization the system could operate in a mode we will refer to as a ``pre-determined'' space-time coding, in which case, the relays do not exchange any instantaneous information as the cooperation is agreed upon a priori. One advantage of this is the removal of the full-duplex requirement on the relays (i.e. the relays do not need to transmit and receive at the same time). In this mode, each relay assumes that the signals of the users it detects have the same reliability as those detected by the cooperating relay. They simply employ these signals to realize the distributed space-time coding. In the implementation of this scheme, however, automatic repeat request in conjunction with cyclic redundancy check might be required to alleviate the problem of performing space-time coding with erroneous data.

Finally, a distinguishing feature of the relay-enabled cooperative schemes (Figures 1 and 2) is that it can be used in any network since they do not require fundamental changes to the user terminals. This constitutes another important advantage of this scheme. The price for deploying the fixed relay-enabled cooperation scheme in a network is the increase in the complexity of relays; although this may translate into a cost higher than that of the conventional relays we argue that the benefits of the proposed scheme far outweigh this cost increment. For instance, it is more economical to increase the cost of few relay stations rather than the cost of changing thousands of terminals. This is in addition to a service that is guaranteed through incentive-free, coercion-free, and secured networks.

7.2.2 Numerical Results

Figure 7-2 shows some results of the relay-enabled user cooperation where the relays use SDMA mode of detection. Each user has single antenna and the destination has one antenna but it can implement Alamouti-type receiver. Results are shown for various numbers of antennas (L) at the relay stations.

For $L = 2$ the new scheme provides each of the two users in the network with an error rate that is not inferior to a single user network. Eight-PSK modulation format is adopted in both networks. This error performance advantage comes at the expense of 33% loss in spectral efficiency. The situation is different for large L. For instance, when $L=3$ the performance of each user in the new scheme is superior to that of the single user network using BPSK. This represents both diversity and spectral (multiplexing) gains. Considering the overall network, an improvement in spectral efficiency of 100% is achieved with the new scheme in addition to the large SNR gain at low error performance. The new scheme achieves an SNR gain as large as 7.5 dB (for $L=3$) and 9 dB ($L=5$) over the BPSK at a BER of 0.001. The margin of the gain is larger in low BER regime. In the uplinks, the base station is the destination. Therefore, the number of antennas could certainly be more than one. The discussions so far, therefore, represent pessimistic results. Figure 7-2 also shows the system performance when the destination uses two antennas. In this case, each antenna of the two (distributed) relays and the two antennas of the destination form a virtual 2x2 antenna scheme that could emulate conventional 2x2 Alamouti scheme (or MIMO channels). We observe a tremendous performance improvement due to the additional antenna at the destination. For instance, and a BER of 0.0001, the virtual 2x2 of the second hop is superior its 2x1 counterparts by about 8 dB. Further performance gains can be obtained if the relays use their multiple antennas to transmit.

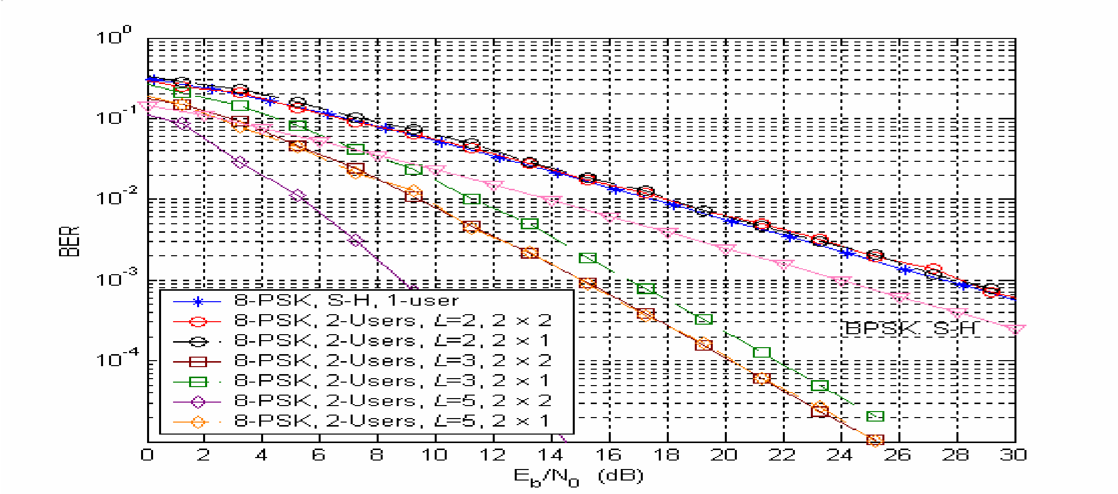


Figure 7-2: Performance of relay-enabled user cooperation in Rayleigh fading channels with one or two antennas at the destination (forming either virtual 2 x 1 or 2 x 2 in the second hop).

7.3 Cooperative Mobile Relaying

As for the case of the Mobile Relay-based Deployment Concepts, Cooperative Mobile Relaying (CMR) has also been referenced in previous deliverables and other occasions. [WIND34] [BaLe05] This section addresses the applicability of CMR to a WINNER-based system and identifies which of the three MR concepts (if any) would be of more gain for the WINNER system. Following the steps in section Annex C there are two perspectives for CMR either the cooperative relaying "advantage" will be used as a criterion to choose the MR concept or first the MR concept will be selected and then we will see if and under what circumstances we will apply CR for that concept.

In section Annex C a short analysis of the connectivity between Type II and Type III mobile relays as well as a connectivity investigation for Type III relays has been performed and what is actually shown is that although MRN Type II could cover a large number of cases it is only Type III that can cover effectively all of the cases. Another issue is the number of hops. As in the case of fixed relays, CRM will be more cost efficient if the number of hops is kept to a minimum e.g. 2. Additionally, complexity is increasing substantially thus in some instances due to e.g. delays, we might be negating the initial rational for deploying CMR.

In general terms and taking into account the discussions in the previous deliverables we can state the following

- Cooperative Relaying is a promising technique that when added on mobile relays could be of gain
- Due to the mobility of MRs an algorithm needs to be in place to evaluate on regular intervals or based on triggers, the needs and the available resources (e.g. number/types of MRs) in order to select the optimum relay(s) for CRM
- Type I is not applicable for CR due to the very deterministic nature/topology of that concept and the very good links between BS-MRN and MRN-UTs. Thus the incremental gain would be quite small.
- In the same lines is also the Type II. Although quite powerful and with favourable characteristics, still the relatively undeterministic movement/distribution within a geographical area, means that CMR will be relatively complex to implement. However, it could be further investigated what are the specific requirements to deploy CMR in Type II and in some occasions (special topologies, distribution of common content/information) they could be of high gain.
- Type III MRs seem to be the most promising out of the three concepts. This is again mainly to the large number of terminals that will be deployed in future networks which means that at any location, at most times during the day, terminals will be available in order to make use of their cooperative mobile relay opportunities. Of course some limitations will exist in supporting complex processes, especially when compared with Type II MRs. However, they are expected to cover the largest percentage of the cases e.g. 70%

- In general, due to the mobility we are aiming to a more opportunistic use of mobile relays i.e. use CRM only if available and favourable characteristics rather than a "deterministic" use of it. As such, CRM could be a "good to have" feature i.e. optional. Thus, under this point of view, by Type III we can come close to a quasi-deterministic system where at any time we expect to find a terminal (optimum or sub-optimum characteristics) in certain locations to be used under the CRM scheme.
- With Type III, however, some disadvantages exist, as highlighted in the general MR section. These are mainly with regards to security and complexity/cost of future terminals. Specifically, we expect that future terminals will incorporate more complex functionalities compared to today's terminals. However, by including more relay functionalities the production and purchase cost could rise increasingly. Additionally, users might not be feeling very secure if their content is passing though another user's terminal. These drawback, however, are not valid for Type I and Type II.

The above crystallise the main finding with regards to the Cooperative Mobile Relaying. The intention is to use these findings a part of the general Mobil Relay analysis, which can be found in Chapter 8.

7.4 Cooperative Relaying Protocols

In [WIND31] a classification of possible cooperative relaying protocols was already presented and most of the shown alternatives were investigated. Based on his analysis Simple Adaptive Decode-And-Forward (AdDF) (and Decode-And-Reencode (AdDR)) presented in [HeZiFe04] shows to be the most promising protocol for Cooperative Relaying. As almost every Cooperative Relaying protocol it is also devided in two phases:

- Phase 1: The source broadcasts its information to both the destination and relay node (probably degraded due to fading and noise). If the source to relay channel has a sufficient SNR the relay decodes the message.
- Phase 2: If the relay decodes the message it sends a newly encoded version of the original message to the destination with either the same code (AdDF) or a different code (AdDR). The destination has now two versions of transmitted source message which it combines using maximum ratio combining. If the relay could not decode the message both the source and the relay will remain silent during the second phase.

The key feature of this protocol is to avoid an unconditional decoding which results – as [LaTsWo04] shows – in worse performance than direct transmission. In comparison to the unconditional decoding protocol (DF), which only achieves first order diversity, the Simple AdDF/AdDR achieves second order diversity through the utilized spatial diversity offered by the source and relay.

An important criterion for the evaluation of Cooperative Relaying protocols is their flexibility, i.e. it should be viewed as a powerful extension which can be easily integrated in the existing system. Another important criterion is the transparency for the mobile terminal which wants to use cooperative relaying. Both criteria gain their importance from the fact that a) Cooperative Relaying must not be a required part of the deployment concept since there need not to be a direct *and* a relay link available and b) the complexity should be shifted to the relay to avoid unnecessary costs and complexity at the UT's side. This supported flexibility and transparency makes Cooperative Relaying very interesting for scenarios where no direct link between mobile and base station is guaranteed.

Figure 7-3 shows a comparison of direct transmission, conventional relaying, transmit diversity (i.e. two antennas at the mobile) and the Simple AdDF/AdDR. For this analysis is assumed that the channel conditions are degraded by Rayleigh fading, pathloss which non-linearly depends on the distance between two nodes and that the RN is located halfway between UT and BS (further effects like shadowing are not considered)

The drawback of this protocol is the fact that it needs orthogonal channels either in time or frequency domain which causes a doubling of the rate used for direct transmission. This is the reason why cooperative relaying protocols gain their benefits more in the higher SNR-regime and the lower rate-regime, respectively. Figure 7-4 shows the beneficial rate region for Simple AdDF/AdDR by plotting its SNR gain over direct transmission at an outage probability of 10^{-2}. This figure points up the fact that for lower spectral efficiency (R=1) Simple AdDF/AdDR outperforms transmit diversity but at higher rates

(R>5) it is even worse than direct transmission. Although it needs not to increase the performance comparing to the direct transmission it performs in any case better than or equally well than conventional relaying.

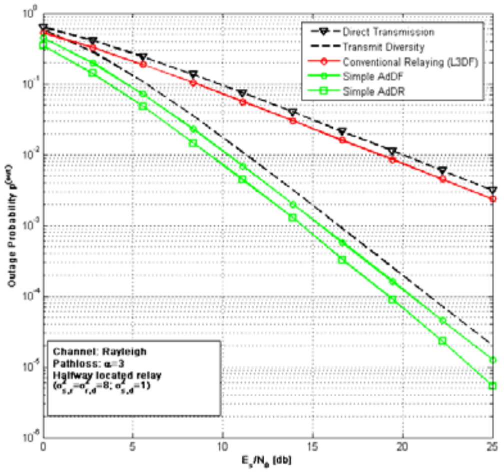


Figure 7-3 Outage probability of Cooperative and Conventional Relaying.

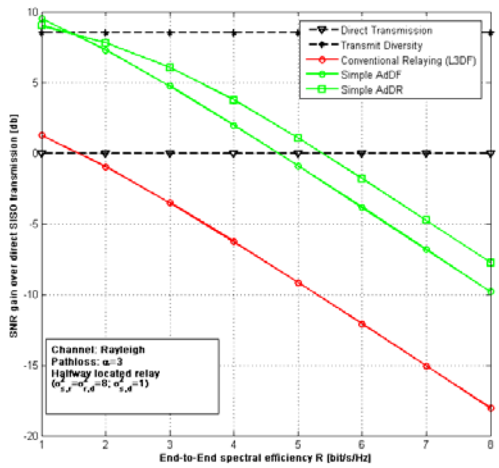


Figure 7-4 SNR loss of Cooperative and Conventional Relaying in comparison to direct transmission.

Beside its flexibility and transparency for the user and its 'ad-hoc nature' in comparison to the conventional relaying it has further properties which make the Simple AdDF/AdDR highly adaptable to the current deployment concept and therefore robust to future developments. Among these properties is the fact that this protocol is not restricted to any number of antennas (see chapter 7.5 which analyzes the MIMO relaying case). Additionally [HeZiFe05] shows that although this chapter only examined the 2 hop case the Simple AdDF/AdDR is easily extendable to multiple serially concatenated relays (even though the protocol achieves its best performance in most scenarios with 2 hops as also [HeZiFe05] shows) and to parallel relays. Another advantage is that the protocol is not restricted to a one-mode architecture but can easily be used for different modes, e.g. between UT-RN and RN-RAP. Furthermore it is not constraint concerning the used orthogonal channels, i.e. it can be used both in an FDMA, TDMA and MC-CDMA system, which actually is the preferred WINNER architecture.

A still opened field of research is the solution of the CR's problems in the low SNR/high rate regime which should be solved in Phase 2 of WINNER. This is necessary since WINNER will operate in a high rate regime where cooperative relaying is only in the higher SNR-regime beneficial. To be able to profit also in the lower SNR-regime – which actually means power and therefore cost savings – it is necessary to overcome the rate-doubling problem.

7.5 Cooperative Cyclic Delay Diversity

A method that introduces artificial *frequency selectivity* and *spatial diversity* in a cooperative relaying wireless communication system is proposed. The artificial frequency selectivity is exploited in conjunction with forward error correction coding to provide a coding diversity gain. Each of the relay nodes consists of one or more antennas. The Base Station (BS) transmits to *M* Relay Nodes (RN) and the User Terminal (UT). The relay forward the information received from a first node (e.g. BS) to a second node (e.g. UT) using cycle delay diversity (CDD)[WiKiLi01][DaKa01][BoHuScCoHa02]. This can be done in either with amplify and forward, decode and forward, or a hybrid.

An illustration of a cooperative relay system is shown in Figure 7-5. In this example, one BS, one UT and *M* RNs are depicted. The relay nodes demodulate and/or decode the signal (decode and forward based relaying assumed) received from the BS and forward the information to the UT using cyclic delay diversity. This is done in two steps:

- Step one: the BS transmits data, which is decoded by 1) the relay nodes where the information is stored 2) by the UT.
- Step two: each relay node encodes the data and applies different cyclic shift on different antennas and adds the cyclic prefix before transmitting the signals. The UT receives the

combined signals and decodes the data which may be combined with the data obtained from step1.

At the BS, the signal is subject to OFDM modulation before the symbol is transmitted over the air. A different or random cyclic shift δ is applied at each relay node to the received OFDM symbol from the BS. The receiver structure at the RN is shown in the top of Figure 7-5. For each receive antenna, the data is first down-converted from the RF-band into base-band and then the Cyclic Prefix (CP) is removed. The data is then subject to an FFT operation and equalized. The data estimates from all receive antennas may be combined using the Maximum Ratio Combining (MRC) method. The coded output data is then stored in order to be processed and forwarded at the next time slot. Here there are two possibilities:

- The coded output data is Modulated and Forwarded (non regenerative relaying).
- The coded output data is Decoded, Re-Encoded, Modulated, and Forwarded (regenerative relaying).

Forwarding is implemented using the CDD method. This is implemented by simply cyclic shifting the OFDM symbol at each antenna. A different cyclic shift δ is applied on different antennas. Following the cyclic shift, a guard interval is applied on each branch. The GI is implemented using the Cyclic Prefix method. Then the signals are up/converted from the base/band into the RF-band and transmitted. An example of the transmitter in the RN is shown identical to the transmitter at the BS. The receiver at the UT is identical to the receiver in the RN.

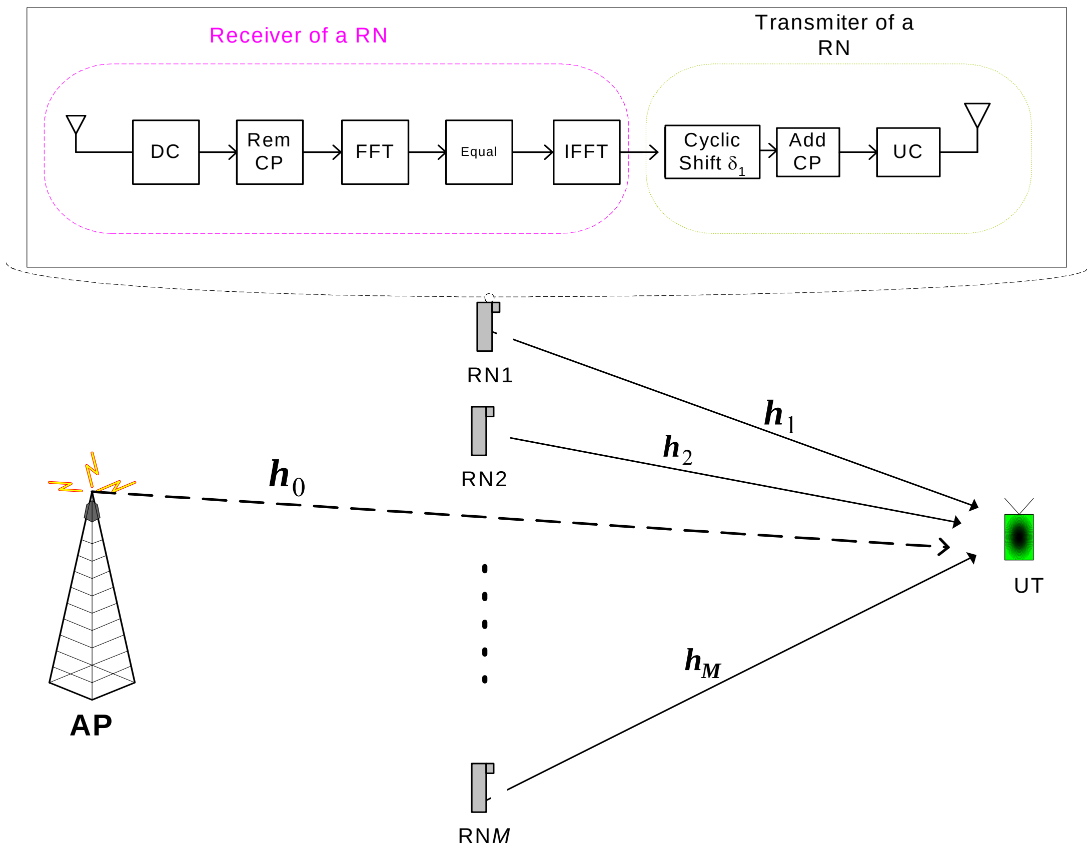


Figure 7-5: Example of CDD distributed relays

7.6 Conclusions and Further Work

We have investigated various cooperative techniques and protocols within the framework of the WINNER deployment concept. All protocols are able to increase data rates and coverage at lower costs. The investigated techniques include multi-antenna cooperative relaying, relay-enabled cooperative diversity techniques, cyclic cooperative diversity while the protocols include adaptive decode and forward and adaptive decode and re-encode strategies.

The above mentioned techniques have been studied extensively and qualitative discussions on their complexity have been motivated in this report and the previous reports [WIND32] [WIND34]. Here, we present the conclusions from the study of the techniques contained in this current deliverable.

The Relay-enabled user cooperative diversity scheme is proposed to circumvent the problems of the user-dependent or explicit user cooperation diversity schemes. The impacts of the new scheme are listed below:

- The scheme can support *heterogeneous relaying mode*.
- The *privacy* of the users is not compromised. Partners do not detect each other's data as is required in the explicit user cooperation.
- *Sanction or reward* to facilitate user cooperation are not required
- Improving network quality and service *does not depend* on any particular user.
- **No** *modifications* are required on current terminals. In contrast, explicit user cooperation scheme requires that terminals be modified to adapt them to their new cooperation tasks.
- Finally, the new scheme provides bit error rate and network spectral efficiency that are *superior to the referenced conventional transmissions.*

Cooperative Mobile Relaying has also been presented. Advantages have been highlighted and especially with Type III (terminals acting as Mobile Relays) gains can be obtained, mainly as a result of the large number of future terminals that will be deployed, their low mobility high connectivity and the moderate complex functionalities that future terminals will have incorporated into them, thus possibly being able to support simple cooperative relaying functionalities. However, issues with security may be the main bottleneck for such a scheme to be taken forward. Still, as stated in Section C.3, the intention is to present cooperative mobile relaying as a criterion for promoting the Mobile relay concepts. Thus, first we need to justify the rational for any of the MR concepts to be included in WINNER and then to see if that concept could be used under the cooperative relaying concept.

A new relaying method called Relaying Cyclic delay Diversity (RCDD) was presented. RCDD introduces frequency and spatial diversity that can be exploited through scheduling, hence improving substantially the SINR in the downlink. RCDD requires only a time delay block module compared to conventional relaying, a minor increase in terms of hardware complexity. It is interesting to note that conventional relaying can be seen as a special case of RCDD. Furthermore, this method can be used together with the simple AdDF. Another major advantage of RCDD compared to a conventional cooperative relaying system is that no additional pilot (signalling) per relay node is required compared to a one hop system.

Simple Adaptive Decode-and-Forward protocol has been shown to be a promising cooperative relaying protocol. This relaying protocol represents a slight modification of the conventional relaying but with excellent return in terms of performance improvement. As explained earlier, the AdDF presents a transparent protocol from the user's perspective as it is also transparent concerning the network architecture since the only add-on is the additional usage of the direct link at the base station. Since the source does not transmit any additional information in the second phase (as it does in the Receive Collision or Complex AdDF protocol), neither additional signalling is necessary nor an added propagation delay or additional problem due to required synchronization will occur. It only requires simple algorithm (software) to combine the direct and relayed signal. Since high computing power is available at the base stations, it can be assumed that this additional computational effort is a minor issue. Therefore additional costs are minimum – if there are any – and it does not claim to be an integral part but an add-on to achieve second-order diversity (assuming the SISO case).

On top of the above mentioned particular benefits of the investigated methods in this report, another advantage of the cooperative relaying protocols is their flexibility and adaptibility to different scenarios. Cooperative relaying can be envisaged as an alternative in those scenarios where no diversity (e.g., time or frequency diversity) can be exploited or the spatial diversity offered by cooperative relaying results in a better performance than conventional methods. Further studies, which remain to be done in Phase II, may include the investigation of the cooperative relaying performance in multi-user/multi-cell scenarios, the integration of cooperative relaying into the WINNER DC and further qualitative and quantitaive investigations on the complexity added by cooperative relaying.

8 Mobile Relay-based DCs

8.1 Introduction

Mobile Relay (MR) – based Deployment Concepts (DCs) and relevant technologies have already been presented so far in previous deliverables [WIND31] – [WIND34] and in other occasions [BaLe05][Bakaimis05]. The intention was to make a top-level analysis of those concepts, highlight initial requirements for each of them, list relations with fixed relay-based technologies, present simulation results and in general make a first assessment of how those DCs could be integrated in a future WINNER-based system. Additionally, a similar analysis was performed which basically included a very concise selection of the main characteristics/idiosyncrasies (advantages, disadvantages, requirements) describing each deployment concept.

Thus, the intention in this section is to conclude this initial high level study by highlighting the main points for each DCs and specific "usage scenarios" that MRs could be used for, including some final analysis/simulation results and making a selection on which of those concepts (if any) should/could be taken forward for future research. In this analysis, a holistic point of view will be taken including aspects of technical performance, complexity, cost, user approval, future trends of cellular systems, impact to a WINNER-based system.

8.2 MR Deployment Concepts

Future 4G cellular systems will aim into two areas.

- Provision of higher bit rates for the UL/DL compared to e.g. 3G/B3G systems,
- Provision of seamless services across any type of environment for any time of the day, i.e. uniform/seamless connectivity.

One of the main targets of the relaying concept is the latter one i.e. provision of coverage/capacity in areas that require that i.e. zero or limited coverage/capacity. Although peak data rates are more important for the operators, from the user's point of view it seems that more important will be the latter one. It is expected, extrapolating from current trends, that users will be more interested in being able to use their mobile phones wherever they are, for relatively simple processes e.g. downloading songs, short video clips, latest goals from football matches, local/tourist information, positioning, weather information etc. which can be covered by low/medium BW requirements, rather than jeopardising this "seamless connectivity" for the sake of the provision/availability of very high bit rates in just designated areas e.g. hot spots. What is more, taking also into account any PAN-related (Personal Area Network) requirements, it seems that this seamless connectivity will be a cornerstone for future telecommunication environments. (Of course, that does not negate the need for provision of high BW.) In order to promote this vision, mobile relays could be used, due to their main advantage, that of being present in different times during the day in multiple locations. A number of aspects for Mobile Relays have been described so far. In TABLE I we present a final selection of the main advantages and disadvantages for the three MR-based Deployment Concepts, incorporating issues raised in [WIND31] – [WIND34].

TABLE I MOBILE RELAYS COMPARISON

MRs	Advantages	Disadvantages	Other comments
Type I	Cover many UTs with one BS, Always on, Continues/Reliable connectivity with UTs, high processing/computational power	Many handovers, limited commercial deployment e.g. countries with no trains	Should be extended to other scenarios/cases
Type II	More commercial cases to Type I, easy to build (e.g. supporting MBMS), lower connectivity with ref. to Type I, high processing power, unlimited power supply, could support cooperative relaying	Not fully predicable movement, large numbers, relatively high cost, problems when hops>2	Could be extended in a fully mesh network, if we manage to integrate with vehicular networks.
Type	Large numbers, low-high complexity, multiple locations, no CAPEX/OPEX	Limited power supply, questionable users acceptance, complexity of	In 10-20 years terminals might have "advanced"

III	cost, higher connectivity compared to Type I, frequent positioning, could support flexible cooperative relaying	algorithms, increased cost of mobile phone, problems when hops>2, security issues,	to that level to "cover" Type II MRs

8.3 Impact in a WINNER-based system

Further to the list of advantages/disadvantages presented in the previous section, a list of WINNER specific issues with reference to MRs has been created (requirements, drawbacks, open issues) as shown in TABLE II.

TABLE II MRS WITH RELATION TO A WINNER-BASED SYSTEM

	Answer: Yes
Impact	• Reusability in multiple Locations/ToD (Time of Day) • Cover cases where mobility is required e.g. ships, buses, trains • Be used on an Ad Hoc basis i.e. when needed • More adaptability - mobile RNs would be beneficial in case of cooperative relaying, however such approach will definitely render the system far more complicated • In general: Provide flexibility to cover multiple locations/in multiple time instances for multiple needs
Require.	• Mobility of RNs must be supported (e.g. RN HO from one serving BS/RN to another needed) • Fast Resource Partitioning at BS or "out-of-band-relaying" needed • Fast routing updates needed
Drawback	• Mobility/Frequent Positioning • Limited availability (time-wise) of RNs Type I/II respectively ◊ many RNs needed to make DC work for Type II MRNs • Added complexity • Completely different sets of protocols might be needed (complex system) for Type II / III. • Multiple handovers especially for Type I • Large numbers Vs Cost for deployment
Open Issues	• Power Supply of mobile RNs Type III • Might be too complex for implementation • Is the protocol overhead small enough? Probably the network can stuck due to permanent changes of the RN? • Number of hops? Flexibility vs Complexity • Types of protocols supported Vs cost Vs number deployed [used for MBMS / Positioning?]

From the above it can be seen that a WINNER-based system could be highly impacted by the introduction of MRs. It is evident that all MR types can increase substantially the scalability, flexibility of the system, but at the same time they increase considerably the complexity of a WINNER system. Thus, under this perspective, two steps could be followed.

- Select and implement one of those concepts
- Select specific functionalities to be incorporated into one of the concepts in order to reduce its complexity, impact to the whole of the system, cost etc.

Both "steps" can be used in a complimentary way to each other. For instance, a MR concept could be selected only because a cut down version of it (following the second point) could be implemented compared to full/non-scalable versions of the other concepts. We will see that in the case of cooperative relaying. (Section 8)

With regards to the second approach, in the previous deliverables we have touched upon some functionalities that could be implemented in a MR in order to simplify its design. A short analysis which is included in the Chapter 13 (ANNEX) points to the following

- Positioning. MR can be used in two ways.
  - Timing measurement provide better accuracy when applied to MRs instead of BSs due to the smaller coverage of MRs thus better channel conditions in the MR case.
  - Increase the pool of techniques to be used in cases when we have limitations e.g. isolated sites. MRs effectively can "mimic" a BS.
- MBMS
  - Mobile relays could be used for MBMS purposes. By supporting only common channels there is no need for e.g. UT feedback, power control issues. Thus, no UL signalling between UT-MRN takes place. MBMS could be both for user-originated or network-originated content. The gain from this is the provision of services (MBMS will be important in the years to come) with reduced impact e.g. interference, signalling, thus freeing capacity, while providing better coverage and capacity in designated areas. More analysis can be found in [Bakaimis06].

The above point to the introduction of only "common" channels within a MR. In the equivalent of a UMTS system only CPICH and P-CCPCH need to be used. The only difference might be the introduction of a UL MRN→BS link, but this is to be investigated. In order to support the above two functionalities, the analysis has also shown that non-fixed power patterns should be used for the Tx power of the MR. This analysis has shown that variable pattern based Tx power levels should be incorporated, effectively making sure that the MR transmits only what is required and not higher or lower to those levels. The criterion to follow is mainly their area of coverage. This is mostly applicable for Type II MRs, where mobility is more predictable, but with some modifications this could be extended to Type III. The final part of the analysis is included in Annex C.

8.4 Discussion and comparison of concepts

In order to select any of the above concepts a number of "technical" factors e.g. complexity, performance, signalling requirements etc need to be taken into account. What is more, the final implementation and deployment will also depend on a number of non-technical factors e.g. user acceptance. Additionally, we have to bear in mind factors with reference to the WINNER project and the impact that MRs will have in relation to the whole system e.g. architecture/design. We have also to bear in mind relations and impact to standardisation, so that we ensure any type of backward/forward compatibility. Thus, we have to consider all of these under a more holistic approach to justify the selection of a concept for further research.

Thus, having into consideration the above we can reach the following conclusions:

- From the technical point of view, Type III seems to be very promising due to the fact that in future networks we will have a large number of already deployed terminals. What is more, if we extrapolate from current trends we can envisage that future mobile terminals will be medium to high end e.g. PDA, with good battery life and high processing power, which means that integration of relaying functionalities will not be such a heavy burden compared to today's terminals. If we combine the above with good incentives from operators (so that users will not disable relaying functionalities from their terminals), mechanisms to make uniform and fair use of those terminals (so as not to drain the battery of only some terminals) and making sure that initially some simple functionalities (positioning, MBMS) are integrated into those terminals (which will also keep terminal cost in moderate levels), then this might be a very promising addition to a WINNER-based system. However, probably the main bottle neck, which appears that will prohibit any future research to be continued (also in the framework of the WINNER project) is with regards to security issues (e.g. user's content passing through another user's terminal and possibly high complexity if we assume that full relaying functionalities are integrated into future terminals.
- Type II seem quite complex mostly with reference to their deployment. Large numbers are required and due to their not fully deterministic (most of the time) movement it is difficult to rely on the existence/presence in certain locations, thus making use of them. Increasing the deployed numbers might be a solution, but then cost will become a major issue.

- Type I seem promising due to the relatively simpler approaches compared to Type II e.g. UT population-to-cover stationary in relation to the MR. The main problem the relatively limited cases e.g. trains that they address and also the high velocity of those MRs which has a direct effect on handover issues and deployment of BSs. For instance, a moving network–tailored WINNER system will need to have BSs next to the railway lines, which means that normal rural coverage will not be optimal and a high number of BSs will need to be deployed. This issue is touched upon in [Bakaimis06]. Similar work to that of Type I is being performed in other for a e.g. IETF [IETFMANET][IETFNeMo] and the AMBIENT NETWORKS IST project [AN], so it would be interesting to see what parallels could be drawn, although they tend to look more on higher layer e.g. IP.

Additional issues that is good to keep in mind is any relations and future extension to cooperative relaying, mobile AdHoc networks, heterogeneous relays etc.

8.5 Conclusion and selection

Based on the above analysis, we concluded on the following ranking of the deployment concepts

- Type II. Not a promising technology due to the large number of issues that need to be resolved
- Type III: Very promising but due to the limited but apparently severe problems like security and complexity, it will not be taken forward.

This means that Type I, although with a relatively limited applicability e.g. only on trains, it is felt that due to the small impact to a WINNER system it will be the one to incorporate more easily.

9 Conclusion and Outlook

The actual work of WINNER is focussed on key technologies for a B3G/4G air interface that allows broadband radio access in several scenarios. One key technology has been the introduction of relays as integral part of the WINNER system concept.

Within this deliverable it was shown that the design of relay based deployment concepts is not only the design of focussed technologies, but is to a large extent also the work on the system architecture. This is underlined by Chapter 5 where the WINNER system archite cture is shown. The WINNER system architecture been designed in a way that the relay nodes are directly visible as own logical node reflecting that relays are an integral part of the system concept.

The protocol work of Chapter 6 shows that the integration of decode and forward relays addresses several layers starting from the MAC layer over the RLC up to the RRC layer. It can be seen that the efficient relaying is to a large extend related to the radio resource management functions as can be found in the RRC and MAC layer. In the solutions shown here the RN has been handled by the RRC in terms of resource partitioning as a BS, but which will need some specific protocols for the distribution of such information.

The MAC plays an important role in the relay based WINNER system as the MAC layer at the RN is responsible to perform an efficient forwarding of user data with as less signalling as possible. In the section on MAC the current status of MAC design has been shown with focus on protocols which the MAC layer has to provide in order to support efficient relaying.

9.1 Deployment Concepts best suited for scenarios

The deployment concept developed in WP3 provides logically a common WINNER deployment concept which is suitable for all scenarios. This means that all scenarios will rely on the same logical nodes architecture as presented in Chapter 5 and use relays to enhance coverage and/or range as presented in Chapter 1. Some differentiations with respect to scenarios have to be made for the physical deployment concepts and for the usability of different relay types as outlined in Section 6.3. Further the positioning of BS and RNs in the specific deployment, which means having antennas above or below rooftop will make the difference for scenario specific deployment solutions.

9.1.1 Local area coverage:

The local area coverage (A.1, A.2 with respect to the table shown in Section 3.3) is regarded to allow the deployment of single cells, which could also be relays enhanced cells, e.g. to overcome walls in the indoor area. The physical nodes architecture would most likely contain a physical BS which comprises the logical nodes ACS, RANG, AR and BS as no direct interconnection with other BS is needed.

In such a scenario relays would be mainly use to cover areas which are otherwise shadowed from the BS or suffer from high pathloss due to obstacles. The RRM functions could be rather simple as the user will have low mobility and only little or no co channel interference is expected. With respect to the missing interference the functions for radio resource control could be simplified as, e.g. no sharing of radio resources between BS is required.

9.1.2 Metropolitan Area Coverage

Metropolitan area coverage is seen from two perspectives. On the one hand a lot of users with low mobility and the need for high data rates are expected (Test scenario B.1 and B.2). Such scenarios will most likely be covered by deploying the RN and BS antennas below rooftop in order to allow LOS connection. RNs can be used to enlarge the coverage area and capacity of a BS and to cover areas, like side alley which otherwise would have been shadowed from the BS by buildings and other obstacles. This scenario would be multi-cellular. Thus the BS is likely to comprise only the BS logical node with the respective WINNER and transport network protocols. The RANG and ACS would then be placed as central nodes serving several BS and RNs inside the RAN to allow for efficient mobility support and radio resource management between the cells.

The second type of metropolitan area scenario will cover the smaller number of users that are going to use the WINNER system in a more mobile scenario travelling with speeds up to 70 km/h (test scenarios C1, C.2, C.3, C.4). In this case a larger cell size would be required. Thus the positioning of BS and RN antennas above rooftop seems to be the better solution, which allows the placement of relays mainly to increase the range of one BS. LOS connection between the BS and RN can be easily realised in such a scenario. Further relays can be used to allow for good indoor coverage, potentially by means of heterogeneous relays as described in Annex B. The physical nodes might look as described for the first Metropolitan area scenario.

9.1.3 Wide Area Coverage

Wide area scenarios are corresponding to the test scenario D.1 and D.2. In this case also large cell ranges are envisaged with antennas above roof top. Indoor coverage could be realised by heterogeneous relays again. Here also Type II relays as described in Section 6.3 could be used to increase the capacity in outer cell regions.

The moving network scenario (D.2) will be addressed by a mobile RN solution as described in Chapter 8.

9.2 Outlook

During the work of WP3 in WINNER Phase 1 a lot of issues regarding deployment concepts and the use of relays as integral part of the WINNER system concept have been addressed. Several solutions have been investigated and partly evaluated.

From the protocol perspective a lot of issues have been addressed which are needed to develop a system concept based on relays as integral part of the deployment. These protocols and protocol functions have to be further specified and investigated to get the full picture of the WINNER system concept in terms of performance and complexity.

One important part for the evaluation is the continuation of the system level simulation work. In Phase I some focussed key aspects have been implemented and simulated based on legacy systems showing first performance results for relay based systems (see Annex E) Further a system level simulation methodology has been driven forward and implemented in the different simulation tools. The next step is the implementation of the key technology as final outcome of WINNER Phase I and continuation of system level simulation to provide reliable performance results of the developed relaying technologies.

To use the cooperative relaying technology as shown in Chapter 7 as add-on technology to further improve the capacity of a relay based WINNER system concept an efficient integration in the current WINNER system concept in terms of protocols and related functions is required. In addition also the cooperative relaying technologies have to be assessed by means of system level simulations.

To support the moving network scenario mobile relays as presented in Chapter 8 have to be further investigated. The concept has to be integrated in the WINNER system concept by allowing mobility support for relays, which might even effect the logical nodes architecture as it might require an additional relay node logical node.

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Annex A Details on System Layers II

A.1 RRC – functional details

A.1.1 Proposed Admission Control algorithm

The proposed AC algorithm is interference – throughput policy based meaning to exploit the soft capacity in system interference systems. By other way the UT plays an important role on the decision when the users preferences, QoS, costs are violated enabling at the end the decision to the user/UT. This way the proposed algorithms is not fully network based rather takes into account the chance the user to express their preferences and choices.

In the proposed AC policy based algorithm each service request has different bandwidth requirements, maximum RTT and delay sensitivity. By these session characteristics is assigned a service class and priority. This sense AC may establish priority schemes according the session requirements which could be based on dropping lower bandwidth sessions to serve handover or new session requests of a larger bandwidth class, thus higher bandwidth session may have a higher priority.

In the prioritising scheme could be useful when multiple WINNER modes are available where connections with a low bandwidth requirements can be reallocated to a network optimised (or low cost) for that particular data rate and service provisioning and thus leave high speed connections free for users requiring high QoS. In the non prioritised AC scheme sessions are treated equally being served on a first come first served fashion.

During mode selection user preferences, network priorities, user-defined QOS parameters and costs are evaluated and weighted in the PDP at the user terminal. A list of best candidate modes are then jointly evaluated with the UT measurements such as cell load, coverage to select the target system mode. The UT might also estimated the cell load of the neighbours in case of dynamic cell coverage is applied allowing a dynamic service cell area aiming to share the cell load in the neighbour region.

Only system modes filling the minimum user requirements are considered for mode selection. If the minimum requirements can not be met even having load room for new session requests UT can decide when to decide or refuse the connection according the user requirements QoS degradation/costs trade-off. If no mode is available in the local to provide for instance fast internet connection then the session may request to be served by any RAT.

Different class of users can be considered according their profile, lets say, gold, silver and bronze users. This is important to distinguish the users priority during the network access.

As in the 3G networks in this algorithm both handover and new sessions are request to be evaluated in terms of the incremented fractional load in the cell. However if cell load does not exceed the defined load threshold and resources availability is scarce then handover requests have higher priority than the new session requests. The idea is to give priority to the ongoing session. In the case new high priority session and not enough resources are available in one mode the session might be request to be admitted in another system mode if available under the user decision based on the expected QOS degradation and costs evaluation at the UT. The basic flowchart admission control algorithm between WINNER modes is described [WINXWPRRM].

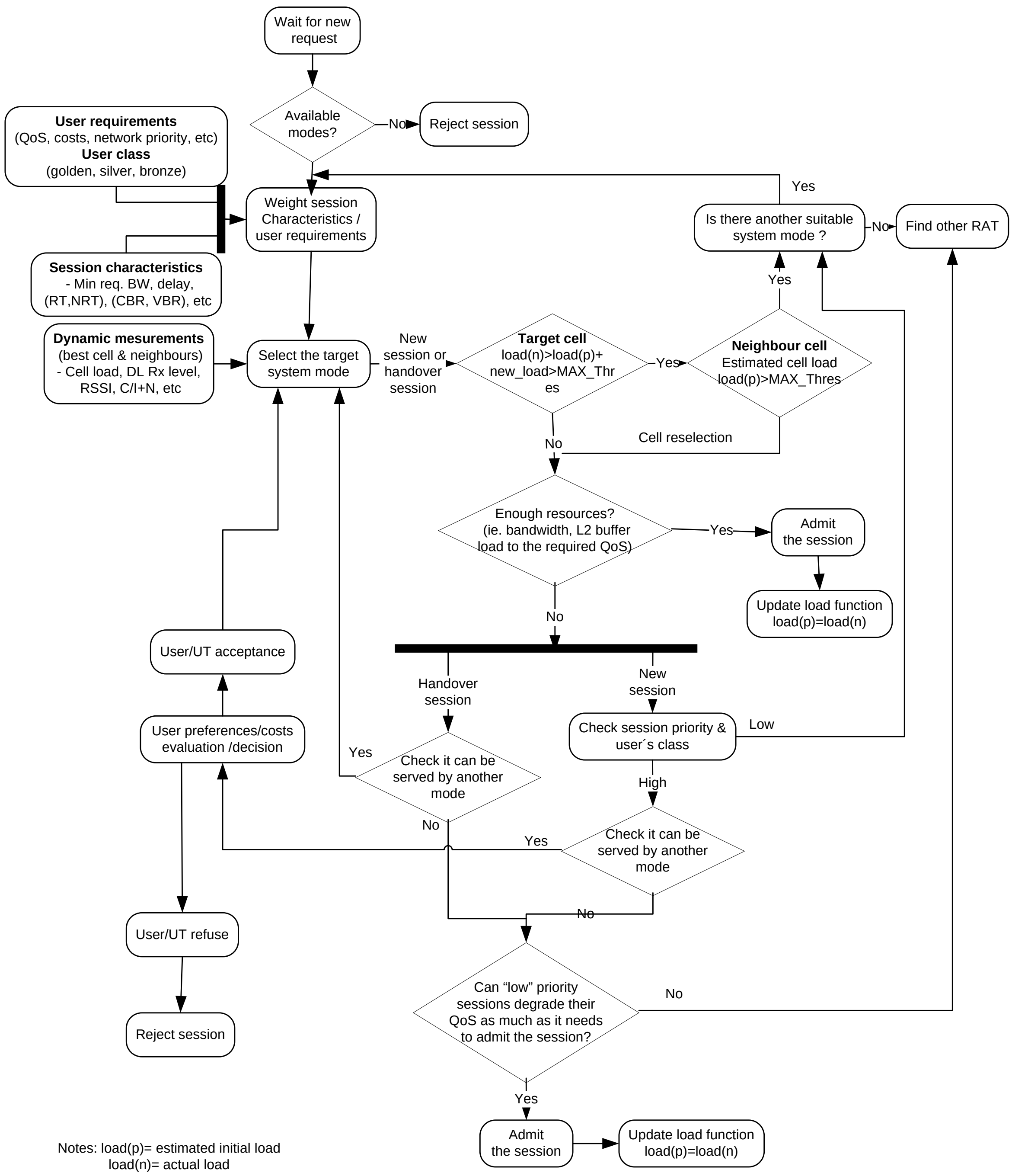


Figure A-1 Flowchart of the admission control between WINNER modes

A.2 RLC Details

A.2.1 Multi-hop ARQ

A.2.1.1 Message Sequence Charts

In the following Figure A-2 an exemplary ARQ message exchange between one BS, two RNs, and one UT is depicted for a DL transmission using Multi-hop ARQ.

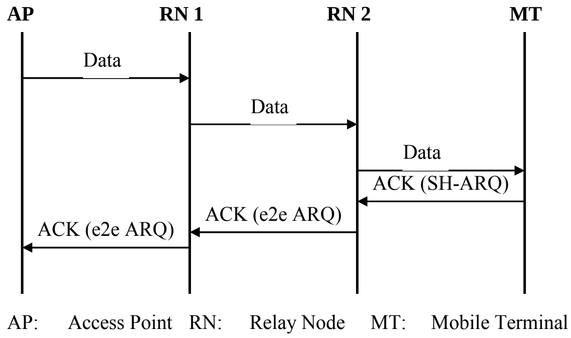


Figure A-2. MSC of M-ARQ for downlink transmission

In the message sequence chart (MSC) the data is transmitted from the BS to RN 1 and is forwarded to RN 2 and finally to the UT. As response to a successful reception of the data at the UT a positive acknowledgement (ACK) is generated by the SH-ARQ "ACK (SH-ARQ) ". This ACK invokes as response a respective positive acknowledgement "ACK (e2e ARQ)" generated at RN 2 by the e2e ARQ protocol, which is relayed to the BS.

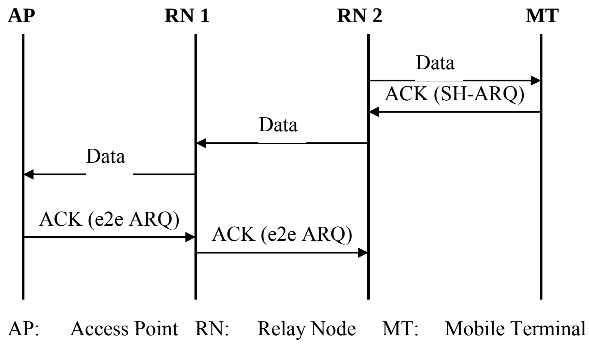


Figure A-3. MSC of M-ARQ for uplink transmission

The procedures for UL transmission are quite similar to those for the DL transmission as shown in Figure A-3. In the UL case the data is stored at RN 2 and will be deleted from the buffer with the reception of the e2e ACK generated at the BS.

A.2.1.2 Advantages

The new approach of two ARQ protocols, one for the spectrally efficient, reliable SH communication and another one for the end-to-end reliability in the MH networks across different number of wireless links is a requirement for the efficient support of handover in wireless MH networks. Moreover, it enhances the reliability of packet delivery over commonly error prone concatenated wireless links. The M-ARQ protocol incorporates the following features and advantages:

1) *Transparency for Mobile Terminals:*

The proposed approach is transparent for the UT. The UT will not recognize a difference being connected to the RN or BS. The point of attachment to the radio access network (RAN) is therefore virtually shifted for the UT from the BS to the RN.

2) *Support of QoS in MH networks:*

The guarantee quality of service (QoS), specifically a residual packet error rate, it is a very important feature in future systems to support acknowledged services. Owing to the new coupled ARQ approach an acknowledged transmission over several hops can be efficiently supported.

3) *Low complexity, low coast Mobile Terminals:*

The buffers for the e2e ARQ protocol are located at the BS and RN, but not at the UT. This reduces the hardware costs at the UT. Moreover, the e2e ARQ protocol state machine only exists at the BS and RN and, hence, reduces the complexity of the UT, too. Specifically, the e2e ARQ protocol requires larger buffers than the SH-ARQ protocol because of the larger round-trip time. With the termination of the e2e ARQ protocol in the RN the complexity is shifted from the UT to the MH RAN, i.e., to the infrastructure. Thus, the storage capacity (costs) is concentrated in the infrastructure and supports cheap MTs.

4) *Flexible integration of legacy Mobile Terminals / different ARQ protocols:*

The new proposed architecture and protocol design allows a flexible integration of legacy MTs and MTs using different ARQ protocols. Since the e2e ARQ protocol is separated from the SH-ARQ protocol that runs in the UT, existing and new ARQ protocols can be integrated quite easily. Only a respective coupling function is necessary that interprets the correct delivery of packets on the last hop between BS/RN and UT. No modifications are needed in the UT to operate in the MH network.

5) *Independent optimization of e2e ARQ and SH-ARQ protocol:*

The e2e ARQ protocol is only active in the MH network comprising the BS and one or several RNs. Because of the unreliable wireless links most probably a second SH-ARQ$_2$ protocol running between RNs and the last RN and BS will be introduced, which is an optional feature (see Figure A-4). This second SH-ARQ$_2$ protocol can be exploited in both transmission directions, for the data and the e2e ACKs, whereby the e2e ACKs can be optionally piggybacked with data.

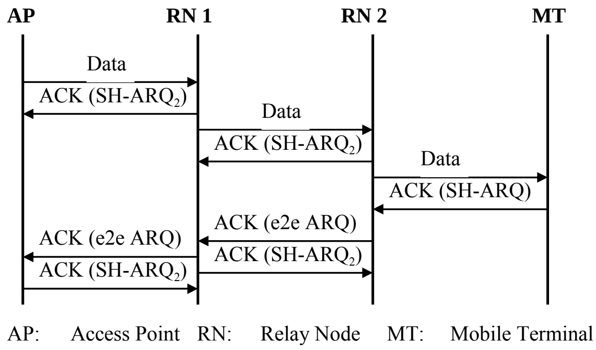
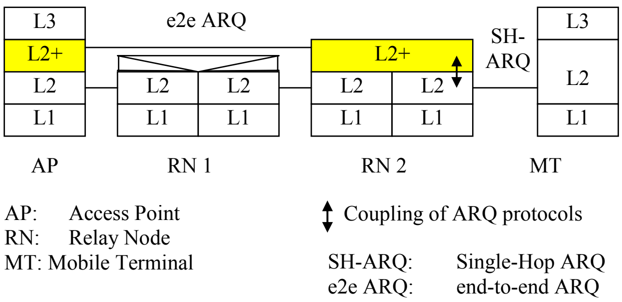


Figure A-4. MSC of M-ARQ for downlink transmission with a second SH-ARQ2 between the BS and RNs

The design of the network can be controlled by the operator and the e2e ARQ protocol and underlying SH-ARQ$_2$ protocol can be harmonized. This can be separated from the task to design and operate an appropriate ARQ protocol on the last hop, i.e. the SH between the BS/RN and the UT. This SH-ARQ



protocol, depicted in

Figure 6-25, can be a different ARQ protocol than the one used between RNs as well as BS and RN. Hence, for each part of the MH network the best suited ARQ protocol can be selected and designed independently from each other. That means, the SH-ARQ$_2$ protocol used for transmission over fixed relay nodes is optimized for line-of-sight (LoS) and stable radio conditions whereas the SH-ARQ on the last hop towards the UT is optimized for unreliable mobile connections, which are characterized many times by non-LoS or obstructed-LoS radio conditions. Moreover, the SH-ARQ$_2$ protocol can be made aware of the overlying e2e ARQ protocol. Furthermore, timing information and status information of the overlying e2e ARQ protocol can be exchanged with the SH-ARQ$_2$ protocol and, hence, the overall performance can be optimized.

In summary, the proposed ARQ coupling can be interpreted as ARQ-splitting approach, where efficient algorithms can be developed for the MH network without the mobile wireless link via separation of the different types of networks (fixed MH network and mobile network).

A.2.1.3 Mobility Support

A handover that takes part between two entities (BS and RN or two RNs) requires an exchange of the ARQ protocol state information of the e2e and SH-ARQ for an efficient operation. The SH-ARQ protocol state information will be passed from the old entity (BS or RN) to the new entity to seamlessly continue the transmission and to avoid requesting of SH-ARQ packets that have been successfully received at the old entity but have not yet been acknowledged. The protocol state information could be, e.g., the sequence number (SQN) of the last correctly received packet. The different possible scenarios for a handover are depicted in Figure A-5.

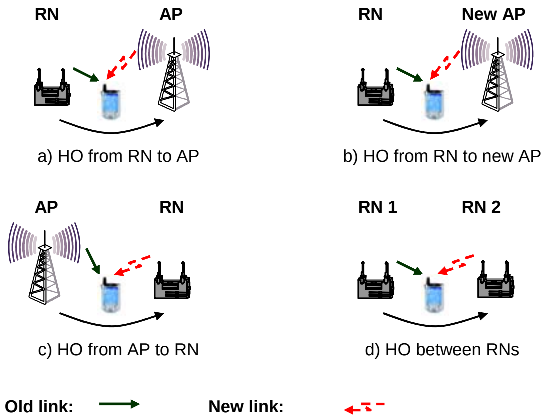


Figure A-5. Handover scenarios

In addition, for each scenario it has to be distinguished between DL and UL transmission. In the following the different possible scenarios for a handover are described.

A.2.1.3.1 HO from RN to BS

After a handover from an RN to the BS there are two different cases that have to be distinguished for the M-ARQ. In the first case if there are still packets missing that have not been correctly transmitted on the DL to the UT, the BS can retransmit these packets since they are stored in the respective queue for the e2e ARQ protocol. In the second case if packets on the UL have not been received at the BS yet, these packets will be retransmitted by the RN until they have been positively acknowledged by the BS.

A.2.1.3.2 HO from RN to new BS

This HO is very similar to the HO from the RN to the serving (old) BS. For DL transmissions all unacknowledged, not correctly delivered packets will be forwarded to the new BS the UT is attached to after the handover. For UL transmissions the packets have to be forwarded first to the old BS and then to the new BS. However, for DL transmissions the packet forwarding is only required if no other means exist in the radio access network (RAN) to support the change of the BS without packet loss, like e.g. simultaneous transmission (simulcast) to both BSs, the serving and the new BS, or context transfer.

A.2.1.3.3 HO from BS to RN

Before a handover from the BS to a RN takes part, the transmitted packets have to be stored in the e2e ARQ buffer unless they are acknowledged by the SH-ARQ protocol running between the BS and UT. This guarantees a seamless continuation after the UT has performed a handover from the BS to the RN. In the DL not acknowledged packets of the e2e ARQ protocol will be retransmitted to the corresponding RN, while in the UL the UT will retransmit to the RN all packets the BS has not yet acknowledged via the SH-ARQ protocol.

A.2.1.3.4 HO between RNs

For DL and UL transmissions the same procedures should be applied like for the handover from the BS to the RN, whereas the old RN takes over the part of the BS and transfers the e2e ARQ state information to the new RN. Packets that have been transmitted in the DL and that have not been acknowledged so far will be retransmitted by the BS. Besides that, the old RN might forward all e2e ARQ packets to the new RN to avoid a retransmission of the same packets by the BS. This forwarding process will only be initiated if the new RN is located more hops apart from the BS than the old RN. In this case the packets have not to be re-transmitted a second time from the BS to the old RN before they can be forwarded to the new RN. In case of UL transmissions the UT will retransmit all unacknowledged packets to the new RN, which will forward these packets to the BS.

A.2.1.4 Conclusions

A new ARQ protocol for multi-hop communication in B3G networks is presented. The mutli-hop ARQ (M-ARQ) protocol realizes reliable end-to-end (e2e) transmission across error prone wireless links that is important to guarantee QoS and efficiently support mobility. Besides a single-hop ARQ (SH-ARQ) protocol operating on the last hop between the mobile terminal (UT) and the access point (BS) or relay node (RN), an e2e ARQ protocol is introduced between the BS and RNs, which is coupled with the SH-ARQ protocol. With this approach an independent ARQ optimization becomes possible since efficient ARQ algorithms can be developed for the MH network without the mobile wireless link via separation of the different types of networks (fixed MH network and mobile network). The M-ARQ protocol inherits the advantages of transparency (the UT will not recognize a difference being connected to an RN or BS), QoS support (acknowledged transmission is supported), low complexity and costs (the complexity is shifted to the MH network, i.e., infrastructure), and many more.

As a consequence of the new proposed ARQ coupling and early termination of the e2e ARQ protocol in the MH system at the RN instead of at the UT, the reliable communication between BS and UT is extended towards the RNs. The unreliable MH connection, which is introduced in a wireless MH system, is made more reliable due to an additional e2e ARQ protocol. Since it can be expected that MH is one important key technology in systems B3G, this approach is a very promising candidate. Consequently, future work will concentrate on the performance evaluation of this new ARQ approach with respect to delay improvements and protocol overhead.

A.2.2 Frame Size and ARQ in Multi-hop Wireless Networks

The communication link between a user terminal and a base station may span multiple wireless links which may have different characteristics. In order to provide reliable data transfer, link layer ARQ is required. However, hop-by-hop ARQ may not be sufficient if data is lost due to handover or buffer overflow in a relay node. One solution would be to provide two layers of ARQ. In GPRS [BeVöEb99, GSM03.60, GSM03.60_97], for example, a radio link protocol operates over the radio interface, between a user terminal (UT) and an access point (BS). On top of the radio link connection, a logical link layer is used, between the UT and a fixed node in the GPRS network. In case of handover between two BSs, only the connection over the radio interface is disconnected and set up again with the new BS. The logical link, on the other hand, remains established and the logical link protocol can recover lost data.

In a multi-hop network, a layered ARQ approach, as in GPRS, could be used, with one ARQ protocol operating over each hop and one over the wireless network, from edge to edge over all hops. One of the drawbacks with this solution is that it is hard to configure the retransmission timers to avoid concurrent retransmissions. In [MeLuWiPa04, WiMeLu05], a co-optimized approach, Relay-ARQ, is suggested to alleviate the problems related to the layered approach. The Relay-ARQ protocol operates both hop-by-hop and edge-to-edge. The co-optimized approach requires that the same link layer frame size is used over all the hops. With a layered approach, on the other hand, the link layer frame size may be different over the hops.

In this chapter, we present TCP throughput simulations in a two hop wireless network using different frame sizes on the hops. In case of Relay-ARQ, these frame sizes are coupled, whereas in the Layered ARQ approach, the frame sizes over the hops can be chosen independently. This study is a first step towards an understanding of the performance merits of the Layered ARQ and the Relay-ARQ approach, but in order to make a fair comparison of the layered and co-optimized link layer ARQ approaches, further work is needed on modelling the link layer overhead. The layered approach results in more protocol overhead, e.g. since both link layers must have sequence number fields.

A.2.2.1 Link Layer Design

Two design alternatives are considered for the link layer ARQ, layered and co-optimized ARQ (also called Relay ARQ) [MeLuWiPa04, WiMeLu05], as illustrated in Figure A-6 and in Figure A-7, respectively.

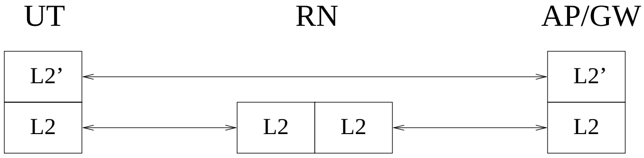


Figure A-6: Layered ARQ approach

In the layered ARQ approach, one ARQ protocol operates over each hop (L2) and another ARQ protocol (L2’) operates over both hops between the BS/GW and the UT at the edges of the radio network.

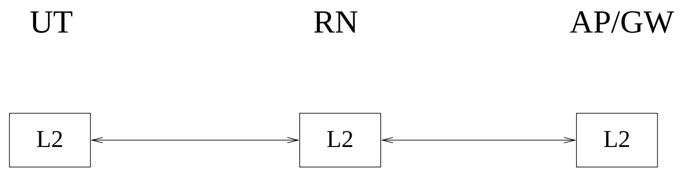


Figure A-7: Co-optimized ARQ approach

In the co-optimized ARQ approach, L2 and L2’ are intertwined which implies that L2 in the edge nodes, the BS/GW and the UT, receive hop-by-hop acknowledgments from the RN as well as edge-to-edge acknowledgments from the node at the other edge. The RN transmits a relay ACK when it receives a frame. The relay ACK corresponds to a hop-by-hop ACK in the layered approach. When the UT (in case of down link transmission) has received the frame it transmits an ACK back to the RN which forwards this ACK to the BS/GW. The ACK from the UT corresponds to an edge-to-edge ACK in the layered approach. For further details on co-optimized ARQ see [MeLuWiPa04, WiMeLu05].

A.2.2.2 Simulation Setup

Figure A-8: Simulation setup

We have used the IT++ simulator [IT++] that is developed within the NEWCOM project. The simulation setup is illustrated in Figure A-8. A user terminal (UT) is connected to a relay node (RN) which in turn is connected over a wireless link to an access point (BS) and a gateway (GW). A L2 protocol that uses selective repeat ARQ operates over each hop. TCP is used between the UT and the BS/GW. The BS and the GW may be located in the same physical node or in two separate nodes. The idea is to model fixed multi-hop homogeneous relaying over two wireless hops. This implies that the RN is stationary and that the radio interface configuration is the same over both hops. This does not imply that the characteristics of the links are completely the same. For the simulations presented in this paper, the physical layer is not simulated in detail, but is represented by bandwidth, delay, and error probability.

Table A-1: Simulation parameters

	Hop 1 (BS↔RN)	**Hop 2 (RN↔UT)**
Bandwidth	5 Mbps	5 Mbps
Delay	1 ms	1 ms
BER	10^{-6}	10^{-6}, 10^{-5} , $0.5{\cdot}10^{-5}$, $0.5{\cdot}10^{-4}$, 10^{-4}
TCP MSS	1460 bytes	1460 bytes
L2 data field	1500, 750, 500, 300, 100 bytes	1500, 750, 500, 300, 100 bytes
L2 control information	8 bytes	8 bytes
Total L2 frame size	1508, 758, 508, 308, 108 bytes	1508, 758, 508, 308, 108 bytes

The parameters used for the simulations are shown in Table A-1 For these simulations, the link between the BS and the RN is assumed to provide more stable characteristics than the link between the UT and the RN, since the BS and the RN are stationary. The bit error rate (BER) is set to 10^{-6} for the BS-RN link. Both links have a bandwidth of 5Mbps per user and a delay of 1ms. The BER of the RN-UT link is varied. The maximum segment size (MSS) is set to 1460 bytes. With TCP/IP control information, an IP packet is 1500. The frame size is varied over the links. The same frame size over both links represent the co-optimized ARQ approach.

A.2.2.3 Simulation Results

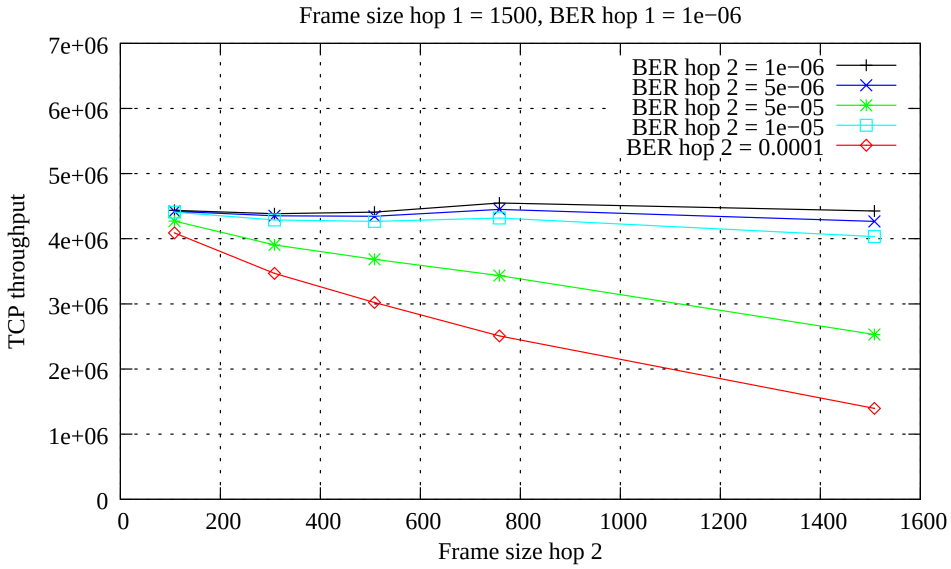


Figure A-9: TCP throughput vs. Frame size hop 2

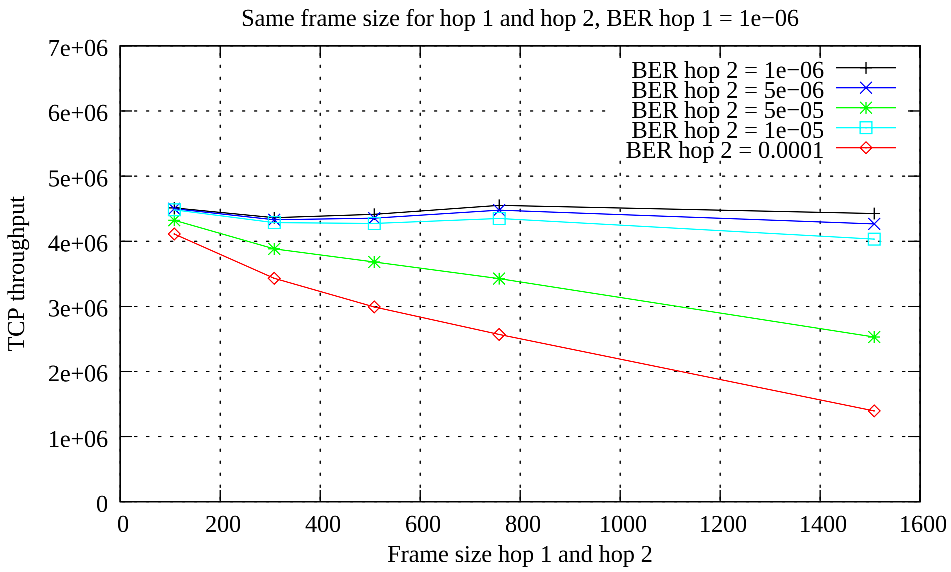


Figure A-10: TCP throughput vs. Frame size, same for hop 1 and hop 2

Figure A-9 shows frame sizes used over the UT-RN link on the x-axis and TCP throughput on the y-axis. Over the first hop between the BS and the RN, the BER is set to 10^{-6} and the frame size to 1500 bytes. The BER values used between the UT and the RN are varied, as indicated in the figure. A BER value of 10^{-6} over both hops results in a TCP throughput of around 4.5 Mbps. For low BER values there are only minor differences in TCP throughput for the tested frame sizes. The TCP throughput achieved with a frame size of 100 bytes over the UT-RN link is as high as with a frame size of 1500 bytes.

TCP throughput decreases when the BER between the UT and the RN becomes higher. As expected, the reduction in TCP throughput depends on the frame size; the larger the frame size, the lower the TCP throughput. For large frames, more data must be retransmitted on the link layer in case of bit errors, which in turn leads to reduced TCP throughput. The highest TCP throughput (for the tested frame sizes) is achieved when the frame size over the UT-RN link is 100 bytes.

In Figure A-10, the same frame size if used over both hops. Also this figure illustrates that TCP throughput remains higher for a larger range of BER values if a small frame size is used. The results indicate that performance is higher if small frames are used over the wireless links. The highest TCP throughput is achieved when the frame size is 100 bytes over both hops.

A.3 MAC Details

A.3.1 The FDD MAC super frame

Figure A-11 shows the structure of the MAC Superframe as defined for the FDD mode. The values depicted are the parameters as chosen for the reference simulation and should not be seen as final decisions.

In the FDD mode, flows to/from half-duplex terminals are assigned to one of four groups: *Group 1* transmits in the downlink the first half of the frame and in the uplink during the latter half. *Group 2* transmits/receives in the opposite way. *Group 3* contains half-duplex terminals that have adaptable and flexible uplink and downlink transmission periods. Full-duplex terminals belong to *Group 4*.

The different UT groups are shown in the first MAC frame of the super frame. Group 3 UTs are devided into 3.1 and 3.2 terminals and can change between them adaptively. Of course the resource allocation mechanisms have to be designed in a way that the terminals are able to receive the resource allocation table early enough. In particular for the *Group 2*, *Group 3.1* and *Group 4* UTs the resource allocation table has to be sent one frame ahead.

The duration of the FDD MAC super frame is similar to the TDD case the same is true for the duration of a single MAC frame and the number of MAC frames in the SF in the current parameterisation[5].

[5] Please not the exect definition of the super frame is a still ongoing activitiy, therefore the current values should only be seen as reference.

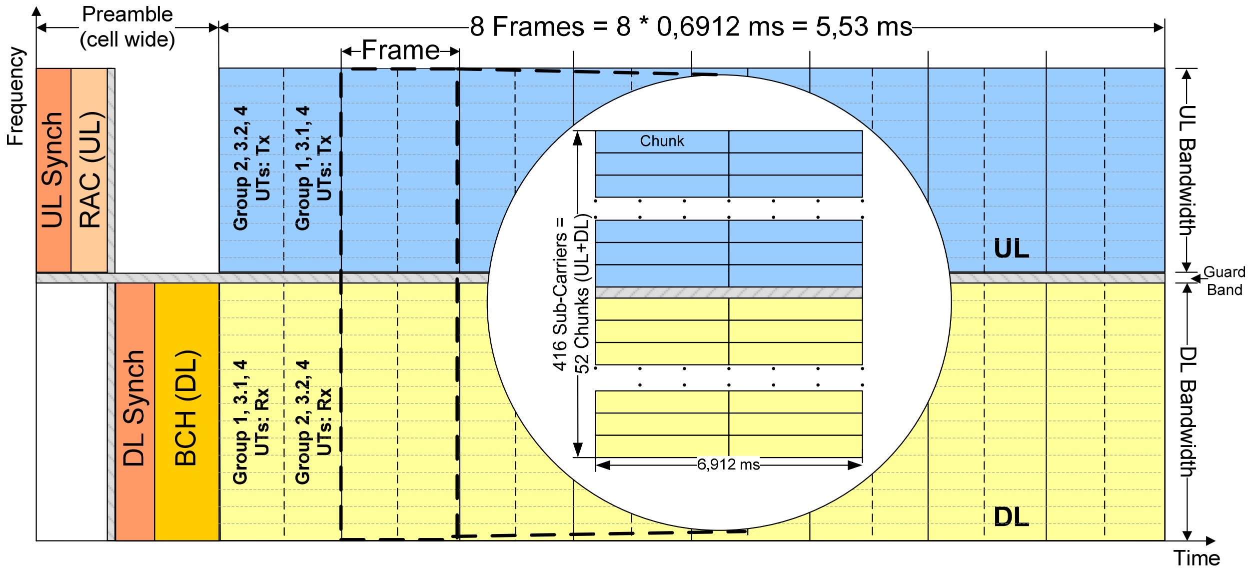


Figure A-11: WINNER MAC Super Frame structure for the FDD case

A.3.2 Overview of main control functions

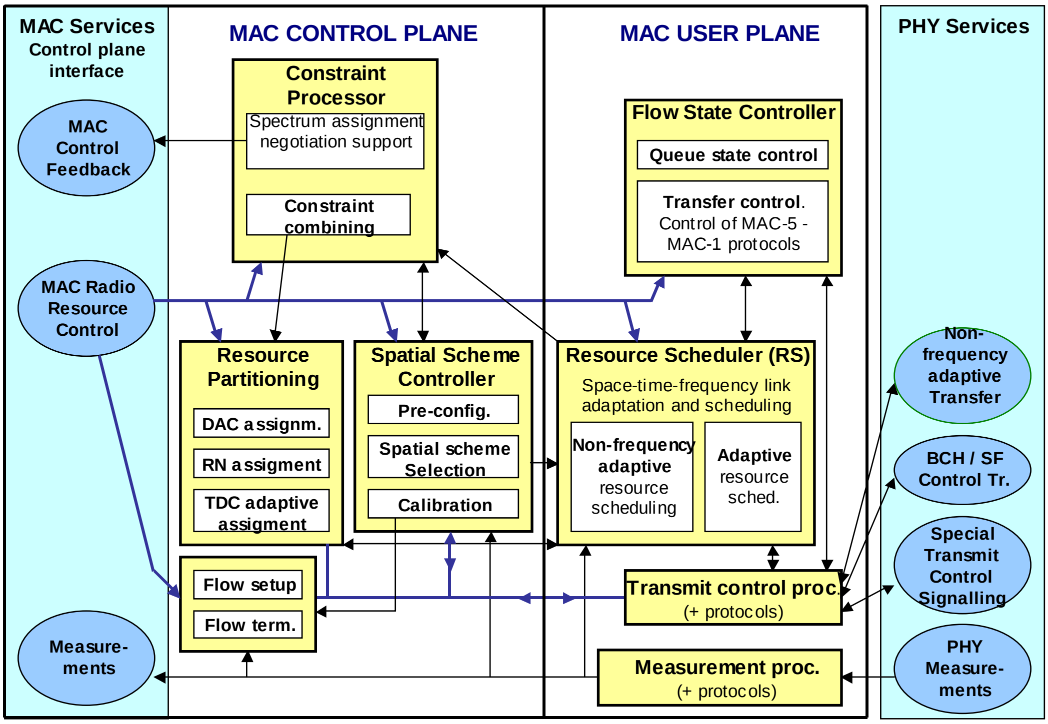


Figure A-12: FDD and TDD cellular MAC control functions: Services and main function blocks

The MAC layer is subdivided into a control plane and a user plane. The PHY layer is not subdivided in that way, as all essential control functions for the physical layer reside in the MAC. The main MAC control function blocks are illustrated in Figure A-12. Control functions that directly control packet transmission on a *slot* time-scale reside in the user plane. Slower functions reside in the control plane.

- ***Resource partitioning***. Partitions the super-frame into sets used for adaptive, non-frequency adaptive and DAC transmission, as well as into chunks reserved for use by RNs, BS-to-RN relay links and as guards for interference avoidance with respect to other cells and operators.
- ***Spatial scheme control***. The appropriate spatial transmit scheme is determined for each flow, and it is held fixed within a super-frame. It is influenced by many parameters, including PLM, deployment, transport channel type, cell load, traffic type, BS antenna configuration, terminal

capabilities, propagation channel, and interference conditions. Further support functions related to spatial processing, like calibration, are invoked if required.

- ***Flow setup and termination*** performs flow context establishment and release over one hop, supervised by the RRM flow establishment and release functionalities in the RLC system layer.
- ***Constraint processor***. Combines constraints on the use of chunks and chunk layers. These arise from interference between user terminals, interference avoidance scheduling with neighbouring cells and spectrum sharing between operators. The output is in the form of chunk masks that define restricted use of a super-frame's chunks. The constraint processor also processes measurements that support the RRM spectrum assignment/negotiation at the RLC system layer.
- ***Flow state controller.*** Controls the segmentation and FEC coding/decoding of packets and monitors the states of RS queues. It also controls the active/semi-active/passive state of flows.
- ***Resource scheduler (RS).*** Includes adaptive and non-frequency adaptive scheduling algorithms and control of spatial link adaptation. Power control in both uplinks and downlinks is performed under the control of the resource scheduler and is integrated into the optimisation of the transmission parameters. The MAC RS cooperates with the RLS layer flow scheduler, Sect.A.3.3.1.

A.3.3 User Plane services and packet processing

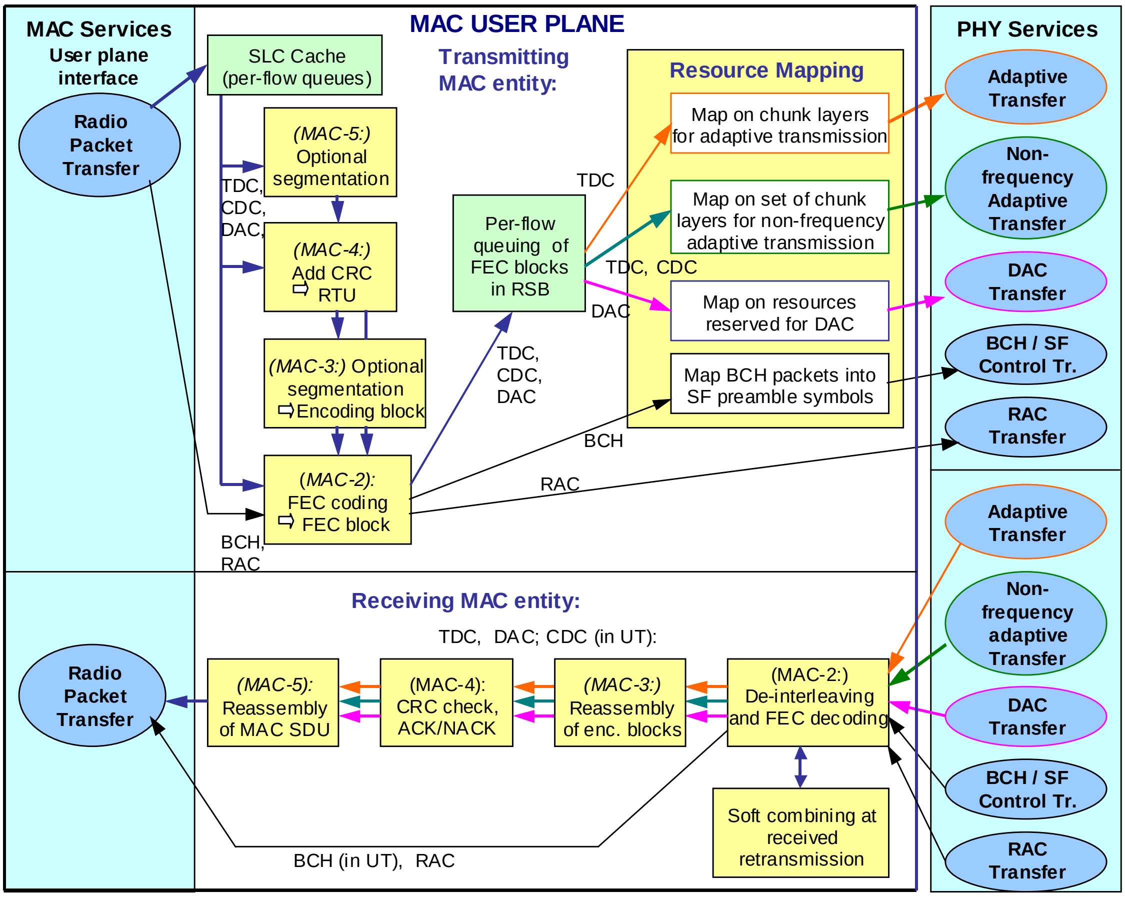


Figure A-13: FDD and TDD cellular MAC: User-plane services and packet processing

The data flows in the user plane, shown in Figure A-13, are processed by three groups of functions:

- ***Transmission: Segmentation, encoding and buffering.*** The flow state controller supervises this sequence. Protocol sub-layers MAC-1-MAC-5 control the transmission and are parameterised to describe the different retransmission options. A MAC SDU is drained from the SLC Cache. If it belongs to the TDC or DAC transport channels, it may be retransmitted. Retransmission can be introduced as an option also for CDC point-to-multipoint flows. The TDC, CDC or DAC packet is optionally segmented. A CRC sequence is added, resulting in a Retransmission unit (RTU). The RTU may optionally be segmented into *encoding blocks* that are encoded separately. Coding and interleaving results in *FEC blocks*, which are buffered in the RS buffer on the transmitter side, in one or several queues per flow. There they remain, until acknowledged or dropped. BCH and RAC-packets are transmitted in the super-frame preamble, without retransmission.

- ***Resource mapping***. At transmission, bits from TDC flows are mapped either on chunks reserved for adaptive transmission, or on the sets of chunks intended for non-frequency adaptive transmission. CDC (multicast) flows should use non-frequency adaptive transmission. DAC packets are mapped on the contention-based physical channel. For TDC and CDC, puncturing of the buffered FEC block may be performed and only a part of a FEC block may be transmitted in a scheduling round that comprises a time-slot. For CDC, TDC and DAC, the resource scheduler controls the resource mapping. For the BCH, the mapping into the preamble broadcast OFDM symbols is under the control of the Resource partitioning function in the control plane, which has over-all control of the multiplexing and modulation of these symbols.
- ***Reception: Decoding and reassembly***. The flow state controller supervises the reception. De-interleaving and FEC decoding is first performed for received FEC blocks belonging to TDC, CDC or DAC packets, followed by the (optional) re-assembly of the RTU. Then, the retransmission unit is optionally checked for transmission errors and a retransmission may be requested. The MAC SDU is finally re-assembled. BCH and RAC packets are received separately in the super-frame preamble and then decoded.

An outline of the transmission and reception algorithm can be found in D2.10 Appendix C1.5 “Transmission and reception”.

A.3.3.1 MAC Resource Scheduling and Multiple Access Schemes

Within the MAC layer, resource scheduling of physical channel resources to flows in a cell is performed per frame. The scheduled flows are allocated to time-frequency-spatial resources available within the super-frame in one of two ways: *Adaptive* RS utilizes the frequency selective fading. This requires processing of CSI/CQI feedback from the PHY layer. *Non-frequency adaptive* RS uses a diversity-based transmission within the frame, see Appendix C.1.6 of [WIND210].

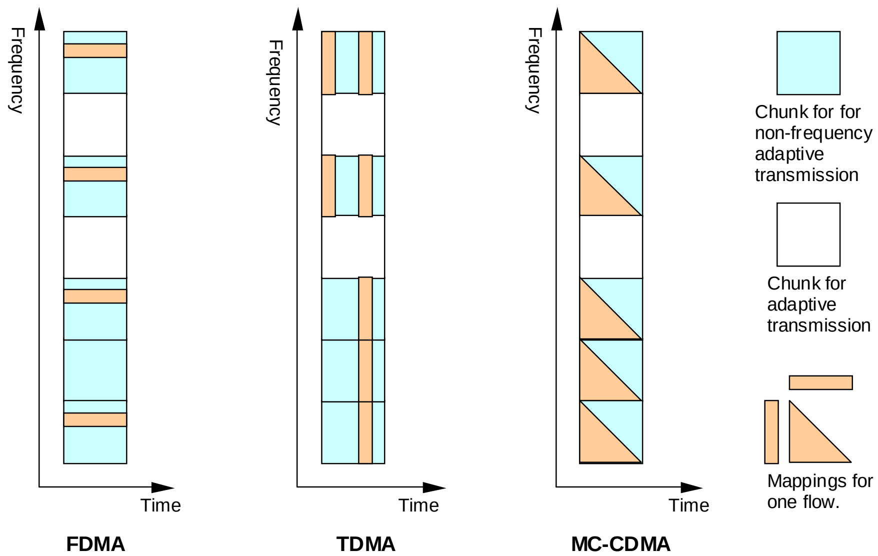


Figure A-14: Multiple access schemes and chunk resource scheduling principles used for adaptive and for non-frequency adaptive transmission.

Flows scheduled for adaptive and non-frequency adaptive transmissions are mapped onto the separate sets of chunks earmarked for these two purposes by the SF resource partitioning (Figure A-14). The mappings use different MA schemes:

- **Adaptive transmission** (DL/UL) uses **chunk-based TDMA/OFDMA**: Flows are mapped onto individual chunk layers with individual link adaptation per chunk layer. The mapping is exclusive within the cells, i.e. each chunk layer carries data from only one flow.

Chunks are designed to work well for adaptive transmission. In a non-frequency adaptive transmission that is based on channel averaging, there would be a problem in attaining sufficient frequency diversity when transmitting small packets when the packets are mapped directly onto chunks. A small RTU or control packet may fill only one chunk. Two schemes that increase the diversity significantly are selected:

- In **non-frequency adaptive downlinks**, **MC-CDMA** is used within the assigned set of chunks. In the *downlink*, spreading may thus be used to code-multiplex the flows onto sets of chunks assigned for non-frequency adaptive transmission. When code multiplexing is used, spreading is performed only within chunks, to minimize non-orthogonality of the received signals. Orthogonal signalling (TDMA per OFDM symbol or FDMA/IFDMA per subcarrier, without code-multiplexing) are special cases of the scheme. They may be used when appropriate.

- In **non-frequency adaptive uplinks, TDMA/OFDMA is used on an OFDM symbol basis.** For non-frequency adaptive transmission in the *uplink*, code multiplexing is not used, to avoid the need for multi-user detection. Instead TDMA is used on an OFDM symbol basis. Several uplink flows with small FEC blocks may share one OFDM symbol (OFDMA) to improve the rate matching (Figure A-14, middle part.). Either OFDM or frequency-domain generated serial modulation can be used in the uplinks

Compared to chunk-based OFDMA/TDMA, these schemes provide increased frequency diversity and resources of shorter duration. The shorter time duration (mapping on individual OFDM symbols/GMC slots) provides increased opportunities for terminals to go into power-saving micro-sleep intervals

There is a possibility to allocate smaller resources in non-adaptive transmission, which could be useful for small (control) messages, but the protocol overhead might be too large for this possibility to be useful. The current assumption is that the basic resource unit is of the same size in non-frequency adaptive transmission as in adaptive transmission, and could be defined by a virtual chunk of the same size as the physical chunk. However, this is for further study.

Further details on the proposed algorithms for the Resource scheduling and spatial user partitioning can be found in D2.10 Appendix C1.6 “Resource scheduling” and D2.10 C.1.7 “SDMA and spatial user partitioning”.

Annex B Fixed Heterogeneous Relay Deployment

B.1 Introduction

This section is devoted at first to describe the motivation for the use of fixed heterogeneous relay nodes (HERN) in a Relay Enhanced Cell (REC), recognizing those scenarios where the inclusion of this type of RN may be interesting for a fast network deployment. In general terms, areas with different propagation conditions are the most appropriate scenarios for the use of HERNs operating in modes tailored for each of the propagation conditions existing in those areas. Some important issues concerning the implementation of HERNs are also analyzed, in particular the super-frame structure, the modes conversion functionality and the resource partitioning problem in heterogeneous relay nodes, that is, relay nodes using a physical layer mode to serve at its associated UTs (RN-UTs links) different to the used for its connection with the correspondent BS (BS-RN link).

First of all it is convenient to bear in mind the definition of relay node. A relay node (RN) is a physical network element serving other relay nodes or user terminal in a given geographical area via its radio access capabilities. It is wirelessly connected to a base station, another relay node and/or a user terminal and forwards data packets between these network elements. Depending on whether its connections (BS-to-RN and RN-to-RN or RN-to-UT) are established with the same radio access technology in the same pool of transmission resources (e.g. RF channels) or not, one may distinguish between homogeneous relay nodes and heterogeneous relay nodes.

So, a heterogeneous relay node is a relay node that uses different radio access technologies (or different modes of the same RAT) using common or different sets of transmission resources (e.g. RF channels) for its links (BS-RN, RN-RN, RN-UT). The radio access technologies that a heterogeneous relay incorporates can be different modes of the same RAT (i.e. in the WINNER context), one WINNER RAT-mode and another (possibly legacy) RAT, or two (legacy) RATs, where the latter case is not in the WINNER scope of research. In short, the HERN is a RN, which simultaneously uses different physical layer modes for the RN-BS link and the RN-UT link respectively. In any case the heterogeneous relay nodes so far contemplated in the project are exclusively those operating in the different physical layer modes, which are being developed in the WINNER project. In this sense and from a general perspective, once defined the input physical layer mode to be used in a HERN (BS-RN link), the two main issues to contemplate in its development would be:

- To define the additional requirements and management of the radio resources for the output physical layer mode to be used by the HERN (RN-UT link).
- To define the functionality of the HERN. This involves more than a simple change in the modulation scheme but less than an IP router.

It is important to note that the heterogeneous relaying topic is really integrated within the WINNER system concept and therefore the requirements, service specification and functionality defined and developed for the WINNER RAN concept are also applicable to the heterogeneous relay case. So we focus on those particularities affecting exclusively to the heterogeneous relaying concept.

B.1.1 Assumptions

In the current analysis regarding the implementation of HERNs in some scenarios, the following general assumptions are taken into account:

- In the analysis it is only contemplated communications using single-hop or two-hops. The first one for the direct transmission between the BS and its associated UTs, and the two-hops for the communications between RNs and its associated UTs.
- The heterogeneous relay node considered in this study is fixed and it is based on the use of a particular mode specifically thought for the BS-RN link (feeder link with TDD duplex scheme).
- High directional antennas are used for the BS-HERNs links.
- Inside the REC each HERN is physically connected to the BS (wireless connection) so that each HERN has only one path towards the BS.
- The HERN is using the FDD physical layer mode for serving at its final users, as well as the BS uses the TDD mode for the communications with its UTs. Then the two modes involved in the operation of the HERN are using two different frequency bands so that the radio resources for BS-HERN and HERN-UT links are totally orthogonal.

- It is assumed perfect synchronization between the network elements and involved modes so that the starting and duration of frames and super-frames are equal.

Taking the HERN's definition into account, this is always a *decode & forward* relay type since the communication between the two elements that has to hold, involve the use of some kind of mapping table for the protocols conversion as well as some interworking mechanisms (congestion control) between the two modes involved in the communications between the BS and UT through the HERN. So before forwarding the data, it is needed to decode the data of the incoming mode, to do the conversion and to encode the data in the other mode. It is important to note that at this moment the homogeneous relay node (HORN) contemplated in WINNER system is also a *decode & forward* relay type, and then the operation in both cases, homogeneous and heterogeneous relays, at least from a digital process perspective is practically the same excepting in heterogeneous case once decoded the incoming data, the information bits have to be encoded using a different physical layer mode. On the other hand in our particular case, it is clear the need to have two transceivers in the HERN since this is working in two different physical layer modes, which operate on different radio resources involving different frequency bands (TDD and FDD).

B.1.2 Requirements

First of all we include here some important and general requirements for any kind of relay node in the context of WINNER system concept:

- Efficient and flexible implementation by means of a flexible protocol architecture based on the multi-mode protocol architecture reference model (see [WIND31] and [WIND32]).
- To support the requested QoS for future mobile radio services and traffic performance.
- Not add additional complexity to the UT so that the multi-hop is transparent for higher layers, above to radio protocols.

On the other hand for the particular case of heterogeneous relay nodes we could add to this list other important requirements:

- Resource allocation information for the two involved modes, assigning two different sets of radio resources to one specific data flow.
- Mobility support for the second mode (HERN-UT mode 2 FDD) and intra-cell handover with simple mechanism for context transfer between HERNs of the same cell.
- Two error detection and recovery mechanisms.
- Retransmission protocols: hop-by-hop for BS-HERN and HERN-UT communications, and end-to-end ARQ mechanism to implement in BS and in UT (optional).
- Secure and reliable packet transfer end to end through two different modes.
- Segmentation and reassembly mechanisms with the same granularity for both modes.
- Resource scheduling for the two involved modes (for the case of BS only in mode 1, for the case of the HERN in both modes).
- Two different super-frame structures involved, one for TDD and another for FDD, properly synchronized.
- To guarantee a certain QoS level in both hops (different modes).
- Link adaptation control (RAPs link not required since the RN is stationary).

B.2 Motivation for the use of HERNs

Firstly and from a general point of view the goals of using HERNs in an infrastructure based deployment concept coincide with those pursued by any kind of relay nodes:

- To optimize the capacity in a given cell area with the possibility to achieve a fair capacity balance over the entire cell area.
- To enlarge the cell area coverage/range by means of the use of advanced antenna technologies such as smart antennas or high directional antennas.
- To improve the radio coverage in areas with heavy shadowing conditions.

Besides the heterogeneous case allow us to provide in a certain far area from a given base station, of one physical layer mode, different to the used by this base station. This possibility has the potential benefit to include in a flexible way inside the coverage area of one cell operating in a certain mode, the use of any other mode without the necessity to deploy base stations of this last mode. On the other hand it is clear that a HERN would be more complex and expensive than a homogeneous relay node, in particular when the involved modes in the heterogeneous case are using different frequency bands and duplex schemes, since the implementation of two transceivers is unavoidable, although from a digital signal process perspective the complexity is similar to the homogeneous case at least if this is of decode and forward type. Certainly the complexity and cost aspects in contrast to homogeneous option, could limit the deployment of HERNs to some particular scenarios.

However in some scenarios where different throughput requirements and propagation conditions are envisioned, the use of HERNs operating in modes tailored for each of the requirements and conditions existing in a certain area could be interesting from an economical and spectral efficiency point of view. So it is important to note that the motivation for the use of heterogeneous relaying is mainly based on the use of different physical layer modes adaptive to different radio environments (in fact the modes in WINNER somehow have been defined for adapting the system to different scenarios and radio environments), but not in the fact to exploit any other different domain for multi-hop purposes, of those which applicable for the case of fixed homogeneous relays.

In any case and according to previous deliverable D3.4 [WIND34], the deployment of HERNs will be more appropriate and useful for those scenarios where the requirements of the different segments which composed a multi-hop communication, are so different as for example from mobility or propagation characteristics points of view. In fact these differences can justify in some situations the use of a network deployment based on heterogeneous relaying, implementing a certain mode for the BS-RN link as well as for the direct communications between the BS and the final users (e.g. short range), and another different mode for the communication of user terminal through the RN (e.g. wide area). Summarizing, mixed environments with different mobility and propagation characteristics like the case of outdoor-to-indoor transitions and vice versa, are in principle clear scenarios for the use of heterogeneous relaying.

Moreover it is important to remark one attractive advantage of the HERN solution concerning the provision in a certain mixed coverage area of different access modes (adapted and convenient for different environmental conditions), in a decentralized manner by means of a proper distribution of heterogeneous relay nodes around the BS. This means that the BS would be providing service to final users directly via a given mode as for example short range (more suitable for the characteristics of the region around the BS), or through the HERN using a different mode as for example wide area (more suitable for the characteristics of the BS's boundary). Moreover, only one connection to the fixed network would be necessary, since provision of the distributed mode in geographical spots far from BS location, is performed by the HERNs, which are wirelessly fed from the BS. This idea is outlined in Figure B-1 where for covering a given geographical area the following network elements are used:

- Four tri-sectored base stations operating in mode 1 TDD (short range).
- Six homogeneous relay nodes operating in mode 1 TDD, tailored for a certain propagation conditions (i.e. low mobility support, short range, and high data rate requirements).
- Six heterogeneous relay nodes bridging mode 1 TDD with mode 2 FDD, tailored for another different propagation conditions (i.e. high mobility support, wide area, and low data rate requirements). In this way the deployment of base stations implementing the mode 2 FDD is avoided, saving the wireline connections with the correspond controller (RANG and ACS).

Besides thanks to the mode 2 (wide area) along with the use of high directive antennas for the links between the BS and the HERNs, it is possible to increase the coverage area of the mixed RECs (composed by homogeneous and heterogeneous relay nodes). Further in order to achieve a fair distribution of radio resources over the complete deployment concept, the resource partitioning of both modes (TDD and FDD) is controlled by a central node according to the hierarchy above the BS (i.e. ACS).

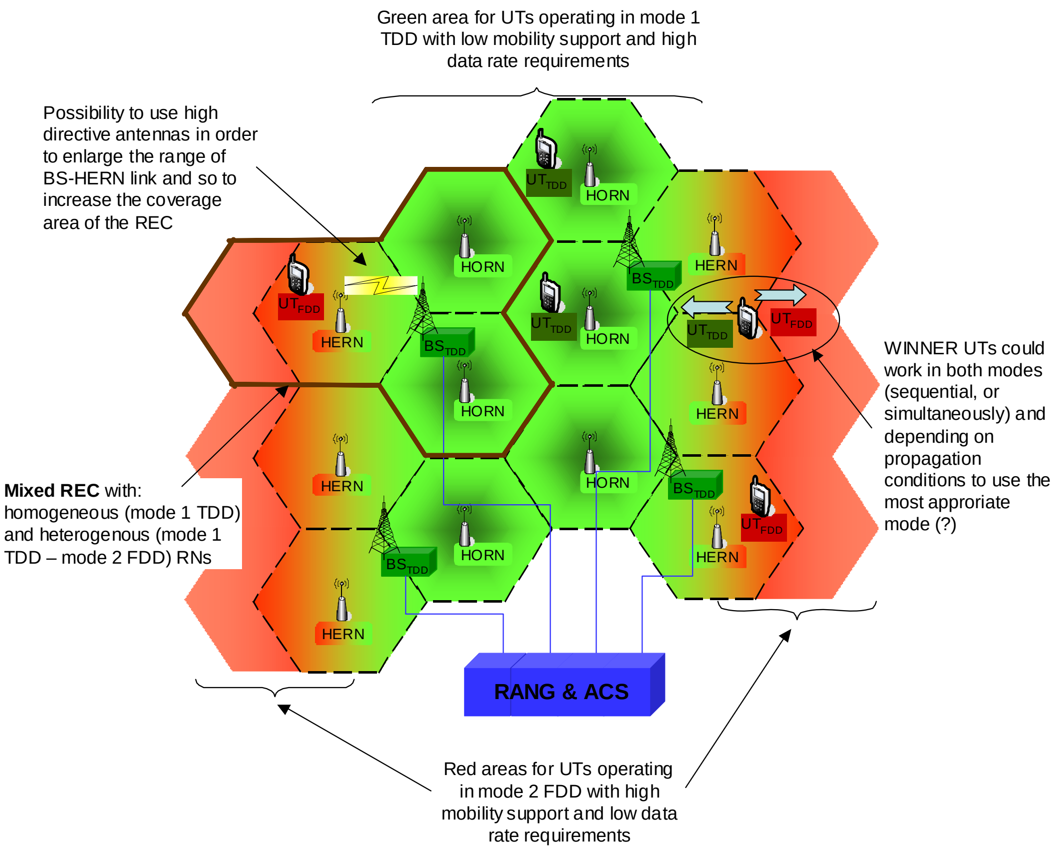


Figure B-1: Motivation for the use of HERNs in areas with different requirements and propagation conditions

Finally in respect of the motivation for the use of HERNs, let's suppose that the geographical area shown in the previous example has two different well-defined zones, and the mode 1 foreseen traffic load ($T_A(t)$) has its rush hour in a certain time interval inside one of these zones, whereas the mode 2 foreseen traffic load ($T_B(t)$) has its rush hour in different time interval inside the other zone. We can mention as real case applications associated to this example, scenarios where there is pedestrian zones, shopping centers or even football stadium with low mobility support needed and high data rate requirements (mode 1 TDD), as long as the surroundings (large avenues and wide streets) have other different requirements such as high mobility support and low data rates (mode 2 FDD). Assuming that the connection of BS (mode 1 TDD) is able to support the accumulated traffic load of both modes ($T_{total}(t) > T_A(t) + T_B(t)$), the deployment of mixed RECs (HORNs in mode 1 TDD, and HERNs in mode 2 FDD) will be less expensive and probably more efficient than the deployment of BSs operating in mode 2 FDD, which need wireline connection with the backhaul network. Moreover with the HERNs deployment in this particular case we can get a best load balancing as well as a fairer distribution of the capacity between the sub-cells of the mixed REC. Anyway so far the costs of these two alternatives (BSs mode 1 with HORNs and HERNs, as opposed to BSs mode 1 and mode 2 both with only homogeneous RNs) are unknown, and then would be necessary to perform in WINNER phase II some detailed deployment cost analysis along with simple simulations comparing these two options in order to determine the best solution from economical and spectral efficiency points of view.

B.3 Implementation issues

First of all it is important to note that the Multi-mode protocol architecture developed in WP3, will make easy the transition between WINNER modes and the coexistence of these modes such as in the case of heterogeneous relay nodes thanks to the cross-stack management supported by the modes convergence manager of a layer or stack as Figure B-2 shows. For details of the multi-mode protocol reference model, which provides the parameterization and the configuration of a particular protocol layer (N) by means of the (N)-Layer Modes Convergence Manager ((N)-MCM), and the cross-stack management of different modes with the help of the Stack Mode Convergence Manager (Stack-MCM), it is recommended to see previous WP3 deliverables ([WIND32,WIND34]) as well as chapter 5 of the current document.

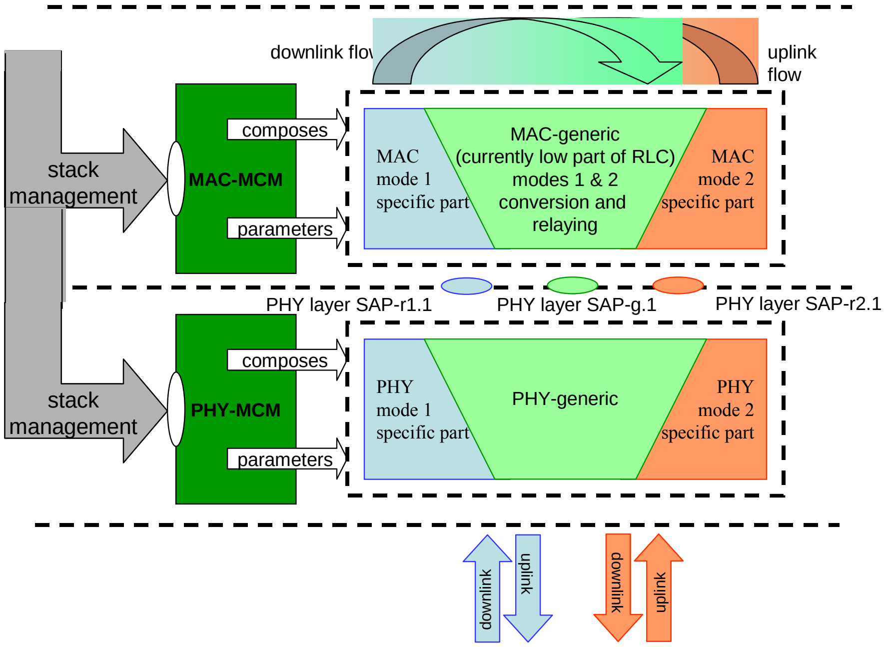


Figure B-2: Example of implementation of a multi-mode MAC layer relay

Hereinafter some important issues concerning the implementation of heterogeneous relay nodes operating in different WINNER physical layer modes are analyzed such as the protocol architecture, the super-frame structure of the involved modes, the resource partitioning problem, and the particular complexity of this type of relay nodes.

B.3.1 Protocol architecture in a HERN

It is important to remark that from a user plane perspective the HERN's operation is identical to the homogeneous case since at this moment the type of relay nodes contemplated in WINNER is *decode & forward*. Therefore the decoding/encoding processes are common to both types of relay nodes, homogeneous and heterogeneous, with the only difference than in the second case once decoded the L2 packet in one physical layer mode, the information bits (payload part) have to be encoded in another physical layer mode. However concerning the control plane, although the multi-mode reference protocol will facilitate the coexistence of both modes in the same logical and physical node (HERN), we will have to include the implementation and the execution of some additional functions such as the resource partitioning function for the second mode used by the HERN for giving service at its final UTs working in this second mode. In other words, different to the homogeneous case the RRM entity for a specific data flow will have to assign two different modes. In the same way and regarding power control function within RRM functionality, according to D3.1 ([WIND31]) two different schemes could be envisioned: centralized where the BS is in charge for the power allocations, and the distributed where every node in a given route is in charge for the power allocation on a hop-by-hop basis. However in a HERN node it seems more appropriate to use the distributed option in order to guarantee the QoS requirements and manage the interference properly, since the BS doesn't know the HERN's activity in its coverage area, in particular when the BS is only operating in the mode used for the BS-HERN link. Otherwise like in the homogeneous case, the decision variables for this function would be basically the induced interference and the available transmitter power.

Figure B-3 illustrates the protocol architecture for control plane and user plane of a HERN in the MAC layer using two different modes (mode 1 TDD and mode 2 FDD). As we can see there are not practically differences with the homogeneous case, except for the control plane where the specific part of RRC protocol for mode 2 (the mode used by the HERN for serving at its final UTs) has to be included, in order to receive and interpret the control messages concerning the resource partitioning (control entity part of RRC layer) of mode 2 coming from the central node. In fact one of the initial ideas for partitioning certain layers in generic and specific parts was to facilitate, by means of the generic parts, the convergence of modes to be developed in WINNER system for covering different situations and scenarios. In this sense the heterogeneous fixed relays try to be useful in areas where the requirements related to each of the segments of a multi-hop communication are so different (e.g. mobility support), that may be justified the use of two different physical layer modes tailored each one for the particular requirements of each segment. Also it is important to note that the main functionalities of MAC layer, in both physical layer modes (TDD and FDD) are identical, making easy this way the modes translation in the implementation of the HERN.

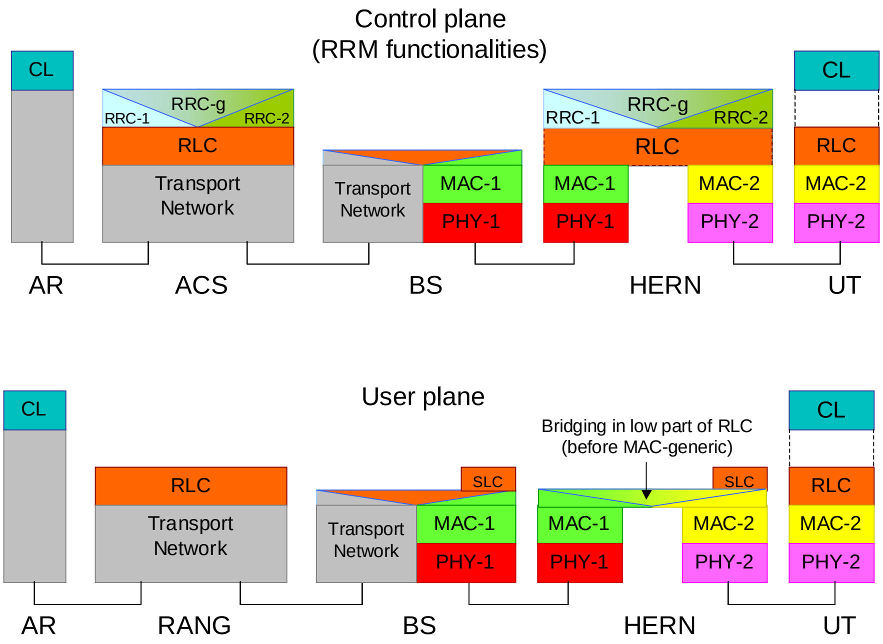


Figure B-3: Control plane and user plane for the protocol architecture in a HERN operating in two different modes

Concerning the location of the Service Level Controller (SLC) in the user plane, note that this is a common aspect to solve in any kind of RN, and at this moment is an open issue to solve (discussion where to locate it, in the MAC layer or in the RLC layer). In both cases, homogeneous and heterogeneous relaying, for a given data flow from BS (mode 1) to UT (mode 1 also for the homogeneous case and mode 2 for the heterogeneous case) through the RN, there are two sub-flows with the possibility to implement different types of ARQ mechanisms, end-to-end and hop-ARQ. The SLC function is in charge of flow scheduling, flow monitoring and flow management of both sub-flows. In the case of a HERN one possible solution would be the shown in Figure B-3: a SLC function in the BS for controlling the sub-flow of mode 1, and another SLC function in the HERN for controlling the sub-flow of mode 2. In the same way the HERN should contemplate two hop-ARQ buffers, one for mode 1 and another one for mode 2: to the HERN, from the BS, arrive coded segments of MAC1-SDU and then these segments should be decoded and encoded in the other mode for forwarding to the UT coded segments of MAC2-SDU. In the particular case of TDD for mode 1 and FDD for mode 2, where different chunk sizes are involved, some algorithm will have to be implemented in order to translate the information bits of the chunks of one mode to the chunks of the another mode, minimizing the wasted bits in the process.

The SLC in the user plane of RLC entity located in RANG logical node has the total responsibility for the control of inter-flow fairness in a slow time scale. The SLC will allocate radio resources of both modes (mode 1 resources pool & mode 2 resources pool) for the data flow, and will forward packets to SLC cache located in BS, which updates priorities. At a time the BS in terms of predicted rate and QoS requirements forwards packets to destination. The RN, which is controlling its own coverage area (sub-cell) forwards packets to UTs, assigning and distributing in a fast time scale radio resources for the different active data flows by means of RS, taking the allocated resources by the RANG into account.

In order to avoid or mitigate inter-cell interference, the resource partitioning (RP) in the control plane of RRC entity will be located in ACS logical node. On the other hand the RP function in the control plane of MAC entity resident in BS, will broadcast this information (control messages coming from ACS) to its UTs in the control preamble of a SF or in the last part of a SF to its RNs.

B.3.2 Frame and super-frame structures in a HERN

In the case of a Relay Enhanced Cell (REC) incorporating heterogeneous relay nodes, which are serving at its final users by means of a physical layer mode different to the used by the BS, there will be two broadcast channels, one for the BS (e.g. BCH in mode TDD), and another one for the HERNs (e.g. BCH in mode FDD). However for the sake of the completeness it would be convenient that both broadcast channels are broadcasting a unique identity cell, different to the rest of the cells, in order to keep fixed the

REC concept, regardless the REC has only homogeneous RNs or both heterogeneous and homogeneous. The question to solve now is what system information messages broadcast in each BCH. The simplest solution would be to use only one set of system information messages common to the different modes to be developed in WINNER system, and depending on the mode used in each network element to process only the messages devoted to this mode in order to extract the information and parameters required for operating in this particular mode.

According to last works carried out in WP2 (see [WIND210]), the time unit used for uplink or downlink transmission only is denoted a timeslot**,** or slot. Slots are used both in TDD and in half-duplex FDD transmission. In the same way a frame will denote one uplink timeslot plus one downlink timeslot in TDD-based transmission. In transmission based on half-duplex FDD, it denotes two timeslots, used for uplink and downlink transmission by a half-duplex terminal. So, taking this definition into account as well as the last proposals about the OFDM parameters used in different physical layer modes and summarized in Section 3.2, Figure B-4 illustrates an example of possible MAC frame structure for modes FDD and TDD, assuming 1:1 asymmetry for this last mode, and from a perspective of physical layer characteristics.

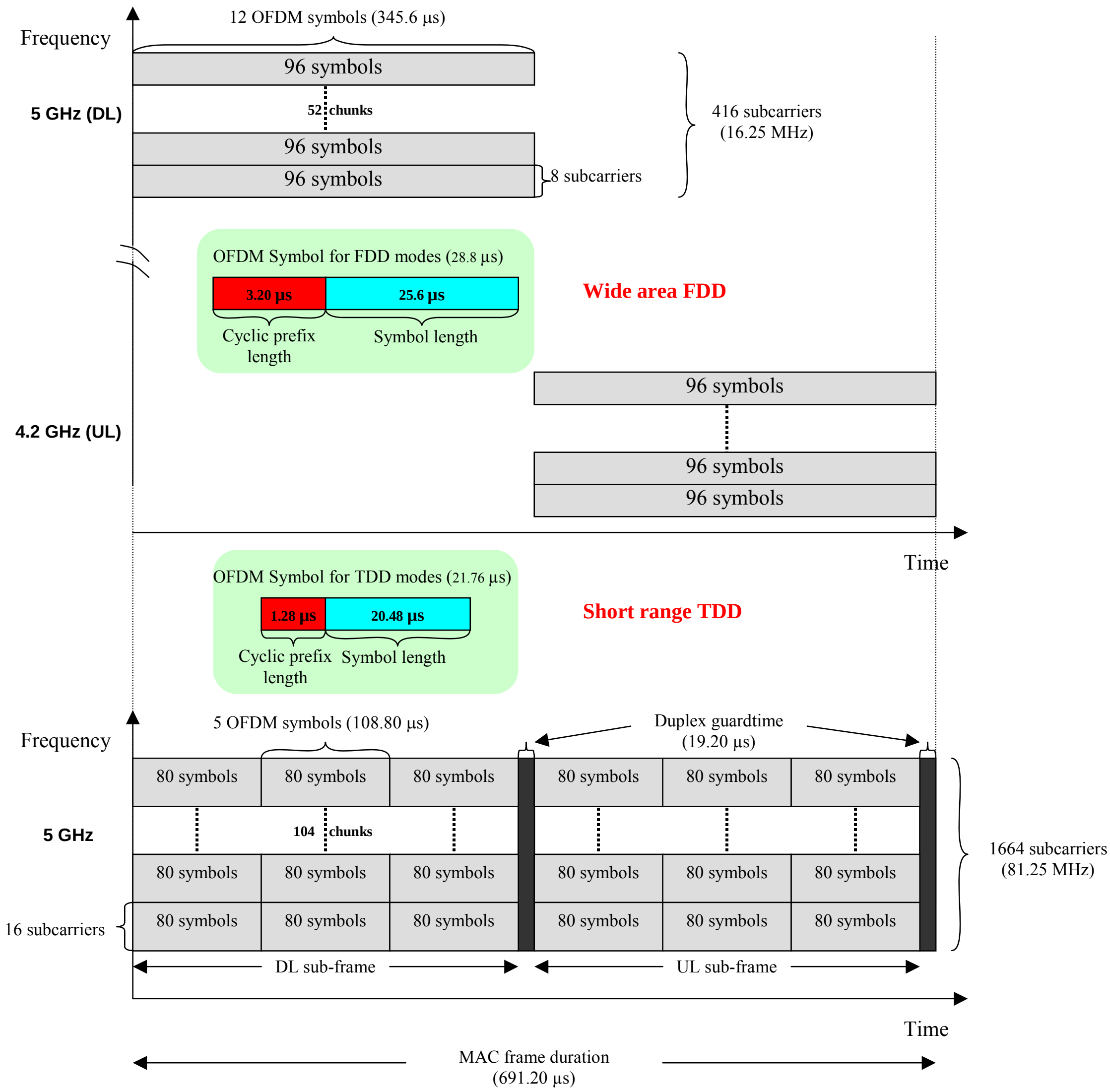


Figure B-4: Proposed MAC frame structure, from a perspective of physical layer characteristics, for physical layer modes in wide area (FDD) and in short-range (TDD), assuming 1:1 asymmetry for this last mode

For the case of physical layer modes based on TDD duplex scheme, Figure 6-26 illustrates the aspect of a MAC super-frame composed by a first part devoted to control signalling on super-frame basis, and 8 frames. So far the duration of these 8 frames is 5.5296 ms, but for the control part we will have to analyze the number of OFDM symbols required for each type of signaling, that is uplink synchronization, random access (RAC) contention based, downlink synchronization and broadcast channel (BCH). Figure B-5 shows the structures for TDD and FDD physical layer modes describing in detail the control preamble parts of both modes. The number of OFDM symbols for DL and UL synch, at least for the TDD mode, is currently fixed to three in each synch part. The other two parts of control preamble (RAC and BCH) are not steady and so in principle they could be changing in each super-frame. Concerning the number of OFDM symbols to be used for DL and UL synch in the mode FDD, although it seems coherent to utilize the same number of symbols than in TDD, this should be analyzed in order to define the most proper number of symbols in the particular case of FDD.

Anyway one important aspect to consider for keeping equal in time the super-frames (control preamble plus a certain integer number of frames) is that the control preamble in both modes should have the same duration. Besides it should be noted that the OFDM symbol duration for TDD is 21.76 µs and moreover we have to add an interval of 19.2 µs as duplex guard time, whereas for FDD the OFDM symbol is 28.8 µs. So, we have to contemplate to include some wasted time in one of the modes in order to fit exactly the control preamble times in both modes ($N_{ControlSymbolTDD} * 21.76 + 19.2 = M_{ControlSymbolFDD\_DL} * 28.8 + X = M_{ControlSymbolFDD\_UL} * 28.8 + Y$, where X and Y are adjustments for equaling both control preambles).

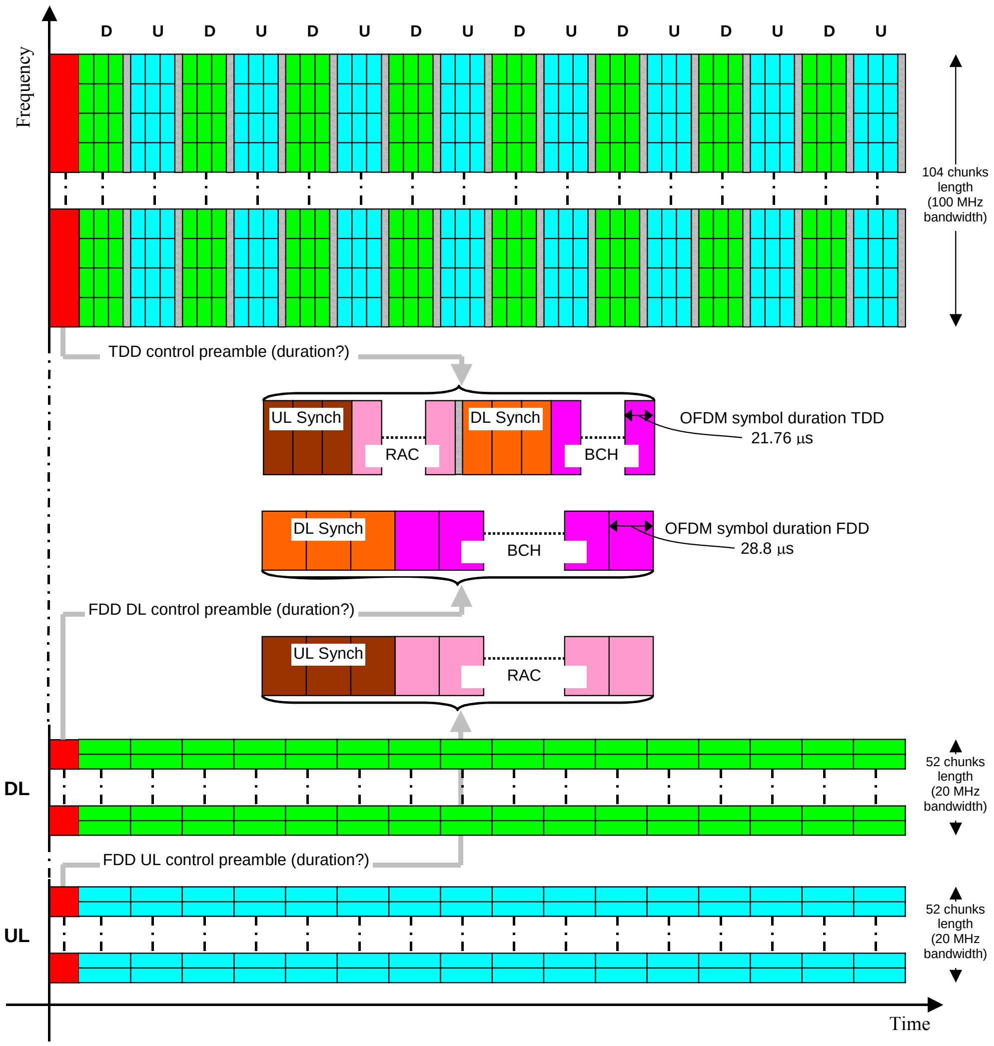


Figure B-5: Detail of control part for TDD and FDD super-frames assuming unpaired 100 MHz with 1:1 asymmetry for the first mode, and 2x20 MHz for the second one

The channel BCH is to be used both for general system broadcast messages and for detailed control of the overall partitioning of resources within the current super-frame. In cells including relay nodes, this channel is partitioned into two parts. One, mentioned above, at the beginning of the super-frame, for transmission either from BS to UT or from RN to UT, and another in the latter part of SF (some OFDM symbols or some control chunks) for control messages from BS to RN indicating the radio resources to be used in the next super-frame (resource partitioning).

The extra control chunk late in super-frame *i* concerns super-frame *i+1*. The BCH at the beginning of super-frame *i+1* describes the overall resource allocation in the rest of super-frame *i+1*. It is computed by the RN partly based on info received by the BS in the extra control chunk in super-frame *i*, and partly by the calculations and prioritisations by its own resource scheduler and the associated feedback from its assigned UTs. It is important to remark that in the particular case of HERNs operating in mode TDD for the BS-RN link and in mode FDD for the RN-UTs links, the BCH at the beginning of the TDD super-frame doesn't need to be shared between the BS and the HERN since this node uses the BCH of FDD super-frame (radio resources fully orthogonal) for broadcasting to its UTs the system information messages and detailed control information about complete resource partitioning within the current super-frame.

B.3.3 Resource partitioning in the context of heterogeneous relaying

This section deals with the analysis of resource partitioning function, in charge of the organization of the super-frame (SF), for the particular case of a heterogeneous relay node (HERN), using the physical layer mode TDD for the BS-RN link, from now on referred to as mode 1, and the mode FDD for the RN-UT links, from now on referred to as mode 2. So the relay node in our case is giving service at its UTs with a different mode and frequency band than the used by the BS for servicing at its UTs and its RNs. In this particular case, we have to contemplate the organization of two different kinds of super-frames:

- The TDD SF organized by the BS. The base station will have to partition the available time-frequency resources between the RNs (homogeneous and heterogeneous) and the UTs associated to it. As it was mentioned in [WINXWPMAC], the resource partitioning resident in the BS assigns sets of chunk layers for adaptive, non-frequency adaptive and DAC transmission, as well as some chunk beams are also reserved for BS-to-RN relay links, for the use by RNs, and as guards for interference avoidance with respect to other cells. One possibility for our particular case, assuming very high directional antennas between the BS and the HERNs may be to use the BS-to-RN relay links for the communications between the BS and its heterogeneous relays through the mode 1.
- The FDD SF organized by the resource partitioning function implemented in a central node with the control of the HERNs, probably the Access Control Server (ACS), which is in charge of the BS as well as the RNs associated to this BS. The ACS node will have to do the partitioning of the available time-frequency resources of the FDD mode between the HERNs of the cell using this mode. Otherwise every HERN will be in charge of the resources distribution, previously allocated by the ACS, between all its final UTs, executing their MAC resource partitioning function for the mode 2.

Firstly it is important to note that the radio resources partitioning for the two involved modes shall be flexible between BS and RNs and from super-frame to super-frame so that the total radio resources of mode 1 (BS-RNs links) are dynamically partitioned into parts used by the BS for the direct communications with its UTs, and parts used by RNs (homogeneous and heterogeneous). Note that in the case of heterogeneous relaying, the BS has to allocate only radio resources for the connection with this kind of RNs since the second hop is implemented by means of another mode and then using different radio resources. In the same way the total radio resources of mode 2 (HERNs-UTs) would be dynamically partitioned from a central node (i.e. ACS) into several parts used by the HERNs for the communication with its UTs, taking mutual interference avoidance between mode 2 sub-cells into account.

According to the terminology used in [WIND210], the fundamental unit for partitioning the super-frame is the chunk layer, which is indexed by three integers (l,m,n), where $l=1,\ldots, Q_c$ denotes the spatial layer, $m=1, \ldots, M$ the frequency bin number and $n=1, .. N$ the time index. For example, in a TDD cell (assuming 100 MHz of bandwidth) comprising a BS sector with two antennas/spatial channels, $Q_{c\text{-}mode1} = 2$, $M_{mode1} = 104$ and $N_{mode1} = 8 \times 6 = 48$. In the same way for the FDD mode assuming 20 MHz of bandwidth and with only one antenna/spatial channel, $Q_{c\text{-}mode2} = 1$, $M_{mode2} = 96$, $N_{mode2} = 8 \times 2 = 16$. Although we could add other integer (w) for indicating the direction ($w = 1$ for downlink and $w = 2$ for uplink), in principle this would not be necessary since for FDD, two allocation tables will be created like outputs of the resource partitioning function, one for the downlink super-frame and another one for the uplink super-frame.

The first distribution of the radio resources of mode 2, done by the resource partitioning function in the ACS, allows us to control the interference problem between the UTs of this mode associated to the HERNs. The problem to solve now is how to send the resource partitioning of mode 2 from the ACS to the HERNs through the BS. In principle the best option would be that the ACS transmit to the BS, just before this broadcast the resource partitioning valid for the next SF, a control message for mode 2 resource partitioning (probably embedded in a message with some field indicating that the message is addressed to the HERNs associated to it). Then during the latter part of SF i of the mode 1, the BS would transmit to all HERNs connected to it a control message indicating the available time-frequency resources of mode 2 that every HERN may use during the SF $i+1$ of mode 2 for its UTs working in mode 2. Besides this message would be transmitted along with the control messaged for resource partitioning of mode 1. In short, the steps of the procedure for the resource partitioning in a cell containing heterogeneous relays (TDD-FDD) may be as follows:

1. During the early part of SF_{mode1} *i*, the resource partitioning function of mode 1 that is implemented at the BS would be executed. The details of this execution with the required input parameters, the resulting outputs and the main steps, are described in the section Resource Partitioning of [WINXWPMAC]. In parallel the resource partitioning function of mode 2 that is implemented at the ACS would be also executed. The execution process in principle could be similar to the resource partitioning function of mode 1, although of course with different values and probably with some changes in the input and output parameters (e.g. for the FDD mode, the input variable *TDD_Asymmetry* wouldn't be necessary).
2. For the latter part of SF_{mode1} *i* a control message is transmitted from the BS to all RNs directly connected to it, in order to inform about the parts of the next super-frame in mode 1 allocated for exclusive or shared use by the homogeneous RNs, as well as the resources for BS-to-RN feeder links (homogeneous and heterogeneous). Assuming a cell with homogeneous and heterogeneous relay nodes, previous to the transmission of this message, the ACS somehow should send to the BS a specific control message for the HERNs with the same purpose but related to the allocation of resources of mode 2 to be used by each HERN. Then this control message may be transmitted from the BS to the HERNs (using the mode 1 since the BS only implements this mode) along with the previous message, which defined the allocation of the next frame in mode 1. Probably the best option to do that is to embed the control message for the mode 2 in a message of mode 1.
3. After receiving these messages every HERN knows:
   - The chunks in mode 1 that it may use during the next super-frame (SF_{mode1} *i+1*) for receiving/sending data from/to BS. Here there are two possibilities: (i) use of exclusive or shared resources (due to the particular characteristics of BS-HERN link, stationary and high directional antennas, the transmission would be adaptive), or (ii) use of resources set aside for BS-to-RN feeder links (of course the best option). Notice that these resources are only for BS-HERN communication since for the HERN-UTs communications are using the resources of the mode 2.
   - The chunks in mode 2 that it may use during the next super-frame (SF_{mode2} *i+1*) for receiving/sending data from/to UTs associated to it.

   Besides each HERN has to execute its resource partitioning function for mode 2 during the last part of the SF *i* in order to determine the resources of mode 2 to be used in SF_{mode2} *i+1* for adaptive and non-frequency adaptive transmission of its UTs.
4. In the control preamble part of the SF *i+1* on the one hand the BS sends updates of the allocation information established in the step 1 to the UTs of mode 1 associated directly to it.
5. On the other hand also in this part of the SF *i+1* but of the mode 2, the HERNs transmit updates of the allocation information determined in the step 3 to its UTs. Note that the radio resources used in the same time slot (control preamble) are completely orthogonal since the two modes are using different frequency bands. However there is a problem to solve here because this part of the SF has to be shared between all the HERNs in the cell. So there are two possibilities: (i) to split the total bandwidth of this preamble as parts as HERNs are connected, and assign orthogonal resources to each HERN, or (ii) to create two groups (reuse pattern of 2), assuring the use of orthogonal resources only in neighbouring HERNs. Of course this last option has the advantage of a best use of the spectrum but has the drawback of the mutual interference between opposite HERNs.

In this analysis it is assumed a perfect and exact synchronization in both modes so that the frame and super-frame structure of BS (SF TDD) and HERNs (SF TDD & SF FDD) starts in the same time and has identical duration, as Figure B-6 illustrates. In this figure are included during the latter part of super-frames the control messages transmitted from the BS to the HERNs (step 2). Although so far the exact position of these messages is not defined we should estimate the required time for the execution of resource partitioning function in order to avoid that this time does not exceed the interval from the reception of the control messages to the beginning of the next SF.

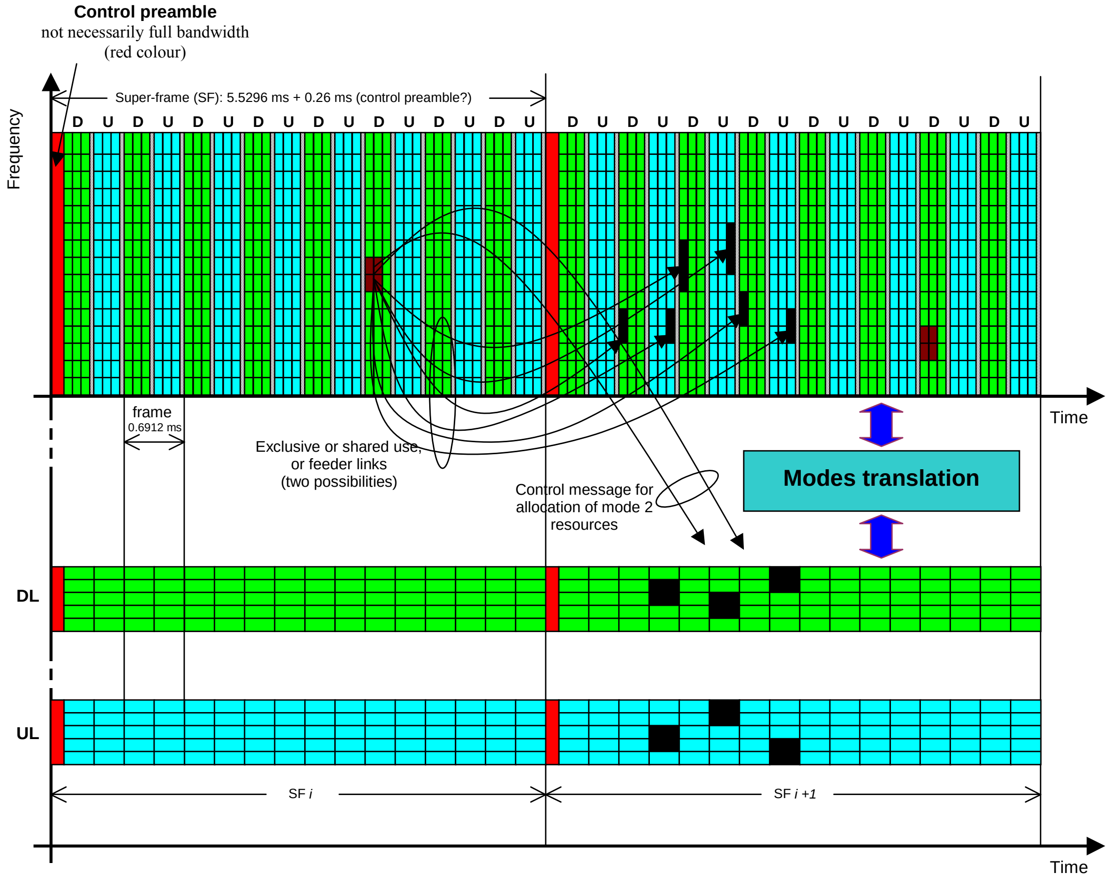


Figure B-6: Mode 1 (TDD) and mode 2 (FDD) super-frame structures in heterogeneous relaying showing the DL control chunks used for the transmission from the BS to its associated HERNs of resource allocation information to be use in the next SF

Summarizing the RRC entity located in a central node (i.e. ACS) distributes radio resources, in principle based on time and frequency domains, to each cell and/or REC associated to it. These resources could be either exclusively used or shared with other cells. The main reason for doing this mid-term resource partitioning (RP) in the ACS is that this central node has a general vision of a certain cluster of cells and/or RECs, and then it can collect all the information required for a fair distribution of the resources in order to avoid or mitigate the inter and intra-cell interferences. Assuming two-hop relaying the resource partitioning messages sent by the ACS to the BS of a given REC, are broadcasting from the RP function in the MAC-CP entity located in the BS to its UTs in the broadcast channel indicating what resources can be used in the current SF, or to its RNs in the last part of the super-frame indicating what resources can be used in the next SF. For the particular case of a heterogeneous relay node, this will receive resource partitioning messages of both modes, the used for the BS-RN link (mode 1) and the used for the communication between the HERN and its final UTs (mode 2). So the HERN once executed the resource partitioning function for the mode 2, in its own broadcast channel (fully orthogonal to the broadcast channel of the BS) will transmit to its UTs information about the radio resources of mode 2, which can be used for each UT. In other words, the best option for a fair resource partitioning, and probably the only one for RECs where HERNs are included, is the fully centralized alternative where the ACS has the total control of the resource distribution between all the cells and/or RECs associated to it.

B.3.3.1 Description and schedule of execution

First of all with regard to the description of resource partitioning procedure explained in [WINXWPMAC], it is important to note the following comments and differences:

- The BS should consider that the resources reserved for the HERNs in the next SF are exclusively for the use of communications between the BS and the HERNs (downlink and uplink), since communications between the HERNs and its UTs are performed by means of radio resources of mode 2, which in our case is using a frequency band different to the used by the mode 1.
- The allocation of resources downlink and uplink to be used by every HERN should contemplate the time required by the HERN for translating the data from mode 1 to mode 2 in the

transmission BS → HERN, and for translating the data from mode 2 to mode 1 in the transmission UT → HERN, as Figure B-8 shows.

- If the BS of the cell implements only the mode 1, in addition to this node and its RNs there will be some central node (i.e. the Access Control Server) involved in the resource partitioning procedure in order to control and manage the radio resources of mode 2 so that the mutual interference between sub-cells (every HERN and its associated UTs) can be controlled and reduced.
- The central node mentioned in the previous point should deliver through the BS, control messages addressed to the HERNs for the control of mode 2 resource partitioning. The BS once received these messages has to forward them (using of course physical layer mode 1) along with the control messages devoted to the resource partitioning of mode 1. As we said before the downlink control symbols conveying these messages will be transmitted during the latter part of a given SF from the BS to all its RNs in order to define the resource partitioning in the next SF.

Concerning the input and output parameters involved in the execution of the resource partitioning function, one possible solution could be as follows:

- The required inputs for this function in the mode 2, resident in the ACS node, would be similar to the used for the mode 1 (TDD) in the BS, but referred to the particular characteristics of FDD (i.e. *TDD_Asymmetry* not needed). In the same way the resulting outputs would be allocation tables (two sets, one for DL and another one for UL) indicating the partitioning of mode 2 resources, but eliminating the *Feeder_assignment* output since we are assuming that the HERNs are only connecting UTs and not other RNs.
- About the resource partitioning function in the BS, the required inputs and the resulting outputs would be the same described in the Resource Partitioning section of [WINXWPMAC], regarding the homogeneous relay case and the physical layer mode TDD.

Figure B-7 shows a first approach for the schedule of time per super-frame for the execution of the resource partitioning function in the different nodes involved in the process, as well as the delivery of control messages needed for informing about the use of time-frequency-spatial resources.

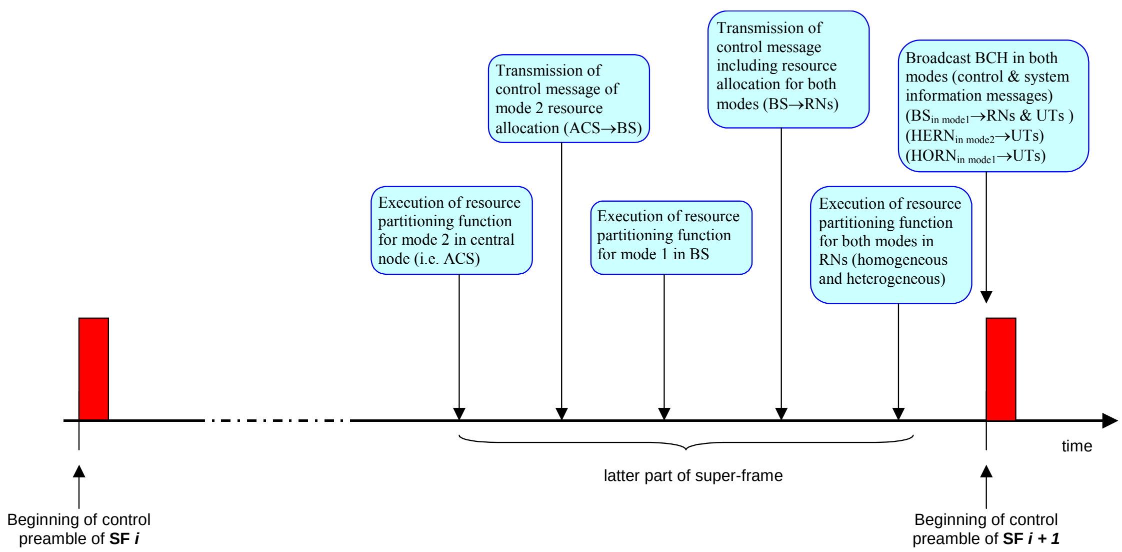


Figure B-7: Schedule for the execution of resource partitioning function and for the transmission of control chunks

B.3.3.2 Modes translation example for voice service

In a particular case of voice service, since the BS sends a radio packet (a certain number of chunks in mode 1) till receives a radio packet of final UT (a certain number of chunks in mode 2), it is necessary two frames (as Figure B-8 shows), that is, 1.3824 ms, and assuming a minimum of 80 payload bits (for lower modulation alphabet) we could get bit rate of 58 Kbps, enough to assure a high quality in a voice communication. The problem now is to know if the times indicated in the Figure B-8 for modes translation are enough. To follow it is described the communication process for voice service between a BS and a given UT through a HERN.

- Frame n:
    - (TDD) In some time-slot of the DL sub-frame, the BS transmits radio packet in mode 1 to the HERN.
    - (FDD) In some of the two time-slots of the UL frame, the UT transmits radio packet in mode 2 to the HERN.
    - The HERN in this frame only receives (radio packet in mode 1 from BS and radio packet in mode 2 from UT). In the HERN is contemplated the possibility to receive simultaneously both packets. Assuming total asymmetry in TDD mode, the minimum time available for decoding the radio packet in one mode and encoding in the other mode would be 128 μs (TDD chunk duration + TDD guard time).
- Frame n+1:
    - (TDD) In some time-slot of the UL sub-frame, the HERN transmits radio packet in mode 1 to the BS. This radio packet corresponds to the transmitted by the UT to the HERN in the previous frame using the mode 2.
    - (FDD) In some of the two time-slots of the DL frame, the HERN transmits radio packet in mode 2 to the UT. This radio packet corresponds to the transmitted by the BS to the HERN in the previous frame using the mode 1.
    - The HERN in this frame only transmits (radio packet in mode 1 to BS and radio packet in mode 2 to UT). In the HERN is contemplated the possibility to transmit simultaneously both packets.

This way the terminal does not need to incorporate a duplexer for transmitting and receiving simultaneously, even only one local oscillator would be enough, and then it would be possible to decrease the costs, the hardware complexity and the power consumption of this element. However from a RN perspective, the fact to have to implement two different frequency bands, one for the TDD mode and another one for the FDD mode, as well as to have to transmit and receive at the same time, increase clearly the HW complexity of this kind of relay.

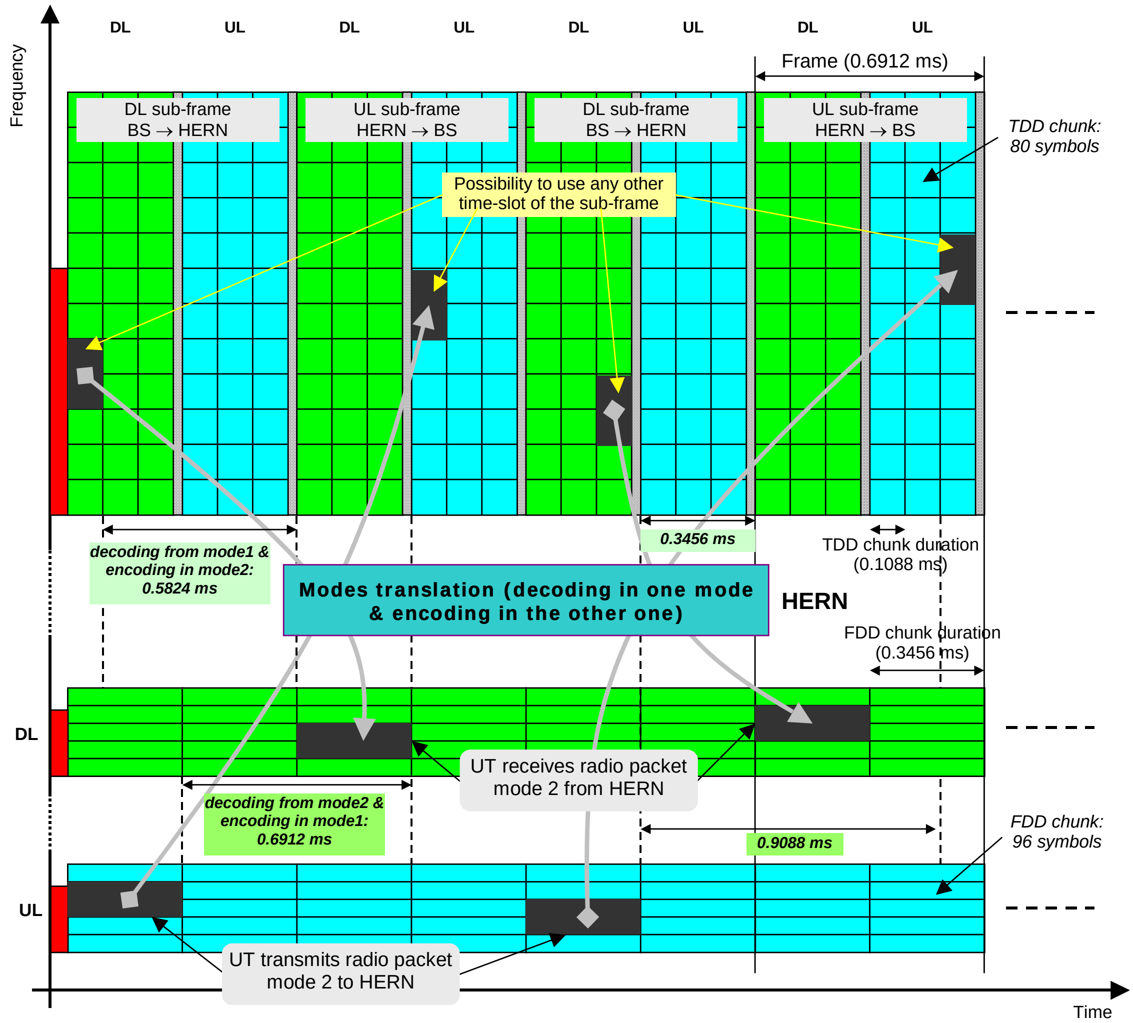


Figure B-8: Time estimation for modes conversion in a HERN for a voice service

B.3.4 Complexity issues

Besides the general complexity topic, common to any kind of fixed RN, the fixed HERN operating in TDD and FDD modes, has additional complexity issues, which are convenient to remark:

- The synchronization problem of both modes so that the super-frames have same duration and start at the same time.
- The problem to implement two different transceivers.
- The design of the antennas system. In the HERN, which is using omnidirectional or sectored antennas for serving at its final UTs and high directional antennas for the communication with the BS, this problem is related with the previous point. However in the case of the BS operating in TDD, if we want also to use directive antennas, we have to analyze the coexistence of these antennas with the used by the BS for serving at its final UTs.
- The adaptation of data rates for the two spans of the multi-hop link taking different chunk sizes into account.

B.4 Conclusion and further works

Inside the WINNER concept in the course of the first phase of the project, two different Physical Layer Modes have been defined and developed in order to cover in an efficient way the different requirements initially stated for the WINNER deployments. These two PLMs are, the TDD transmission over unpaired band and the FDD transmission over paired bands supporting half-duplex terminals. In spite of the fact that any of these PLMs could be configured for any type of scenario, the TDD has been envisaged for short-range cellular deployments as metropolitan areas, whereas the FDD has been identified as more appropriate for wide area cellular scenarios. However it is clear that in certain mixed scenarios with different data rate requirements and radio propagation conditions, both modes will have to be deployed in the same area, having to deploy either in the same or in different physical nodes, TDD and FDD base stations. For those cases the use of heterogeneous relay nodes instead of base stations deployment, could be a fair candidate to minimize deployment costs and to increase the spectral efficiency.

The deployment concept described in this section is based on the use of heterogeneous RNs and it composed by the following logical nodes in star/tree topology: RANG, ACS, BS, HERN and UT. The BS is connected to the backhaul network through the RANG and the ACS in the user plane and control plane respectively. Otherwise some UTs are connected to the BS by means of a multi-hop (two hops) communication through the HERN. Depending on the particular radio propagation scenario for what this DC is intended, the physical location of previous logical nodes might be different (ACS, RANG and BS co-located in the same physical node or far away each other).

The multi-mode reference protocol contemplated from the first of WINNER project will enable the heterogeneous relaying. Although from a user plane focus, the operation of a HERN is practically identical to the homogeneous case, in the control plane is envisioned the modification and adaptation of some RRM functionality as for example the resource partitioning and power control.

Finally although it is clear the complexity increase of a HERN relatively to homogeneous case, in some scenarios under certain requirements and propagation conditions, the deployment of heterogeneous relaying could be useful. For this reason in WINNER phase II, it is proposed to follow on investigations concerning this type of relay nodes, being the main research items as follows:

- To identify the commonalities between the two modes involved in a HERN in protocol layers with generic and specific parts.
- To specify the interfaces between mode-independent and mode-dependent functions.
- To clarify the role or mission as well as the interaction with the mode independent and dependent functions of the Stack and layer MCM.
- To estimate the delay in MAC frames for additional process in a HERN. Besides decoding/coding it should be implemented the functions for controlling modes conversion.
- Estimation of round trip time (RTT) for hop-by-hop and end-to-end, taking into account the low error probability in the BS-HERN link due to the particular characteristics of this link (fixed and optimized by the use of high directional antennas or beamforming).
- Preparation of simple simulations in particular scenarios (different data rate requirements and radio propagation conditions) where the use of the HERN could be useful.
- Estimation of exact cost of a HERN as compared to a base station.
- To investigate the heterogeneous relaying between one WINNER RAT-mode and another legacy RAT (e.g. UMTS, WiFi, WiMAX), identifying the benefits and drawbacks of this alternative.

Annex C Mobile Relay-based Deployment Concepts

Here we include a simple analysis on some of the concepts that MRs could be used, as a cut down version

C.1 Positioning

As we have stated in [WIND33], an area that relays (fixed and mobile) are envisaged to help in future network deployments is that of positioning. It is widely known that no single positioning technique can give very good accuracy levels for all scenarios/cases. For instance, GPS is very accurate, but does not perform well inside buildings. Timing techniques in cellular systems e.g. RTT/OTDOA are also quite promising, but in general favourable channels conditions need to occur e.g. LOS. The main limitation for timing techniques is the large cell coverage which results in NLOS, bad channels conditions, fast/slow fading all of these impacting the accuracy. Exactly in those areas mobile relays could be of help. Specifically, mobile relays can be seen in the following dimensions.

C.1.1 MR are used instead of BSs to perform positioning

Due to the fact that the coverage of mobile relays is far smaller to that of BSs, positioning techniques/timing measurements will be done for shorter distances and in more favourable channel conditions e.g. LOS, which results in higher accuracy for the location of the UT. For instance, our gain with regards to the Cell Id technique, when performed by a BS and a MR with $R_{BS}=2R_{MR}$ is $A=\frac{\pi R_{BS}^2}{\pi R_{MR}^2}\xrightarrow{R_{BS}=2R_{MR}}A=4$, i.e. four times better accuracy. This is shown in Figure C-1. This analysis can be extended for the case of 2 and 3 cells for Cell Id, RTT and OTDOA techniques (techniques standardised in 3GPP). The area defined by the intersection of the circles of the MRs/BS is indicated by the inclined /horizontal lines respectively for Cases B/C.

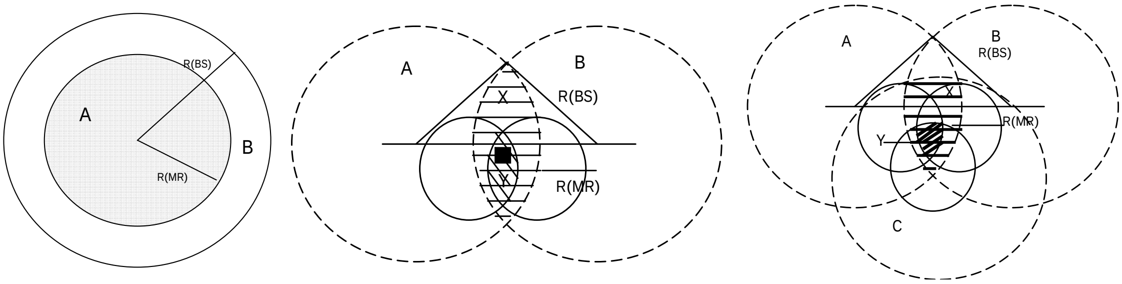


Figure C-1 (a/b)Cases for Cell Id positioning with BSs and MRs for 1/2/3 cells (A/B/C)

C.1.2 MR are used to increase the "pool" of positioning techniques in the absence of BSs

The second dimension is actually using a MR to bypass limitations that exist in certain deployment cases.

Assume an isolated site (only one BS), where the only techniques available are Cell Id and RTT. However, with the use of a Mobile Relay, we can mimic another BS which means that we can enhance the accuracy levels from Level I, to Level II with one MR and to Level III with 2 MRs (TABLE III).

TABLE III PARAMETERS FOR SCENARIO I

			1MR			**2MRs**		
			Cell Id	RTT	OTDOA	Cell Id	RTT	OTDOA
Sites	**Technique**	**Accuracy Level I**	**Accuracy Level II**	**Accuracy Level II**	**Accuracy Level II**	**Accuracy Level III**	**Accuracy Level III**	**Accuracy Level III**
1	Cell Id	Area A	AreaB	Line/Arc	N/A	*AreaC*	1 Point	Hyprbola
	RTT	Circle	Line/Arc	2 points	2 points	Line/Arc	*1 point*	2 points
	OTDOA	N/A	N/A	2points	Hyperbola	AreaB	2 points	*1 point*

Comments:

- Area A/B/C →Circle/Common area of two circles/Common area of three circles respectively
- The "2MRs" case has two realisations: Two separate MRs or one MR in two different positions.

- We are not restricted to use only one MR and due to the mobility/large number of MRs we can select the best MRs in every time instance e.g. MR in the intersection of streets

C.2 Tx power levels

Closely related to the MBMS case and Positioning (supporting CPICH and CCPCH) is the issue of power control and specifically Tx power levels of a MR, which define the coverage of a MR. (Applicable for Type II /III MRs. For Type I we assume the power is always fixed – area of coverage always the same) As it has been investigated in [BaLe05], MRs do offer gain in terms of better reception of signal levels at the UT. However, problems occur if the MR employs fixed power. There is no need and we may induce additional problems. Thus, it has been investigated that pattern-based or variable pattern-based Tx power levels should be employed by MRs to try and "provide" uniform power reception levels at the UT. This is achieved by the MR employing, for instance, "anti-Gaussian" type of Tx power as they move in/out of the area of coverage, if we assume movement on a straight line with constant velocity. Results in [Bakaimis05] show that with these simple schemes we succeed in having a more subtle "operation" for the MRs and "lessening" the impact in the network. Figure C-2 shows the relevant results. Cases C1/3 represent the fixed power concepts, Case C4/5/6/7 are the pattern-based and C8 is the variable pattern-based concept. The main measure is the difference between max-min values which are in the area of 4dB/3/1dB respectively for the three schemes. This analysis is applicable for single frequency networks, but could be applicable to e.g. OFDM networks.

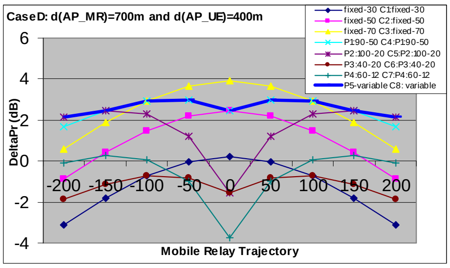


Figure C-2 (a/b) MBMS provided by UT in its close vicinity

C.3 Cooperative Mobile Relaying

C.3.1 Connectivity Investigation

In [WIND34] we investigated the connectivity for mobile relays. What we found out is that a number of parameters need to be taken into account when selecting the MR(s) for cooperative relaying. The more parameters we take into account, the more complex, but at the same time, the more accurate/reliable the systems becomes. Effectively, we monitor some statistics of the MRs e.g. number of connections, connection profiles, etc which could be used for performing a more accurate MR selection. Thus, extending the investigation of [WIND34], where also the system model is included, here we provide some further analysis. In Figure C-3we present (a) the connection profile (number of connections per device per time instance) and (b) the accumulated sum of those connections per device. However, even those results are not enough in order to select the best MRs in terms of the connectivity. Specifically, in terms of connectivity two seem to be the main parameters i.e. connection times and duration of each connection interval (or average connection duration). We want to minimise/maximise them respectively and this we portray in Figure C-4(a). We assume an interval (i.e. Total Connection Time) of e.g. 100 sec (X axis). We assume that this is split to a number of connections (1,….,N), that we have the same number of connection/disconnection intervals, that each Connection/DisConnection (C/DC) interval has the same duration of A/1 secs respectively, with A=AverageConnectionInterval. (For reasons of simplicity we assume same length for each "C" interval) Thus, based on the above we have

$$
\begin{aligned}
&TotalConnectionTime(100\,\text{sec}) = \\
&= NoOfConnections * AverageConnectionInterval + NoOfDicConnections * AverageDicConnectionInterval = \\
&= NoOfConnections * A + NoOfDisConnections * 1 = Connections * (A+1)
\end{aligned}
$$

This analysis, tries to simulate the ideal/optimum case i.e. for a specific number of connections what is the max average duration for all connections and this is depicted by the blue line (A0 case) in Figure C-4(b). Any value below that line represents a non-deal/sub-optimum case. Cases "A1-A7" represent the 7 devices portrayed in Figure C-4Figure C-3(a). What is portrayed is the average length of each connection interval which effectively indicates the efficiency of each interval. Cases B1-B7 represent the total connection time for those devices, with reference to Cases A1-A7. One interesting outcome is that even though Cases 2/5/7 have the same connection time (NoOfConnection*AverageConnectionTime=45secs), we should select Case2 due to the smaller number of connections. This could be extended in choosing a case with slightly smaller Total Connection Time, but with smaller number of connections. However, this has to be dealt on a case by case scenario taking into account e.g. type of applications to run e.g. voice is not the best due to the high number of disconnections, although packet based services could cope with some abnormalities. Additional information could be e.g. standard deviation of the duration of each interval, due to the average values sometimes giving misleading results.

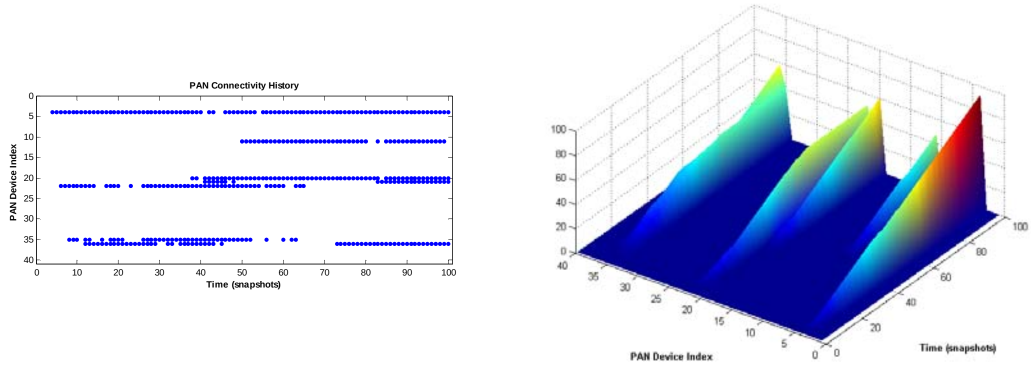


Figure C-3 (a/b) Connectivity results per device per time instance

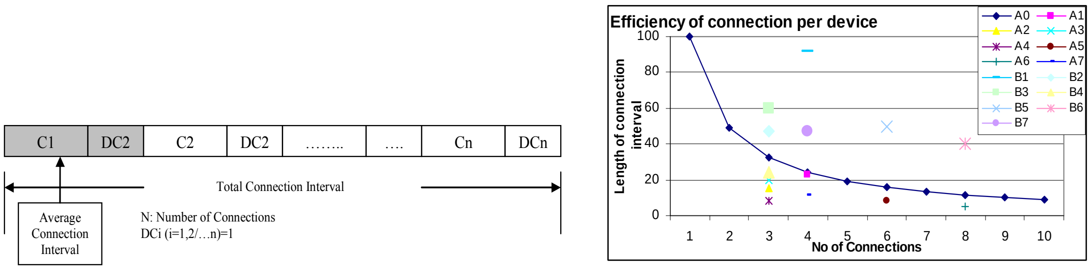


Figure C-4 (a/b) Efficiency of each device

C.3.2 Connectivity comparison of Type II/III MRs

In the previous section we discussed the connectivity aspects for Mobile Relays Type III. In this section we will compare the connectivity of MR Type II and III for a very simple scenario just to get an indicative idea of their relative performance, as it has been highlighted in [Bakaimis05]. We assume that both MRs are moving on a straight line with constant velocity. Thus, in Figure C-5 we present the *maximum* velocities that MRs are required in order to be able to support for a number of different cells (i.e. diameters of cells (X Axis)), connection times of 10/100/1000 seconds. This is a quite simplistic example, but it gives a good initial understanding of those connection times. Thus, if we assume that an average cell in a urban area is 250m (cell diameter =500m) then in order to support connection times of at least 120 secs, we can use a MR Type II (max v=12 or 4.8 km/h) or Type III (max v=2.4km/h), whereas for connection times between 300-600 sec (5-10 min) we are restricted in using MR Type III. (max v=2.4km/h). In general for cells of 200m→d=400m for applications of more than 2-3 minutes, it seems that the best selection is either a slow moving Type II or (a better solution) a Type III MR. Of course other parameters, like type of applications to support, power supply of the MR, functionalities it can support etc should be taken into account. So, the conclusion is that even though Type II can actually provide decent connection times, it is Type III which can support all types of cells and applications.

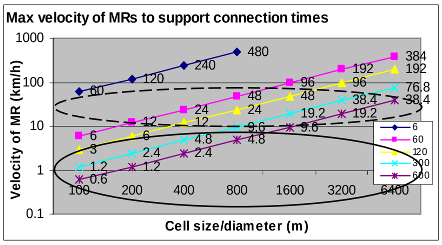


Figure C-5 Connectivity results for MRs Type II and III

Annex D Multi-constrained routing and cooperative relaying

The multi-constrained approach to routing in WINNER in the form of the *Multi-Constrained Optimized Link State Routing protocol (MC-OLSR)* was first proposed in [RFC3626] and then further investigated in [WIND31] and [WIND32]. However the main disadvantage of this solution was the lack of support for cooperative relaying. Therefore it has been extended appropriately recently on the basis of the work performed within the confines of [WIND24], where the *Adaptive approach to antenna selection and space-time coding*, was analysed. Consequently, the multi-constrained routing protocol enhanced with space-time coded cooperative relaying, based on virtual antenna arrays, is described here.

D.1 Introduction

The conventional *Optimized Link State Routing protocol* belongs to the proactive group and it features a *Multipoint Relay selection mechanism (MPR)* for the needs of control messages dissemination, which keeps the protocol overhead minimum [RFC3626]. It is also equipped with other enhancements, such as the possibility of specifying the willingness of distinct nodes to forward traffic. There are different levels of willingness, making it possible to decide which nodes or class of nodes can act as BSs, which of them cover the RN functionality only and which may be simply excluded from relaying (eg. UTs). There is also another advantage, namely since it is a link state class protocol, each node is aware of its two-hop neighbourhood. This information can be exploited very efficiently at the downlink for the purposes of relaying in two hops, which are envisaged the optimum number in case of the WINNER system. Taking all those issues into account it was decided to tailor this protocol more accurately to WINNER requirements and in result its core functionality was further extended with the multi-constrained routing algorithm [WIND31], as the means of enabling the *Quality of Service* routing. The idea was to select a number of distinct paths between the source and the destination nodes and allow an application to choose one of them on the basis of the required parameters such as bandwidth, delay, etc. In the following subsection another improvement is introduced, which will allow to exploit the additional diversity offered by space-time coded cooperative relaying with the use of virtual antenna arrays. To this end additional information is exploited, which is provided by the modified *MPR selection heuristic*.

D.2 Multipoint relay selection and virtual antenna arrays

The core *Multipoint Relay* selection heuristic is aimed at finding a subset of one-hop neighbours, which cover the whole strict symmetric two-hop neighbourhood [RFC3626]. In result control messages are disseminated with the use of this optimized set of neighbours, and therefore the network overhead is reduced.

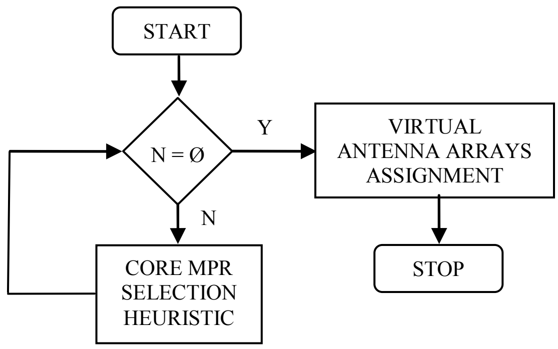


Figure D-1: Extended MPR computation heuristic featuring VAAs selection

It was noticed that this mechanism can be also applicable for the purposes of selecting the nodes that may act as the elements of virtual antenna arrays. More specifically, redundant MPR sets are introduced i.e. secondary MPR set, ternary MPR set and so on in addition to the generic MPR set. Then the original MPR set selection algorithm is performed iteratively as long as all the nodes, which can act as MPRs, are subdivided into the generic and the additional MPR sets. The additional MPR sets contain information about nodes that can be used for the purposes of cooperative relaying with the use of virtual antenna arrays, whereas the generic MPR set remains unchanged. This information is then processed accordingly and advertised in the one-hop neighbourhood with the aid of *Hello* messages. The extended version of the

MPR selection algorithm is depicted in Figure D-1, where *N* represents the whole set of *one-hop* neighbours.

D.3 Conclusions

For the purposes of evaluation of the proposed solution, a cross-layer simulation environment was developed. The specific functionalities were grouped into physical, link, network and a common higher level layer in order to make the joint optimisation more efficient. The simulation assumptions pertaining to the link layer, as well as the details concerning the simulated scenario and achieved results, can be found in the Multi-Link and System-Level Assessments chapter in the deliverable D2.10.

Annex E Evaluation of Key Aspects

E.1 Introduction

This Annex contains system assessment results of the WINNER system concept in accordance to the common agreed assumptions. The performance evaluations have been done for various aspects of the system concepts and give an overview over their benefits and drawbacks. Central elements of the investigations are the relaying concept and the adaptive transmission. A general study on the effects of adaptive transmission for system level simulations is performed in Section E.2. Section E.3 presents performance estimations of the WINNER system concept as proposed by [WINXWPMAC] in different cellular deployments.

Different relaying approaches have been investigated, decode-and-forward (Sections E.4, E.5) and more sophisticated delay diversity based distributed schemes (Section E.6). In all cases, performances gains can be achieved through the use of fixed relay stations, although different deployments also lead to situations where relaying is not beneficial. Mobile relays, which are considered as an add-on technology to the basic system concept, have not been considered in this chapter.

All those results together contribute to reach a deeper understanding of the performance properties of the common WINNER system concept. System Level Simulation tools have been developed by individual partners to investigate certain aspects of the system concept as described above. The work achieved on the system level simulation tools provides a solid basis for further in-depth investigation in the upcoming second phase of the project, enabling to base the overall system concept on simulative evaluation of the proposed concepts.

E.2 Evaluation and system-level aspects of adaptive transmission

E.2.1 Motivation/Introduction

One major part of the WINNER system concept is the envisaged adaptive mode. Various theoretical as well as semi-analytical investigations assign a good benefit, i.e. capacity gain, over existing approaches. First attempts for adaptivity have been introduced in current 3G systems utilizing so called fast power control. The latest attempt to further increase the adaptivity has been introduced with the HSDPA transmission technology. WINNER does further make another step towards more adaptivity by introducing predictive algorithms for channel-state evolution. It is known that such attempts are only advantageous for certain conditions and do not bring benefits over the whole range of operation. Predictive algorithms do depend on the channel coherence-time, i.e. the channel needs to be sufficiently stable for prediction. As the channel coherence time is inversely proportional to the users speed, predictive and adaptive transmissions are expected to perform well only for low speed users. Adaptivity is also a key aspect of MIMO transmissions.

This section contains considerations concerning adaptive transmission from system-level point of view. Throughout the WINNER project a common set of simulation assumptions have been agreed. However for the purpose of these simulations, some additional assumptions are necessary. Those assumptions that are additionally needed to the common one are described in the following and mentioned in the text where necessary.

E.2.2 Scenario Description

The scenario chosen for simulations is a macro-cellular urban deployment. The cells are hexagons of 250m edge length and the clusters are of clover-leafs shape. Both situations of frequency reuse of one and three – as depicted in Figure E-2 – have been simulated. The used deployment is depicted in Figure E-2 and the associated antenna diagram is depicted in Figure E-1. The antenna is only modelled in the horizontal direction, a dependency to the elevation angle is not considered.

The statistics are collected from the inner three cells only, i.e. the inner cluster. Two interferer rings surround this so called reference area.

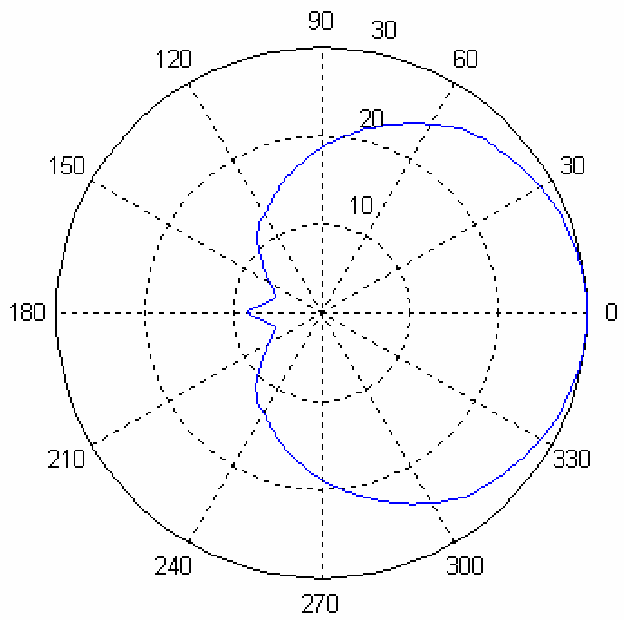


Figure E-1: Used antenna radiation pattern

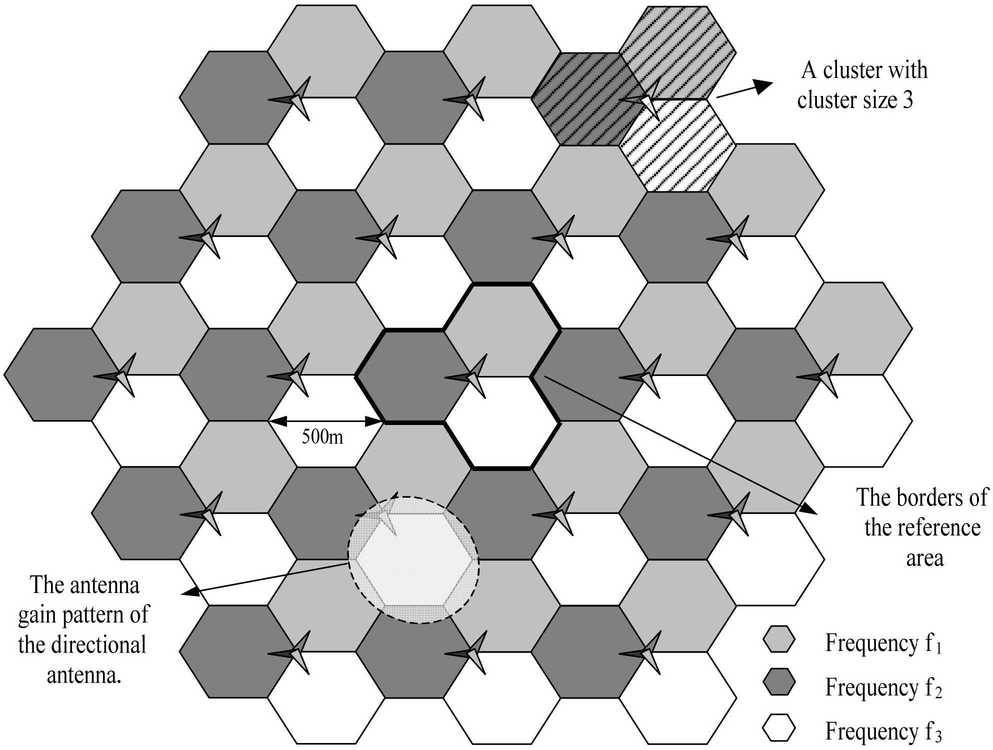


Figure E-2: The simulation area

A Markov-mobility model with 3 and 30 km/h user speed is used. However 30 km/h are somehow optimistic for adaptive systems and label the upper bound. The arrival is a modified Poisson process with a uniformly distributed starting position within the environment. The user is assigned to the cell with the best propagation conditions including antenna gains and average over channel measures, initially and in certain intervals of the simulation of a couple of sub frames. The traffic model is modelled by a web-browsing session with scaled-up traffic demand. The following parameters characterize the probability density function of the packet sizes:

$$P(x) = \begin{cases} \dfrac{\alpha \cdot k^{\alpha}}{x^{\alpha+1}}, k \le x < m \\ \beta, x = m \end{cases} \tag{1}$$

with $k = 500.000.000$, $m = 1.500.000.000$ and $\beta = 1{,}1$. A user keeps alive in the system during its session duration, users do not get dropped. The session consists of a geometrical distributed number of packet calls. Each packet call by itself consists of a geometric distributed number of packets, see Figure E-3 for details. The packet call inter arrival time is geometrical distributed.

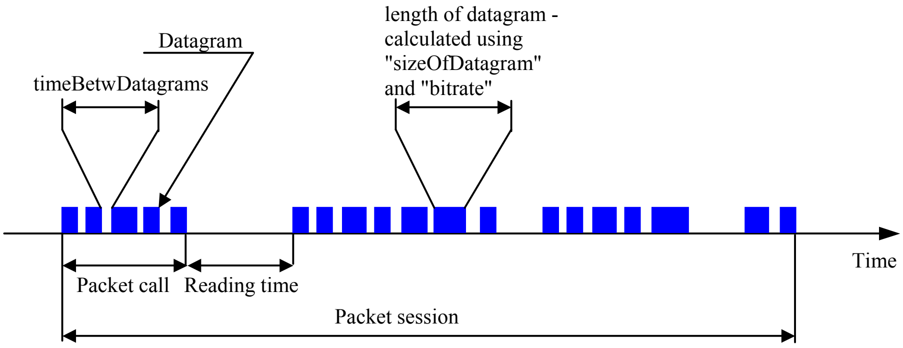


Figure E-3: WEB-browsing traffic model

The system efficiency is measured at a load, where 98% of the users are satisfied. A user is said to be satisfied, if the average throughput over the session is at least 10% of the nominal throughput.

E.2.3 Scheduling and Link performance

As we are interested in the general upper bound to be achievable through adaptive transmission, we assume perfect channel knowledge at the transmitter. This means that the transmitter is able to perfectly adapt to the channel conditions and utilize the channel capacity fully. It is furthermore assumed that the channel remains constant during the transmission of the burst for simplicity. This can be achieved through a fast physical layer signalling, i.e. the used modulation and coding/puncturing is in-band signalled to the receiver. For the further discussion and the performance evaluation the needed constant signalling overhead is not considered anymore.

The receiver is modelled by an optimal decoding procedure, i.e. perfect channel knowledge as well as optimal decoding performance (iterative turbo-decoding) is assumed. The impact of synchronization bias and imperfect channel estimation is modelled by a degrade of the SINR. It is known that the loss of SINR for a given small frequency error Δf is given through:

$$\mathrm{SNR}_{\mathrm{loss}} = \frac{10}{3\ln 10} (\pi T \Delta f)^2 \frac{E_S}{N_0} \tag{5}$$

The channel estimation bias is more difficult to model, as there are a lot of options for different algorithms with different performance. We concentrate on the Minimum Mean Squares Estimation (MMSE) for this purpose and simplicity. Lets assume a pilot sequence $P = (P_1, \ldots, P_N)$ of length N and the noise represented by $n = (n_1, \ldots, n_N)$. Then the error of the channel estimates can be expressed as

$$e = \frac{n \cdot P}{\|P\|^2}. \tag{6}$$

It can be seen, that the estimation error only depends on the noise and the pilot symbols. Thus we do not depend on the E_S / N_0 ratio. Hence for this particular case, we can model the channel estimation bias by a simple degrade of the SINR.

The applied scheduling is a proportional fair scheduling strategy. All users share the same bandwidth equally.

E.2.4 Result stability& Birth Process

The Poisson arrival or birth process has been modified as mentioned above. This was necessary, because the simulations have shown to be unstable, i.e. will either become completely empty or overloaded. It turns out, that the stability of the results very much depends on the packet sizes and the channel conditions. To overcome this problem, the arrival process has been modified. The arrival rate has been increased artificially, thus much more users are born. In order to not overload the system, the number of admitted users to the system is upper bounded by the total number of users in the system. All those users that are rejected from the system are not considered for the user-satisfaction evaluation.

The reason for this instability can be seen out of Figure E-4 and will be explained in the following. Let's consider a session that will be started when the channel conditions are good, i.e. the achievable throughput due to adaptation is rather high. This on the other hand reduces the transmission time and the session can be closed nearly immediately after start. For the other users in the system this means that in average the experienced interference is lower. And even sessions starting with bad channel conditions will achieve better throughputs and thus shortened transmission times. On the other hand considering a session starting with bad channel conditions will take more time and thus causes more interference. The trends are in opposite direction, but do not compensate them, thus the system will end up in an unstable condition.

This effect becomes dominating when the packets sizes are low, i.e. the standard deviation of the session duration compared to the mean duration becomes large. For big sized packets, this ratio is small, thus the effect will have lower impact with larger packet sizes. This makes the choice of the traffic model parameters clear.

A further implication to this effect is the channel stability. All the above discussed issues need to be in relation to the coherence time, i.e. a short session length needs to be considered as short compared with the underlying coherence time.

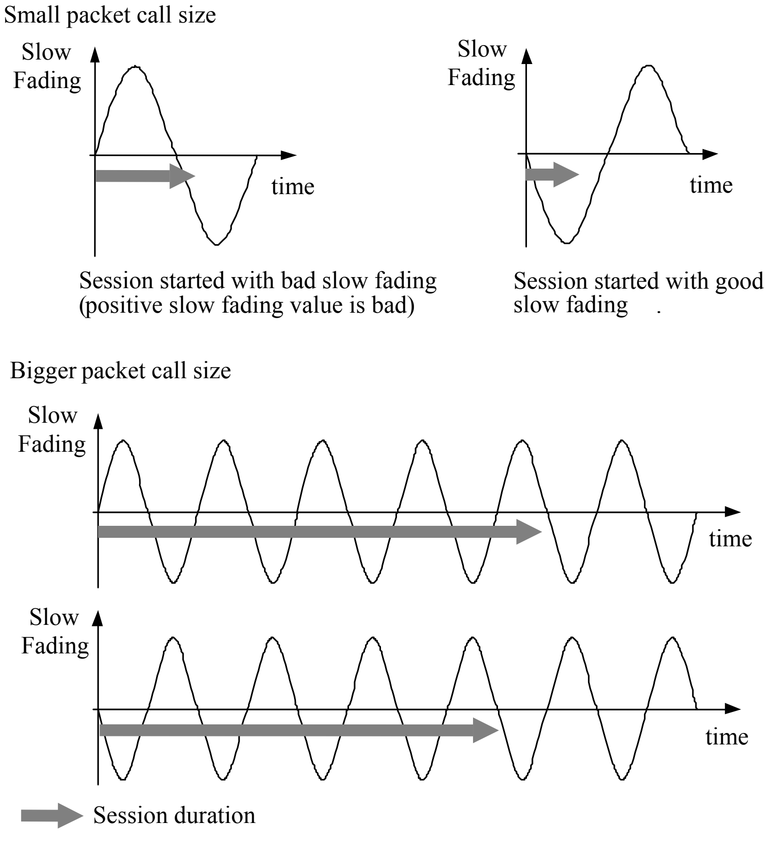


Figure E-4: Plausibility sketch of the expected session duration

E.2.5 Impact of user speed

The impact of the user speed to the session duration is depicted in figure @ below. The astonishing result shows that with increasing speed the mean session duration will be shortened. In other words, with increase of speed, the user will receive a higher mean throughput. This of the first glance surprising effect comes from the channel coherence time and the adaptive scheduling strategy. A user with higher speed will more often see different channel conditions, whereas a user with lower speed will see a slow varying channel. Furthermore users with higher speed will leave their initial geographical area and do not remain at the same cell or cell border, but change the cells more frequently, which lead to a better load balancing between the cells.

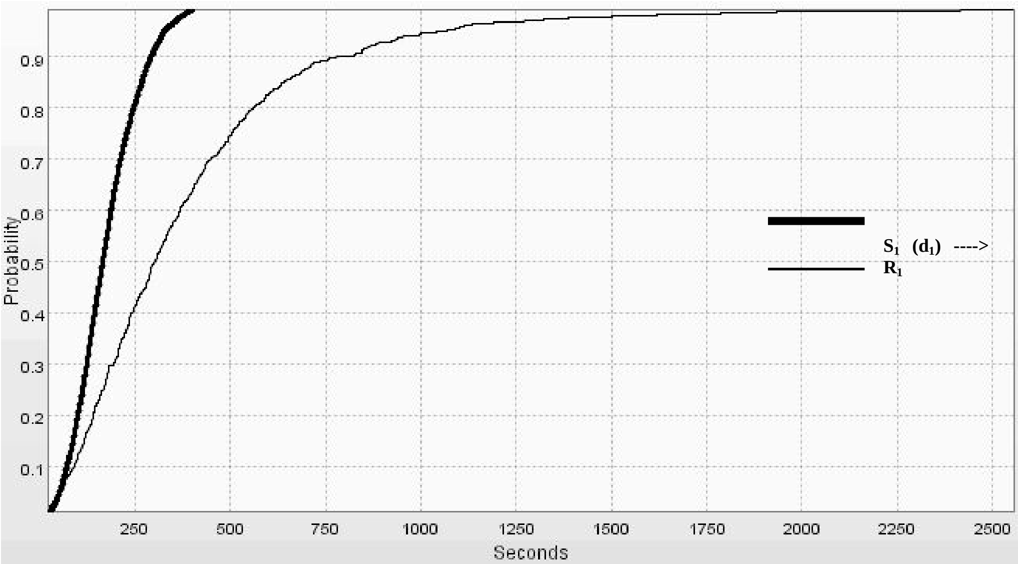


Figure E-5: Session duration for different speeds

The introduction of relay nodes to this scenario is a pending investigation. The benefits for these simulations are very much dependent on the used scenario, i.e. the propagation conditions. It is thereto no trivial task to investigate relaying in that context. The outcomes do depend strongly on the chosen assumptions and no consistent picture can be drawn at the moment, even though some gains could be expected from the result depicted in Figure E-5.

E.2.6 Achievable system capacities

The fully and optimally adaptable system would temporarily achieve throughputs that would need modulation and coding schemes that are beyond practically desirable ones. To model practical restrictions, e.g. 256QAM, we introduce an upper bound of exploitable capacity. I.e. the maximum useable capacity can never exceed this bound, although the actual channel capacity would allow higher throughputs. Figure E-6 depicts the system capacity, measured in bit per second and Hertz (bandwidth) and cell as a function of the upper limitation of the maximum throughput. The three curves represent the respective capacities with degradation of the SINR.

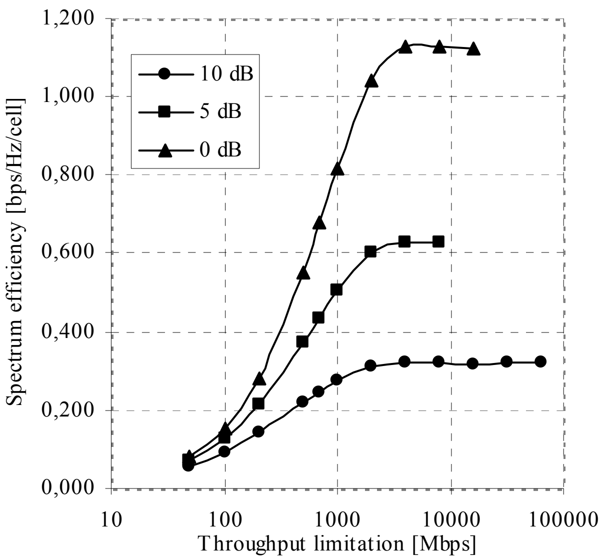


Figure E-6: The achievable efficiency as a function of upper bounding of the used modulation and coding scheme.

As can be seen, the maximum achievable efficiency approaches saturation. Further increase of the upper throughput bound does not increase the system capacity anymore. This saturation appears regardless of the degradation, only the efficiency, but not the limitation changes. From this we can conclude, that modulation schemes higher than 16QAM do not give any further gain for fully loaded systems.

E.3 Estimation of initial capacity figures for the WINNER Air interface concept in a multi-cellular deployment

E.3.1 Motivation and Modelling Approach

The motivation of this study is to obtain initial baseline performance figures to estimate the capabilities of the WINNER air interface proposal that was elaborated by the XWP MAC working group. The focus of the investigation is on IP throughput and IP packet delay figures achieved by the WINNER air interface.

The simulations model the envisaged WINNER TDD mode operating in 100 MHz bandwidth in short-range scenarios using non-frequency-adaptive transmission. Chunk- Frame- and Superframe-structure have been modelled according to the suggestions in [WINXWPMAC]. The protocol structure used throughout the simulations is displayed in Section 5.2.2.

The simulations are intended to derive baseline figures suitable for quantifying the performance gains resulting from all kinds of enhancements to be studied in the upcoming phase. The simplifications assumed in these simulations were:

- No connection admission control mechanism to prevent overloading of the system
- One traffic class only
- No terminal mobility
- No coordination between adjacent cells for interference avoidance
- Random Subchannel allocation
- First-come first serve granting of resources
- Hop-wise ARQ error correction only, no end-to-end ARQ
- No upper limit on the number of ARQ retransmissions (i.e. packets can be queued “forever”)
- No adaptive modulation. Fixed Modulation and Coding set to 16QAM ½.

E.3.2 Simulations and Results

Parameters common to all simulation runs presented in this section are summarized in the following tables.

Table E-1: Propagation Assumptions used for the Metropolitan Scenario

Parameter	Value	Units	Reference
Carrier Frequency	5.0	GHz	[WIND52]
Pathloss Model	$28.3 \times \log_{10}(d/m) + 53.5$	dB	[WIND32]
Minimum Coupling Loss	70	dB	[WIND32]

Table E-2: WINNER Air Interface Configuration used for the Metropolitan Scenario

Parameter	Value	Units	Reference
Duplexing Scheme	TDD		[WIND76]
Multiplexing Scheme	TDMA / OFDMA		[WIND76]

Modulation Scheme	OFDM / 16QAM		[WIND76]
Number of Subcarriers	2048		[WIND76]
Number of Data Subcarriers	1664		[WIND76]
Number of OFDMA Subchannels	104		[WIND76]
Subcarriers per Subchannel	16 (1 Chunk)		[WIND76]
Subcarrier spacing	50	kHz	[WIND32]
Symbol Period	22.5	µs	[WIND32]
Turbo Code Rate	½		[WIND32]
ARQ	Type 1 H-ARQ Selective-Repeat		[WIND32]
Power Control	None		
Signalling Overheads	Not modelled		
Handover	Not modelled		
UT Noise Figure	9	dB	[WIND32]
BS Noise Figure	6	dB	[WIND32]
RN Noise Figure	6	dB	
TxPower per Subchannel[6]	10 dBm		

The performance of the transmitter-receiver chain has been modelled by means of mapping functions shown in Figure E-7. The coding performance of the ½ rate turbo coder together with the 16QAM modulation leads to an approximate minimum SINR requirement of about 5 to 6 dB.

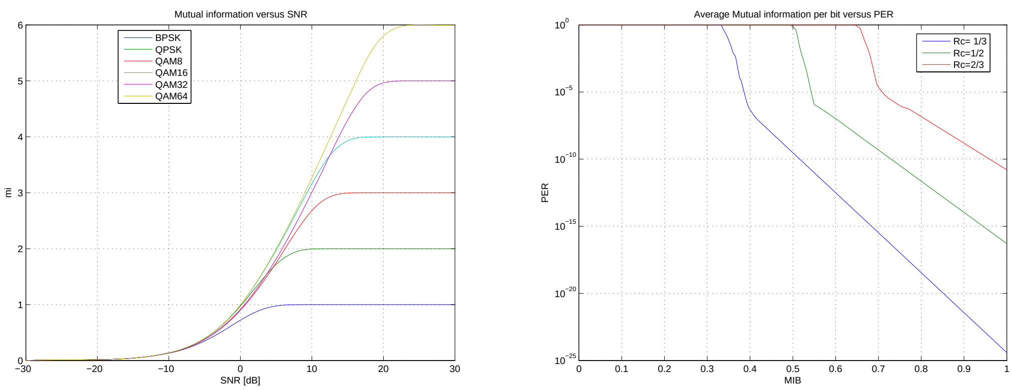


Figure E-7: left: Mapping from SNR to Mutual Information Domain for different modulations right: error performance of assumed FEC schemes

The traffic offer for all the simulations was specified per cell. IP packets in both UL and DL direction had a fixed header size of 60 bytes (typical TCP/IP header) and a variable payload between 0 and 11520bit, resulting in evenly distributed packet sizes between 480bit and 12000bit. The mean packet size is therefore 6240bit. Packet Inter Arrival Times (IAT) for each individual user followed a negative exponential distribution with the mean value calculated according to (7), where *Traffic* refers to the total traffic offer per Relay Enhanced Cell (REC) and *nUT* refers to the number of User Terminals in the REC. Each user maintains individual UL and DL generation processes, leading to the factor of 2 in the equation.

[6] Note that for 104 OFDMA-subchannels (á 16 subcarriers) this yields a theoretical max. output power per station (when all 104 subchannels on all 1664 subcarriers are in use) of ~30 dBm

$$IAT = \frac{\overline{PacketSize \cdot 2 \cdot nUT}}{Traffic}. \quad (7)$$

The following table gives an overview about all performed series of simulations.

Table E-3: Simulations Overview

Name	# Hops	# APs	#RNs	#UTs	Cluster Sizes	Traffic Range [Mbps / REC]	Cell Radii [m]
One Hop Singlecell	1	1	0	50	1	20-100	50,100,150
Two Hop Singlecell	2	1	6	210	1,3,7	4-60	150
One Hop Cluster	1	7	0	50	1,3,4,7,12	20-100	100
Two Hop Re-use	2	7	42	1470	1,3,7	4-60	150

In the following, the results from each simulation run will be presented in a separate subsection together with the specifics of the regarded scenarios.

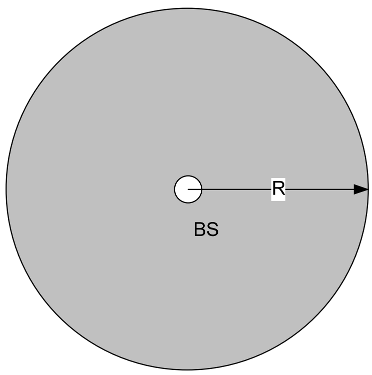
E.3.2.1 One-Hop Singlecell

The objective of this simulation run was to determine the transmission range / cell radius and the achievable capacity of an isolated cell.

E.3.2.1.1 Scenario Description

The simulation setup involved a single isolated cell. 50 static users were evenly distributed over a circular area of radius R. The radius was set as described in Table E-4. Traffic has been varied between 20MBit/s and 100 MBit/s cell load.

Table E-4: One-Hop Singlecell scenario



Cell Radius	Subchannels per Cell
R=50 m	104
R=100 m	104
R=150 m	104

E.3.2.1.2 Results

To estimate the achievable transmission range in the ideal case, we investigate the CDF of the SINR at which the BCH transmissions are received by the UTs (Due to individual subcarriers allocated for the BCH at all BSs, the BCH is received un-interfered). It shows that the target SINR of about 5 dB can only just be met for cells with R=100m. The case R=150m exhibits ca. 50% of users out of communication range

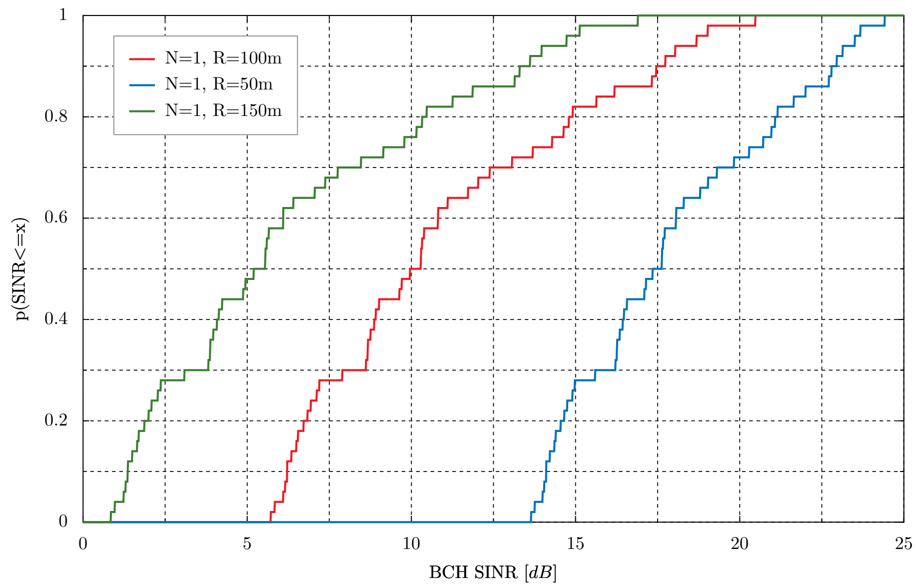


Figure E-8: CDF of the SINR of the received BCH transmissions for different cell sizes.

The next figure shows the IP packet error probability, another proof for cell size estimation presented above. In the case of R=150m the error prob. exceeds tolerable levels.

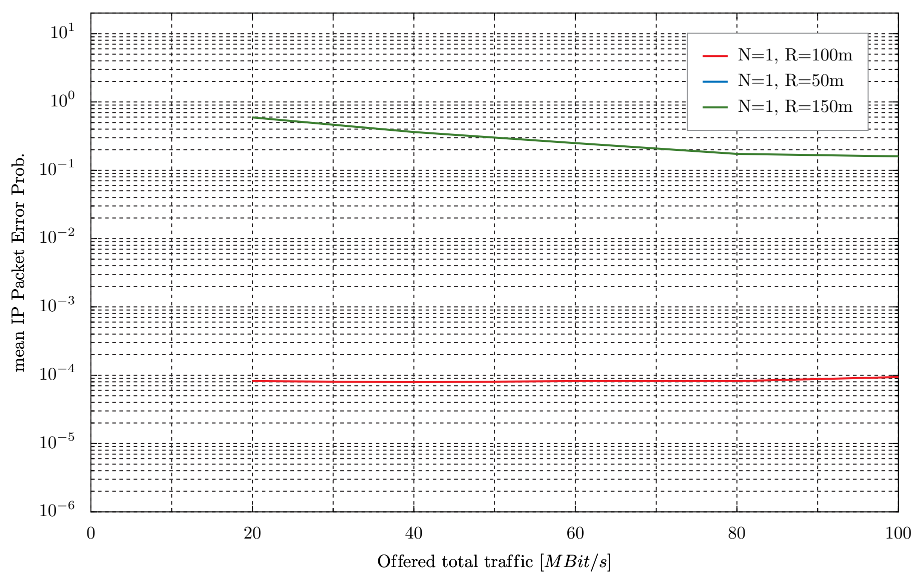


Figure E-9: Mean IP packet error probability vs. offered traffic

To show the IP traffic capacity of the isolated cell, the traffic offer was increased in subsequent simulation runs to reach the saturation point of the cell. The figure shows that for the cases with R=50m and R=100m, the cell starts becoming congested beyond 80 MBit/s. In the R=150m case, only 50% of the offered traffic is carried in all cases since 50% of the users are out of communication range. The fact that (i) no admission control is performed and (ii) no upper limit is put on the number of ARQ retransmissions leads to the rapid degradation of cell throughput in the overload case.

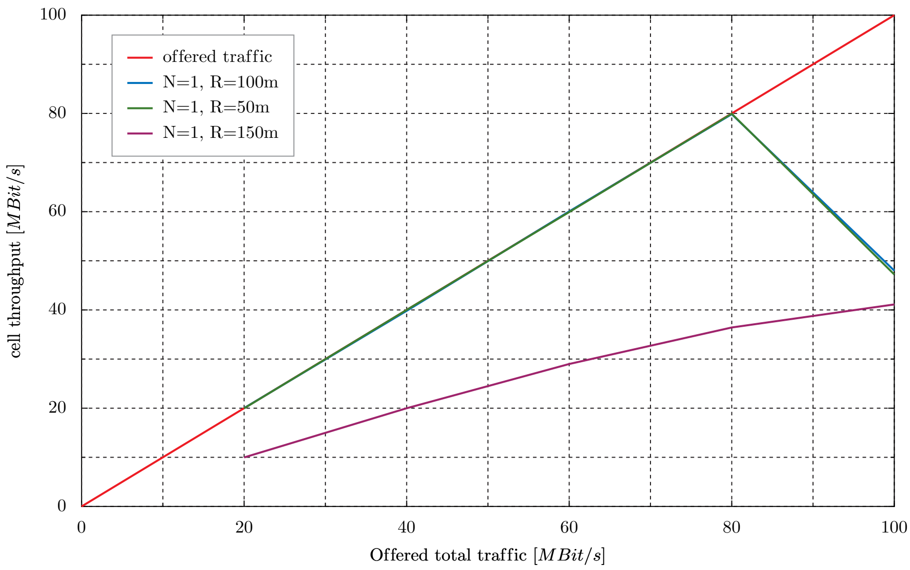


Figure E-10: Total cell IP throughput vs. offered traffic for different cell sizes.

The next plot shows the CCDF of the delay experienced by the IP packets in the case of 20MBit/s traffic load and cell radius R=100m. Reasonable delays can be guaranteed up to a cell load of 80MBit/s, with only little dependency on the traffic offer. For 100MBit/s, the congestion becomes clearly visible as delays significantly increase.

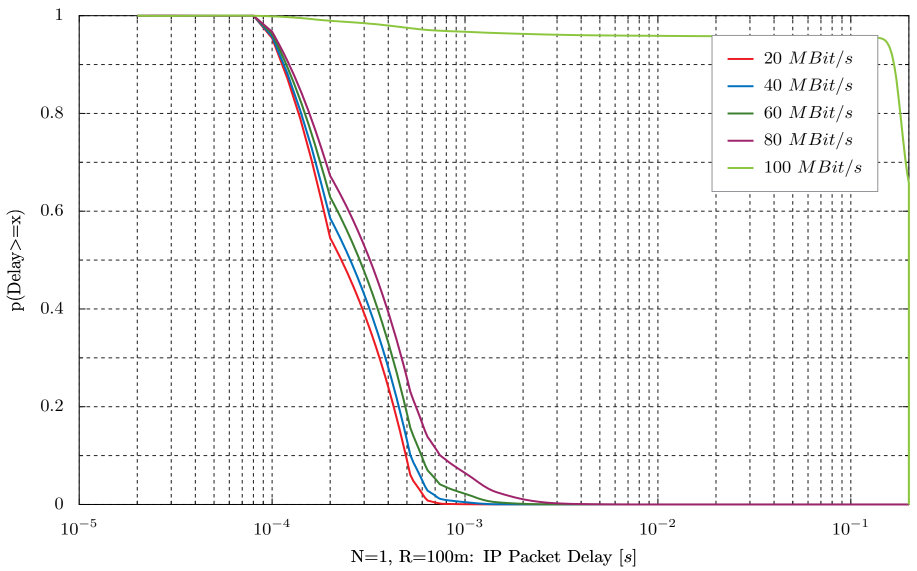


Figure E-11: CCDF of IP packet delay for cell size R=100m and varying offered traffic.

E.3.2.2 Two-Hop Singlecell

The objectives of this simulation run were to show that by means of using relay stations, the coverage range of the cell can be extended. To show the trade-off between cell capacity and range, another goal was to estimate the capacity of the REC, once again in the isolated cell case.

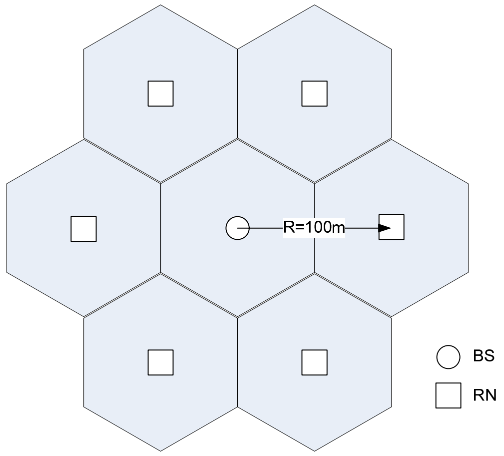
E.3.2.2.1 Scenario Description

The scenario setup is given by a single BS, supported by a tier of 6 RNs located at a distance of 100m, which, according to the investigations above, represents the maximum communication distance using the fixed settings of modulation and coding assumed. The subcells served by the BS and the RNs have a radius of ~60m. The resulting overall radius of the REC is ~150m, which represents a gain in range of 50% and a gain in coverage area of 125%.

The resource allocation in the relay enhanced cell is assumed to be coordinated by the base station in such a way that the full set of OFDMA subchannels is available at all nodes, but a certain subchannel may only be used by one node at the same time. Hence there exists no intra-REC interference.

Deploying such cells in a clustered fashion would require partitioning the available resources in such a way that only a subset of the available subchannels is available in a REC. The effect of reduced cell capacity has also been investigated (see Table E-5), but is not evaluated here.

Table E-5: Two-Hop Singlecell scenario



Cluster Size	Subchannels per REC
N=1	104
N=3	33
N=7	13

E.3.2.2.2 Results

The capacity of the isolated cell shows the same properties as in the single-hop case. The cell saturation throughput is reduced by 50% compared to the single hop case, which meets the expectations. Saturation behaviour is also similar to the single-hop case owing to the absence of traffic control functionality.

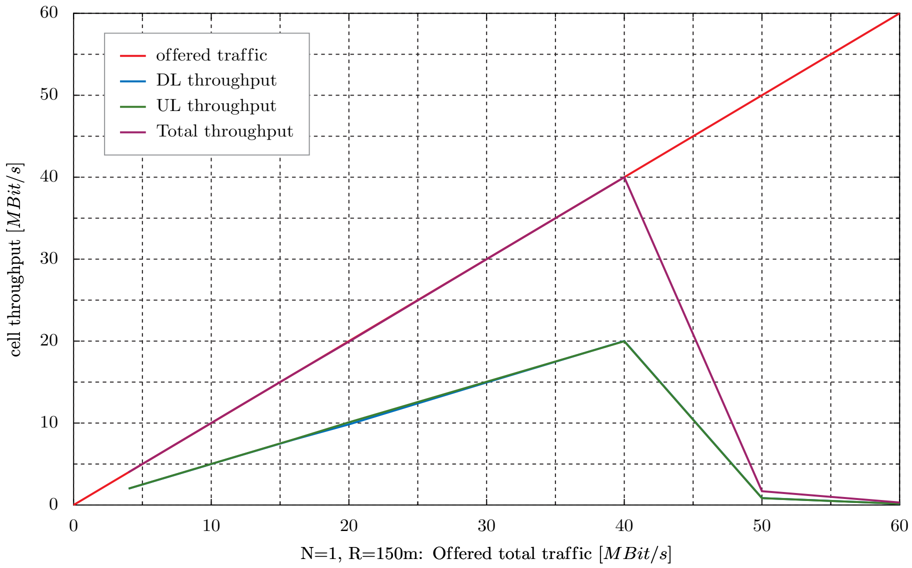


Figure E-12: UL, DL and total cell IP throughput vs. offered traffic.

The IP packet delay has roughly doubled compared to the single-hop case. It has to be noted that the simulation model does not account for the processing delay contributed by the FEC stages. Decoding and re-encoding at the RN can be expected to further add to the shown delays. Note that the curve for 60MBit/s offered traffic is not shown because owing to the congestion no packets have been successfully transmitted throughout the simulation.

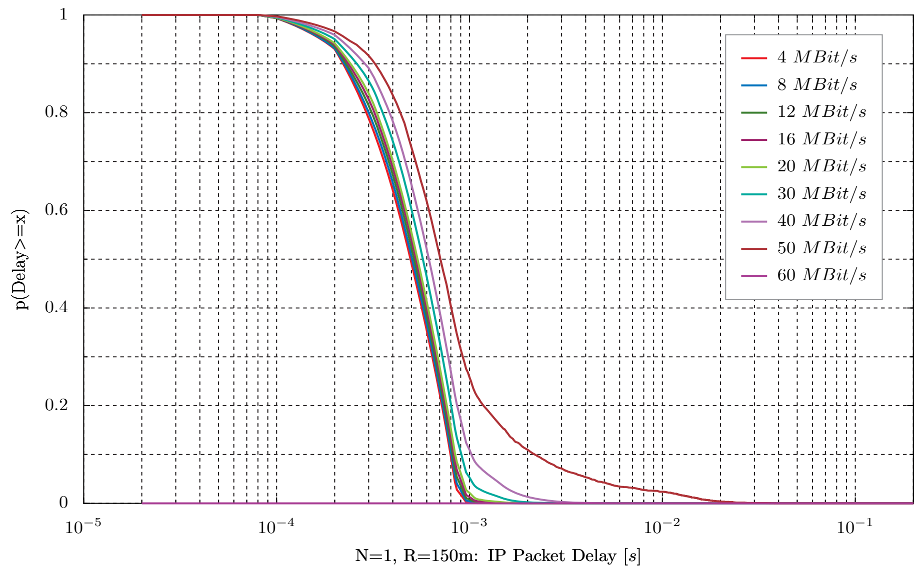


Figure E-13: CCDF of IP packet delay for cell size R=150m and varying offered traffic.

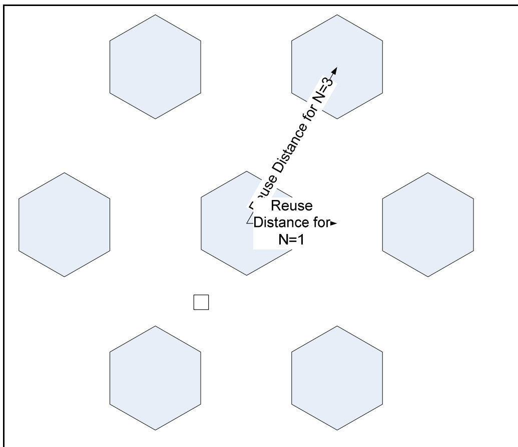
E.3.2.3 One-Hop Cluster

The objective of this simulation series was to investigate the capacity loss under co-channel interference respectively estimate the necessary re-use distance for proper operation of the system in a multi-cellular deployment. Another aim is to show the capacity reduction due to fixed partitioning of resources to multiple nodes in a cluster.

E.3.2.3.1 Scenario Description

The scenario consists of 7 cells, each having a radius R=100m. Depending on the cluster order, the BSs are separated by according reuse distances given by Table E-6. Also given there are the number of resulting OFDMA-subchannels available in each cell of the cluster. Statistical evaluation is performed in the center cell only to avoid border effects.

Table E-6: One-Hop cluster scenario



Cluster Size	Subchannels per Cell	Reuse dist.
N=1	104	100m
N=3	33	300m
N=4	26	346m
N=7	13	458m
N=12	8	560m

E.3.2.3.2 Results

The simulation results for the achieved cell IP throughput exhibit two different effects. The first effect is that of the co-channel interference, limiting the cell throughput in the case of N=1. For higher cluster oders, starting from N=3, the maximum throughput appears to be limited by the number of OFDMA-subchannels available. The saturation throughput gradually decreases with increasing cluster size.

Another visible effect is the loss of trunking gain, since the saturation throughput in the case N=3 is notably smaller than one third of the saturation capacity in the singlecell one-hop case with N=1.

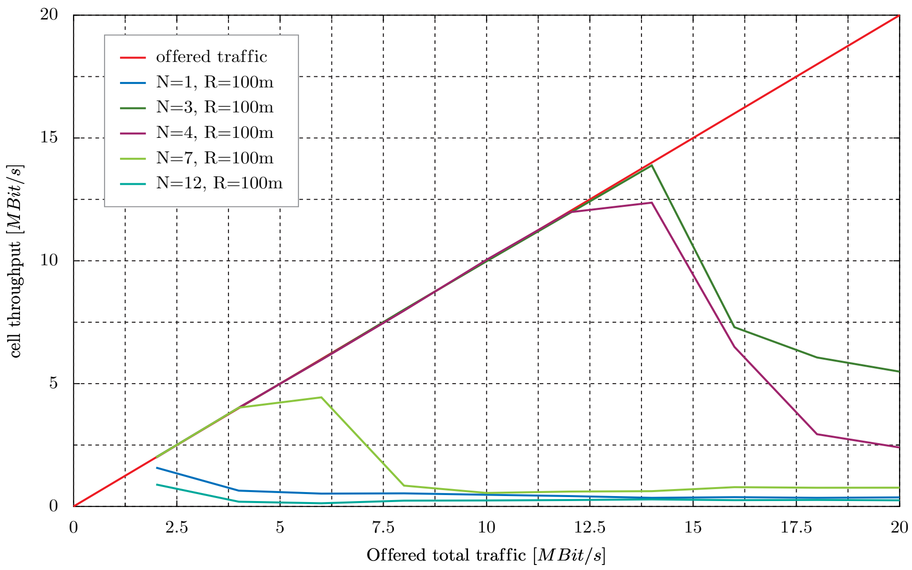


Figure E-14: total cell IP throughput vs. offered traffic for different cluster sizes.

The statement that for cluster sizes starting from N=3 the system seems to be limited by noise rather than interference is backed by the CDF of the SINR of the transmitted data packets, which shows no SINR gain for cluster sizes bigger than 3. At the same time, the curve for N=1 shows that at such tight re-use - without any inter-cell coordination, 90% of the users are in outage.

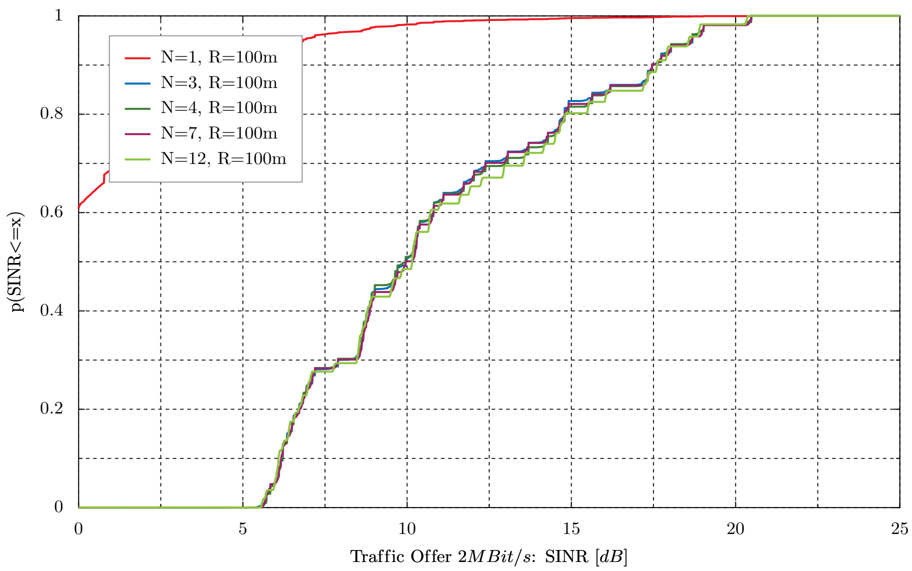


Figure E-15: CDF of SINR of received chunks for different cluster sizes.

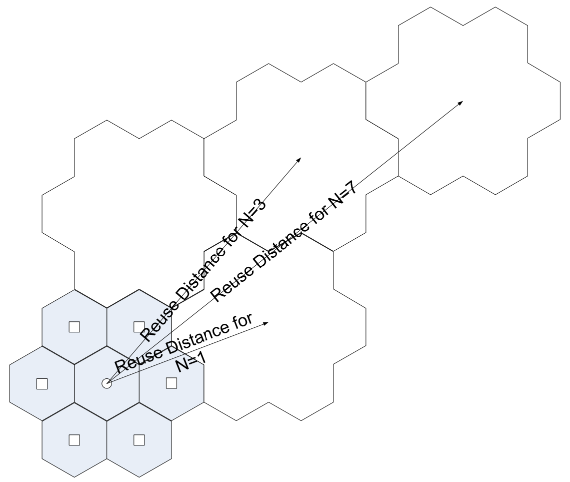
E.3.2.4 Two-Hop Reuse

The goal of this final simulation series is to investigate the capacity loss of RECs under co-channel interference in the multi-hop case and to determine the necessary re-use distance for multi-hop cells without coordination. All cells were assumed to have the full 100MHz BW, resp. 104 Subchannels, as opposed to the one-hop cluster investigation presented before.

E.3.2.4.1 Scenario Description

The scenario is composed out of 7 RECs which are placed as a tier of 6 interfering RECs around a central REC which is used for statistical evaluation. The RECs are identical to the ones in the Two-Hop singlecell investigation. Simulations have been performed for three different re-use distances. The distance between the central BSs in adjacent co-channel RECs can be obtained from Table E-7. The more important property is however the closest distance between co-channel nodes that can occur in such a setup, which is the min RN-RN distance, also to be obtained from Table E-7.

Table E-7: Two-Hop re-use scenario



Reuse Factor	Subchannels per REC	BS-BS dist.	Min RN-RN dist.
1	104	264m	100m
3	104	458m	264m
7	104	700m	519m

E.3.2.4.2 Results

The results from Section E.3.2.3 indicate that a reuse distance of 300m is necessary for un-interfered operation of co-channel cells. This is ensured only in the N=7 case. The IP throughput figures support this statement. While a re-use of N=1 leads to unacceptable interference and yields no throughput at all, N=3 still shows significantly reduced throughput compared to the isolated cell. Only for N=7 we find performance comparable to the isolated cell case.

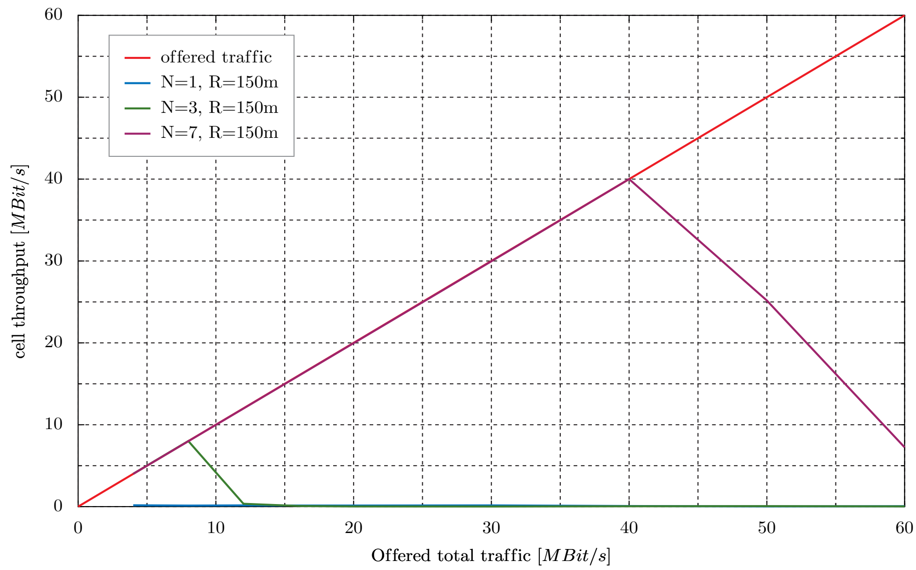


Figure E-16: total cell IP throughput vs. offered traffic for different reuse distances.

The IP packet delay, here shown for a traffic offer of 20MBit/s per REC shows that only for N=7, the delay has a reasonable upper bound, which is in around 10ms. For smaller reuse distances delay figures increase beyond tolerable measures.

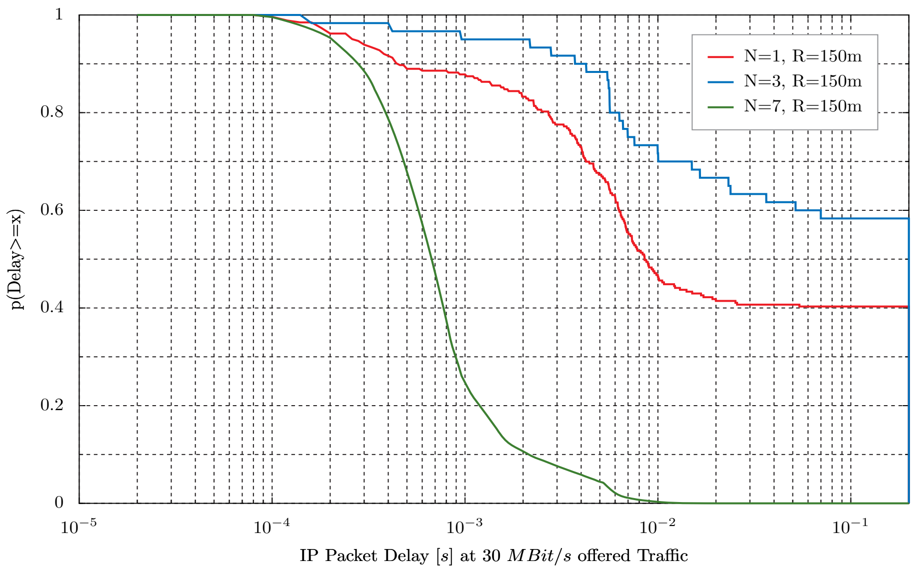


Figure E-17: CCDF of IP packet delay for different reuse distances

E.4 Coverage improvement through relay based deployment

E.4.1 Motivation

We are interested in the benefit that a repeater network might bring. In particular, we are interested in the improvement in coverage (i.e. throughput) that a repeater network would give to areas at the edge of the current cells that are currently limited by intercell interference. However, the repeaters themselves cause interference, and will consume capacity that could otherwise be used to carry traffic as we assumed that the backhaul from the RNs was in-band. Hence there is a cost associated with using a repeater network. We wish to explore whether the benefits of increased cell-edge coverage outweigh the additional interference and loss of capacity from using in-band backhaul.

E.4.2 Scenario Description

The metropolitan scenario is characterised at the macro cell layer by a regular hexagonal deployment topology. Each site is normally tri-sectored, with the nominal coverage area of each site resembling a clover-leaf shape. This can be seen in the reference deployment area used for the evaluation described in this section, which is shown in Figure E-18 below.

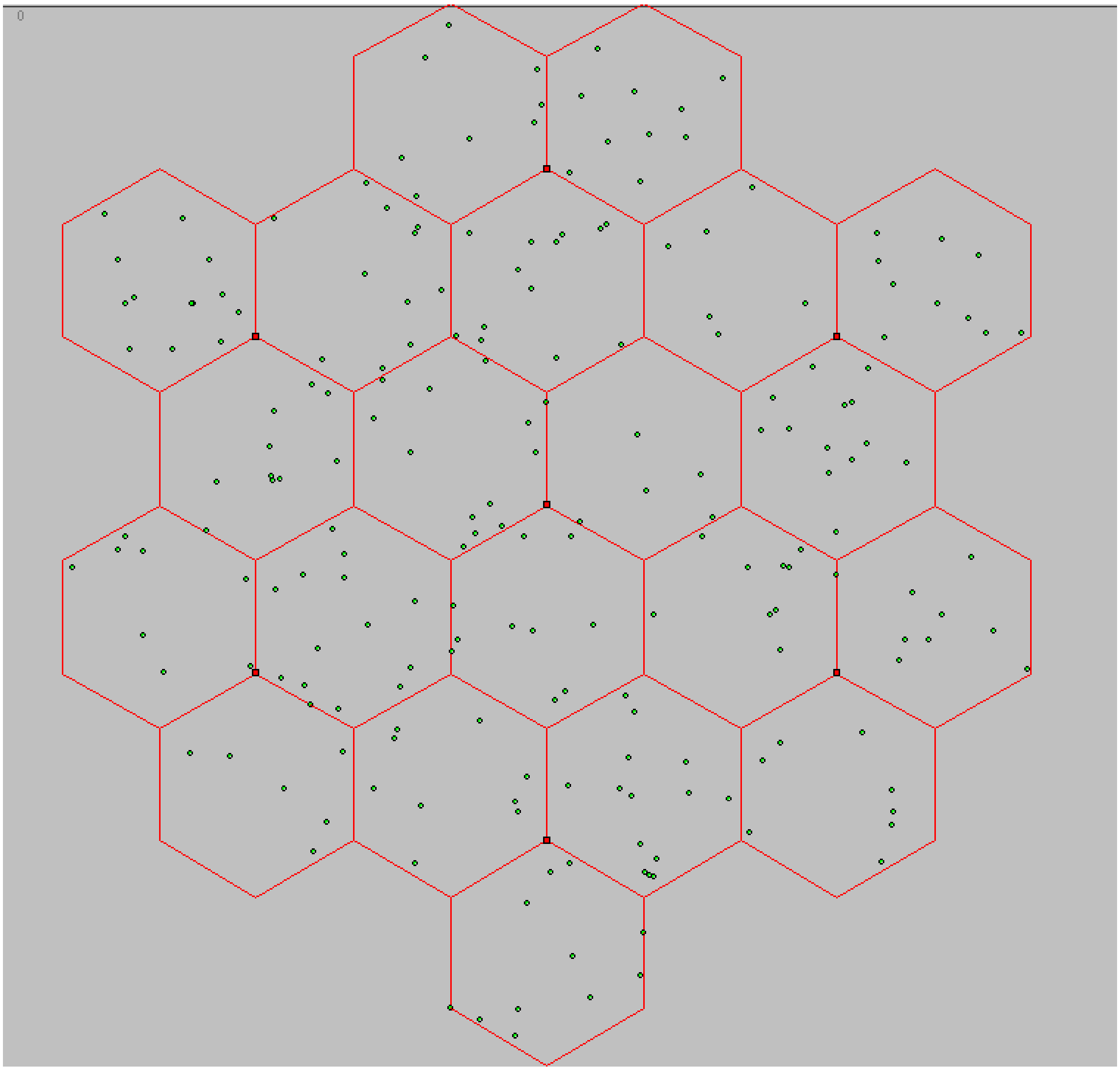


Figure E-18: Reference Deployment Topology used for the Metropolitan Scenario

Each red square represents a tri-sectored base-station site, with the nominal coverage area of each sector being represented by the adjacent hexagonal area. The cell radius was set to 200 m. One tier of surrounding cell sites was modelled, giving a total of 7 sites, and thus 21 base-stations. Each BS was assumed to have a maximum transmit power of 40 dBm.

The green circles represent the deployed user terminals. A total of 210 UTs were randomly deployed over the whole deployment area, giving an average of 10 UTs per sector. Note that the random nature of this deployment means that the number of UTs served by each sector was not equal, with some sectors more heavily loaded than others. All UTs are stationary, even though they are modelled as having a non-zero speed for the fast fading calculations.

The following propagation environment was assumed:

Table E-8: Propagation Assumptions used for the Metropolitan Scenario

Parameter	Value	Units	Reference
Carrier Frequency	5.0	GHz	[WIND52]
Pathloss Model	$28.3 \times \log_{10}(d) + 61.5$	dB	[WIND52]
Shadow Fading Std Dev	5.7	dB	[WIND52]
Minimum Coupling Loss	70	dB	[WIND32]
Fast Fading Model	Urban Macro SISO		[Baum04]
UT Speed	3.0	km/h	[WIND32]

The base-station was modelled as having a single antenna with a boresight gain of 16 dBi, a 3 dB beamwidth of 90º and a Gaussian shape around the boresight angle. The linear gain of the antenna is thus given by the equation:

$$g(\theta) = 39.8 \cdot e^{\left(\frac{-\theta^2}{2921.457}\right)} \tag{4}$$

where θ is the azimuth in degrees. Only two dimensional propagation modelling was used, and hence the variation in antenna gain with elevation was not modelled. Each UT was assumed to have a single omni-directional antenna with a gain of 0 dBi.

A randomised traffic arrival model was used, with packets arriving at each BS being scheduled to the associated UTs on a round-robin basis. UTs were associated to the BS from which they received the strongest median signal strength, taking into account shadow fading and antenna gains. Only downlink traffic was modelled, with the arriving traffic per BS being set to 12.9 Mb/s. When a packet was received with errors at a RN or UT, it was retransmitted unchanged by the BS or RN at the next scheduling opportunity (Type 1 H-ARQ). No combining of re-transmissions at the RN or UT was modelled. The simulation logged only the number of correctly received packets when computing the achieved throughputs.

For the purposes of the reference simulations, the WINNER air interface was configured as follows:

Table E-9: WINNER Air Interface Configuration used for the Metropolitan Scenario

Parameter	Value	Units	Reference
Duplexing Scheme	FDD		[WIND76]
Multiplexing Scheme	TDM		[WIND76]
Modulation Scheme	OFDM / QPSK		[WIND76]
Number of Subcarriers	512		[WIND76]
Number of Data Subcarriers	416		[WIND76]
Subcarrier spacing	50	kHz	[WIND32]
Symbol Period	22.5	µs	[WIND32]
Coding Rate	½		[WIND32]
ARQ	Type 1 H-ARQ		[WIND32]
Power Control	None		
Signalling Overheads	Not modelled		
Handover	Not modelled		
UT Noise Figure	9	dB	[WIND32]

The output of the reference simulation was primarily the achieved mean BS and UT throughputs. These results are reported in Section E.4.3 below. Note that, due to the simplified packet arrival model, the simulator could not estimate packet delay (as packets are not queued at the BSs).

The repeater network considered is shown in Figure E-19 below, where the repeater sites are denoted by the blue squares. One repeater per BS was modelled, with that repeater being located half-way between the associated BS and its nearest neighbour. Note that, due to the positioning of the directional antennas at the BSs, the RNs are associated (i.e. served by) the BS within whose nominal hexagonal coverage area they are located. It is for further investigation as to whether alternative locations for the RNs (either further from or closer to the serving BS) would be more optimum.

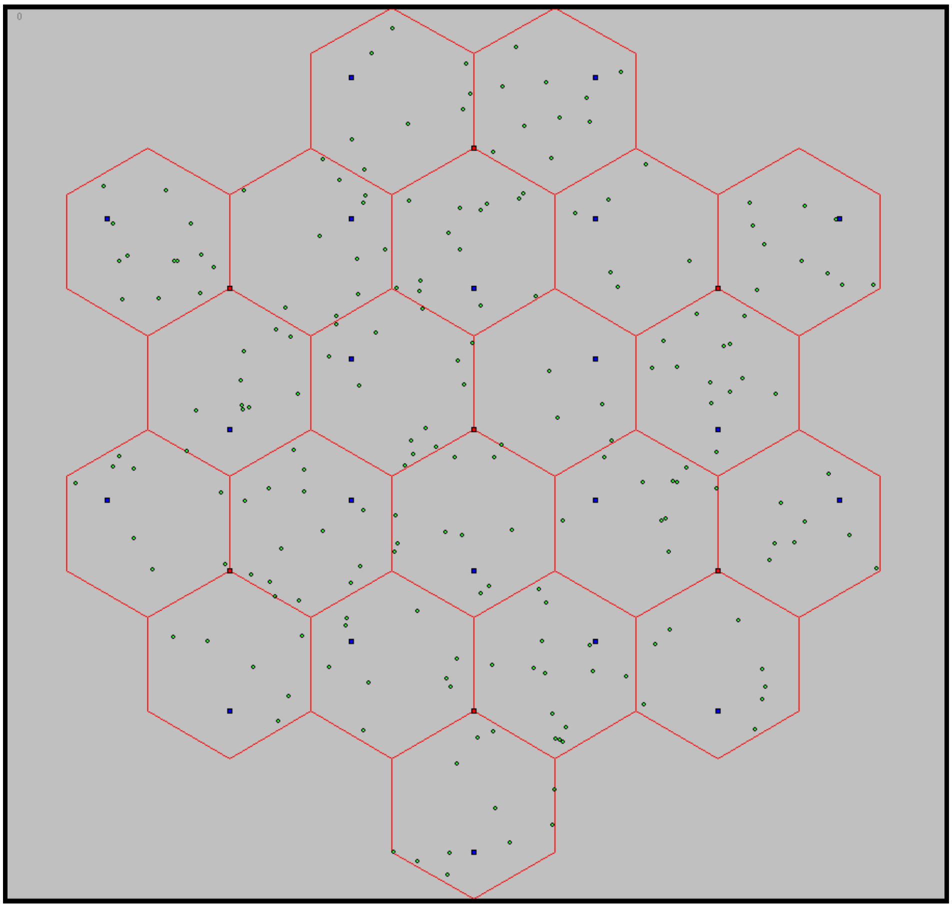


Figure E-19: Reference Deployment Topology for Metropolitan Scenario

Each RN was assumed to have a transmit power of 40 dBm (the same as the BS) and use a single omni-directional antenna with a gain of 10 dBi. This antenna is used for both receiving packets from the associated BS, and for forwarding these packets on to any associated UTs. Although a separate dedicated directional antenna could be used for the link to the associated BS, it was decided to investigate the use of a common omni-directional antenna as this is a cheaper solution, both in terms of hardware and site installation costs. It is for further investigation as to whether a separate dedicated antenna for the backhaul would give sufficient benefit to justify the additional cost.

Because RNs cannot transmit a packet to an associated UT in the same timeslot as they are receiving a packet from the associated BS, the scheduling algorithm was designed to avoid scheduling to the RN in a timeslot that would be used by the RN to forward a packet it had previously received from the BS. However, the BS was allowed to schedule to one of its own associated UTs in this timeslot, and hence the BS and associated RN were able to transmit simultaneously.

As for the reference case above, the primary output of the repeater network simulations was the achieved mean BS and UT throughputs. The throughput for a given BS included successful packets sent by its associated RN to a UT, but excluded packets sent between the BS and RN. As with the reference case, it was not possible for the simulator to estimate packet delay.

E.4.3 Results

The scenario being modelled is described in Section E.4.2 above. It consists of a reference scenario, where no repeaters are used, and a modified scenario where a number of repeaters have been deployed. The goal is to establish whether the benefits of increased cell-edge coverage offered by the repeaters outweigh the additional interference and loss of capacity from using in-band backhaul between the RNs and the BSs.

We can see the increased coverage offered by the repeater network if we examine and compare the CDFs of the received signal strength at the UTs in the scenarios with and without repeaters. This is shown in Figure E-20 below.

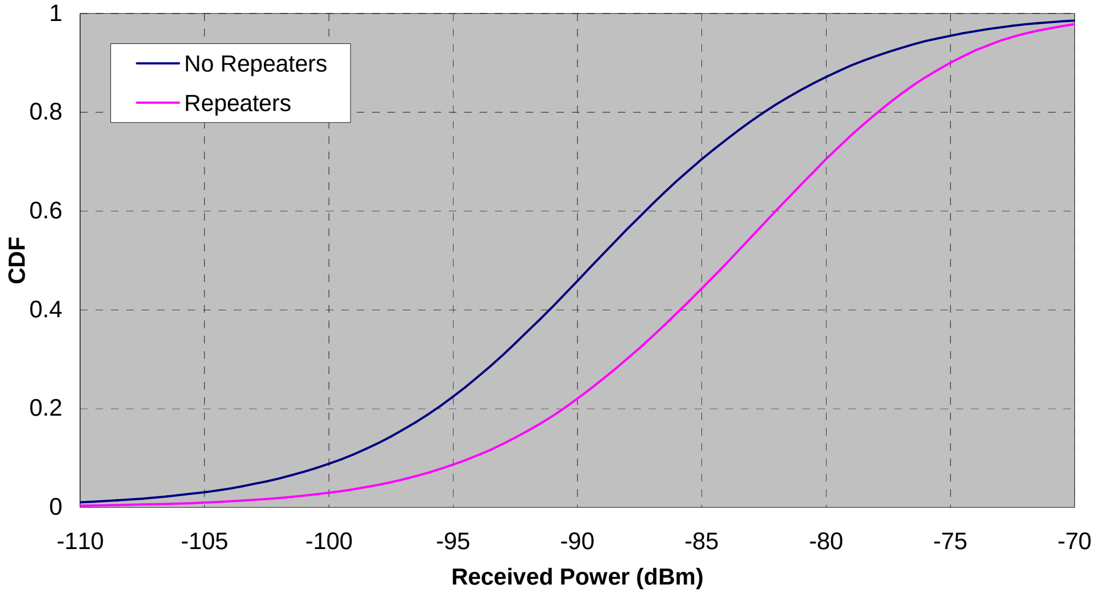


Figure E-20: Comparison of Received Subcarrier Signal Power

It can be seen that the repeater network increases the subcarrier signal power received at the UTs by around 5 dB. However, as noted above, the repeater network will also generate additional interference. This can be seen if we examine the received subcarrier interference powers for the two scenarios. This is shown in Figure E-21 below.

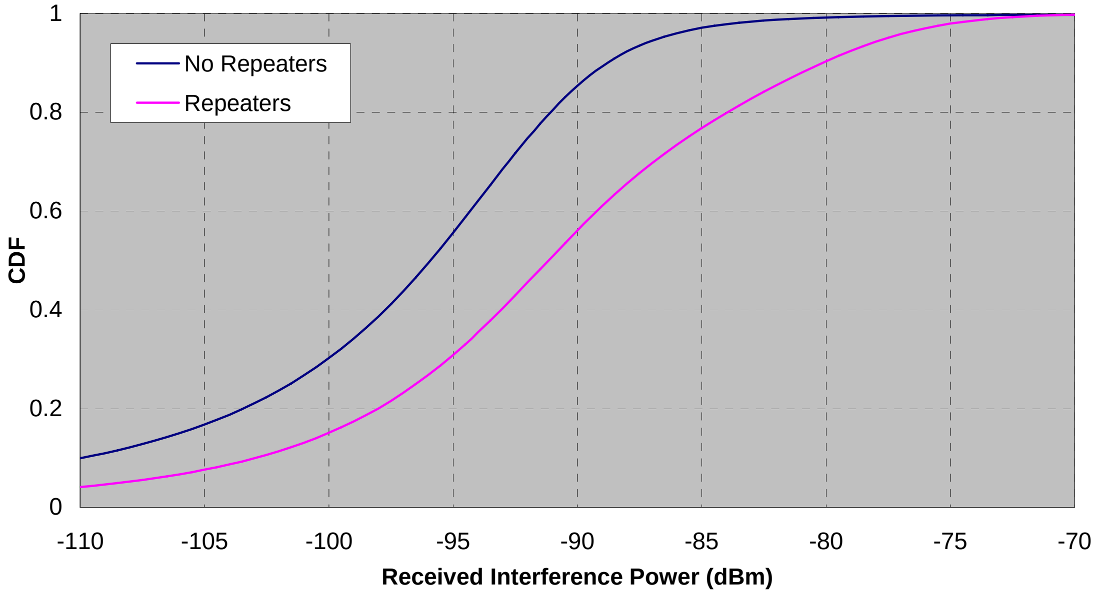


Figure E-21: Comparison of Received Subcarrier Interference Power

It can be seen that the repeater network increases the interference received by a large minority of UTs by more than the expected 5 dB increase in received signal power. This would be expected to have an adverse effect on the received SINR, which is indeed what is seen if we compare this metric for the two scenarios. The measure of SINR used here is effective SINR, as described in [Alexiou04].

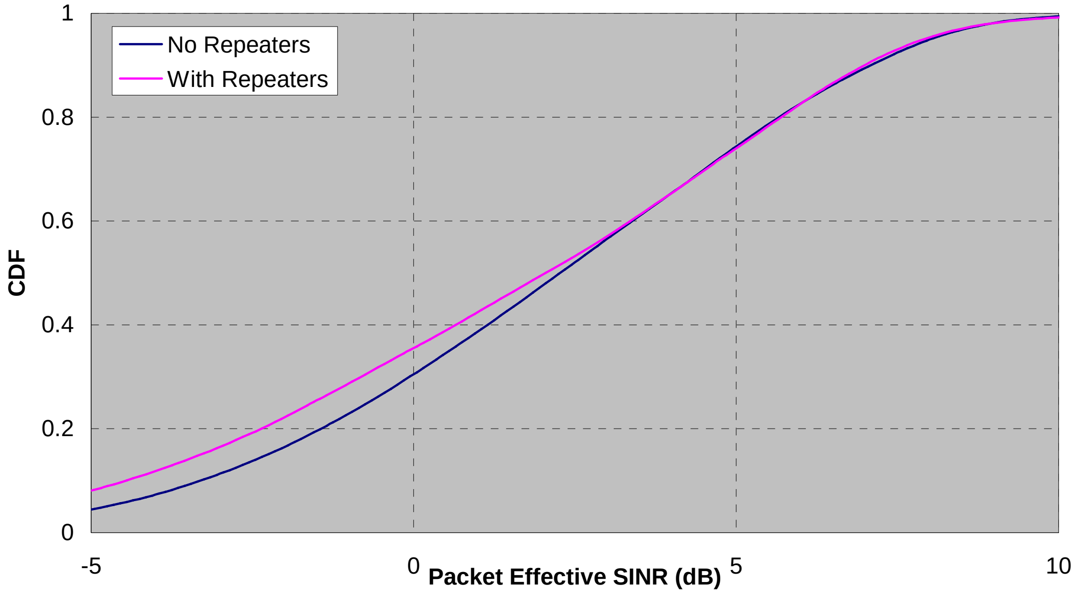


Figure E-22: Comparison of Packet Effective SINR

It can be seen that instances of low effective SINR occur more frequently when the repeater network is used, whereas the frequency of instances of high effective SINR is largely unchanged. This would suggest that the repeater network does not provide an overall gain in throughput, due to both the reduced effective SINR and the loss of capacity in serving the repeater network. This can be seen by comparing the aggregate throughputs of the BSs in both scenarios. This is shown in Figure E-23 below. Note that, for the scenario where repeaters are used, the aggregate throughput of a BS is the sum of throughputs to UTs directly served by the BS plus the throughputs to UTs served by the associated RN.

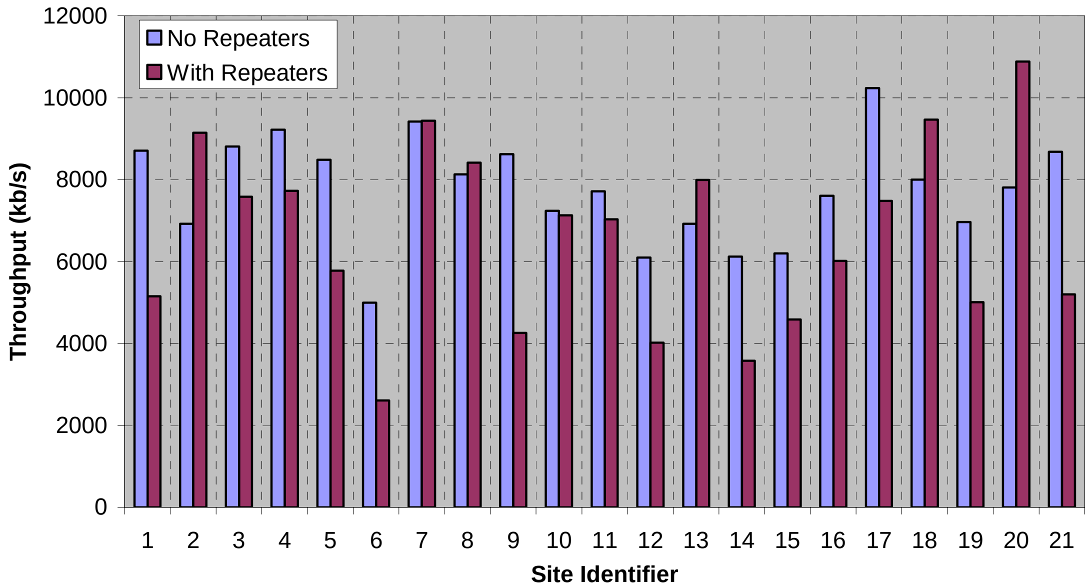


Figure E-23: Comparison of Base-Station Aggregate Throughputs

It can be seen that some BSs experience increased throughput, whereas others experience reduced throughput. Typically, however, the throughput decreases. The average throughput per BS for the scenario with no repeaters was actually 7.8 Mb/s, whereas the average throughput when repeaters were deployed was only 6.6 Mb/s, a reduction of 15 %. This would tend to suggest that, overall, it would not be beneficial to deploy repeaters using in-band backhaul in this type of scenario in order to improve the overall aggregate throughput. Note that, interestingly, the reduction in total throughput in the network with repeater nodes is less than the mean data rate required to support the links to the RNs. Hence the overall data throughput of the network has been increased by the presence of the RNs, but not by enough to compensate for the use of in-band backhaul.

However, note that one of the motivations for deploying repeaters was to increase the coverage (i.e. throughput) to UTs located at the edge of the cells, which would otherwise achieve only a low throughput due to their typically worse interference situation. Hence it may be that the use of the repeater network reduces the variance of the throughputs achieved by different UTs by reducing the number of UTs achieving either very high or very low throughputs. This is examined in Figure E-24, which compares the CDFs of achieved UT throughputs for the two scenarios.

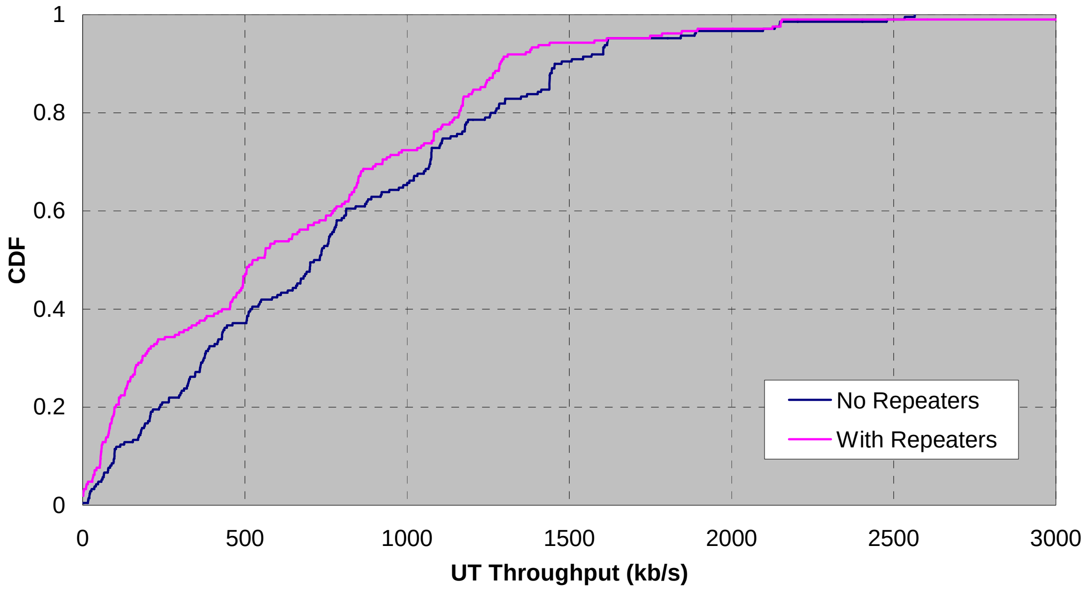


Figure E-24: Comparison of UT Throughputs

It can be seen that this is not the case, as there is no overall reduction in the variance of the UT throughputs, only a reduction in the mean. The mean UT throughput for the scenario with no repeaters present was 776 kb/s, whereas the mean throughput when repeaters were present was only 660 kb/s. This reduction is not uniform for all UTs, as can be seen from Figure E-25 below. Most UTs experience either a small increase or decrease in their achieved throughputs, with only a small minority experiencing large increases or decreases. Typically, UTs that were at the edge of a cell do experience an increase in throughput when repeaters are deployed, but this is offset by the reduction in throughputs to UTs that are nearer the centre of the cell, and hence now receive significant interference from either the BS or the RN. It may be that a more intelligent scheduling algorithm would prevent this.

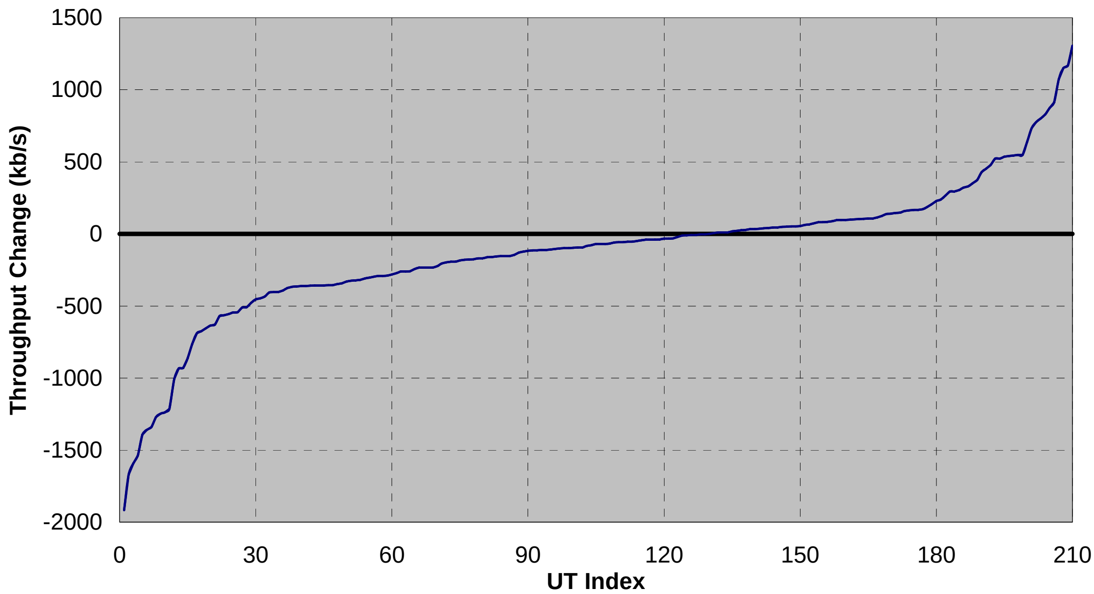


Figure E-25: Comparison of Change in UT Throughputs

Clearly, the relay nodes benefit some UTs, mainly those near the edge of the cell, but are detrimental to others. The net effect in the scenario considered, however, is a reduction in system throughput. Hence it is concluded that, for this scenario, the benefits of improved coverage offered by the relay nodes are outweighed by the loss of capacity required for in-band backhaul and the increased interference generated by the relay nodes.

It may be that the benefits provided by the relay nodes can be increased by considering modifications to the above scenario. Further simulations are required to determine the effect of enhancements such as dedicated antennas for the BS to RN link (either at the BS or RN or both), modified positioning of the RNs (either further from or closer to the BS), reduced BS and RN transmit powers and improved signal processing techniques, such as co-operative relaying.

E.5 Urban macro cell deployment enhanced by low transmission power fixed relay nodes

E.5.1 Motivation / Introduction

The goal of this study is to find out how close to cell border a low transmission power relay node can effectively operate, before inter-cell interference and range problems become dominant. Moreover, the study provides initial results on the delay induced by introducing two-hop communications. The throughput values are also shown, but the main purpose is to illustrate the performance of a relay node, when the distance between relay node and base station is varied. Although given throughput figures could also be interpreted as a worst case performance assessment of a deployment with relays. As will be explained later link adaptation has not been considered for this investigation. It should be kept in mind that enabling link adaptation can change the results significantly.

E.5.2 Scenario Description

The scenario studied is a metropolitan area macro-cell layer, where decode-and-forward relays based on WINNER relay concept are introduced inside macro cells. As a reference, the same scenario is considered without the relay nodes. Deployment is based on omni-directional antennas at each base station. The deployment scenario is illustrated in Figure E-26 (relay nodes are marked with a star):

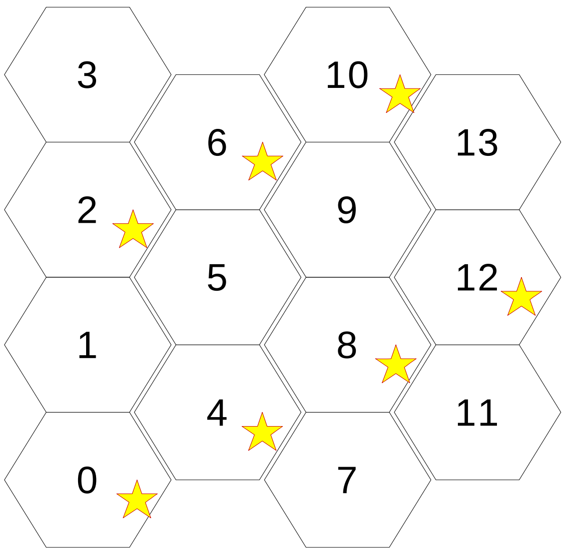


Figure E-26 Deployment scenario

Cell ranges are set to 133m. The scenario consists of 13 cells, and the centre most cells have been used for data gathering.

Transmission powers of base stations were set to 43 dBm, and for user terminals to 23 dBm. Relay node deployment was intended to be based on low transmission power, small and cost effective nodes. Although the equipment was assumed to be similar to UTs, a constant power supply slightly easier thermal management lead to less constrained transmission power for relay nodes. Thus, value of 30 dBm was used for relay nodes. The propagation modelling employed is summarized in Table E-10.

Table E-10 Propagation parameters

Parameter	Value	Units
Carrier Frequency	5.0	GHz
Path Loss Model	$28.3 \times \log_{10}(d) + 61.5$	dB
Shadow Fading Std Dev	5.7	dB
Minimum Coupling Loss	70	dB
Fast Fading Model	Urban Macro SISO	
UT Speed	3.0	km/h

The number of UTs in the network was 60. Each UT was moving at a speed of 3 km/h. Higher mobility was left for further study.

As traffic model was a Constant Bit Rate model was used. The offered load was 53 Mb/s/ user, in 0.5 Mb files. Round robin scheduling was implemented in base-stations and relay nodes. Relay nodes scheduled traffic independently inside the assigned set of resources. The resource partitioning scheme gave a static part of the frame (1/3) to each relay node for both uplink and downlink traffic. QoS handling was not modelled.

The air interface follows WINNER TDD mode simulation assumptions, and employed parameters are shown in Table E-11. Modulation used was 16-QAM, with coding rate ½. The link adaptation was not modelled, since it would further complicate interpreting the results. However, absence of link adaptation also leaves out significant potential of gains from relay node deployment rising from hop-by-hop link adaptation. One needs to understand better the link adaptation algorithms, and potential problems they cause, therefore combining the link adaptation to studied scenario was left for further study. The simulator models transmission and reception of allocation information, and random access requests.

Table E-11 Physical layer parameters

Parameter	Value	Units
Duplexing Scheme	TDD	
Multiplexing Scheme	TDMA	
Modulation Scheme	OFDM / 16-QAM	
Number of Subcarriers	2048	
Number of Data Subcarriers	1664	
Subcarrier spacing	48.828	kHz
Symbol Period	22.5	µs
Coding Rate	½	
ARQ	Type 1 H-ARQ, selective repeat	
Power Control	Constant power	
Signalling Overheads	BCH + allocation information (x %), RAC (x %)	
Handover	Path loss based, lossless	

E.5.3 Relay node positioning

Figure E-27 shows user throughputs for deployment scenarios with relay nodes being 40, 60 and 80 meters from the base station. The throughput is measured on top of layer 2 in the receiver. Data has been gathered only from connections that were routed through a relay station. The throughputs for relay distances of 40 and 60 meters are very similar, but when the RN is moved to a distance of 80 metres from BS, there is a significant drop in the achieved throughput.

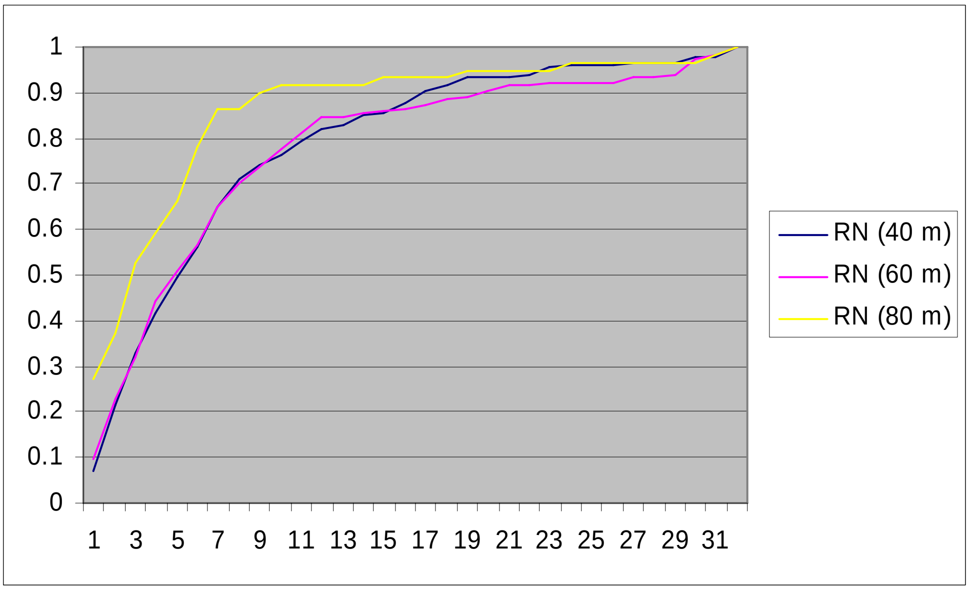


Figure E-27 CDF of downlink user throughputs for RNs deployed at three distances from BS (Mb/s)

Figure E-28 shows delay performance of the system. Again, data has been gathered only from connection going through a relay station, and delays have been measured at the top of layer 2 in the receiver. The delay performance of relay based deployments with RN located at 40 and 60 metres from the BS are again very similar. However, as distance is increased, the packet error probability increases, and number of re-transmissions of the RN-BS hop start to be more of a problem. Thus, there seems to be a clear drop on delay performance when the relay is moved towards the border of the cell. Differences in the order of magnitude of the delay figures presented here and in Section E.3.2 are owing to a number of factors that have been modelled differently: (i) the simulations presented here assume the frame structure defined in the calibration case [WIND32], which is approx. 1 order of magnitude higher than that of the WINNER MAC frame defined by [WINXWPMAC]; (ii) the traffic model in this case assumed a fixed IP packet size of 12000bit while Section E.3.2 assumed a maximum packet size of 12000bit; (iii) and most important: the overall traffic offer per cell in the simulations presented here was much higher than that assumed in Section E.3.2, which means that the probability of congestion is dramatically increased. This different point of operation of the system consequently leads to much higher packet delays, as visible from Figure E-28 and Figure E-30.

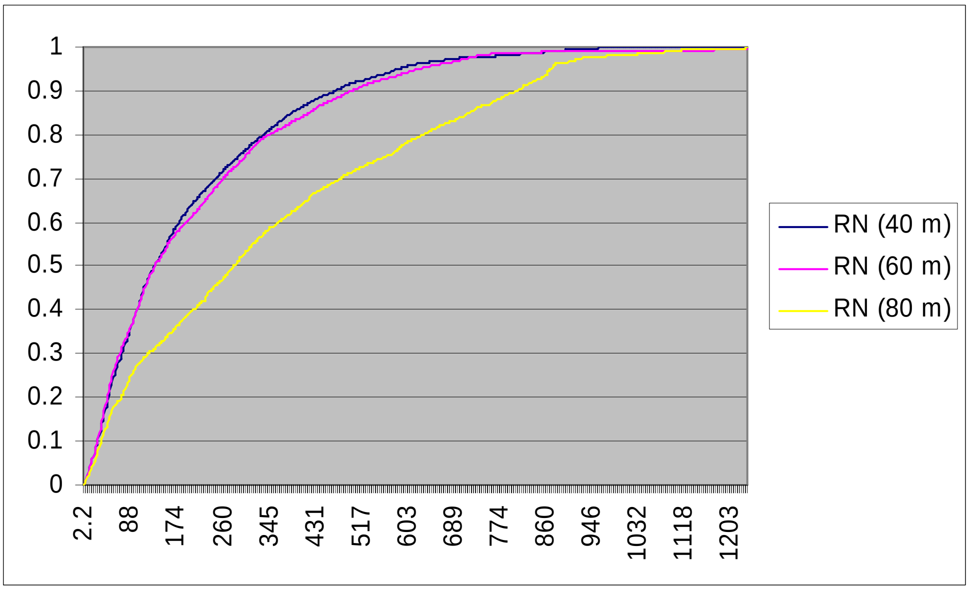


Figure E-28 CDF of downlink packet delays for RNs deployed at three distances from BS (ms)

E.5.4 Comparison of performance with and without relay nodes

In the following comparison of relay based deployment and conventional single-hop deployment, the distance between relay node and base station was set to 60 m. In Figure E-29 the results of relay based deployment scenario have been divided into two groups. The blue line shows the throughput for connections, where UTs are attached to cells without RN. Magenta shows the throughput for connections occurring in BSs which have RN deployed. The low mobility used in scenario makes this kind of division possible. In addition, results are shown for the subset of connections, which were using RN, and for the reference, results from the conventional single-hop deployment.

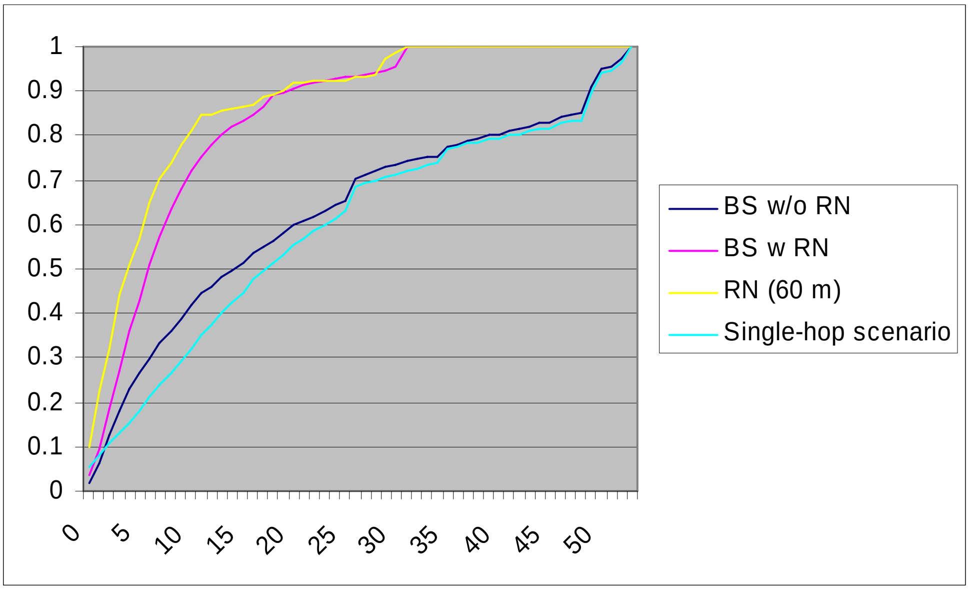


Figure E-29 CDF of downlink user throughputs (Mb/s)

The user throughput of BSs having RN deployed is rather similar to subset of connections going through a relay node. On the other hand, the user throughputs in the cells not having RN are close to single-hop reference scenario.

The packet delay results in Figure E-30 for relay based deployment have been divided into two groups: blue line shows connections going through RN, and magenta line shows results for single-hop connections. The reference single-hop results are shown by yellow line.

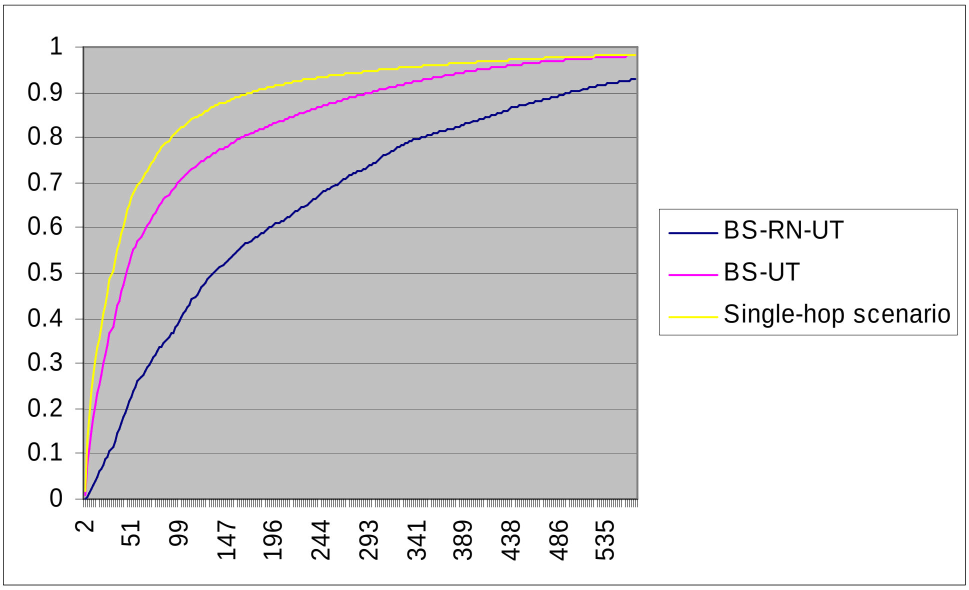


Figure E-30 CDF of downlink packet delays (ms)

It can be concluded that the single hop deployment offers better performance, especially the maximum throughputs are significantly better. However, the similarity of results for total throughput of BS having a RN, and the connections actually going through a RN indicates, that the throughput performance difference is mainly due to resource partitioning used, where 1/3 of the frame is always allocated to the relay node. The difference in delay performance is expected effect of decoding and forwarding the packets in a RN. Moreover, the resource partitioning contributes to worse delay performance for RN based scenarios. As discussed above, the potential gains from hop-by-hop link adaptation are no taken into account.

In order to illustrate the potential advantages of relay nodes, Figure E-31 shows the SINR values measured at UT for different scenarios. The data has been gathered from all connections completed during the simulation. There is a clear improvement of SINR values for the relay based deployments, which has not been utilized in current simulation setup.

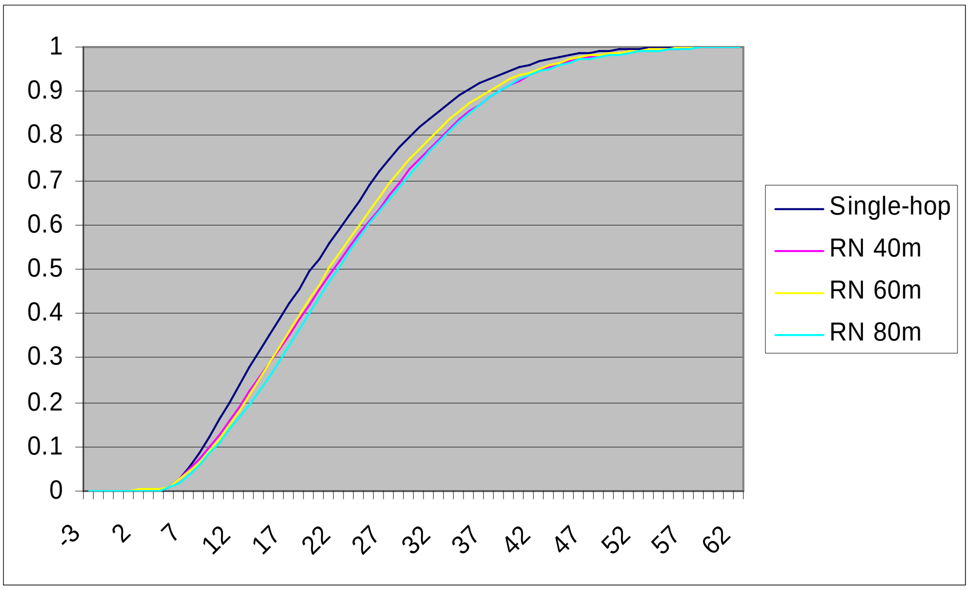


Figure E-31 CDF of SINR values for different deployments (dB)

E.6 Evaluation of Relay Cyclic Delay Diversity

E.6.1 Motivation / Introduction

The goal of the study is to evaluate the system and user throughput of the relay cyclic delay diversity (RCDD) concept in a two hops radio network and compared to a one hop system. Note that the RCDD concept is described in cooperative diversity section. The following related aspects are investigated:

- Max-SINR scheduler in combination with Relay cyclic delay diversity.
- The placement of the RN and the RN selection mechanism relay for RCDD
- Interference management for the RCDD in case of a 2-hop.system.
- The system and user throughput gain of RCDD

The evaluation is made via system simulator that focuses on the physical layer, radio networks deployment and algorithms (such as link adaptation and scheduling). Whereas the higher layer signalling is not considered, the extended spatial channel model is considered.

The simulations were conducted by mean of a dynamic system level simulator. The radio network cell plan is simulated over a certain number of snapshots. Each snapshot consists of a large number of super-frames which itself is composed of a certain number of radio frame. Each radio frame consists of a number of OFDM symbol. Based of a desired radio network planing (Radio elements positions, antenna configurations, cell radius etc..) the cell plan is created and remains unchanged over the whole simulation time. After, for each super-frame new users are created, placed in the cell plan and attached to an BS. On the frame level, first the users will moved according to the chosen mobility model then the propagation channel conditions will be generated (i.e. path loss, slow and fast fading). Thereafter the traffic model will generate the data packets for all active users. Then, the scheduling will decide the users to be scheduled. The signal to noise ration is computed afterward. In case of an adaptive system, the users' modulation and coding scheme will be decided by the link adaption. The link to system interface will allow to compute the block error rate before the correctly received bits can be known. This process will be repeated until the desired number of snapshots is reached.

E.6.2 Scenario Description

The simulated cell plans consisted of 7 sites each site is tri-sectored with a regular hexagonal deployment topology as shown in Figure E-18. The BS points which are represented by red circles that are placed at the intersection of three hexagons so each will be considered as a sector. The relay nodes are placed in an arc of circle which radius is relative to the cell radius and has the BS as origin. As seen from Figure E-18, six RNs, represented by the star sign, per sector were assumed. Note only two of then will be relaying the signal to the UT.

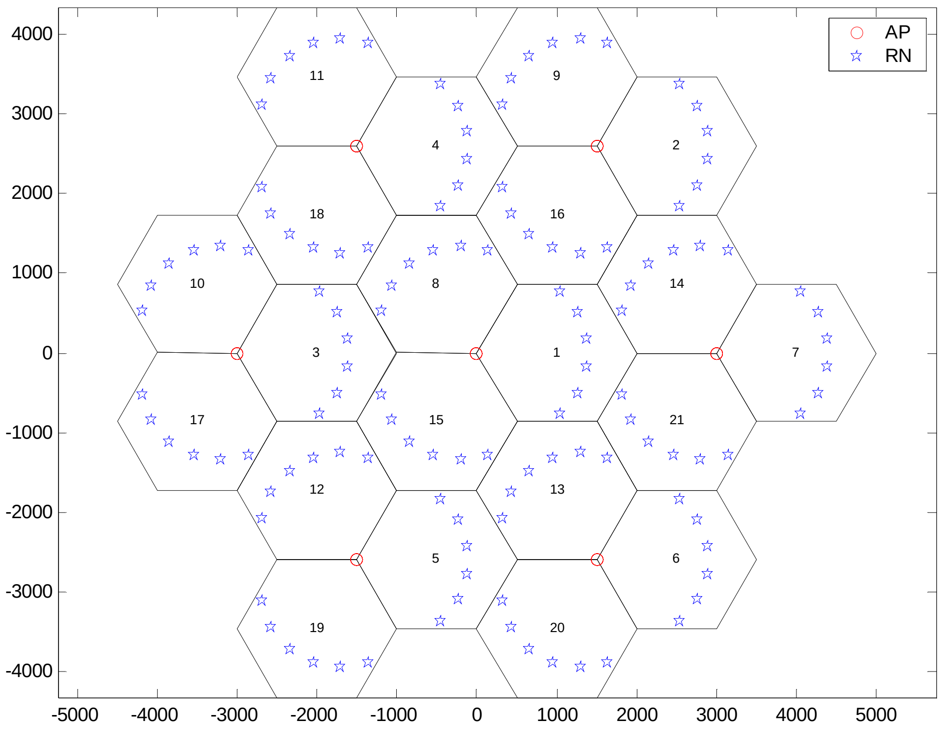


Figure E-32: Deployment Topology used for RCDD concept.

An average number of 10 UT per sector is generated. A simple data traffic model is used as the interest is in the relative performance of the simulated schemes and not in the traffic modelling performance per se. Generated users have full buffers, ready to transmit when they are scheduled.

The main parameters of the simulated air interface are as follows:

Table E-12: Air Interface Parameters

Parameter	Value	Units/Type
Multiplexing Scheme	TDM	
Modulation Type	[4 16 64 128 256 512]	QAM
Number of Subcarriers	512	
Number of Data Subcarriers	416	
Sub-carrier spacing	39.062	kHz
Symbol Period	28.8	µs
Frame Length	12	OFDM symbol
Super Frame length	2	Frame
Coding Rate	[1/3 1/2 2/3 8/9]	
Power Control	None	
Signalling Overheads	Not modelled	
Handover	Hard	
Scheduling	Max- SINR	

The most relevant simulation parameters are also summarized in the Table below:

Table E-13: Simulation Parameters.

Parameter	Value	Units/Type	Additional Info
Number Of Sites	7		
Number Sectors Per Site	3		
Wrap Around	Yes		Modelled
BS power	43	dBm	
RN power	37	dBm	
BS Antenna	14	dBi	Tri-Sector
RN Antenna	10	dBi	Omin
UT Antenna	0	dBi	Omni
UT Speed	3	m/s	
UT Noise Figure	9	dB	
Channel Estimation Error	10	dB	

As it was stated previously, the main objective is to compare a 1 hop OFDM system with a 2 two hops OFDM RCDD system. The latter system consists of a two transmission phases (i.e. hops): in the first hop the BS will transmit the OFDM symbol, while the RN and the UT receives. In the second hop the RNs belonging to the same BS will act a distributed antenna system. On each of the RNs'antennas a random cyclic shift is applied to the received OFDM symbol on the first phase before retransmission.

E.6.3 Results

The cdf of the SINR of the 1 hop and 2 hops system for 2 various cell radii are shown in Figure E-33. The proposed method (i.e. RCDD) yields an impressive 15 dB SINR gain for the cdf median value. The cdf for the received interference and desired signal power are shown respectively in Figure E-34 and Figure E-35. While the interference of RCDD increased slightly compared to a 1-hop system, the received power improves greatly due to the gain in terms of path and the additional diversity provides by the cyclic shift that was exploited by the Max-SINR scheduler.

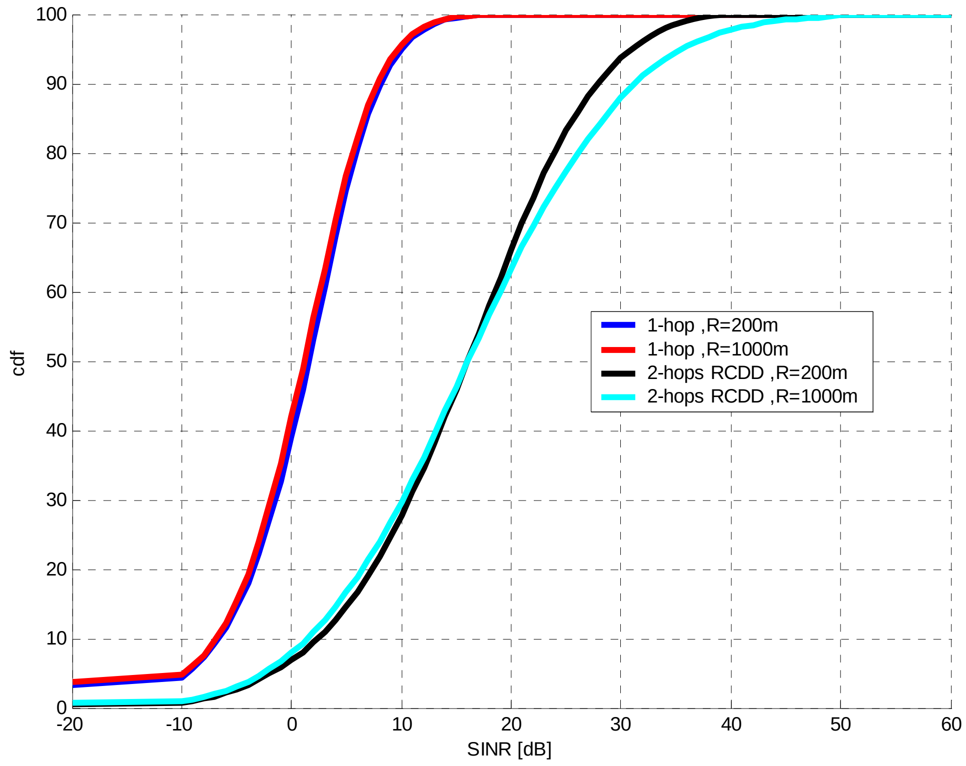


Figure E-33: cdf of the received SINR of the 1 hop and 2 hops schemes.

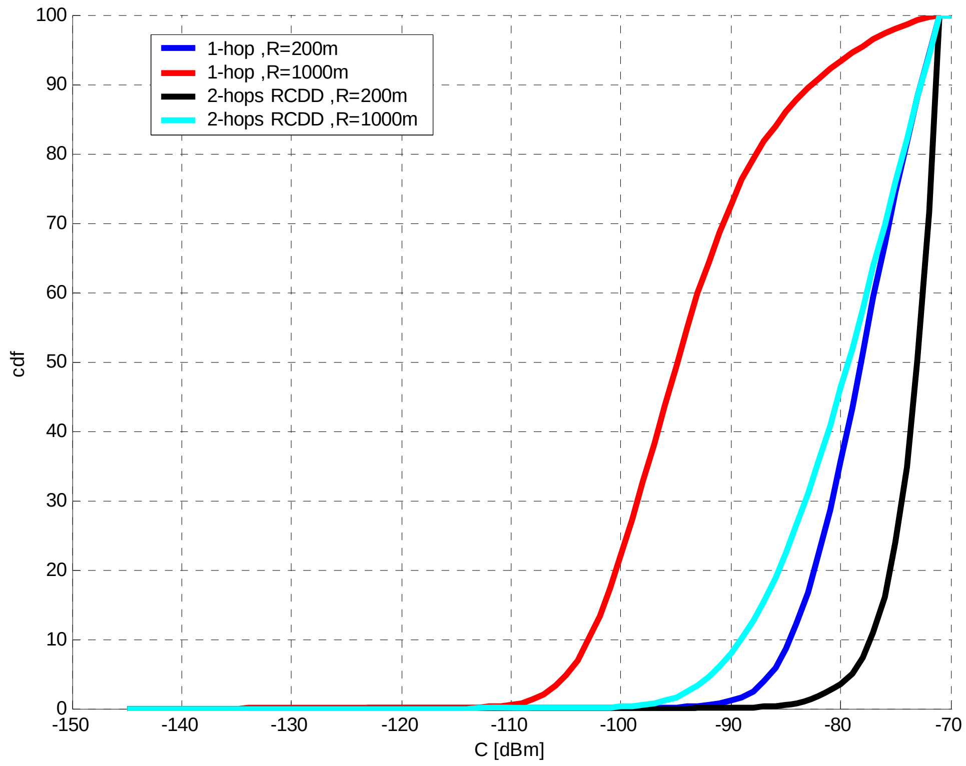


Figure E-34: CDF of the received Sub-carrier Signal Power of the 1 hop and 2 hops schemes.

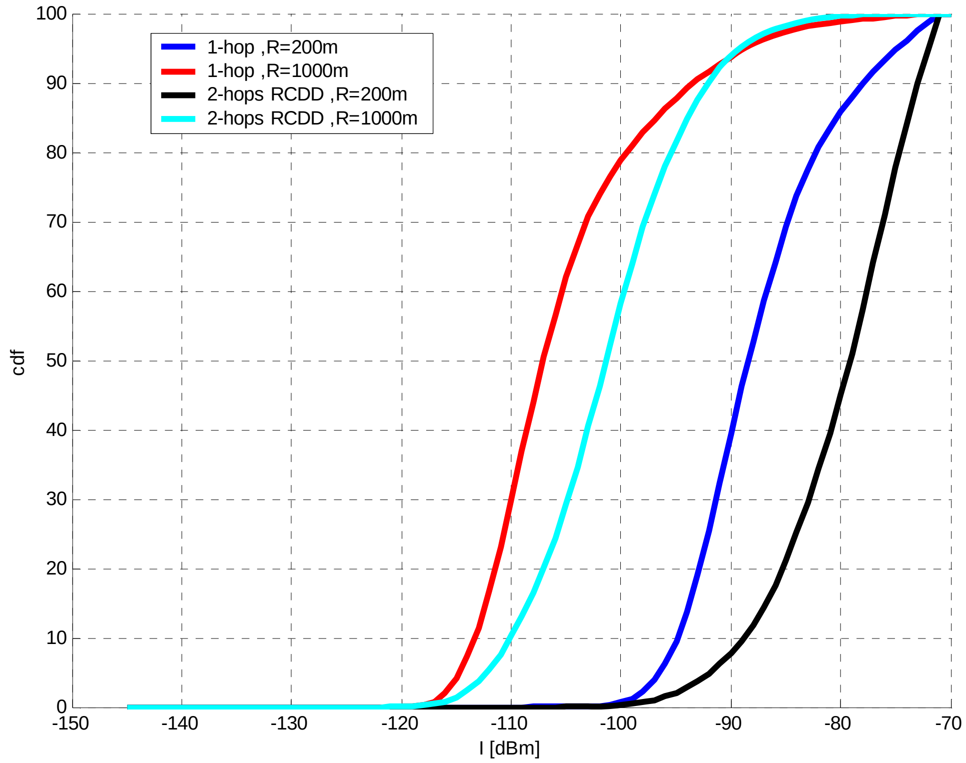


Figure E-35: CDF of the received Sub-carrier Signal Interference of the 1 hop and 2 hops schemes.

The impressive SINR gain for the RCDD translates into a substantial cell throughput gain as it is shown in Figure E-36. In fact for a cell size of 200m, the Relay cyclic delay diversity yields 2.1 to 2.4 times cell throughput gain compared to a one hop system.

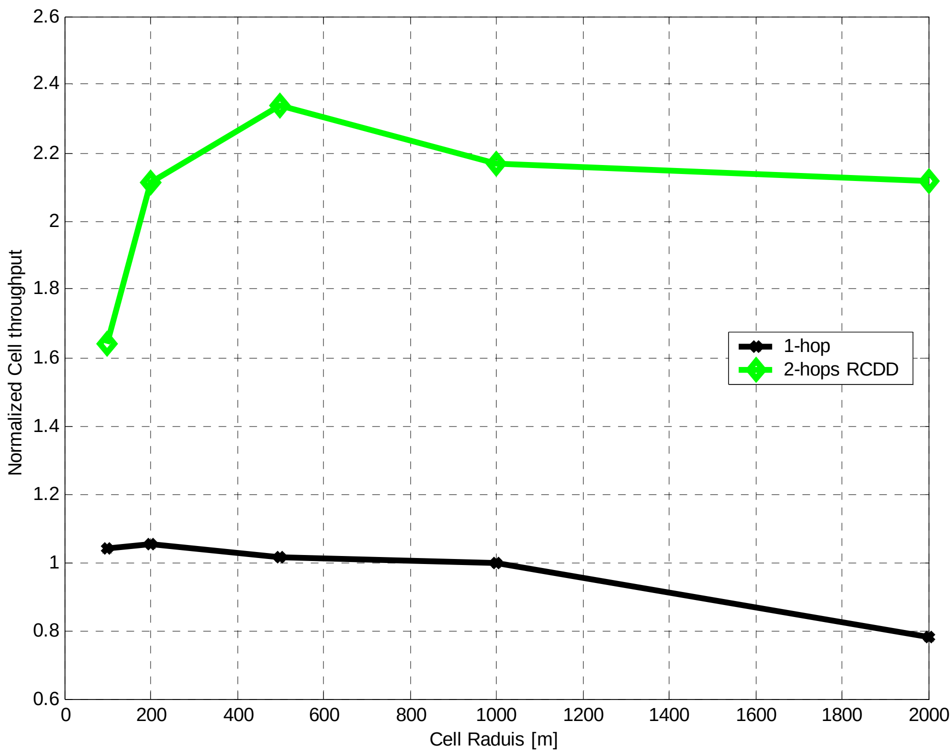


Figure E-36: Relative Cell Throughput Gain of 2hop RCDD compared to a 1 hop system for various cell Radii.

The RCDD also improves the fairness for the user throughput since the artificially introduced selectivity in the channel allowed at least some chunks of any users to be scheduled. This can be deduced by examining Figure E-37, which compares the CDFs of achieved UT throughputs for the two scenarios. For instance the median user throughput for a cell radius of 1km increases from 200kps to approximately 1.2Mbps. In fact at least 75% of the UTs got a chance to be scheduled compared to 60% for the 1 hop system.

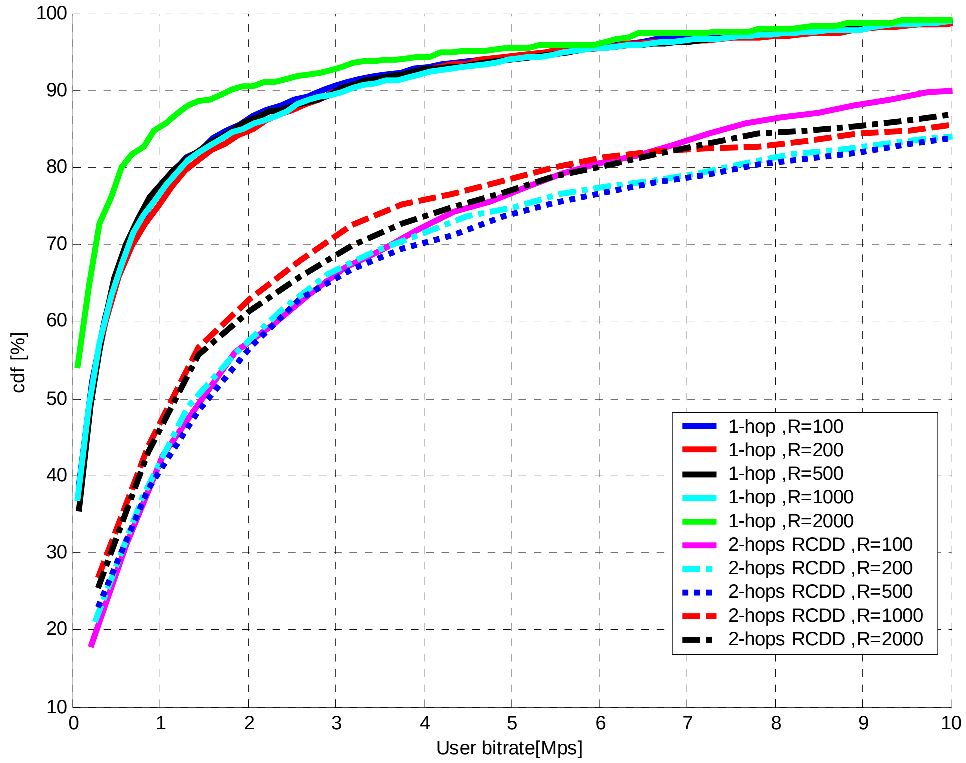


Figure E-37: cdf of the user throughput of 2hop RCDD compared to a 1 hop system for various cell Radii.