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Description of identified new relay based radio network deployment concepts and first assessment by comparison against benchmarks of well known deployment concepts using enhanced radio interface technologies

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Abstract:

Following the WINNER vision of a ubiquitous radio system providing wireless access for a wide range of services and applications across all environments D3.2 has consequently continued the work of D3.1 towards a WINNER deployment concept. To cover the requirements of a WINNER RAN the flexible node architecture has been further elaborated. The envisaged WINNER modes are taken into account by multi-mode protocol architecture that allows to exploit commonalities between different modes leading to a very flexible protocol and a close integration of the WINNER modes.

The first steps towards concepts harmonisation are presented by categorising the large number of deployment concept proposals coming from D3.1 into logical groups. Further results on the comparison between single-hop and multi-hop are shown in D3.2 accompanied by the presentation and assessment of related routing strategies. Also the cooperative relaying concept has been driven forward and new results have been achieved in this field.

Next to multi-hop, the single hop based deployment concepts have been elaborated partly with respect to new air interface technologies. Finally D3.2 describes a system level simulation methodology to simulate the deployment concepts based on the new WINNER air interface taking the agreed scenarios into account.

Keyword list:

Relay based deployment, deployment concept, multi mode radio network architecture, modes convergence, logical node architecture, fixed relay node, cooperative relaying, mesh network, multi-hop deployment concept, system simulation methodology, fixed relay nodes, mobile relay nodes, retransmission protocol, single-hop vs. multi-hop

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Executive Summary

To make the WINNER system concept economically successful, the costs per bit have to be minimised. The development of efficient deployment concepts plays an essential role to provide efficient ubiquitous radio coverage.

A deployment concept for a wireless system based on the WINNER system concept has to integrate several modes related to different scenarios, which most probably partly overlap. These modes have to be integrated in one flexible protocol architecture as well as they have to be fit into one flexible WINNER RAN architecture, as presented in Chapter 2, “Architectural Concepts and Generic Functions“.

Also in Chapter 2 the concept of the multi-mode protocol architecture which allows implementing the idea of several WINNER modes efficiently is further developed in continuation of D3.1. The presented multi-mode protocol architecture is based on the idea of partitioning the protocol functions into generic and mode specific parts.

Chapter 3, “Identification and preliminary assessment of Deployment Concepts for Harmonization”, presents the process of concept harmonisation that has been started by listing and categorising the deployment concepts investigated in D3.1. As starting point the deployment concepts are grouped into categories based on the chosen concept. Therefore five categories have been defined starting with single hop deployment concepts, going over to the four different categories of multi-hop groups. The multi-hop groups are categorised as multi-hop concept based on one air interface mode with fixed relays called fixed homogenous multi-hop concepts, as fixed heterogeneous multi-hop concepts, as mobile multi-hop concepts based on mobile relays and as cooperative multi-hop. The groups are further narrowed which results in only one fixed multi-hop group including different relaying technologies.

The multi-hop concepts as described in Chapter 4, “Multi-hop deployment concept”, are complementing the work started in D3.1 by new simulation results for fixed as well as for mobile relays. The work on cooperative relaying has been continued showing that relay based deployment concept can be efficiently exploited as a distributed antenna system. Following the description of multi-hop concepts some enhanced single-hop concepts partly based on new air interface technologies are presented in Chapter 5, “Advanced single-hop deployment concepts”.

In Chapter 6, “System Simulations Methodology”, finally, the issue of assessment is tackled, which already focuses on the assessment of deployment concepts based on WINNER air interface technologies. It shows Methodologies for simulator design to allow system level performance evaluation of WINNER deployment concepts based on appropriate modelling of the new WINNER air interface modes.

1. Introduction

The WINNER follows the vision of a ubiquitous radio system providing wireless access for a wide range of services and applications across all environments, from short range to wide area with one single adaptive system concept for all envisaged environments. Therefore, the WINNER system concept aims at adapting to multiple scenarios by using different modes of a common technology basis. To make the WINNER system concept economically successful, the costs per bit have to be minimised. The development of efficient deployment concepts plays an essential role to provide efficient ubiquitous radio coverage.

In the context of WINNER a radio network deployment concept has been defined as a description of 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 thereby WINNER modes) and

c) where physical network elements are deployed according to the radio propagation scenarios for which the deployment concept is applicable.

A deployment concept for a wireless system based on the WINNER system concept has to integrate several modes related to different scenarios, which most probably partly overlap. These modes have to be integrated in one flexible protocol architecture as well as they have to be fit into one flexible WINNER RAN architecture.

Further the deployment concept has to make the envisaged high capacity of the new WINNER air interface available in a large area in a cost-efficient manner and support all envisaged service capabilities. Thereby the range for broadband air interfaces that aim at covering densely populated areas is expected to be very limited due to severe attenuation in the expected high frequency band (3.4GHz-5GHz), the limited transmission power (regulatory constraints) and the unfavourable radio propagation conditions, e.g. in urban areas.

Finally the developed deployment concept has to be assessed in terms of traffic performance by means of system level simulations modelling the capabilities of the assumed WINNER air interface technologies as developed by WP2.

To solve the challenges of a WINNER deployment concept WP3 has presented several multi-hop concepts in D3.1. They show the potential of relay based deployment concept in terms of bringing the capacity of on AP into the area, balancing the capacity distribution in the area and providing coverage to otherwise shadowed areas. Also, first steps to the RAN- and protocol-architecture have been provided.

The work of D3.1 has been consequently continued in D3.2. With the aim of integrating the different modes on the one hand and the new network elements to support the several relaying concepts on the other hand. In Chapter 2 , “Architectural Concepts and Generic Functions”, a flexible node architecture for the WINNER RAN is described. Also in Chapter 2 the approach of a multi-mode protocol architecture which allows implementing the idea of several WINNER modes efficiently is further developed in continuation of D3.1. The presented multi-mode protocol architecture is based on the idea of partitioning the protocol functions into generic and mode specific parts. The idea of the generic protocol aspects in the context of the multi mode protocol architecture is to exploit commonalities between different modes by grouping functions common to the different modes in re-usable generic parts of the protocol. These commonalities may be found on many different levels. Chapter 2 is complemented by the description of several control plane functionalities, like radio resource management and user plane functionalities, like, e.g. the proposal of an ARQ algorithm tailored for efficient retransmission in relay based networks.

In Chapter 3, “Identification and preliminary assessment of Deployment Concepts for Harmonization”, the process of concept harmonisation has been started by listing the deployment concepts currently investigated in WINNER and categorising them. As starting point the deployment concepts are grouped into categories based on the chosen concept. Therefore five categories have been defined starting with single hop deployment concepts, going over to the four multi-hop groups. The multi-hop groups are categorised as multi-hop concept based on one air interface mode with fixed relays called fixed homogenous multi-hop concepts, as fixed heterogeneous multi-hop concepts, as mobile multi-hop concepts based on mobile relays and as cooperative multi-hop. The groups are further narrowed which results in only one fixed multi-hop group including different relaying technologies.

The multi-hop concepts as described in Chapter 4, “Multi-hop deployment concept”, are complementing the work started in D3.1 by new simulation results for fixed as well as for mobile relays. The work on cooperative relaying has been continued showing that relay based deployment concept can be efficiently exploited as a distributed antenna system. Chapter 4 presents the first approach to integrate two WINNER physical layer modes namely the TDD and half duplex FDD. The multi-hop concepts are complemented by the presentation of appropriate routing mechanisms to forward the packets in relay based systems.

Depending on the envisaged scenario, also single-hop deployments are in the focus of WINNER and are further developed taking new technologies like adaptive antennas into account. Such enhanced single-hop deployment concepts are shown in Chapter 5, "Advanced single-hop deployment concepts".

Finally, the issue of assessment is tackled, which already focuses on the assessment of deployment concepts based on WINNER air interface technologies. Therefore Chapter 6, „System Simulations Methodology", introduces the ongoing work towards system level simulations. It shows methodologies for simulator design to allow system level performance evaluation of WINNER deployment concepts based on appropriate modelling of the new WINNER air interface modes.

In addition to the deployment concept, RAN protocol-architecture and -functionalities and the introduced simulation methodology, the Annexes provide valuable additional information on how the current results presented in D3.2 have been achieved and outline the calibration case which has been developed to allow comparison between the different simulation results for a fair assessment of the many proposed concepts.

1.1 Terminology and Definitions

One **AP** coverage area can be divided into several cells. An AP might be geographically distributed. A **cell** is defined by the geographical coverage area of the broadcast channel. E.g., the cell as defined by 3GPP [1] is a radio network object that can be uniquely identified by a User Equipment (UT) from a (cell) identification that is broadcasted over a geographical area from one *UTRAN AP*.

Radio waves are received and transmitted by antennas that have certain antenna diagrams (antenna gain and phase as function of angle). A **beam** is radio waves received or transmitted by an antenna system with a certain antenna diagram. **Beamforming** is a function that determines the antenna diagram.

WINNER Connection Service (CS_W): The CS_W is the sum of network elements that are required to allow interworking between different Access Equipments (AE, see below), e.g. for inter-AE handover. Thereby the AEs can also be of different WINNER modes. The CS_W is also the interface to the backbone network, most likely the internet. *Optional (not in the scope of WINNER WP3):* The CS_W might also enable the direct interworking with other (non WINNER) AEs

WINNER Access Equipment for Mode X (AE_{WX}): The AE_{WX} as shown as enlargement in Figure 1-1 comprises always one AP and can be extended by one or several Relay Node (RN) that can be either fixed (FRN) or movable/mobile (MRN). The AE_{WX} is assumed to be WINNER mode X specific. The AP-UT connection can involve several RN. Whether or not AP and RN will share common broadcast channels or not is subject to further study.

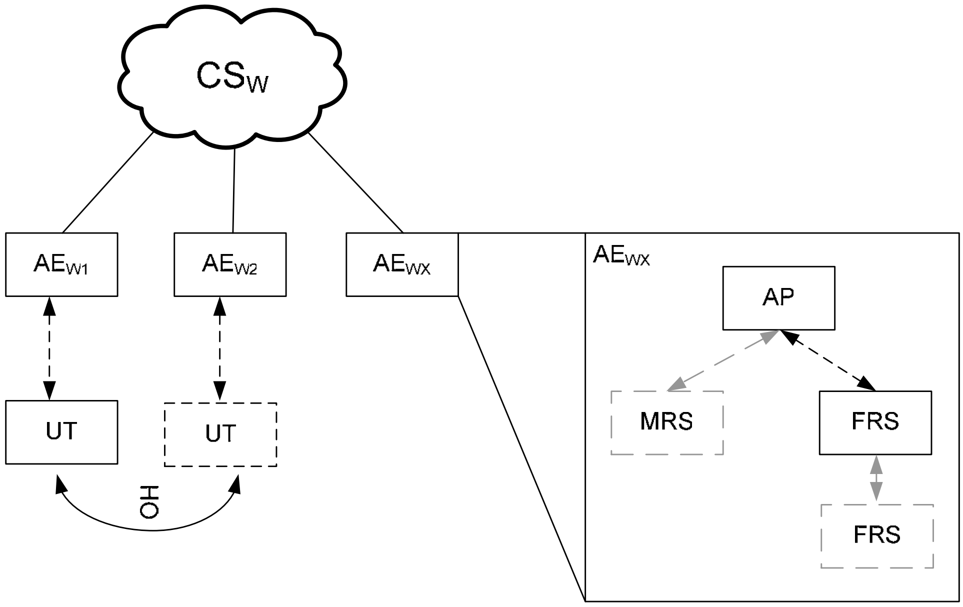


Figure 1-1 Proposed WINNER High-Level Access Network Architecture including relays

The following definitions have been extended from those given in D7.1 chapter 10 [1]:

RAN

Current 2G and 3G cellular radio system comprise the core network and the Radio Access Network (RAN). A RAN terminates all protocols involved in managing and maintaining radio connections towards UTs. Example: in the case of UMTS the RAN is called UTRAN (Universal Terrestrial Radio Access Network) and is a conceptual term identifying that part of the network which consists of RNCs and Node Bs, see [1]. The RNC (Radio Network Controller) is a logical node in the RNS (Radio Network Subsystem) in charge of controlling the use and the integrity of the radio resources. An RNS offers the allocation and release of specific radio resources to establish means of connection in between an UT and the UTRAN. A Radio Network Subsystem contains one RNC and is responsible for the resources and transmission/reception in a set of cells.

Radio Link

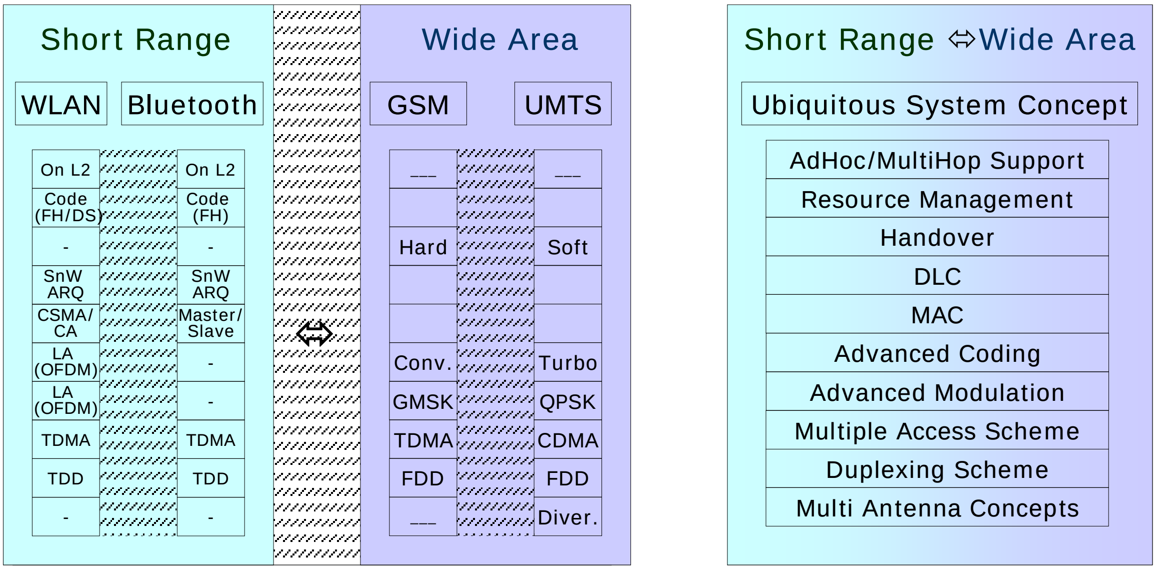
The radio link is a logical association between a single UT and a single UTRAN AP. Its physical realization comprises one or more radio bearer transmissions. The radio bearer provides the services by the RLC (Radio Link Control) layer for transfer of user data between the UT and Serving RNC.

RAT

The Radio Access Technology is the air interface that is used to allow the link between end user terminal and AP of the RAN. This includes also the links between RN and AP. In current cellular systems, one RAT is usually associated to a RAN, e.g. UMTS RAN uses radio access technology CDMA. The RAT has influences on the functionalities and nodes (logical and physical) required in the associated RAN.

Mode (WINNER)

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. The figure below shows the "one adaptable RAT" concept in contrast to different RATs.



Code:	Selection of different spreading-codes to separate user groups FH: Frequency Hopping, DS: Direct-Sequence
SnW:	Stop-and-Wait Automatic Repeat Request
LA (OFDM):	Link Adaptation for the OFDM-based systems

Figure 1-2 WINNER Common Radio Interface

Feeder system (WINNER)

The WINNER feeder system is the system used to feed the WINNER APs. The distinctive characteristic is that WINNER users can not connect to this network. The transmission technology used by the feeder system could be wireless or wireline and is irrelevant and transparent for the final user.

Radio access system (WINNER)

The WINNER radio access system is the wireless network that is used to connect the WINNER UT. The elements of this network are the WINNER access elements (see above, consisting of AP and the relay node), WINNER UTs may function as mobile relays and thus become part of the radio access system. WINNER RAT and its modes are used by the WINNER radio access system.

Homogeneous Relay

The homogeneous relay node is a network element with relaying capabilities that is wirelessly connected to an AP, another RN and/or a UT and that uses the same radio technology (RAT_{modex}) for all its connections.

Heterogeneous Relay

A heterogeneous relay node is a network element that is wirelessly connected to another relay node or an AP (Access Point) by means of a given radio access technology, and serves to another relay node or to a UT using a different radio access technology. 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 is not inside the WINNER scope of research.

Logical and Physical Nodes

A logical node denotes an entity in the (radio access) network that terminates a certain set of protocols. A physical node denotes a physically existing device in the network that incorporates certain functionality, thereby representing one or possibly even more logical nodes.

2. Architectural Concepts and Generic Functions

2.1 Introduction

A challenging task for WINNER system architecture design is to realize the vision of ubiquitous system concept providing access in different deployment scenarios with one single system for all envisaged radio environments. In this chapter, architectural concepts are proposed to brake down the problem into smaller and manageable pieces.

A fundamental principle is to support several distinct modes on a common radio system technology basis. Each mode is further tailored to the needs of different deployment scenarios and radio environments. The ubiquitous radio system is thereby envisioned to efficiently adapt to different deployment scenarios and radio environments by using intelligent mode selection and switching between modes. Another important key feature is the distinction between generic and mode specific functions. Generic functions are referred to such functions that are common for all modes and they are not tailored to the needs of any specific scenarios, i.e. their operation is independent of mode selection. Generic functions are aimed to keep the complexity of ubiquitous radio system concept at a reasonable level. The main objective of this chapter is to describe these architectural concepts and identify some possible generic functions.

The rest of the chapter is organized as follows. Section 2.2 introduces the WINNER architecture followed by the identification of generic radio interface functions. Sections 2.3 and 2.4 discuss generic control-plane and user-plane functions respectively. Section 2.5 finally draws some conclusions and summarizes the chapter.

2.2 WINNER Architecture

2.2.1 Introduction

This section proposes architectural concepts for building, operating, and maintaining the WINNER radio interface for multiple deployment scenarios spanning from short-range to wide-area coverage and from conventional cellular to multi-hop networks. Section 2.2.2 outlines requirements on the architecture. Section 2.2.3 presents a flexible node architecture model that aims to support all envisioned deployment scenarios while still keeping the number of logical nodes and interfaces at a manageable level. Section 2.2.4 describes a concept of multi-mode protocol architecture that attempts to make use of commonalities between different modes. It moreover provides a unified interface towards upper layers such that the heterogeneity of different modes is hidden from upper layer protocols. Section 2.2.5 introduces a channel structure where the same channel types are used for all modes. The proposed structure is simplified by using a distinction between logical and transport channel types.

2.2.2 Requirements on the Architecture

The purpose of this document is to outline a first proposal for the WINNER Architecture, including the overall functions and the inter-relations of logical nodes in this network, as well as a high-level protocol architecture. Important requirements for the WINNER architecture are:

1. The architecture shall allow for scalability as well as optimizations for a given scenario, i.e. it shall allow flexibility in network deployment[1]. Drivers that facilitate this flexibility are ease of configuration (plug and play) and low-cost equipment.
2. The architecture shall preserve layer transparency by providing to upper layers a unified interface for transparent utilization of the various accesses. However this interface towards upper layers needs to have the capability to inform these layers about the capabilities and options (in terms of QoS) that the current configuration of the lower layers provides.
3. The number of logical network nodes and interfaces shall be sustained as low as possible (supporting requirement number 2). Moreover, in the proposed architecture, the choice of radio interface capabilities or radio link modes can be based on the actual situation in the radio environment, the properties of the involved network nodes and the requested services (i.e. mode switching should be supported).
4. The Architecture needs to allow interworking in between different modes as well as with other (new and legacy) radio technologies. It should therefore integrate seamlessly into the Multi-Radio-Access (MRA) Architecture defined within WWI-Ambient Networks (AN)

[1] For instance, it should enable full-scale networks with centralised functionality as well as stand alone networks (e.g. similar to a WLAN AP with decentralised functionality) to be deployed.

To support this integration, the architecture has to support

a. the transfer of protocol status information of existing connections between the different modes. The potential applications for this are multi-mode terminals or multi-mode access points performing inter mode handover.
b. information gathering about existence, status, capabilities of different modes for efficient decision making. This includes cross-mode and cross-system measurement procedures.

2.2.3 Node Architecture

This section will briefly outline the current view of the node architecture (some more details, e.g. envisioned physical implementations, are given in Annex I sections 8.1.1 and 8.1.2, but for a detailed description and protocol termination points, the reader is referred to [2].The main goal of the logical node architecture model presented in this section 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 WINNER architecture. Even though the logical node architecture includes protocol termination points, they do not suggest a certain physical placement of functional entities. Figure 2-1 proposes a preliminary logical node architecture model. Here, dashed lines indicate control relations whereas solid lines indicate user plane data transport. N.B. the Access Router (AR) is outside the primary focus of WINNER.

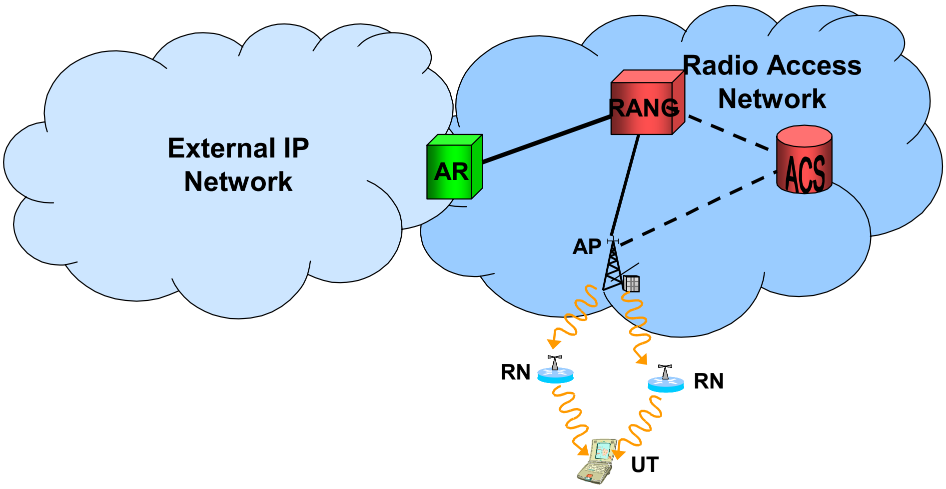


Figure 2-1 Logical nodes

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

Access Point Logical Node (AP_{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.

Radio Access Network Gateway Logical Node ($RANG_{LN}$) is a logical network node terminating the data link layer. It terminates generic data link layer user plane protocols.

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.

Relay Node Logical Node (RN_{LN}) is a logical network node with relaying capabilities that is wirelessly connected to an AP_{LN}, UT_{LN} or another RN_{LN}. Hence, one major difference to an AP_{LN} is that it does not terminate the transport network layer protocols. In many cases this classification is not sufficient and 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 is performing forwarding on (e.g. classified as a RN with layer 3 routing capabilities ($RN_{3,LN}$)). The number of necessary logical RNs is not currently known and is left for future work. In the remainder of this section the more detailed partitioning will be used only in those cases where it is deemed necessary.

Access Control Server Logical Node (ACS_{LN}) is a logical network node that controls the access to the radio interface resources. It terminates generic control plane protocols.[2]

Consequently, the radio interface is, on the network side, terminated in the ACS_{LN} and in the $RANG_{LN}$.

The subsections below further explain the roles, functions and interfaces of the different nodes.

Note that as the definition of the WINNER system and related protocols progress refinement of the above definitions is likely. The aim of the above definitions is to afford the nodes a distinct identity.

2.2.4 Protocol Architecture

This section presents a protocol reference model that allows for the efficient integration of multiple WINNER-modes in a complementary way, taking maximum benefit of the commonalities between the modes. Hence it facilitates the coexistence and the cooperation of different modes in all logical nodes of a future radio access network, (i.) the UTs the (ii.) access points and (iii.) the relay nodes. This section focuses on the realization of multi-mode capability of user and control planes and the related management of the protocols for a flexible air-interface. The related work has been performed in cooperation between WP2 Task 6 and WP3 Task 3.

The separation of the protocol software into generic and specific parts is the basis for the multi-mode reference model that is outlined in Section 2.2.4.3. Optimized transition between modes and coexistence of different modes is realized with the help of a hierarchical management structure on the level of (i.) the complete protocol stack and (ii.) a single layer as introduced in Section 2.2.4.3. An example for generic protocol functions of the data link layer that realize a flexible air-interface through parameterizable modules can be found in Annex 8.1.3.2.

2.2.4.1 Related Work on Generic Protocols

From the software engineering perspective, there are generally two possibilities for approaching the generic protocol functions: (i.) Parameterizable functional modules and/or (ii.) inheritance, depending on the abstraction level of the identified protocol commonalities. The focus of this section is on the modular approach, as introduced in [135], while the inheritance-based approach is in considered in [136]. Additionally, [137] takes up the idea of a generic protocol stack in focusing on a generic link layer for the cooperation of different access networks at the level of the data link layer. However, not only the link layer protocols but also higher layer functions as for instance the control and management of the radio resources as well as mobility have to be considered in a multi-mode capable network.

2.2.4.2 Separation of the Protocol Stack into Generic and Specific Parts

The reference model presented in this section is based on the widespread perception that radio interface protocol functions can be divided into two sets of functionalities:

I. Mode-/System-specific functions: These are protocol functions that are unique to a certain kind of air-interface mode and can not be found in any other mode of the same or any other radio interface. Examples for such mode-specific functions are the allocation of a dedicated physical resource and parameters for dimensioning a mode related to the local communication environment.

II. Generic (common) functions: The view that is usually taken in standardization is that "generic" functions are not fully specified and have to be enriched by specifying missing parts. This view does not apply in our case. It has to be noted that in our context, the "generic" functions are assumed to be the identified set of common (mode independent) functions of a set of air-interface modes. This means they are "generic" from the viewpoint of the modes, but not from the viewpoint of functionality. It is assumed that these functions can be adapted to the use in any of the targeted modes through proper parameterization. They are generic in the following sense: In most cases, they will have to rely on additional mode-specific functions and / or parameterization to provide the full functionality in a certain protocol layer of a certain air-interface mode.

2.2.4.3 Multi-Mode Protocol Architecture Reference Model

Figure 2-2 illustrates the architecture of a multi-mode protocol stack for a flexible air-interface. The layer-by-layer separation into specific and generic parts (where appropriate) enables a protocol stack for multiple modes in an efficient way: The separation is the result of a design process that is referred to as

[2] N.B. certain generic control plane functions may also be allocated to the $RANG_{LN}$ and/or AP_{LN} under the supervision of the ACS_{LN} (see [2]).

cross-stack optimization, which means the identification and grouping of common (generic) functions. The generic parts of a layer, marked green in Figure 2-2, can be identified on different levels as further elaborated in Annex I section 8.1.3.2. The generic parts are reused in the different modes of the protocol stack. The composition of a layer out of generic and specific parts is exemplarily depicted in Figure 2-2.

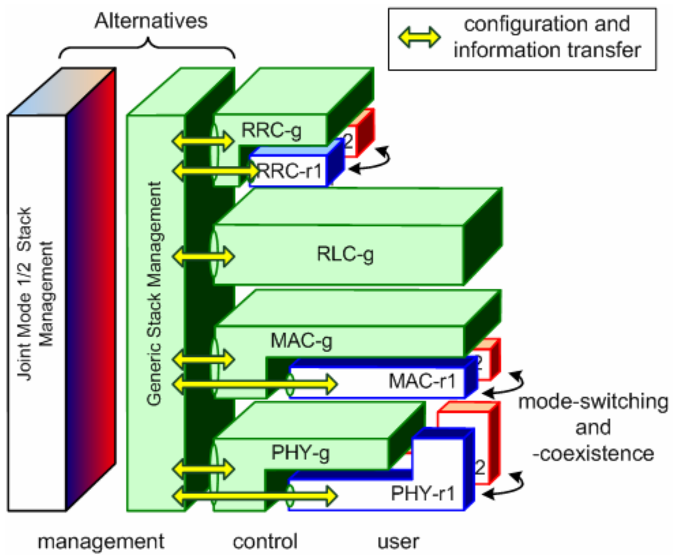


Figure 2-2 The Multi-mode protocol architecture, facilitating transition (switching) between modes (inter-mode handover) and coexistence of modes (e.g. in relay nodes connecting different modes) by way of the cross-stack management supported by the modes convergence manager of a layer or stack.

The Management and the joint handling of the protocol stack operating in different modes is performed by the Stack Management, as shown in Figure 2-2. When multiple modes are operated, this can be regarded as Cross-Stack Management and it is envisaged to be performed by a Stack Modes Convergence Manager (Stack-MCM) which controls the Management functionality in the respective protocol layers (N-Layer Modes Convergence Managers, (N)-MCM) in a hierarchical manner. A more detailed description of the hierarchical stack management approach and the functional elements in the reference model may be found in Annex I section 8.1.3.3.

The introduced hierarchical management structure differs between the complete stack and single protocol layer. The counterpart of the Stack-MCM is the (N)-MCM which exists once in each layer as presented in this section.

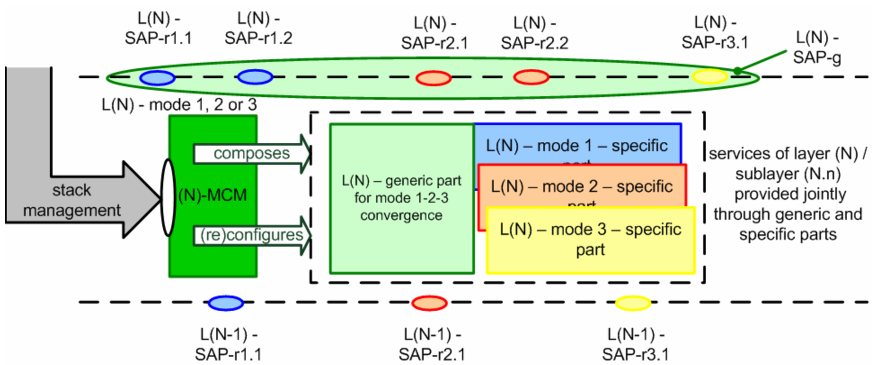


Figure 2-3 Composition of a layer (N) from generic and specific functions. The composition and parameterization is handled by the (N)-MCM. The (N)-MCM is controlled by a layer-external stack management entity, namely the Stack-MCM

Figure 2-3 shows the general structure of a protocol layer conforming to our reference model from above. It is assumed that the functionality inside the layer is always composed of a generic (common to all modes) part and mode-specific parts, which jointly provide the modes' services of the layer via SAPs. Through this, layers can be operated in one mode at a time. Modes can also coexist temporarily or permanently. The specific SAPs of a layer are defined via the currently used mode or set of modes. This does not preclude the possibility that SAPs of different modes can be accessed by higher layer entities in a common way as visualized by L(N) SAP-g.

The composition and parameterization of the layer is taken care of by a layer-internal instance, the (N)-Layer Modes Convergence Manager ((N)-MCM), which resides in the management plane. It is the layer's counterpart of the Stack-MCM as introduced above. The (N)-MCM enables that a (N)-layer provides multiple modes and makes functionality of one mode or common to several modes available. An instance of the (N)-MCM serves in each layer of the air-interface protocols. The (N)-Layer Modes Convergence Managers ((N)-MCM) thereby achieve the following

- Protocol Convergence – Integrating different modes in one protocol layer, also allowing coexistence of modes
- Layer Composition and Parameterization – arranging functionalities and setting mode-specific parameters
- Data Preservation and Context Transfer – to facilitate seamless transition / switching between modes

2.2.4.4 Conclusion

The introduced multi-mode protocol reference model facilitates the structuring of an arbitrary layer into generic and specific parts. In providing guidance for understanding this structuring it marks up optimization potential in questioning the necessity of indicated differences. In this way, an increased protocol convergence is reached enabling an efficient multi-mode protocol stack for future wireless systems. A clear separation of current, mutually complementing research efforts in 4G is fulfilled and commonalities are identified.

2.2.5 Channel Structure

This section presents a channel structure for the radio interface. For the sake of simplicity and clearness, a distinction between logical and transport channels is made. The channel types are defined as follows.

Logical Channel: A logical channel is an information stream dedicated to the transfer of a specific type of information over the radio interface. Logical Channels are provided on top of the MAC layer.

Transport channel: The channels offered by the physical layer to data link layer for data transport between peer physical layer entities are denoted as Transport Channels. Different types of transport channels are defined by how and with which characteristics data is transferred on the physical layer, e.g. whether using dedicated or common physical channels.

Logical channels are furthermore divided into two main categories, which are control channels and traffic channels. Control channels are used for transfer of control plane information whereas traffic channels are used for transfer of user plane information. Following logical and transport channels are identified for the radio interface.

Logical channels:

Control channels:

- Broadcast control channel (BCCH) for control information to all terminals in a cell
- Paging control channel (PCCH) for paging terminals
- Dedicated control channel (DCCH) for dedicated control messages
- Common control information (CCCH) for point-to-multipoint control messages

Traffic Channels:

- Directed traffic channel (DTCH) for point-to-point traffic
- Common traffic channel (CTCH) for point-to-multipoint traffic

Transport channels:

- Broadcast channel (BCH) for broadcasting system information etc to all terminals inside the cells coverage area.

- Contention based random access channel (RACH) for initial access to a master device
- Contention based direct access channel (DACH) for using efficiently resources, speeding up connection establishment, peer-to-peer communications
- Common data channel (CDCH) for point-to-multipoint communication
- Targeted data channel (TDCH) for point-to-point communication

Observe, that there are two different types of scheduled data channels — common (CDCH) and targeted (TDCH) transport channels. The motivation for this choice is that point-to-multipoint communication and dedicated communication have different physical channel requirements concerning, for example, power control. In addition, there are two contention-based channels — random access (RACH) and direct access (DACH) channels. Normally contention-based channels are used for small amounts of data only, such as association requests. The current view in WINNER is that contention based channels may speed up data traffic also for large amount of data and thereby separate transport channels may be useful. Whether two contention-based channels are actually necessary is a subject for further studies.

2.3 Control Plane Functionalities

2.3.1 Introduction

The control plane of a WINNER system is envisioned to be a harmonized/integrated concept which could accomplish efficient Radio Resource Management (RRM) for different WINNER modes. To realize such an integrated concept capable to function under different modes, apparently, on one hand, mode-specific RRM functions should be studied in the context of individual modes; on the other hand, generic (mode-independent) issues should be investigated as well for effective management and coordination of mode-specific RRM entities.

The main objective of this section is to address the generic issues for the harmonization/integration of WINNER control plane functionalities. The discussion is organized in terms of RRM functions: for each key function, firstly, a brief overview is presented on what mode-specific functions are under investigation within WINNER project. Afterwards, a detailed discussion is given on the important harmonization/integration issues, such as: the general architecture of this RRM function (centralised or distributed), and the interactions between the given RRM function and others.

This study is continuation for the previous work in [1] where following generic control-plane functions are identified;

- Mode selection
- Cell selection and handover
- Admission control
- Control interval of admissible power allocations
- Traffic policing
- Congestion control and buffer management
- Packet scheduling and flow classification
- Routing

In the rest of this section, following functions are studied in more detail;

- Handover
- Queue management
- Routing
- Scheduling

2.3.2 Radio Resource Management functions

2.3.2.1 Handover

Handover is the process to support terminal mobility, which occurs when a terminal changes the access terminal through which it is communicating. Various solutions have been proposed, but seamless handover – low loss and low latency handover, is still a challenge in mobile telecommunication.[89]

2.3.2.1.1 Overview on the related algorithm/protocol developments in other WPs or sub-tasks

Depending on the deployment scenario the handover might involve both the IP layer and the link layer, more details can be found in Annex I section 8.2.1. In this section, IP layer procedure is described first and then the link layer procedure is followed. A two-level decision algorithm is used in link layer procedure to make a final optimised decision.

IP Layer Procedure in Handover [90]

IP mobility investigates the IP packets routing updates due to user terminal's mobility. It is generally agreed that IP terminal mobility can be separated into two complementary types – macro-mobility and micro-mobility and they need two different solutions. The macro-mobility refers to the mobility between different administrative domains (AD). The micro-mobility means that mobility within the same AD.

It is generally accepted that Mobile IP (MIP) will be used in macro-mobility, which will use the home agent and tunnel to locate the user when it is not at home network. For micro-mobility, all the route update signalling schemes come to the similar point that path updates are localised which only travel between the cross-over router (the router at the cross point of the old path and new path) and the old and new ARs. This will reduce the signalling load and also ensures that the path update process is quick. The requirement of seamless handover is very critical for real time applications, like Voice over IP (VoIP). In order to reduce the latency the IP handover procedure should take place at the same time as the link layer handover – proactive IP handover. This requires the user terminal to be able to detect the new AR and configure the new Care of Address (CoA) (the terminal's IP address assigned by a router) while the current link connection is still working.

Soft handover is the best way to do seamless handover, but it requires user terminal to connect with two APs simultaneously and requires very tight timing alignment of traffics between two APs, which is not always possible. Some semi-soft handover scheme has been proposed in mobile IP which requires the user terminal to connect to two APs at the same time, but allow the misalignment of the traffics between them. This can be achieved by bi-casting. Also in order to reduce the packet loss a tunnel between the old AR and new AR should be set up which will forward the packets in-flight to new AR after the old connection has been cut.

Link Layer Procedure in Handover [91]

Link layer handover can be described in four steps: Link layer triggers; Link layer measurement and other information collection; Decision making based on different criteria; Signalling and Contexts transfer between different APs (N.B. the necessity of signalling and context transfer depends on the deployment scenario but will always be employed if IP mobility is used).

In conventional cellular mobile system like UMTS, the mobile can make a measurement of radio link signal to noise ratio and make a decision of handover by using handover algorithms based on the measurements. This is adequate because this handover decision is only based on the coverage condition. Furthermore, in a heterogeneous network environment with different radio interface modes available, the measured link level signal matrix need to be mapped correspondingly in order to make a right decision of handover among them.

It is foreseen that the WINNER system will operate in different modes within the same coverage area, i.e. same area might be covered by different modes offering different data rate, QoS, and prices. Furthermore, wireless relays will be a significant part in the WINNER wireless network. The RN could also be a point of attachment for a user terminal. The differences between a RN and an AP in terms of measured link quality, QoS, security, et. al. should be studied carefully when making a decision to handover from a AP to a RN.

In the WINNER system, several handover triggers can happen at the same time. The decision algorithm for individual triggers should be supervised by a Joint Arbitration Algorithm [142] to prioritise the individual algorithm and optimise the final handover decision. The overall link layer handover architecture is given in Figure 2-4

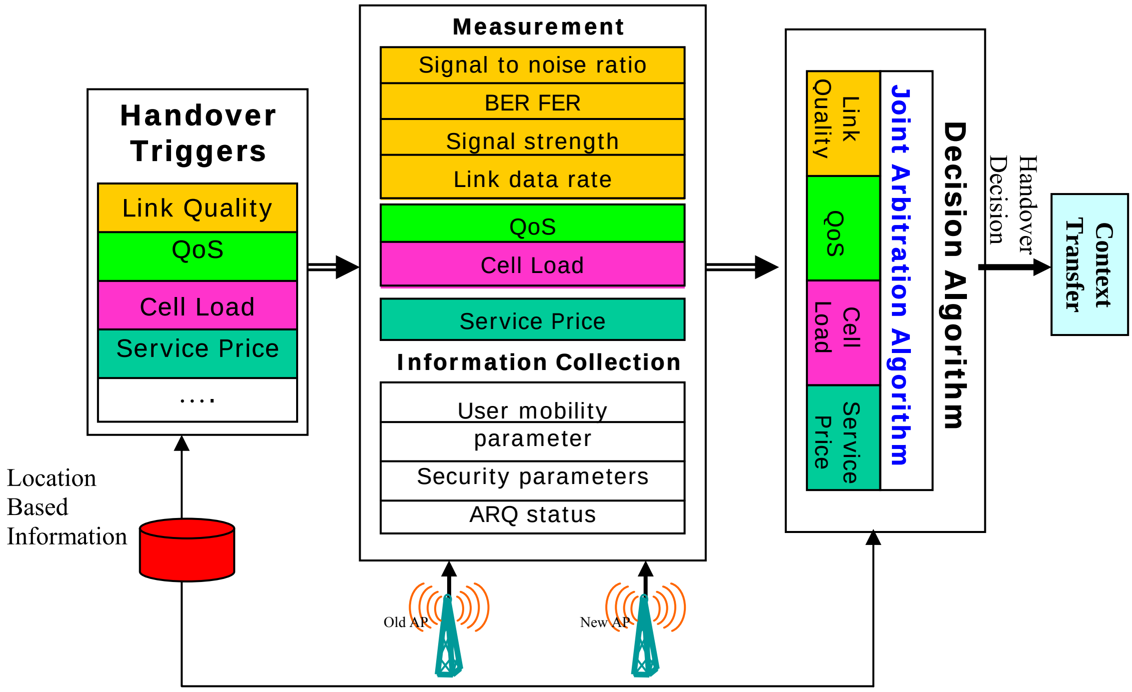


Figure 2-4 Link layer handover procedure

2.3.2.1.2 Problem definition for integration/harmonization

The handover process in the WINNER system will be complicated by multiple handover choices among multiple WINNER modes. As in the future WINNER system, the radio service with different characteristics might be offered in the same area, handover between different modes with different data rate, QoS, link quality, different price will be quite common. In order to achieve a seamless handover, these parameters need to be kept consistent. As the cells offering high data rate usually have small coverage, this will increase the probability of handover. Fast and reliable handover between the APs, AP and RN are very important for the future WINNER system. How to make a right handover decision in terms of radio link connectivity, QoS, price under different mobility condition is very challenging.

2.3.2.1.3 Tentative integrated concept

The integrated concept is to design a proactive handover algorithm which can work in a multi-mode scenarios.

2.3.2.1.4 Conclusion and future work

Handover in Winner system is complicated by the multiple handover criterias and multiple handover choices. It is essential to design an algorithm which can derive the indications of link connection quality, QoS, cell load, et al from measurable parameters and compare the parameters among different modes. An arbitration algorithm is needed on top of individual algorithms to prioritise and optimise the decision.

2.3.2.2 Queue Management

A queue management scheme for the WINNER system should ensure both high link utilization and low queuing delay. In order to provide congestion control, both queue management and packet scheduling are required. The purpose of queue management is to control the queue length and to determine which packets to drop when the queue becomes full.

According to [83], a traditional First-In-First-Out (FIFO) scheduled queue with a tail drop policy may not be sufficient. In some situations one flow may lock out other flows, due to the tail drop policy. Lock out can be avoided by applying a drop front or a random drop policy instead. Another problem with a FIFO queue is inefficient link utilization. When the queue becomes full and packets are dropped, rate-adaptive flows (e.g. TCP) reduce their transmission rates in response to the dropped packets (which are taken as an implicit signal of congestion). After the queue has emptied, the link utilization may be low if there are only few packets available for transmission. Delays could be minimized by properly adjusted buffer sizes. Over-buffering should be avoided, since this leads to increased queuing delay and TCP is degraded when the round trip times becomes longer and more variable.

Various active queue management (AQM) schemes have been proposed [85]. The purpose of AQM is to reduce the congestion in the network before the queue becomes full and packets are dropped due to buffer overflow. Many AQM schemes are modifications to the Random Early Detection (RED) scheme [83][84]. The RED scheme uses the average queue length as an estimate of congestion. When the queue size becomes larger than a threshold value, incoming packets are dropped with a certain probability. The problem with RED is that it is hard to parameterize.

2.3.2.2.1 Overview on the related algorithm/protocol developments in other WPs or sub-tasks

In D4.1 by WP4, radio resource management architectures for inter-RAT communication are examined. Queuing is discussed briefly in relation to scheduling. A mix of priority queuing and weighted fair queuing is mentioned as a possible solution. The alternative of using a queue with threshold dropping instead is also discussed. The choice of mechanism to use is proposed to depend on the desired service, e.g. weighted fair queuing for real-time traffic and priority queuing for traffic with different service levels.

2.3.2.2.2 Problem definition for integration/harmonization

The AQM should be optimized for the WINNER modes. The AQM scheme should work efficiently in relation to inter-cell and inter-mode handover. Multi-hop communication over RNs must also be considered.

Over-buffering should be avoided, since this has a negative impact on rate-adaptive flows [79][86][87][88]. Over-buffering leads to long per packet delays, increased timeout values for TCP, unfair sharing of the available bandwidth due to lock out of new connections, timeout during TCP connection establishment, and unnecessary transmission of stale data (e.g. when the stop button is pushed in the Web browser).

On the other hand, in order to ensure high link utilization, the buffer must be sufficiently large to ensure high link utilization [86]. Therefore, the drop policy should avoid dropping multiple packets in the same TCP window, since this leads to timeout and reduction in the transmission rate, which in turn may lead to under utilization of the wireless link.

2.3.2.2.3 Tentative integrated concept

In [86], the Packet Discard Prevention Counter (PDPC) scheme is presented, which is one of the more promising AQM schemes for wide-area wireless networks, such as GPRS and 3G. The wireless link is assumed to be the bottleneck link with a higher latency than fixed links. Hence, the wireless link constitutes most of the bandwidth-delay product. Furthermore, per user buffering is assumed, which gives a low degree of statistical multiplexing. The assumptions made would probably apply also to the WINNER system.

In contrast to RED, PDPC operates on the instantaneous queue size and drops packets deterministically. If the queue reaches a threshold value, which is based on the bandwidth-delay product, but stays below the maximum queue size, then a single packet is dropped. A number of packets is transmitted over the wireless link before a single packet may be dropped again. This avoids dropping multiple packets from the same window.

2.3.2.2.4 Conclusion and future work

By applying an AQM scheme in the WINNER system, the effects of congestion could be minimized. AQM schemes used in the fixed network are, however, not optimized for the characteristics of wireless networks and multi-hop communication over RNs. In a wireless, it is important to provide both high link utilization and short queuing delays. The PDPC scheme could serve as a starting point for designing an AQM scheme for the WINNER system.

2.3.2.3 Routing (relay node selection)

2.3.2.3.1 Overview on the related algorithm/protocol developments in other WPs or sub-tasks

The following routing algorithms/protocols are under investigation within WINNER WP3 T3.2 and T3.4, [2].

- A hybrid routing scheme combining location-based routing and on-demand routing
  The objective of this algorithm is to combine the characteristics of on-demand and location-based routing algorithms to come up with a better scheme avoiding the drawbacks of both at the same time.

- WiFR (Wireless Fixed Relay Routing)
  WiFR is a centralized routing strategy for wireless networks with fixed RNs. The routing decision-making is carried out flow by flow based on precise knowledge of global network status.
- Multi-constrained routing
  This scheme is based on the generalized Dijkstra's algorithm. The basic idea is to find a set of paths based on one main criterion, then every single application may select a path from the resulting set to best meet any other criteria, i.e., multiple constrains are applied during the route determination.
- OLSR (Optimized Link State Routing)
  One of the main advantages of the OLSR is that it uses only a set of selected neighbour nodes for the transmission of control messages, whereas all other neighbour nodes also receive and process the broadcast messages, but do not retransmit them.
- Routing/forwarding mechanism in a HiperLAN/2 evolution
  An initial approach has been investigated on how the routing/forwarding can be realized in a multi-hop system like HiperLAN/2.
- Routing/forwarding in the context of heterogeneous relaying
  An initial investigation has been carried out on how routing/forwarding can be performed at the GLL level in the context of heterogeneous relaying. Noteworthy, in this study, a path/route is explicitly defined as the nodes and *modes* traversed from source to destination.

2.3.2.3.2 Important Issues for integration/harmonization

It is envisioned that the following issues are very important for the integration/harmonization of routing algorithms of WINNER systems, and thus should be paid attention to:

- Centralised and distributed routing

From the architecture perspective, two main routing strategies are envisioned in WINNER systems: centralised and distributed routing. The centralised routing strategy is mainly used in multi-hop cellular networks (MCN), and its major advantages are: 1) since the route calculation is mainly performed by APs, as a result, no complex routing tables need to be maintained within individual UTs, moreover, complex procedures for route discovery, maintenance, and repair are not necessary either; 2) due to the fact that the processing capability of APs are more powerful than normal network nodes like UTs, more sophisticated routing algorithms could be applied to optimise the QoS. In a distributed routing strategy, the routing decisions are jointly made by all network nodes along the route from source to destination, based on local information on network status, therefore, complex route discovery, maintenance, repair procedures are inevitable, and this strategy normally performs worse than a centralised one in terms of QoS optimisation (throughput maximization and delay minimization etc.) etc., however, it could function in the places where UTs have difficulties to reach APs and it thus impossible to use a centralised strategy.

Based on aforementioned discussion and considering the objective of WINNER systems (the high requirements on the throughputs of different scenarios), the centralised routing strategy is envisioned as the main focus of the research on routing algorithms/protocols, whereas the distributed strategy could be considered as a complementary solution to be applied in the extension of infrastructure-based network.

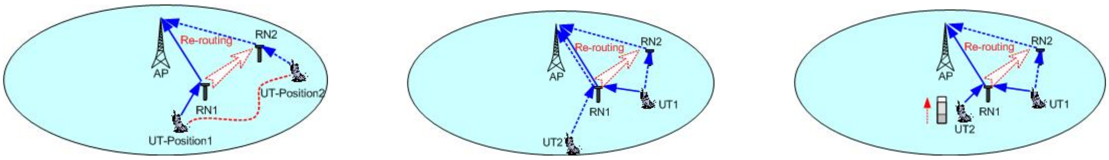
- Routing and re-routing

In WINNER systems, due to the multi-hop capability of network nodes, when a session is about to be established, routing needs to be carried out to set up its transmission route. During a session, re-routing may need to be carried out from time to time to effectively maintain the quality of the route, due to the following reasons:

1) Mobility of UTs (or mobile RNs)
   During the course of a session, when the UT moves (or the involved mobile RNs move), the original route set up at the beginning of the session may become unsuitable (or even invalid). An example is shown in the case (a) of following figure.
2) To effectively accommodate new coming sessions
   In the course of a session, when new sessions are established, the route of existing sessions may need to be changed to accommodate the newly established ones. A simple example is illustrated in case (b) in the following figure, where there are two RNs in a cell: RN1 and RN2, and both of them are assumed to be only able to relay one UT (due to the limitation of relaying time slot or code etc.). When UT1 first comes into the cell, we assume that it has difficulties to reach the AP, and therefore use RN1 to relay its signals to the AP. When UT2 comes into the cell, we assume it cannot reach the

AP directly either, but it could reach RN1 (not RN2). Apparently, in this situation, we need to perform re-routing for UT1 and hand it over to RN2, and then UT2 could connect to the AP via RN1.

3) To effectively accommodate traffic volume fluctuation
Normally bandwidth, time slots, or codes etc. limit the capacity of RNs. As a result, the traffic volume fluctuation of a multi-hop UT may affect the routes of those who are sharing the same RN with it. A simple example is shown in case (c) in the following figure. Initially, both UT1 and UT2 are using RN1 to reach the AP. Afterwards, there is a big jump on UT2's queue size, as a result, the total traffic volume of UT1 and UT2 at this moment exceeds the capacity of RN1. In this situation, to deliver the traffic promptly, we could do the re-routing for UT1 to hand it over to RN2, and then leave RN1 dedicated to UT2.



(a) (b) (c)

Figure 2-5 Re-routing scenarios

Based on aforementioned discussion, routing is necessary not only at the establishment of sessions, but also in the course of them. However, it is worth mentioning that re-routing improves system performance, but meanwhile causes signalling and processing overhead as well. Therefore, a proper balance needs to be made between the overhead and performance paybacks when we chose the period of re-routing.

➢ Interactions between routing and radio resource allocation

In WINNER systems, the routing and radio resource allocation are highly coupled: on one hand, the allocation of radio resources should be based on user transmission routes, since delivering the same amount of traffic via different routes normally costs different amount of radio resources; on the other hand, the selection of user transmission routes should be based on the available resources of APs and RNs. An example illustrating the benefits of considering routing when performing the resource scheduling and vice versa is given in Annex I section 8.2.2. It should however be noted that even though the solution outlined in the Annex uses a (joint) centralised approach similar benefits are expected for a (decoupled) distributed approach as long as the interdependency between routing and scheduling is kept.

➢ Interactions between routing and admission control

The interactions between routing and admission control are: when a new session comes, routing is carried out to find out its best route, based on which admission control decides whether accept the session or not.

➢ Interactions between routing and handover

The interactions between routing and handover are: for each user, routing is carried out first to find out the best routes in different modes (inter-mode handover) or cells (intra-mode handover), and then based on the best routes, handover determines whether the user should be handed over (or whether its active set should be updated).

➢ Interactions between routing and fast power control

Due to some effects like error accumulation (digital relaying) or noise amplification (analogue relaying), the overall BLER of a multi-hop link is likely to be higher than direct transmission ones. To maintain the same overall BLER for multi-hop links, normally higher SIR targets are needed for fast power control, so that fast power control SIR targets should be adjusted when a user is switched from direct transmission to multi-hop transmission (or vice versa). As a result, the fast power control module should be aware of the results of routing in order to determine whose SIR targets should be tuned and how much.

2.3.2.4 Scheduling architecture

This section will present a scheduling architecture that is based on the scheduling architecture presented in D3.1 [1]. As in D3.1[3], the basic hypothesis for the scheduling architecture will be that scheduling is partitioned into two levels (although combined integrated solutions may also be envisioned and is not

[3] N.B. some terminology have been altered as compared to D3.1: what was denoted packet scheduling in D3.1 is now denoted service level controller (SLC), while the channel dependent resource allocation of D3.1 will be denoted resource scheduling (RS).

prevented by the proposed scheduling architecture), of which one belongs to the mode-independent MAC-g. Motivations for this split may be found in Annex I section 8.2.3.1. The related work has been performed in cooperation between Task 4 and Task 6 within WP2 and Task 3 within WP3 (see [138] for more details).

The proposed scheduling architecture will be introduced by explaining the downlink case for scheduled data channels first. The more general case that also accommodates the uplink and relaying case is given in Annex I section 8.2.3.2.

Figure 2-6 gives an overview of the proposed scheduling architecture for the downlink case. In the upper (generic) part of the link layer, per flow buffering of incoming packets[4] is performed, with flows ordered by service class and terminal/user. The buffers are here provisionally denoted service level control buffers. It is envisioned that admission control and active queue management are used to control the queue lengths. Moreover, to facilitate inter-mode handover and quality control over multi-hop links, a separate retransmission protocol functionality (e2e ARQ) [139][5] may be used for the packets queued in the service level control buffers. It is assumed that the packets queued in the service level control buffer are IP packets.

The main task for the service level controller (belonging to MAC-g) is to requests resources by issuing requests to the resource scheduler and forward a whole or a part of a packet into the lower per-flow buffers in Figure 2-6. These buffers, the MAC-r queues, are here called resource scheduling buffers. The granularity of this copying operation is a parameter to be determined. The control strategy of the service level controller takes inter-flow fairness, service level contract agreements, total delay constraints and how much data that is already queued for transmission in the resource scheduling buffers into account. Moreover, to support QoS requirements it is envisioned that the service level controller will assign different priorities to different flows. Furthermore, even though not shown in the figure, it is assumed that the service level controller may perform inter mode scheduling/mode selection (see D3.1 [1] for details). Hence, the service level controller may control the queue lengths of the resource scheduling buffers. The service level controller works on a time scale characterized by the packet arrival rates, i.e. in the order of 10-1000 ms.

In the lower, mode specific, part of L2 (MAC-r), we thus perform per-flow buffering in the resource scheduling buffers. The resource scheduler determines which queues are to be drained. They work on the time-scale of the channel slot length[6]. The queues are drained with a granularity defined by the scheduling unit (SU)[7] (i.e. the MAC-r PDU). After coding and bit interleaving of an SU, we have formed a L1 packet. L1 packets are mapped onto the physical channel resource units, henceforth denoted chunks[8]. Finally, a link ARQ functionality performs retransmissions of failed scheduling units.

Within the resource scheduler, two basically different allocation strategies can be used: adaptive (opportunistic) and non-adaptive (averaging). These two types of transmission differ fundamentally, and they require resources with different properties. Adaptive transmission uses channels state information to utilize the channel and disturbance/interference variations for different terminals. By selecting the best resources for each flow, the throughput may be increased. Averaging strategies, on the other hand, are designed to combat and reduce the effect of the variability of the signal-to-noise and interference ratio (SINR), by e.g. interleaving, space-time coding and diversity combining. Non-adaptive transmission is required when fast channel state feedback is unreliable due to e.g. a high terminal velocity or a low SINR, or when the terminal does not support adaptive transmission. It is hence a task of the resource scheduler to partition the resources between the flows that are governed by the adaptive resource scheduler and those that are governed by the non-adaptive resource scheduler. Two different principles may be envisioned for this: i) the set of chunks are pre-allocated to the two schedulers, in proportion to the expected requirement over a future time interval that comprises several scheduling rounds (the disadvantage with this strategy is that when a set of resources is split rather than pooled, some trunking

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[4] It is currently assumed that these incoming packets are IP-packets.

[5] N.B. If Relay-ARQ (see section 2.4.2.2.2) is used, some form of cross-layering is needed.

[6] Time interval for which the resource scheduler allocates time-frequency-antenna resources. Equals one or several chunk durations (see below).

[7] The size, in payload bits that is allocated by a resource scheduler. Constitutes the link retransmission unit.

[8] A chunk consists of a rectangular time-frequency area that comprises a number of subsequent OFDM symbols and a number of adjacent subcarriers. A chunk contains payload symbols and pilot symbols. It may also contain control symbols that are placed within the chunks to minimize feedback delays. Moreover, the channel slot length equals the time duration of one or several chunks.

gains will be lost); and ii) the adaptive resource scheduler is assigned a certain percentage of the chunks and hence selects the chunks that provide the best transmission for its flows (according to some performance criteria), and leaves the rest to the non-adaptive scheduler. The second alternative will result in the highest total cell throughput, and the highest performance according to the whatever criterion used in the adaptive resource allocation, with little reduction of the performance of the non-adaptive allocation. It should therefore be the recommended approach.[9]

It should finally be noted that: if resource schedulers can be designed to control buffer levels relative to target levels, then the transmission delays within the MAC-r can be controlled locally. Furthermore, the service level controller may then handle an increased traffic demand for a flow by simply increasing the inflow to the resource scheduling buffer. The resource scheduler would then automatically increase the transmission rate by assigning more resources to this flow.

A good overview of possible service level controller and resource scheduler protocols and algorithms may be found in [140].

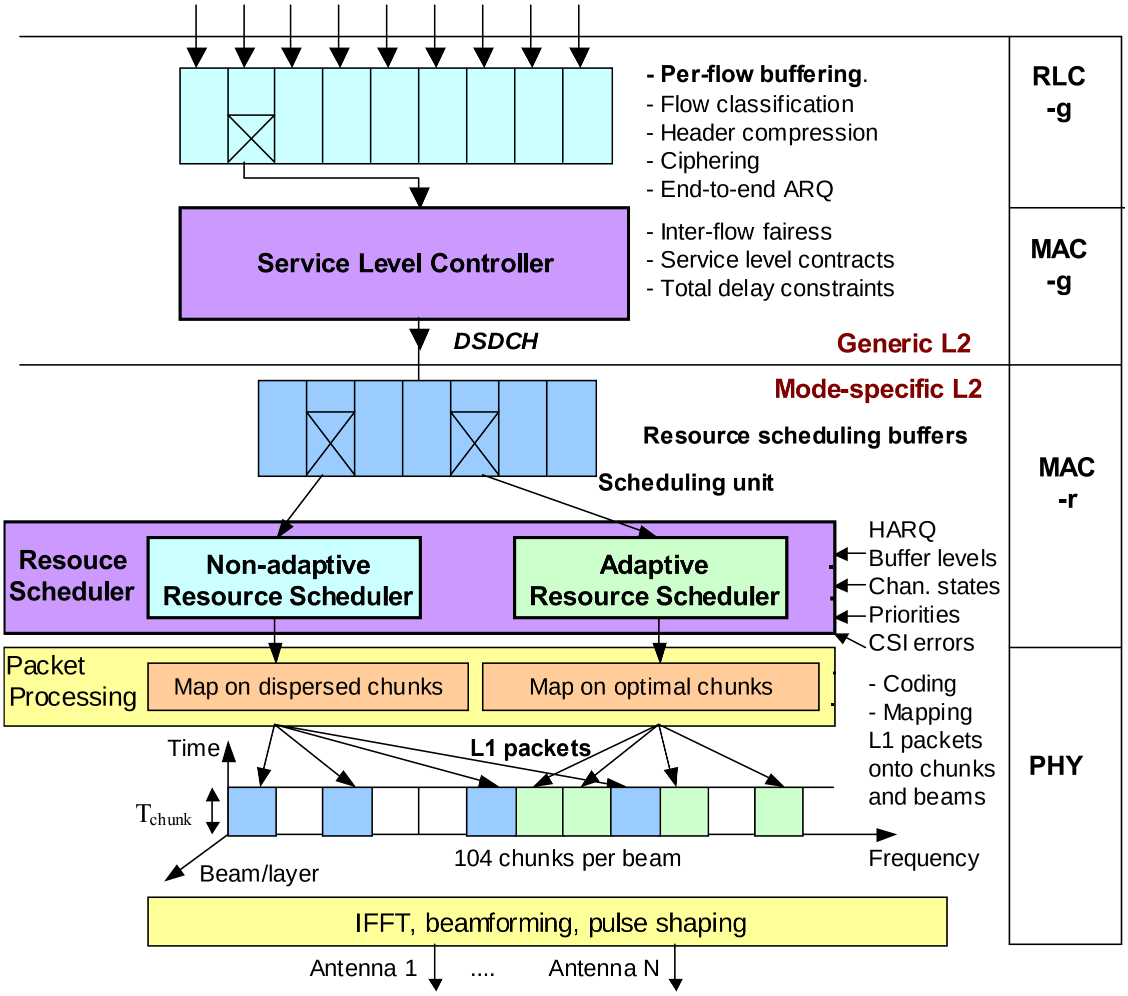


Figure 2-6 Scheduling architecture proposal for the downlink case

2.4 User Plane Functionalities

2.4.1 Introduction

The user-plane functionalities in radio access network are usually referred to a group of protocols and algorithms that facilitate higher layer packet transfer through the radio interface. In this section, generic

[9] N.B This pooling of resources is just one part of the overall partitioning of time-frequency-spatial resources, e.g. between different transport channels (see Annex I section 8.2.3.3 for more details).

user plane functions are studied. This work is continuation for the previous studies in [1] where following generic user plane functions are identified;

- Header compression
- Segmentation and reassembly
- Retransmission protocols
- In-sequence delivery
- Link security
- Flow control

In the rest of this section, following user plane functions are studied in more detail;

- Retransmission protocols
- Broadcast/multicast protocols

2.4.2 Retransmission Protocols

2.4.2.1 Introduction

An important issue that needs to be taken into account is that the information transfer over the radio interface is always exposed to uncertainties and thus error correction techniques are essential. Reliable information transfer is especially important when carrying transmission control protocol (TCP) traffic, which may mistakenly interpret radio interface errors as congestion and therefore unnecessarily reduce the transmission rate.

A widely used error correction technique is the class of retransmission protocols where the receiver sends requests for the sender to repeat data unit transmissions whenever errors are detected [70], [71]. In current standards, multiple retransmission functions are often located at different protocol layers on the top of each other [75]. Lower layer retransmission aims to correct transmission errors on the physical channel whereas higher layer retransmission ensures reliable information transfer over the radio access network and its different interfaces. Similar kind of functions is also proposed for the WINNER radio interface where separate hop-by-hop and end-to-end acknowledgements are found to be useful — especially for different types of relaying scenarios [76]. The objective of this section is to further develop the proposed approach in [76] to a common and flexible framework that is suitable for WINNER scenarios and within which any specific retransmission protocol may be used. Moreover, following requirements are identified for the framework.

- It should be able to handle worst-case scenarios (bad links, multiple hops, frequent inter-cell and/or inter-mode handovers due to high mobility and/or mobile relays, …)
- It should support resource (e.g. memory) optimization features.
- It should be able to operate independently of other radio interface functions, but also be able to take advantage of side information from other functions if available (e.g. RTT estimation for timer settings).
- It should support all envisioned deployment scenarios (including e.g. relaying and distributed antennas).
- It should not need any interaction with higher layer protocols, but could take advantage of such side information.
- It should provide an interface towards higher layers being as wire-like as possible, under the delay constraints; in order to hide non-congestion related data losses (e.g. for the sake of correct TCP operation).
- It should support both reliable and non-reliable relays.
- It should support cooperative relaying.

As indicated above, there should be the possibility to use extremely robust configurations that can handle difficult situations where nothing is reliable and everything may change quickly, e.g. high user mobility, mobile relays, several unreliable hops, inaccurate measurements, resource allocation uncertainties etc, i.e. we may have situations where we cannot rely on the RNs and their management schemes. In the other extreme, there should be the possibility to disable the relaying ACKs in case of highly reliable RNs and/or cooperative relaying, where a packet is simultaneously transferred through several parallel paths and therefore an error in the first hop is not fatal because the final receiver may receive several copies of the same packet and/or perform packet combining resulting in a successful decoding in the end node.

2.4.2.2 WINNER ARQ Framework

In this section a retransmission protocol framework is outlined tailored for the WINNER radio interface. Section 2.4.2.2.1 discusses a traditional layered retransmission protocols for error recovery, whereas section 2.4.2.2.2 proposes a cross-layer solution for error recovery. Both solutions fulfil the identified requirements.

Throughout these sections we will use a simplified protocol architecture in which we do not distinguish between RLC, MAC-g and MAC-r (i.e. we do not distinguish between mode-specific and generic functionalities as this makes it easier to describe the core of the two proposals). The mapping of these two solutions onto the proposed protocol architecture will be dealt with in section 2.4.2.3. It has furthermore been assumed that all retransmission functionalities are placed in the data link layer, however there is nothing that prevents us to move some or all of the functionalities to other layers (e.g. put the lower layer ARQ functionality in the physical layer, especially Hybrid ARQ type II schemes).

2.4.2.2.1 Layered ARQ for error recovery

The most straightforward solution to embody what has been concluded above 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 2-7.

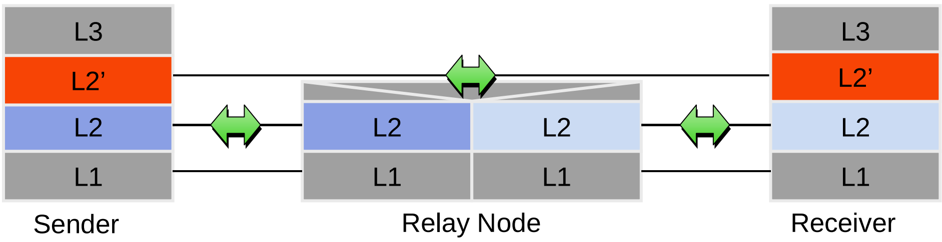


Figure 2-7 Layered ARQ for error recovery.

However, such a simple and straightforward approach has some inherent problems: i) ARQ protocol layering may potentially cause harmful interactions.[10] 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 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 Annex I section 8.3.1.1.2.

2.4.2.2.2 Relay-ARQ for error recovery

A retransmission protocol overcoming many of the aforementioned drawbacks of layered retransmission protocols will be 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 2-8). 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.

[10] 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.

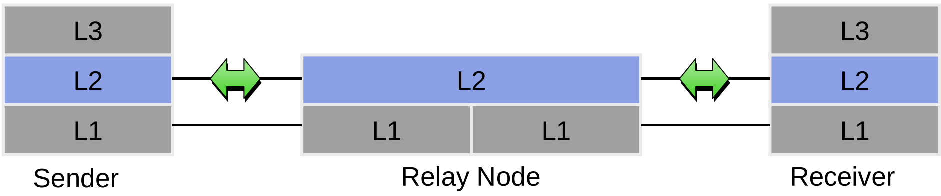


Figure 2-8 Common ARQ layer for complete route.

As in the preceding cases, the retransmission of lost data is separated into a two-stage error recovery process. When an intermediate RN 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 RN may take back the retransmission responsibility in case the next RN drops out of the connection. If on the other hand a RN joins the route they build up a soft-state ARQ protocol state for this connection.

Figure 2-9 depicts more details of the Relay ARQ operation in a two-hop scenario. A single relay acting as a receiver towards the sender and as a sender towards the receiver as depicted in Figure 2-9. It maintains a local ARQ window as well as receiver- and transmitter-side status bitmaps. Moreover, the RN receives status messages from the next node (RN or receiver) on the way towards the receiver, and it sends status messages to the next node (RN or sender) on the reverse path towards the sender.

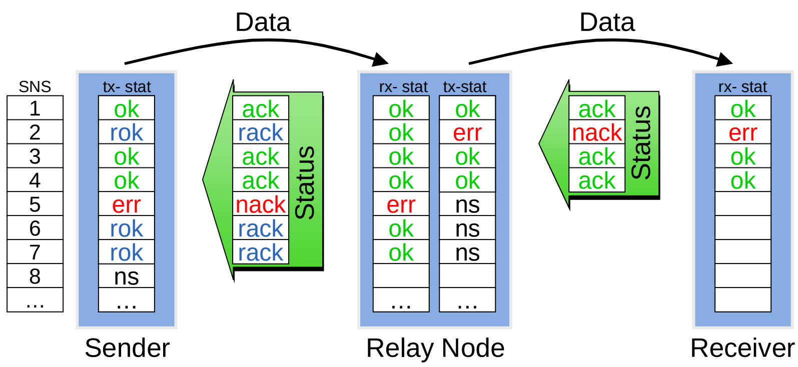


Figure 2-9 Basic Concept (Relay-ARQ and 3-State Bitmaps)

Figure 2-9 depicts how a status message sent by the final receiver is updated by the RN and forwarded to the transmitter (the various states in the transmitter- and receiver- windows as well as in the status messages are explained in Table 8-2, Table 8-3 and Table 8-4 in Annex I section 8.3.1.1.3). As can be seen in the figure, the main difference to a layered approach is that the RN takes the state of both the sender and receiver window into account to build their status messages (see Table 8-6 in Annex I section 8.3.1.1.3 for details).

More details about Relay-ARQ and its potential benefits may be found in Annex I section 8.3.1.1.3.

2.4.2.3 Mapping onto the WINNER Architecture

The ARQ framework should rely on the logical node definitions as defined in the WINNER Architecture Proposal in [2] and summarized in chapter 2.2.3. In this chapter we discuss the mapping of the ARQ functions onto these logical nodes. Note that the overall ARQ performance depends crucially on the mapping of these logical nodes onto physical nodes: co-location of certain logical nodes in the same physical node, the characteristics of the links between the physical nodes and the reliability of the physical nodes. The appropriate mapping of the logical nodes onto envisioned physical nodes in targeted deployment scenarios for the different WINNER modes is a subject for further in depth studies, and is briefly addressed in chapter 2.4.2.4.

The ARQ framework provides error recovery both end-to-end and over each wireless hop. In this section, the two protocols are mapped onto the logical nodes and the functionality of the different protocols is discussed (for notational simplicity, we drop the LN subscript on the nodes).

2.4.2.3.1 Layered ARQ

In the layered ARQ approach, retransmissions are performed both by RLC-g and MAC-r. RLC-g is responsible for the end-to-end error recovery, regardless of the number of hops. It is placed in the ARQ end points, the UT and the RANG, and is used to recover from data losses due to handover and buffer overflow. The MAC-r protocol can be the same (homogeneous relaying) or different (heterogeneous relaying) at each hop, and it terminates in the peer nodes of each hop, as illustrated in Figure 2-10.

Mode specific recovery of transmission errors over each hop is performed by a hybrid ARQ protocol in MAC-r. The Hybrid ARQ protocol could be completely specific for a certain MAC-r, but it is likely that at least a part of the link level Hybrid ARQ scheme optimized for a specific WINNER PHY mode can be made generic for all WINNER PHY modes (i.e. part of the Generic Link Layer, GLL).

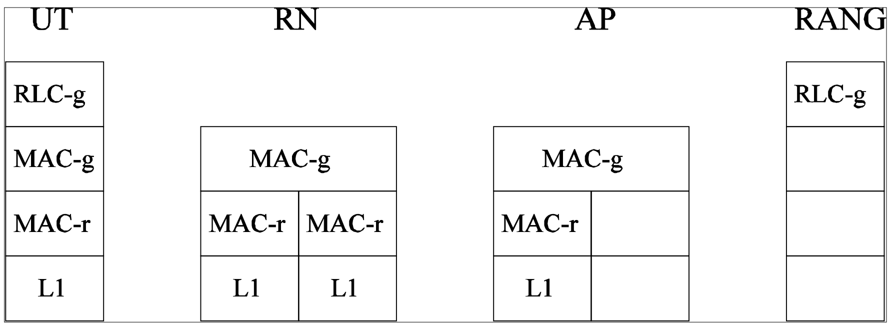


Figure 2-10 Layered ARQ, logical nodes and protocol entities

2.4.2.3.2 Relay-ARQ

In the Relay-ARQ approach, the retransmission functionality used in end-to-end and hop-by-hop is combined and optimized for multi-hop communication. The combined retransmission functionality is placed in the end points (UT and RANG) and in all intermediate nodes. Figure 2-11 illustrates the placement of the retransmission functionality corresponding to RLC-g and MAC-r. End-to-end error recovery is achieved, since hop-by-hop retransmission feedback is propagated to the next node in the communication path. In a multi-hop scenario including an RN, when RCL-g in e.g. the UT receives a correct frame, then it transmits an ACK to the RN which propagates the ACK to the AP which in turn propagates the ACK further to the other end point, the RANG.

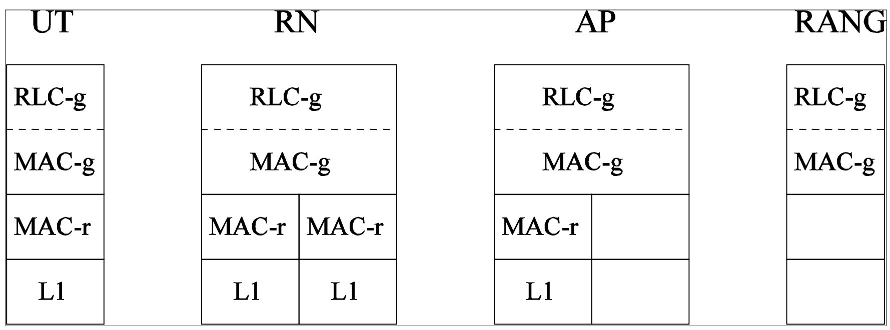


Figure 2-11 Relay-ARQ, logical nodes and protocol entities

More details about mapping of the ARQ framework onto the WINNER architecture can be found in Annex I section 8.3.1.2.

2.4.2.4 Envisioned Configuration in Different Deployment Scenarios

The basic characteristic of the WINNER RAN will be the multiple operating modes and deployment scenarios. For instance, in a short range scenario, ACS, RANG and AP could be co-located in the same physical node. In a wide area scenario with high mobility and less radio resources, on the other hand, the ACS and the RANG could be centralized and serve many APs. A problem arises, since a common retransmission scheme for all modes and scenarios may often result in very conservative system design with excessive signalling and buffering requirements. One possible example is the selection of termination points of the end-to-end retransmission scheme. As the UT may perform handovers between RNs that are associated with different APs appropriate termination points for the end-to-end retransmission scheme seem to be the UT and RANG. However, the requirement for a successful retransmission during handover is that the ARQ termination points should be one hop further than the handover point so as to avoid any loss of data. As the UT may handover between RNs or APs the ARQ termination points should be capable of being transferred from between different types of nodes such as an AP and the RANG according to the deployment scenario. Consequently, signalling burden over the transport network may be reduced. In addition, end-to-end ARQ may take advantage of the reduced number of intermediate nodes.

Moreover, in the uplink direction the issue of data losses during handoffs does not arise since even if the sender (UT) is handed over, the path to the receiver is already established and the next hops should be able to forward the data units. A simpler retransmission scheme could be therefore applied whereby the retransmission window advances and buffered data of a node are released upon the receipt of an acknowledgment from its next hop. That way the signalling and buffering requirements are minimised. An important requirement for the employment of this scheme is the reliability and functionality of the RNs and therefore a periodic assessment of the RNs might be needed for each deployment scenario by a management entity.

Another particularly interesting and promising proposal within the WINNER framework is to exploit the concept of cooperative relaying. One possible retransmission protocol candidate for cooperative relying requires two phases, at least initially, while further improvement can be achieved by avoiding the second phase if the destination has successfully decoded in the first one. Using feedback from the destination, additional information is provided by the relay only upon explicit request. The source transmits half of its message during the first time slot of duration in a broadcast manner to the relay(s) and destination. Relay(s) and destination then both try to decode the message and send feedback on their decoding status in a broadcast manner to the other nodes. Consequently, all nodes have all necessary information to act appropriately during the (optional) second time slot. In case the destination was not able to correctly decode the source message while the relay was able to decode it, the latter will send additional redundancy in the second time slot. The destination then assembles the complete codeword and retries decoding. In case of decoding failure, a block error is declared. The same occurs when destination and relay simultaneously fail to decode.

A more detailed description of the proposed techniques is given in the Annex I section 8.3.1.3.

2.4.3 Broadcast/Multicast Protocols

2.4.3.1 Introduction

The broadcast/multicast service (BMS) will be an important service in future wireless mobile telecommunication. The broadcast service is to send the information to a group of users based on their location. The coverage area is defined by a set of cells/sectors which is defined by the radio technology. This could be considered as “un-addressed” messaging. Examples could be location based services, such as local shop information, local traffic information. Multicast is the ability to broadcast information to a specific set of users based on their subscription to user group. Example could be all students registered in a university department. The whole end-to-end service provision involves multi-layer functionalities, which is beyond the scope of this protocol specification. In this section, only functions in layer 2 are described.

Broadcast/Multicast services (BMS) are intended to provide flexible and efficient mechanisms to send common information to multiple users. The motivation for these services lies in the need to utilize the most efficient method of delivery for the use of air interface and network resources when sending the same information to multiple users. The only service offered in 2G and current 3G is Short Message Service (SMS). The protocol model in 3G is shown in Figure 2-12. [92] .

In a mobile system with multi-hop relay actually adds another possibility of the radio resource allocation for Broadcast/Multicast Control (BMC). The Broadcast/Multicast packet can be delivered to the user directly or through a multi-hop path. Whether using single hop or multi-hop depends on the overall available radio resource, broadcast/multicast service requirement and the choice of the more efficient way to use radio resource. Broadcast/Multicast also requires routing function to send the packet efficiently, only duplicate the packet when the paths diverge close to the broadcast/multicast members.

The transmission at the last hop usually use Unacknowledged mode, because it would be too cost to collect all ACKs from all users in the cell and repeat transmission for just individual failure. But this conclusion might change when only a few users are in the cell. ARQ may be used in the transmission among the RNs and AP.

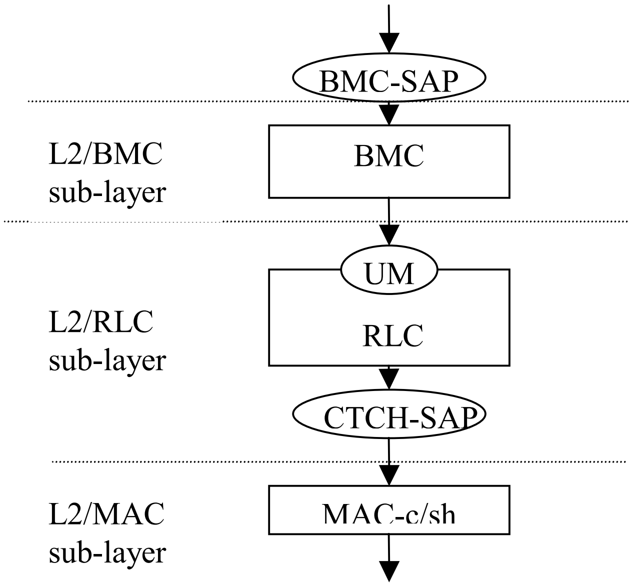


Figure 2-12 3G BMC protocol model

The functions of BMC protocol are:

- Storage of Cell Broadcast/Multicast Messages.
- Scheduling of BMC messages
- Traffic volume monitoring and radio resource request
- Delivery of Cell Broadcast/Multicast message to upper layer

The transmission at the last hop usually use Unacknowledged mode, because it would be too cost to collect all ACKs from all users in the cell and repeat transmission for just individual failure. But this conclusion might change when only a few users are in the cell. ARQ could still be used in the transmission among the RNs and AP. From the above model we can see an unacknowledged mode (UM) is used in RLC sub-layer. Broadcast usually is originated from Cell Broadcast Centre (CBC). Multicast can be originated either from CBC or from a user. The service users can be invoked through paging or broadcast schedule information. The multicast service can be limited through using layer 2 encryption.

2.4.3.2 Problem identification and possible solution

BMS need to be delivered with high reliability and efficient use of radio resource. The strategy can be varied under different conditions.

Repeated transmission of the same message over a period of time will reduce the transmitting power requirement by using time diversity, which may improve the spectrum efficiency. This method can only be realized for the messages with long delay tolerance. According to different delay tolerance, the repetition time period can be adapted to the different requirement.

For delay sensitive BMS, more intelligent radio resource management methods are needed to improve the radio resource efficiency. A layered multicast protocol has been proposed in Internet broadcast/multicast to adapt the dynamic change of the available bandwidth in the network and the data-rate capability of the receiver.[93] Usually the multimedia traffic is coded in multi-layers, with higher layers offering better quality of the information but requiring higher bandwidth to transport. Lower layers carry basic information with higher priority. Multi-layers can be transported in multiple traffic flows simultaneously. The adaptive schemes are to change the number of layers transported in the network, matching the condition of the available bandwidth of the network.

This principle can be extended to the wireless link which has much quicker dynamic change of channel condition. Each layer with different priority can be transmitted with different power or different coding/modulation scheme. The overall radio resource for one information session can be budgeted according to the available radio resource in the cell and priority of the session compared with other sessions. The average radio resource usage could also depend on the slow feedback indication of link quality. But how to utilize the feedbacks from many users remains a challenge. One possible solution is to use the user's location information to derive the average radio resource.

The nature of broadcast/multicast is geographically limited. Conventional link adaptation technology is designed for single user, which optimise the radio resource usage by adapting the transmitting power, time slot, bandwidth, and coding /modulation scheme according to individual channel condition. Multi-user diversity can be used with above scheme to further improve the overall spectrum efficiency. But for broadcast/multicast service the above scheme is not applicable, this is due to no feedback signalling link for broadcast/multicast. Nevertheless, the above mentioned Layered Multicast algorithm can be used by each user to adaptively select the number of layers broadcasted/multicasted in the cell to match its individual channel condition with minimum loss of the quality.

Also the Broadcast/Multicast usually requires much higher transmitting power in order to guarantee the coverage of the cell. By using the knowledge of the user's location distribution the transmitted power may be optimised.

In the WINNER system, the broadcast/multicast may be offered through different radio modes. When user moves from one cell to another the service may suffer from the change of the available bandwidth. How to optimise the radio resource allocation in the handover scenario is very critical in terms of 'seamless' handover. The above mentioned Layered Multicast algorithm may be optimised in this scenario.

As several BMS may happen at the same time in one cell, the priority of different BMS are different. How to schedule the different BMS is needed in order to optimise the radio resource utilization.

Finally, in case only a couple of users are in the cell, whether to use multicast or single cast needs to be investigated to optimise the radio resource usage.

2.4.3.3 Conclusion and future work

It can be envisioned that a lot of broadcast/multicast services will be available in the future. How to adaptively use the radio resource available in the AP for broadcast/multicast and how to select the proper data rate according to the channel condition needs to be studied. How to support the mobility of Broadcast/Multicast in case of handover is very challenging.

2.5 Conclusion and Future Work

In this chapter, architectural concepts and radio interface functions are proposed to realize the vision of ubiquitous system concept providing access in a large variety of deployment scenarios with one single system. An integral part of the WINNER radio interface architecture is the introduction of distinct radio interface/system modes that are aimed for different scenarios and radio environments. Another important feature is the concept of generic radio interface functions that aims to simplify the overall system concept. The architectural concepts are proposed in section 2.2 whereas generic functions are discussed in sections 2.3 and 2.4.

The radio interface architecture studies indicate that the vision of ubiquitous radio system concept may be realized with relatively simple architectural concepts and building blocks. It is shown, that one single and flexible node architecture model may support multiple deployment scenarios with a manageable number of logical network nodes and interfaces. Moreover, a multi-mode protocol stack together with a unified interface towards upper layers may facilitate seamless interworking between multiple modes and hide the heterogeneity of modes from upper layer protocols and functions. Finally, a simplified channel structure with a relatively small number of channels (and channel types) may handle multiple radio interface modes and deployment scenarios.

The investigations of radio interface functions show that some functions may be generic. Some parts of the scheduling functions may be generic thus simplifying the overall scheduling problem. Furthermore, a common retransmission protocol framework may be used by multiple modes and envisioned deployment scenarios. The proposed frameworks and generic functions may be also easily configured to fit the needs of different technical solutions and radio environments.

Even though the studies so far indicate that the ubiquitous radio system concept can be realized with relatively simple building blocks, there is probably a trade-off between performance and simplicity. An important topic for further work is to study the performance of the concept. Furthermore, the integration of different relaying based concepts into the architecture has not been carried out yet and their impact on the generic functions is moreover unknown. Consequently, some architectural definitions are still rather preliminary and they need to be developed further. In particular, the definition of logical RN and its relation to other nodes — especially logical APs — is a subject for further studies.

3. Identification and preliminary assessment of Deployment Concepts for Harmonization

3.1 Introduction

The objective of this chapter is to summarize some results from the ongoing deployment concept harmonisation work in WP3, focused on:

⇒ collection, classification, grouping of "Basic Concepts", derived mainly from D3.1

⇒ first analysis on the categories of basic concepts, in order to come to a more consistent and organized view on "Deployment Concepts"

⇒ positioning of Deployment Concepts with respect to scenarios

⇒ first examples of "architectural harmonization work", i.e., identification of significant cases of coexistence and complementarity of different concepts within a reasonable scenario.

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.

The deployment concepts investigated in WINNER have been evaluated for some selected scenarios, but should allow enough flexibility for deployment in all real environments.. Specific aspects related to physical and protocol functionalities of each concept, as well as their specific harmonization, are out of the scope of this chapter.

The analysis reported in this chapter is qualitative and preliminary. It can be considered as a first (but necessary) step of the processes of Deployment Concept definition, selection, positioning and harmonization. This first step consists of collecting the basic concepts reported in D3.1 (in order to guarantee continuity and consistency along the WP3 deliverables), organizing them, performing a first analysis about priorities and inter-relationship, positioning with respect to the main WINNER scenarios. In summary, the proposed analysis aims at leading from the multiplicity of concepts studied in D3.1 to a more focused and organized view on prioritized concepts and eventually identify missing deployment concepts and missing parts of some specific deployment concepts.

Many of the concepts recalled in this chapter are further investigated in this deliverable. The "wide overview" proposed here should be considered as complementary to the focused analyses on single concepts reported in chapters 4 and 5. The classification, abstraction and simplification of concept categories is performed in this chapter for the exclusive purpose of harmonization and does not preclude that the concepts and categories are studied and assessed from other viewpoints, within the project, however it is expected that investigations will continue on the harmonized concepts when available, in order to meet the overall WINNER goals.. Consistently with what was stated above, this contribution to D3.2 should be considered just as a first report of the results of the overall deployment concept harmonization activity, which is still on going. In the future it is required to take the new WINNER air interface technologies as identified by WP2 into account.

3.2 List and classification of Basic Concepts

In this section, the main basic concepts included in D3.1 are synthetically recalled, as a starting point for the concept assessment, reorganization and harmonization. The D3.1 approach will be reconsidered in the following section, in order to come to a more consistent, general and at the same time simplified view.

The Basic Concepts are classified into five main categories:

- Single Hop Concepts
- Fixed Homogeneous Multi Hop Concepts
- Fixed Heterogeneous Multi Hop Concepts
- Mobile Multi Hop Concepts
- Cooperative Multi Hop Concepts

3.2.1 Single Hop Basic Concepts

The "Single-Hop" solution can be seen by itself as a major Deployment Concept, that can be realized in a multiplicity of different ways, many of which are already existing/evolving (indeed, the main systems currently deployed are single-hop; WINNER explicitly distinguishes the single-hop feature because it also might be taken into account by multi-hop solutions).

In D3.1 an analysis was performed not on different "single-hop concepts", but, on existing, evolving and emerging single-hop systems/technologies, that seemed worth being analyzed as possible "sources" of specific "ideas" (e.g., in terms of MAC, LLC, RLC, RRC functionalities and protocols) to be possibly kept and properly combined with new ideas in the context of the development of the new and original WINNER air-interface. This is the sense that should be assigned to the following items, i.e., "background references" whose features (or even enhancements) can be usefully exploited in the context of WINNER; being clear, however, that no existing unchanged technology will be used *per se* as a WINNER mode.

Few words are recalled for each "background technology"; more detailed analyses were reported in D3.1 [1].

- **3G / HSDPA-Enhanced Uplink** should be considered, in the current and short-term evolutionary landscape, as the background reference for wide-area cellular (3G) and data-oriented "public hot-spot" (HSDPA-Enhanced Uplink) application scenarios. 3GPP has recently started to develop a new OFDM-based physical layer; and can be seen as an emerging and still evolving technology.
- **WiFi / 802.11** should be considered, in the current and short-term evolutionary landscape, as the background reference for data-oriented "indoor coverage" (small hot-spot application).
- **HiperLAN2** has the same target as 802.11, but is less deployed than 802.11. However, in the context of HiperLAN2 some interesting advanced solutions are studied, which should not be lost. These solutions may be considered as valid ideas to be properly exploited in the context of the development of the new WINNER air-interface. An example is the so-called Centralized Ad-Hoc Network Architecture (CANA). This concept is suitable to short range scenarios (indoor and hot spot) and fits in the context of the WINNER WP3 relay based concepts. This architecture is capable of supporting very high bit-rate applications, allowing for a dual mode of operation in the traditional HiperLAN/2; the frequency channels at 60 GHz offload data traffic in dense environments with low mobility while 5 GHz can always be a back-up frequency when users move faster.
- **802.16 / WiMAX** should be considered, in the short/medium term, as a reference solution for the "Wireless MAN" application scenarios. WiMAX is related to a known concept (standardized by 802.16-2004), but it is not yet a deployed network. <u>802.16-2004</u> is mainly a Fixed Broadband network technology, i.e., a Wireless DSL system, providing data services to the final user. It can be also seen as a "WiFi extension", in the sense that it ensures not only indoor but also outdoor coverage over cells of dimensions comparable to the mobile network ones. Initially, stand-alone WiMax User Terminal (UT) will be deployed, with costs and dimensions low enough to be suitable to residential customers; it is envisioned that UT will evolve towards miniaturized cards to be integrated in PC/laptops. Large public hot-spots up to cellular coverage of entire urban areas will be supported, ensuring <u>stationary portability</u>: the possibility of connecting to the broadband wireless data network from “anywhere”. The further <u>802.16e</u> step, currently being defined, will then ensure <u>nomadicity</u>: the possibility of keeping session continuity while moving. It will also support full mobility (including inter-cell seamless handover), if connected to a proper transport/core network structure. To conclude, the foreseen roadmap for WiMax is as follows: it will initially address fixed broadband, then (potentially) nomadic access, and then even mobile wireless access. <u>From the viewpoint of technology profile</u>, the 802.16 physical and MAC layers incorporate some of the more advanced technologies; e.g., the physical layer is based on OFDM, intrinsically open to evolve by including adaptive techniques, smart antennas, antennas diversity and MIMO. It is also open to support Mesh architectures. This item is for the main aspects converging with the wireless multi-hop concepts. Therefore, some specific aspects of 802.16 solutions might become candidates to be evaluated as a basis for specific aspects of the new WINNER air interface.

3.2.2 Fixed Homogeneous Multi-Hop Basic Concepts

The Fixed Homogeneous Multi-Hop Deployment Concepts are an essential category in WINNER. The main concepts of this category that were taken into account in D3.1 [1] are briefly recalled below.

- **MAC-frame based Fixed Relay in the Time Domain**

The concept is based on the **time-domain** discrimination of Multi-Hop (MH) and Single-Hop (SH - to be intended here as "LastHop") links within a frame. It is focused on significant application scenarios; e.g., it seems suitable to "Manhattan grids", in the realization where a single cluster is composed by one AP plus four FRN (with directive antennas at the relay for communication with the AP and omni directional antennas at the relay for communication with the UT). In general, it seems suitable to urban area, high bit-rate scenarios, including hot-spots. Concept enhancements in order to deploy it in other scenarios need to be further investigated. This applies also to the exploration of space shadowing concepts in which shadowed FRN transmit simultaneously in the same resource (e.g., time, frequency, code). Common for this concept is the control of the MAC frame (e.g., scheduling), which is centralized in the AP.

- **Fixed Relay in the Time Domain through TDMA clustering**

This concept is based on the time-domain as well, but, unlike the concept described above, each AP and each relay has dedicated resources assigned and fully controls these, in terms of scheduling. Deployment is based on the definition of sub-networks with one AP and a number of relays per sub-network where Multi Hop communication takes place between the AP and the UT utilizing the RNs. This concept supports different kind of scenarios and hierarchical PMP (tree-topology) as well as Mesh architectures. On the other hand especially the relays become more complex from a higher layer perspective since more protocols and functionalities need to be supported.

- **Fixed Relay through "hierarchical PMP", tree-topology, in 802.16 TDD mode**

This concept is similar to the previous ones, although based on the specific 802.16a technology.

- **Fixed Relay through "Mesh" in 802.16 TDD mode**

This concept is similar in principle to the previous ones. It is based on the specific 802.16 technology, and includes the possibility of Mesh CS (centralized MAC control) and Mesh DS (distributed MAC control).

- **Fixed Relay through combined Time- and Frequency-domain**

This concept introduces a new principle. It exploits not only the time-domain, but also the frequency domain, in order to allow the coexistence of multi-hop and last-hop links. In fact, different frequency bands are used for multi-hop and last-hop. In this context, it is even possible to conceive dual-band multi-hop. A simple significant example of this concept is the coexistence of 802.16 Mesh and PMP in a particular access network deployment, assigning to Mesh and PMP different frequencies. The concept is also valid for heterogeneous relaying (e.g., based on 802.16 and 802.11).

- **Fixed Relay through combined TDMA and OFDMA concepts**

This deployment concept brings the principle of using different "dimensions" to discriminate between Multi-Hop and Last-Hop even further. In this case, the frequency dimension is represented by the sub-carriers of OFDM, which are used to discriminate between last-hop and multi-hop links.

The basic feature of this concept is the dynamic (although on a semi-static timescale) assignment of OFDMA sub-carriers to MH and SH, that allows to take into account the clear distinction between MH and SH links, characterized by different requirements. Some further enhancements of this concept (spatial frequency reuse, fixed directional antennas) can be considered.

- **Fixed Relay enhanced through Smart Antennas (space dimension)**

This concept can be considered as a very advanced and interesting case, where the introduction of the smart antennas allows the full exploitation of the space domain, besides the time and frequency domains. In principle, this means to have a time plus frequency plus space domain based system. Actually, it potentially brings advantages by allowing simultaneous transmission by more (all) FRN. In other terms, the situation obtained by lucky shadowing effects, in case of omni directionally emitting FRN, is here a deterministic effect achieved through Smart Antennas.

- **Temporarily Fixed Relay**

This concept is referred to relays that, while characterized stationary during operation, are temporarily deployed in a given position through some kind of vehicles. This fact allows a more flexible deployment with respect to the needs, available “on-demand” e.g., for big events; typical examples are provision of coverage at football grounds, accident scenes. It is a complementary/additional solution with respect to the classical fixed relay concept. In D3.1, this concept was defined "Movable Relay". In the classification proposed here, the conceptual similarity with fixed relaying has been highlighted. Nonetheless, some differences can be identified with respect to the other fixed relaying concepts, e.g., the increased need for self-configuration capabilities, in order to allow fast and easy re-deployment.

3.2.3 Fixed Heterogeneous Multi-Hop Basic Concepts

The category of "fixed heterogeneous multi-hop deployment concepts", according to deliverable D3.1 [1], includes cases that are conceptually different from one another. The common aspect of all the heterogeneous multi-hop cases is the fact that the access network is composed by two or more "segments", implemented by different wireless technologies.

- **Heterogeneous Fixed Relay**

This concept refers to the "heterogeneous relaying", strictly speaking. It allows a multi-hop connection through AP and RN, the RN being characterized by two different modes. This implies a more complex RN since it requires support for two different modes. It can be useful if (when) the requirements related to the different spans of the multi-hop link are so different (e.g different motilities) to justify a separate optimization by two different technologies.

- **Feeder deployment concept**
- **Radio over Fibre**

These two concepts are recalled here according to the D3.1 classification. However, they can be included in the heterogeneous relay category only in a very wide sense. Therefore, they are just briefly recalled.

When the feeder segment is implemented through a wireless link, e.g., by deploying a microwave point-to-point or point-to-multi-point system, a similarity with the previous case apparently arises. However, the feeder is a "transport network concept" and not a "multi-hop concept" in the sense of WINNER. In fact, in this case, a terminal has not direct access to the feeder wireless link.

The Radio over Fibre (RoF) concept is even farther from the WINNER "multi-hop concept". The "digital Radio over Fibre" can be seen, from a logical viewpoint, as splitting the AP into two parts, a "central" one plus a "remotized" one, connected through a high bit rate digital interface over optical fibre (this interface is under standardization by the CPRI consortium). A further potential advantage of RoF is the improved possibility for fast coordination between access points (sites) e.g. to allow fast inter-cell interference scheduling also for cells belonging to different sites.

3.2.4 Mobile Multi-Hop Basic Concepts

"Mobile Multi-Hop concepts" described in D3.1 are briefly recalled in this section.

- **Multi-hop through dedicated Mobile Relays Type I**

This concept consists of a Relay which is stationary with respect to a moving vehicle, where some UTs are travelling. The Relay is moving, in absolute terms (with respect to the earth) but it is stationary with respect to the terminals it is serving. It is the case for example of mobile relays associated to trains, serving the terminals in the train, that are stationary with respect to the relay (while the ensemble "relay + terminals" is moving). It should be noted that this case will most likely employ some form of heterogeneous relaying. This deployment concept is interesting and different from the other cases; it is well focused on a specific realistic scenario, i.e., a "moving network".

- **Multi-hop through dedicated Mobile Relays Type II**

A conceptually different situation with respect to the previous cases is here considered: the relay is moving (not stationary) while working; it is also moving with respect to the terminals it is serving. The objective is to allow a very flexible deployment with respect to the needs. This concept seems to be complementary/additional with respect to the fixed relay concept, rather than as a stand alone solution. It may be fitted on buses and other public transport means to provide coverage in parks, streets, etc.

- **Multi-hop through Terminals acting as Mobile Relays**

In this concept, the UTs themselves are enhanced to provide possibly the relay functionality. It is clearly a concept that potentially allow the maximum of "spreading" and "diffusion" of the relaying option. However, it may be actually affected by several and severe limitations: for example, not guaranteed presence; non homogeneous distribution in space and time; critical power issues; switching off by the user; non availability of acting as relay by the user, etc. Moreover, this kind of RN requires auto-configuration capabilities. On the whole, it is seen as an 'additional" concept. It is not considered suitable as a basic option to deploy a mobile network. An application example is a hot spot/urban area, where medium/ high end terminals are used to provide relaying functions to other low/medium-end terminals.

- **Pure Ad-Hoc Deployment**

In this concept, the terminals act as relays, and perform direct (peer to peer) communication, in a completely independent manner (i.e. without involving any infrastructure), or are assisted by a network node at least for transport of signalling messages. Pure Ad-Hoc networking should not be considered, further as being outside the scope of WINNER.

3.2.5 Cooperative Multi-Hop Basic Concepts

- **Cooperative Relaying**

"Cooperative Relaying" combines the basic ideas of multi-antenna systems and of relay-based networks. The term "cooperative" is due to the fact that in this context user terminals are capable and allowed to "cooperate", in order to build *distributed antenna arrays* able to overcome the drawbacks of correlation and spatial limitations. In cooperative relaying, spatial diversity achieved through the network topology itself. In general, cooperative networks employ relay nodes that receive signals from a source and resend a processed version of these signals to the intended destination. The destination node *combines* the signals from source and relays, thereby exploiting useful information that is unnecessarily discarded in conventional relaying or sometimes even regarded as interference. In principle, significant performance improvements may be achieved through cooperative relaying, to be assessed versus the related increased complexity.

While cooperative relaying might be considered under a long-term perspective, it is important to be aware of this concept and to provide means for implementation so as to enable a future evolution of WINNER.

In this preliminary analysis, the cooperative relaying is considered one basic concepts, although many possible "sub-concepts" can be in principle identified.

3.3 Overview on Concept Categories and identification /synthetic assessment of Deployment Concepts

In this section, a synoptical view inside each of the "categories of Basic Concepts" is proposed. The objective of this qualitative analysis is to highlight positioning and reciprocal relationships among concepts belonging to the same category. In each category, similarities and differences are identified.

The overall "map" of concepts is simplified, by reorganizing them according to a more abstract and consistent view, in order to identify relevant "Deployment Concepts" (DC). The proposed simplification will allow keeping trace of "lower priority concepts", to avoid the risk of losing some specific advanced aspects that might be worth being reconsidered as enhancements in further harmonization steps.

Some preliminary remarks about the priority of DC within a category are proposed ("intra-category priority"), as well as about the positioning and priority of an entire category ("inter-category priority").

It should be highlighted that a more detailed assessment will be carried out in the continuation of WP3, in particular by evaluating further parameters related to the deployment characterization, such as topological deployment and type of installation of RN, number of antennas, etc.
The preliminary analysis can be seen as a necessary step along the process of Deployment Concept definition, selection and harmonization.

3.3.1 Synoptical View of Single Hop Concepts

It is proposed to classify the systems reported in section 3.2.1 in the following Deployment Concepts:

1a. Traditional Cellular Concepts (including 3G, in particular HSDPA-Enhanced Uplink)

1b Traditional Wireless LAN Concepts (including 802.11 / WiFi and HiperLAN2)

1c. Wireless MAN concepts (including 802.16 / WiMAX)

The DC 1a. and 1b. are "traditional" concepts; the third one, is an emerging concept. These concepts should be taken into account, at least in the background, for the following reasons.

- All of these concepts are already existing/emerging technologies. It is widely agreed that the WINNER system concept will comprise a Single Hop solution as very important deployment. The Single Hop deployment case has however not been the main focus of WP3 in the past since no new deployments are expected or proposed in this area for the envisaged WINNER scenarios.
- Some of the mentioned technologies are conceptually well known, but not yet completely deployed, or still in phase of evolution, including an "enhancement roadmap", so that they cannot be simply considered as "legacy" (e.g., WiMax).
- Existing/emerging solutions might be exploited as source of ideas for the implementation of specific functionalities, to be included and enhanced in the new WINNER air interface (this is true for Single-Hop, and generally for relay systems too). In the continuation of task 3.3, the mentioned "background technologies" will be benchmarked against from the viewpoint of "functional techniques" (e.g., link adaptation and fast resource scheduling, centralised/decentralised scheduling, network topology), and advantages and disadvantages of different solutions will be discussed.

About priorities, 3G HSDPA-Enhanced Uplink, WiFi and WiMax are to be considered as state-of-the-art technologies, i.e. basic references for the scenarios of fully mobile data-oriented system, indoor hot-spots, and fixed/nomadic high capacity data oriented wireless access, respectively. HiperLAN2 is a secondary option with respect to the widely deployed 802.11, but should not be neglected being the potential source of interesting advanced solutions.

The new WINNER air interface will likely adopt parts from the traditional Deployment Concepts if still “state of the art” (e.g. coding). It is envisioned that the WINNER Single Hop deployment will match the cases given above, however as long as the development of the new WINNER air interface will make progress, its positioning with respect to the mentioned DC needs to be validated and addition of new DC may be considered if required.

3.3.2 Synoptical View of Fixed Multi-Hop Deployment Concepts

In section 3.2.2 and 3.2.3, according to the D3.1 classification and the structure of WP3, two separate categories "Fixed Homogeneous Multi-Hop" and "Fixed Heterogeneous Multi-Hop" concepts have been considered.
However, the preliminary harmonization analysis suggests a different approach.

Let us focus on the "Heterogeneous Multi-Hop" category, first. This includes, according to section 3.2.3, the following concepts:

1. Heterogeneous fixed relay
2. Feeder concept
3. Radio over Fibre (RoF) concept

Feeder and RoF concepts are to be clearly distinguished from all other ones relaying concepts, being conceptually different from the WINNER multi-hop relay principles. A specific protocol harmonization among these two concepts and all the other ones would hence be meaningless. Therefore, for the purpose of concept harmonization only, "Feeder" and RoF" will no longer be considered. Obviously, this fact does not exclude that these concepts (that are anyway important) are investigated within the Project in other contexts (e.g., from the viewpoint of "architectural harmonization").

The Fixed Heterogeneous Relay needs to be clearly considered as one major concept. It should be noted that the "heterogeneous relay" may itself require some sort of functional harmonization between different mode pairs that are used in the HERN. Since the fixed heterogeneous relay can be conceptually considered close to the Fixed Homogenous Multi Hop concepts given in the section above one major category has been identified therefore, from this point on called **"Fixed Multi-Hop Deployment Concepts"**.

This category includes the following concepts for further simplification:

1. MAC-frame based Fixed Relay in the Time Domain
2. Fixed Relay in the Time Domain through TDMA clustering
3. Fixed Relay through "hierarchical PMP" tree-topology network in 802.16 TDD mode
4. Fixed Relay through "Mesh" in 802.16 TDD mode
5. Fixed Relay through combined time- and frequency-domain
6. Fixed Relay through combined TDMA and OFDMA concepts
7. Fixed Relay enhanced through Smart Antennas (space dimension)
8. Temporarily Fixed Relay

It is here proposed to reorganize the basic concepts in the following Deployment Concepts.

2a. Time-Domain based Fixed Relay Deployment Concepts

including the concepts 1, 2, 3, 4 of the abovementioned list

2b. Combined Time- and Frequency-Domain based Fixed Relay Deployment Concepts

including the concepts 5, 6 of the abovementioned list

2c. Combined Time- Frequency- and Space-Domain based Fixed Relay Deployment Concepts

including the concept 7 of the abovementioned list

This means that the way of discriminating between MH (Multi-Hop) and SH (Single-Hop, to better interpreted in this context as "Last-Hop") links has been chosen as the main classification variable (TIME or FREQUENCY or SPACE or hybrid solutions including more than one variable).

In principle, the Deployment Concept 2a. should be built by combining the best of the four sub-tended basic option. In other words, it is recommended to carry out a further "specific functional harmonization" among the basic concepts 1. to 4.

Further classification variables are:

- type of control scheme (centralized vs distributed)
- topology (PMP/ hierarchical PMP vs Mesh)

An additional variable is the nature of fixed RN: static vs auto-configuring. This is independent from the other variables, in the assumption that each of the mentioned deployment concepts can be in principle upgraded through a "functional package" to acquire autoconfiguration properties.

Movable fixed relays will likely need this functionality and may require, if deployed, at least the support of autoconfiguration functions in the fixed relay elements. It is likely that movable fixed relays are tailored to a specific "niche" application scenario only (although they may not be neglected).
The further sub-category **2d. Temporarily Fixed Relay Deployment Concept** is at this stage kept (and included in the tables of the next section), although the possibility of including it as a special case within categories 2a. - 2c. is taken into account, in sight of the future harmonization activity.

About priorities, only preliminary remarks can be given at this stage. If criteria related to "overall performance" are considered, a priority should be probably assigned to 2c., (combined time- frequency-space-domain based concepts), then 2b., then 2a. This analysis will be completed, in the continuation of task 3.3, by a more detailed analysis about "complexity" and "cost" (related to covered areas and requested capacities).

3.3.3 Synoptical View of Mobile Multi-Hop Deployment Concepts

In this case, each of the basic concepts is considered to be a "Mobile Multi-Hop Deployment Concept".

3a. Multi-hop through Mobile Relay serving "stationary" terminals

3b. Multi-hop through dedicated Mobile Relays

3c. Multi-hop through UTs acting as Mobile Relays

Some remarks are here proposed about the positioning and priorities of the mentioned concepts.

In general, due to the "mobility" feature, more severe constraints arise than for fixed multi-hop network. For example, it might be beneficial (or even required) to restrict the number of hops to two (AP→Relay, Relay→UT) or maximum to three.

In the perspective of a gradual evolution, a first focus is on simpler solutions, easy to deploy/maintain and cost efficient (thus low cost). From a preliminary analysis taking into account issues like complexity, concept feasibility, deployment and maintenance cost concepts 3b. and 3c may be seen as viable solutions to improve capacity/coverage in some capacity/coverage limited areas.

Mobile relays serving stationary terminals (3a.) address an important application scenario, sometimes called "moving network". It is worth of further studies. It may bring several advantages, e.g.,

- for the connection between the fixed AP and the moving AP/RN there is only one channel to be estimated/predicted, adapted to and scheduled on in the AP
- multiplexing gains can be achieved (many aggregated flows to the same destination point)
- the link between the UT and the moving AP/RN will be a short range link saving battery in the UT and avoids communication through the Faraday cage created by the vehicle.
- a moving AP/RN with external antenna has a better channel than, and is not energy limited as, a UT.

<u>By considering fixed and mobile relaying together, the following remarks can be done</u>.

- In general, mobile relay can be considered of lower priority than fixed relaying.
- More specifically, mobile relaying concepts 3b., 3c. should be seen as additional (not substitutive) with respect to fixed relaying. They can potentially be very useful to improve capacity in already covered networks, for the single user (e.g., by lowering the overall path loss) as well as for the network as a whole (e.g., by lowering the overall interference levels).
- On the contrary a medium/high priority should be assigned to the concept 3a. because it addresses a new and interesting application scenario (moving networks).

- A specific harmonization between mobile relaying, on one hand, and fixed relaying, on the other hand, should not be a focus of the functional harmonization process. Instead, mobile relaying will probably require a set ("package") of "additional" functions with respect to fixed relaying.
- Mobile relays should be "combined" with fixed relay concepts into reasonable "scenarios" in the context of a architectural harmonization, with the possible objectives of improving coverage or extending coverage, or both.

3.3.4 Synoptical View of Cooperative Multi-Hop Deployment Concepts

The "Cooperative Relaying" has been considered in section 1.2.5 as one Deployment Concept.

5a. Cooperative Multi-Hop Relay

This DC includes several "sub-concepts" as described in D3.1 [1]. They differ in protocol nature (static vs. adaptive), forwarding strategy (amplify-and-forward vs. decode-and-forward) and other parameters.

Adaptive decode-and-forward protocols are promising. Their adaptive nature limits error propagation by forwarding only when a certain quality level (e.g. SNR, CRC check etc.) can be guaranteed. Their decode-and-forward property makes them similar to conventional store-and-forward relaying, which is attractive from an implementation point of view.

The system connectivity analysis has shown that for amplified relaying and decoded relaying without error propagation the priority is to maximize the connectivity of the destination terminal, while for decoded relaying with error propagation the priority is to equalize the connectivity of the destination and relay terminals. In this respect, using multiple antennas at relays can significantly enhance the performance. Implementations using selection combining require only a single RF chain (receive or transmit); therefore, they offer a good trade-off between cost and performance.

Using just two hops is well-known for its simplicity with respect to routing and resource allocation. In addition, even under ideal conditions with respect to combing and capacity exploitation, the two-hop approach is optimal for a wide range of targeted rates and path loss exponents in fading channels as it provides a trade-off between diversity gains and path loss reduction on the one hand and rate increase and repetition coding on the other hand.

3.3.5 General remarks on Deployment Concept Definition and synthetic assessment

Some preliminary and qualitative results have been achieved by the overview analysis on Deployment Concepts that has been reported in the previous sections. Summarizing remarks are reported below.

- The main existing and emerging technologies, in the field of **single-hop deployment concepts** (here classified as "Traditional Cellular, Traditional WLAN, WMAN), are studied - at least as sources of ideas for the functionalities of the new WINNER interface. In particular, the technologies that are still evolving (leading to "enhanced single-hop solutions") should be considered for this purpose. Moreover, the understanding of the evolutionary relationships among the existing/evolving systems is important in order to properly positioning the WINNER solution and outline a consistent and effective migration path from current to future access networks.
- A high priority should also be assigned to the overall **Fixed Multi-Hop Relaying** category, taking into account the intrinsic innovation of its concepts, which outline a new structure (not yet deployed) for the wireless access network. This category includes concepts that can be implemented through either Homogeneous or Heterogeneous relaying, where heterogeneous relaying is especially referred to the combination of WINNER mode pairs (it can be noted that heterogeneous relaying concepts are intrinsically based on a kind of harmonization). The stand-alone categories of "heterogeneous relaying" and “fixed homogeneous relaying” are no longer considered for the purpose of harmonization.

  The heterogeneous Feeder and RoF concepts will not be taken further into account for the harmonization purpose.

- Focusing on the Fixed Multi-Hop Relaying category, the view is simplified by grouping the basic concepts into three major Deployment Concepts: "time-domain based relay", "combined time and frequency domain based relay", "combined time, frequency and space based domain relay". In this case, a "specific harmonization work" is necessary to harmonize the related sub-concepts. **Combined frequency and time domain concepts** (especially the "*hybrid TDMA and OFDMA system*") potentially achieve a good trade-off between high performances and acceptable complexity. ***Time-domain concepts*** should be considered as the first evolutionary step, while ***Combined time,***

frequency, space domain concepts are important as longer term solutions and belong to a major evolutionary trend.

- The category of **Mobile Multi-Hop Relaying** should be considered as a complementary/additional perspective with respect to fixed relaying. A specific inter-category protocol harmonization between fixed and mobile relaying is not a priority. Rather, the functionalities related to the different Mobile Relaying concepts should be grouped into "packages of additional functions" to be gradually added to the basic package of Fixed Relaying functionalities. A particular importance should be assigned to the DC ***Multi-hop through Mobile Relay serving "stationary" terminals***, because it addresses the "Moving Network" application scenario.
- The category of **Cooperative Relaying** offers a variety of concepts, ranging from fully distributed virtual antenna arrays to simple extensions of conventional relaying as discussed within WINNER. While all of these concepts have received strong interest in the research community, it can be concluded that only some of them are relevant for the objectives in WINNER. In particular, simple adaptive decode-and-forward protocols have been identified that are attractive from an implementation point of view, while still offering tremendous performance improvements for scenarios in which conventional spatial or temporal diversity is not available or exploited.
- Finally, it should be noted that the general Deployment Concepts considered in this section do not coincide with "modes". Once the harmonization of concepts have been completed, the resulting final concept will probably be specialized into more than one mode, tailored for different scenarios by optimally considering variables such as FDD vs TDD. About these topics, WINNER is active in WP2 especially on the PHY and MAC layer, while WP3 will focus on L2 protocols.

3.4 Deployment Concepts with respect to scenarios

As a pre-requisite for the selection/harmonization process, the considered Deployment Concepts must be mapped to relevant application scenarios. A preliminary, synthetic qualitative overview of the relationships between Deployment Concepts and main WINNER scenarios is proposed in this section through five simple tables "concepts vs. scenarios", one per each concept category.

The Deployment Concepts are all the ones listed in the previous section, classified into four categories:

- Single Hop
- Fixed Relay
- Mobile Relay
- Cooperative Relaying

This section aims on mapping the different approaches and their respective sub-divisions to the following scenarios which have been chosen by D7.2 as main WINNER focus:

- Scenario A.1: Indoor Hot-Spot
- Scenario B.1: Hot Area (wide area but non-ubiquitous coverage), typical urban
- Scenario C.2: Wide area (ubiquitous coverage), typical urban
- Scenario D.1: Rural (ubiquitous coverage)
- Scenario B.5: LOS Stationary Feeder for hot area

The reported tables summarize the "concept vs scenarios" mapping. A column "Notes & Characteristics" is added, including some remarks about systems, cell size (and AP-RN distance), antenna position (of AP and RN). It should be noted that all the given values are to be considered as assumptions only and that the tables are simplified and likely incomplete.

For the Single Hop deployment three different categories have been derived:

- Cellular Deployment
- Wireless LAN Deployment
- Wireless MAN Deployment

These Single Hop deployments are currently based on enhancements of existing systems (3G-HSDPA-Enhanced Uplink, IEEE 802.11x, HiperLAN/2, IEEE 802.16x) described in D3.1 which are potentially applicable to the WINNER air interface. However as they have already been optimized for this kind of deployment only few developments have been made in this area. Since enhancements to existing systems are not within the WINNER scope. The respective Air interface technologies envisaged by WP2 (e.g. FDD, narrowband options vs. TDD broadband) needs to be considered in the future.

Table 3-1 Single Hop Deployment Concepts

	A.1 (Indoor)	B.1 (Hot Area)	C.2 (Wide area)	D.1 (Rural)	B.5 (LOS Feeder)	Notes & Characteristics
Cellular Deployment		X	X	X		3G cellular: typical cell size some kilometres, antenna positions above roof top e.g. at already acquired MNO sites Hot Area deployment: some 100m, Antenna positions below roof top
Wireless LAN Deployment	X	X				802.11, Hiperlan/2: typical cell size some 100 meters Antenna positions below roof top e.g. indoor, at lamp masts or traffic lights, characterized by small local deployments (e.g. hotel, airport terminal or pedestrian zone) with rather simple resource planning or auto configuration
Wireless MAN Deployment		X	X	X	X	802.16: Typical cell size few kilometres Antenna positions below roof top for Hot-Spot scenario, Antenna positions above roof top for Wide Area, LOS feeder scenario characterized by larger deployments with higher resource planning demands or auto configuration capabilities

For the fixed relay deployment three different categories have been derived based on the different discrimination options:

- Time-domain based concepts
- Combined time- and -frequency domain based concepts
- Combined time-, frequency- and space-domain based concepts

and the sub-category "Temporarily Fixed Relay"

Table 3-2 Fixed Relay concepts

	A.1 (Indoor)	B.1 (Hot Area)	C.2 (Wide area)	D.1 (Rural)	B.5 (LOS Feeder)	Notes & Characteristics
Time-domain based concepts	X	X	X		X	Cell size few 100m AP – RN Distance few 100m Antenna positions below roof top Hot Area scenario Antenna positions above roof top in Wide Area and LOS feeder
Combined time- and -frequency domain based concepts	X	X	X			Cell size few 100m AP – RN Distance few 100m Antenna positions below roof top Hot Area scenario Antenna positions above roof top in Wide Area
Combined time-frequency- and space-domain based concepts	X	X	X			Cell size few 100m AP – RN Distance few 100m Antenna positions below roof top Hot Area scenario Antenna positions above roof top in Wide Area
Temporarily Fixed Relays	X	X	X			Cell size few 100m AP – RN Distance few 100m Antenna positions below roof top Hot Area scenario Antenna positions above roof top in Wide Area

For the Mobile Relay concepts four different categories have been derived:

- Mobile Relays serving stationary stations (dedicated mobile relays type I)
- Dedicated Mobile Relays (dedicated mobile relays type II)
- Terminals as Mobile Relays

Table 3-3 Mobile Relay concepts

	A.1 (Indoor)	B.1 (Hot Area)	C.2 (Wide area)	D.1 (Rural)	B.5 (LOS Feeder)	Notes & Characteristics
Dedicated Mobile Relays Type I			X	X		The terminals are stationary with reference to the Mobile Relay, mainly considering a moving system (train) AP-RN distance up to some kilometres RN-UT distance 100m-200m Antenna positions RN inside and outside vehicle potentially at large distances Low/medium/high velocity, no power constraints, medium/high computational power, medium/high level of complexity and incorporated functionalities, medium numbers deployed, possible operator or other provider (train company) ownership, always available for use
Dedicated Mobile Relays Type II		X	X			Mainly considering a moving relay fitted on a carrier (bus, car) AP-RN distance 500m up to some kilometres RN-UT distance up to 500m, Antenna positions RN on top/outside vehicle potentially at large distances, medium velocity, no power constraints, medium/high computational power, medium/high level of complexity and incorporated functionalities, medium-large numbers deployed, possible operator ownership, always available for use, relatively low cost due to large numbers
Terminals as Mobile Relays	X	X	X			AP-RN distance up to the cell edge RN-UT distance up to some 300 meters, Low velocity/stationary, power constraints, low/medium computational power, low level of complexity and incorporated functionalities, large numbers deployed, user-ownership, not-always available (technical or user-related restrictions)

The "Cooperative Relaying" is conceptually a further step ahead with respect to the previous ones since spatial diversity achieved through the network topology itself. Cooperative relaying is considered here as one basic concept, however many possible "sub-concepts" can be in principle identified and merged with the previous relaying techniques.

Table 3-4 Cooperative Relaying

	A.1 (Indoor)	B.1 (Hot Area)	C.2 (Wide area)	D.1 (Rural)	B.5 (LOS Feeder)	Notes & Characteristics
Cooperative Relay	X	X	X			Exact characteristics need to be determined from the system that it is applied to.

3.5 Architectural Harmonization of Deployment Concepts - Preliminary examples

In this section, a different point of view on "harmonization" is taken, and preliminarily analyzed.
As already mentioned, the identification and selection of advanced Deployed Concepts, on one hand, can be finalized to a specific harmonization of different concept, to build the basis for the development of the new optimized functional protocols (in particular, the MAC) for the new WINNER interface. On the other hand (in a complementary way) a useful "harmonization" can be outlined at a higher level: i.e., the coexistence / synergy of different concepts within coherent realistic scenarios (e.g., coexistence of Fixed and Mobile relays in urban areas). This kind of higher layer harmonization is here defined as "Architectural Harmonization". A wide although qualitative view on "architectural harmonization" can help in the proper positioning of the new WINNER solutions versus application scenario landscape.

Three examples of "architectural harmonization" have been analyzed

- HSDPA-Enhanced Uplink & WiMax

- Feeder & Radio Remotization & Multi-Hop concepts
- PMP and Mesh coexistence in Multi-Hop Wireless Concepts

An outline of the mentioned analysis is reported Annex II: Examples of "Architectural Harmonization" of Deployment Concepts

3.6 Conclusions and further work

In this chapter, the first step of the Deployment Concept definition, selection, harmonization processes has been documented. The results of the analyses carried out up to now (although still preliminary) have been reported.

A simplification and clarification of the landscape of concepts has been carried out, starting from the multiplicity of concepts considered and described in detail in the deliverable D3.1. A preliminary positioning of Deployment Concepts vs. Scenarios has been proposed.

Some first examples of "architectural harmonization" have been studied too, having recognized that the "architectural harmonization" is an important part of the overall harmonization process.

As foreseen in WP3 Task 3, the work on the considered aspects will continue along two directions: the finalization of Deployment Concept definition and selection and a more specific harmonization on the selected concepts (protocol and L2 functional viewpoints) will be performed. Further studies on Architectural Harmonization will be carried out.

Interactions with other WPs (particularly WP2) will take place in order to specialize the general selected Deployment Concepts in "modes" tailored to specific scenarios.

4. Multi-hop deployment concept

4.1 Introduction

In the last few years, the interest in multi-hop wireless systems has witnessed an incredible increase. These systems present many challenging aspects. Despite the fact that they have to be treated differently according to the scenario in which the concepts are deployed, the potential performance improvements and the economic fallouts that they are likely to provide thrust this communication domain in the limelight.

Within the WINNER project it has been decided to investigate the homogeneous and heterogeneous deployment concepts in parallel, in order to deeply analyze the different features they can offer and to later merge them in a comprehensive system able to cover any possible scenario.

This chapter addresses many homogeneous and heterogeneous relaying issues. The performance of some RAN protocols that had been partially presented in the previous deliverable are discussed. Particularly, crucial mechanisms such as routing and medium access scheme are described and assessed through simulations. Further, a comparison between single-hop and multi-hop deployment concepts is performed. An important part is also represented by the study on the cooperative relaying where different characteristics are detailed. Finally the heterogeneous relay and wireless feeder concepts are explained in the WINNER context. Besides, some deployment examples, based on the use of heterogeneous relays with same and different duplex schemes, are outlined according with the preliminary definition of new air interface modes developed and proposed by WP2. All of them utilize the specific feeder links mode for the communication between the AP and the RN, allowing the development of advanced relaying techniques based on the optimised AP-RN link.

4.2 Comparison of a Multi-hop with a Single-hop Network Approach

4.2.1 On the Efficiency of Using Multi-hops in Relay Based Networks

4.2.1.1 Introduction

In [1], we have introduced the Multi-Hop Criterion (MHC) to determine in which conditions a single-hop fixed radio link can be efficiently (from a spectral efficiency perspective) replaced with a chain of (multi-hop) links:

$$\gamma < n^{\frac{p}{n-1}} \tag{1}$$

where γ is the single hop SNR, n the number of hops, and p the path loss exponent. If for a given single-hop link condition) (1) is not met, a n-hop link replacement with a better overall spectral efficiency does not exist, no matter where the relays are located. Note that the MHC does not consider shadowing. The inequality represents a quantitative criterion which can be used to decide in which situation a multi-hop link could be used.

In the simulation results presented here we demonstrate the impact of using MHC over the average network spectral efficiency, for a network architecture using fixed relays. Also, we show that MHC works as well in the presence of normal shadowing.

The computer simulation model is described in Annex III: Additional Information on Multi-hop Deployment Concepts.

We consider 4 methods in which a message can be routed to its destination relay:

- Single-hop transmission
- Greedy multi-hop transmission
- Using multi-hop criterion
- Exhaustive search

4.2.1.2 Method 1 – Single-hop Transmission

The message is sent to its destination relay directly in a single hop. For each link, the computer generates the log-normal shadowing component, calculates the path loss, the SNR and maximum spectral efficiency, then the message transfer time. An example of network topology in this case is shown in Figure 4-1a).

4.2.1.3 Method 2 - Greedy multi-hop transmission

This method increases the number of hops to the maximum extent possible. To obtain efficient and short routes, and since the communication from the AP to the relays takes place in a “radial” direction from AP, the difference between the ranks of relays communication over any hop is always 1, i.e., the two relays are located on adjacent concentric tiers, see Figure 10-1.

The routing algorithm is a modified Bellman-Ford method [41], in which we always force the rank difference between relays communicating directly over one hop to be exactly 1. The cost for each link is done by its MTT.

The resulting network transmission topology is a radial tree without loops, see the example in Figure 4-1b). In the diagram, which considers only downlink transmission, transmitters are symbolized by hexagons, while receivers are symbolized by triangles. As it can be seen, the transmission tree limbs are in general short, between relays located in adjacent positions. The multi-hop links are organized in hops which are always between relays located on adjacent tiers. As the algorithm “goes for” short links with high SNR, and high spectral efficiency, it can be considered greedy (or opportunistic).

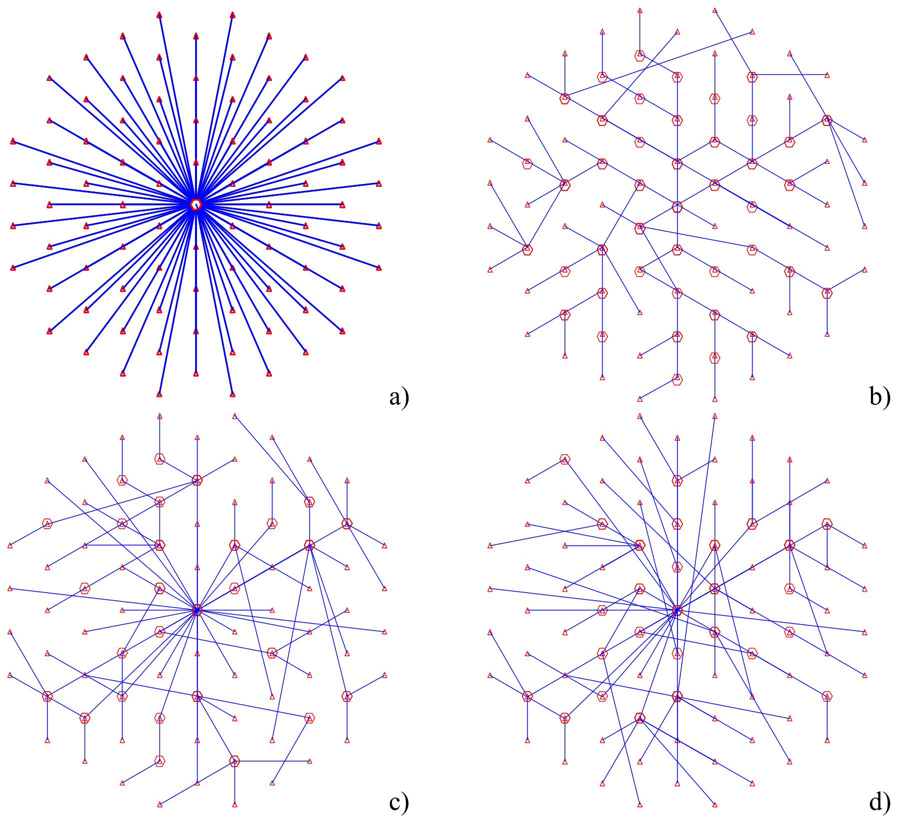


Figure 4-1 Network topology: a) Direct single hop; b) Greedy multi-hop transmission; c) Transmission using MHC; d) Exhaustive search

4.2.1.4 Method 3 – Multi-Hop transmission using Multi-Hop Criterion (MHC)

In Method 3 the number of hops is dictated by the MHC. Instead having a hop for each time a packet is crossing from one rank to the next (or from a hexagonal tier to the next), here the hops may be longer and pass over one or more ranks.

Once the SNR for the single hop link between the AP and the destination relay is known, using MHC equation (2) we can find the number of hops at which a more efficient multi-hop link becomes feasible. Knowing the target number of hops for the multi-hop link, and the rank of the destination relay, we calculate the size of each hop, in "rank units". We use the following formula:

$$HopSize[ranks] = \mathrm{int}\left(\frac{Rank_{dest}}{n_{MHC}}\right) \qquad (2)$$

where $Rank_{dest}$ is the rank of the destination relay, n_{MHC} is the number of hops according to MHC criterion, and the *int* function returns the closest integer to its argument. Once the hop size is known, we use the same algorithm as in the Method 2, with the following difference: assuming the destination relay has rank R_{dest}, instead of choosing the "best" intermediate relay from the relays located in the tier of next lower rank R_{dest}-1, we select the "best" intermediate relay from the tier of rank $R_n – HopSize$. The complete logical flowchart for the algorithm are shown in Appendix 7.

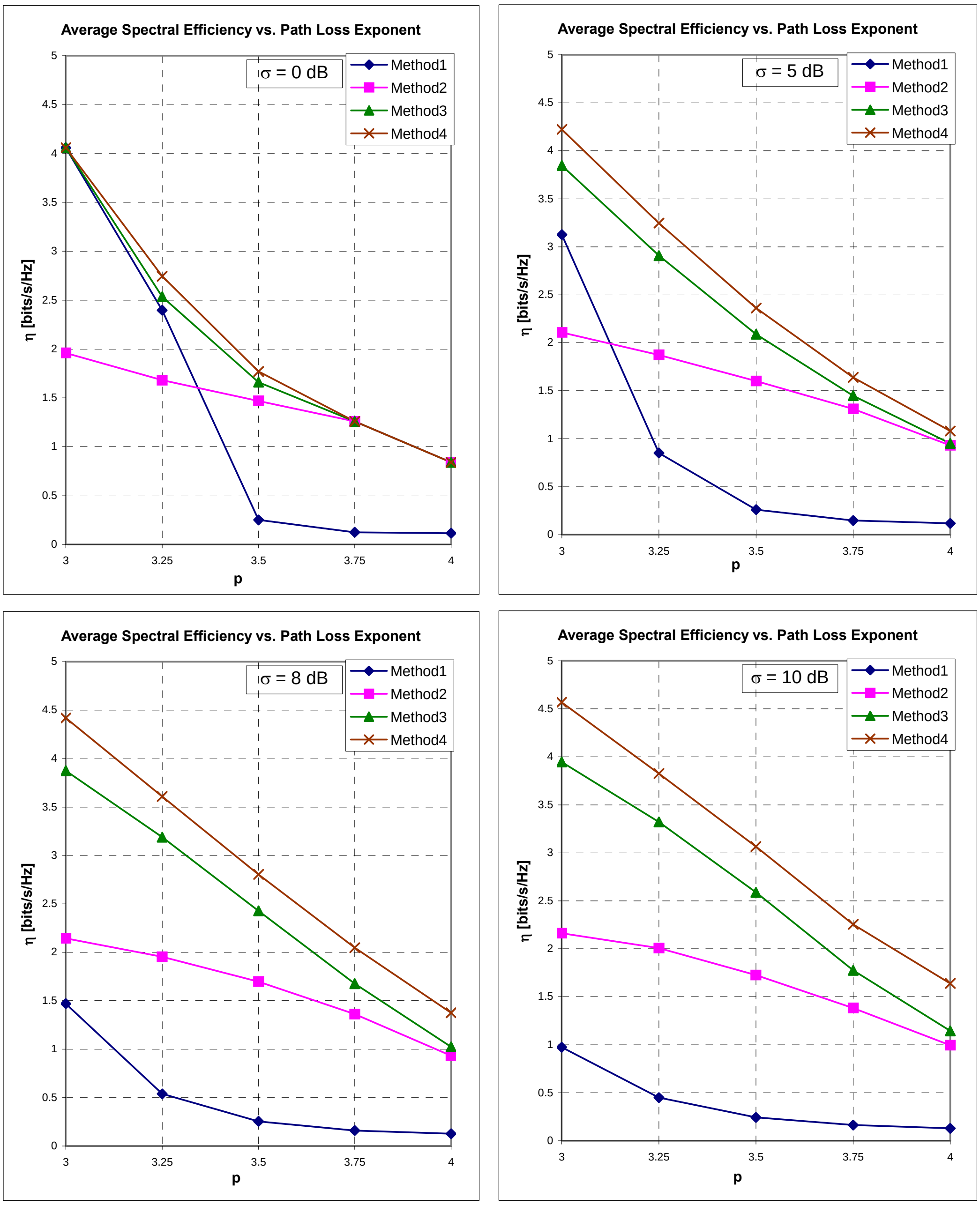
The resulting network topology, an example shown in Figure 4-1c), is a loop-less tree. With similar propagation parameters, the tree has longer limbs; some relays of rank larger than one are connected directly to the AP – in these cases, the MHC indicated the single hop link achieves a better aggregate spectral efficiency.

4.2.1.5 Method 4 – Exhaustive search

Same as the previous methods, this search method assigns costs to each relay, starting from the AP and working toward the network periphery. A link cost is represented by the MTT across a hop between two relays. The AP is assigned zero cost. For each relay i, we calculate the sum between the cost of any relay k of equal or lower rank (already available) and the cost of the link i-k. We repeat, calculating the sum for all intermediate relays k. The smallest sum obtained is assigned as the cost for i, and the chosen relay k is part of the route from the relay i toward the AP.

The algorithm ensures that the selected routes are optimal and the cost assigned to each relay (actually the total MTT from the AP to the relay) is the smallest possible. The complete logical flow chart for the algorithm is shown in section 10.1.1.2.

A example of network configuration resulted from using method 4 is shown in Figure 4-1d). In appearance, the topology is the most “disorganized” from the 4 methods. This is due to the fact that the algorithm has the freedom to choose from a wider pool of intermediate relays. Direct comparisons can be made between network topologies in Figure 4-1, b), c), d), as the same shadowing values are used for all.



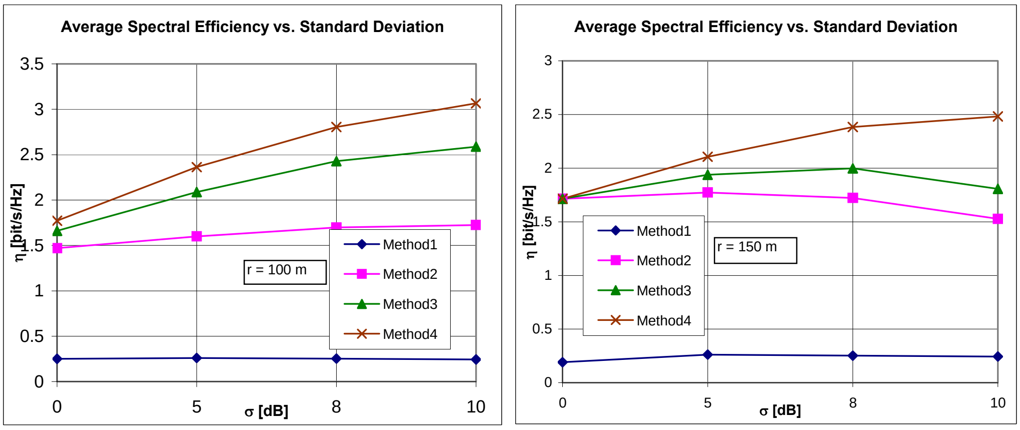
R	r	F	B	Pt	NF	Gt	Gr	d0	p	σ	α
800 m	100 m	5 GHz	25 MHz	1 W	5 dB	10 dB	10 dB	2 m			1

Figure 4-2 Average network spectral efficiency, relay service radius 100m

4.2.1.6 Simulation Results

The average network spectral efficiency was plotted versus path loss exponent in **Figure 4-2**, for different values of the log-normal shadowing standard deviation. The following observations can be made:

1. Although on a per link basis the spectral efficiency can go as high as 8bit/s/Hz, the average network spectral efficiency is much lower, (close to 1 bit/s/Hz for high values of path loss exponent p), regardless of the routing method used. This is due to repeated resource assignment, to carry the same information (radio packet) over a number of hops to the relays located close to network periphery.
2. For high values of the path loss exponent p, the Method 1 using single-hop links has the lowest performance. Its average spectral efficiency performance tends asymptotically toward zero as the path loss exponent increases. Since the capacity decreases toward zero, we conclude the communication is not possible using Method 1 at high path loss exponents. However, at low values of the path loss exponent, and especially when there is no shadowing, the direct single-hop link performance is optimal. This leads to the possibility of deploying feeder systems with fixed relays installed within LOS of access point, where particular terrain conditions would allow such deployment; with low path loss exponent and no shadowing, feeder systems of this kind would function very well using single-hop direct links, i.e., no multi-hop.
3. At low propagation exponent values (i.e., high SNR) the Method 2 (greedy multi-hop) has the lowest spectral efficiency, due to unnecessary hoping. The highest possible modulation scheme considered allows for a hop spectral efficiency of 8 bit/sec/Hz, even when the hop SNR is higher than 35.5 dB
4. The shadowing has a minimal impact on the performance of Methods 2, 3 and 4, while the direct single-hop transmission method is heavily affected. As described in the previous section, the algorithms for methods 2, 3 and 4 avoid hops with heavy shadowing. This is the core advantage of multi-hop technology.
5. At high path loss values and low shadowing, the algorithms 2, 3 and 4 converge to the same solution. For low SNR, the MHC (Method 2) will recommend using multi-hop links with the maximum number of hops (Method 3), which in this case is the optimal solution (Method 4). For higher shadowing, the random path losses for various links advantages the Method 4 which has more flexibility in choosing the routes.
6. An increase in average network spectral efficiency can be noticed for larger values of the standard deviation of the log-normal shadowing, see Figure 4-3. This can be explained considering that the mathematical model for log-normal shadowing allows for variations from the median in both directions, i.e., for higher or lower path losses. Whenever a below median path loss is realized in some link, Methods 2, 3 and 4 have the ability to exploit it and use it to minimize the path costs to that relay.



R	r	F	B	Pt	NF	Gt	Gr	d0	p	σ	α
800 m		5 GHz	25 MHz	1 W	5 dB	10 dB	10 dB	2 m	3.5		1

Figure 4-3 Average Spectral Efficiency vs. Standard deviation

4.2.1.7 Conclusions and Future Work

Using computer simulation, we have compared a routing algorithm based on the MHC, with three other routing methods: single-hop direct transmission, greedy multi-hop transmission, and exhaustive (optimal) search. We concluded the MHC method outperforms in all situations the single-hop and the greedy multi-hop algorithms, having performance (in terms of average network spectral efficiency) close to or on par with the exhaustive search method.

The work presented makes a strong case in favor of using relaying as an essential component of 4G systems. Under the initial assumption, we have considered for analysis only the feeder link between the access point and the relay providing access service to user terminal (this assumes the link AP – UT has at least two hops). Even under such “lighter” requirements we have shown that in most cases the spectral efficiency of multi-hop relaying systems outperforms the single-hop techniques, provided that specific deployment strategies for relays are followed:

- Every effort must be made to increase the inter-relay distance, and reduce the number of relays to the minimum, provided the coverage objectives to provide the required grade of service are met.
- The feeder system parameters must be chosen such that the maximum data rate can be achieved between adjacent relay locations.
- When a chain of relays is to be deployed between a source and a destination, the ideal locations are along the straight line from start to end, at equal intervals.
- The feeder system antennas should be installed within LOS of each other whenever possible.

More refined radio channel models for communication between fixed relays are needed to fully characterize feeder systems. Most channel models currently used in literature are borrowed from mobile communication channels. Beside the differences related to the “fixed” nature of the channel (i.e., no frequency dispersion, much less Rayleigh fading), other aspects should be included, for example the conscious selection by the system designer of good relay locations with unobstructed radio links back to the servicing access point. Measurement campaigns or simulation on real high resolution GIS/ terrain databases are needed o characterize such effects, which could “skew” significantly the shadowing statistical distribution parameters.

4.2.2 Performance comparison between Single and Multi-Hop Deployment

In D3.1 a system concept with relaying capabilities has been proposed. A deployment based on this concept in a Manhattan structure is given in the following sections and investigated in terms of achievable mean throughput from a theoretical point of view. To compare the results a traditional Single-Hop network covering the same area as the Multi-Hop network has been developed. Both networks were investigated and their respective theoretical mean throughputs have been derived.

4.2.2.1 Assumptions and boundary conditions

Single- and Multi-Hop deployments differ in general in the number of network elements necessary to cover a certain area with similar capacity. Since the goal of this investigation was to ensure a fair result the cost perspective needed to be considered as well. The following assumptions are the basis for the investigations:

- Comparison should be fair.
- The deployment for both Single-Hop and Multi-Hop transmission should be realistic and should offer their respective best capacity.
- The number of APs per unit area should be kept constant since the fixed data connection is regarded as major cost driver.
- No other than standard technologies (e.g. no Beamforming and no Advanced Antennas) are considered at the relay in order to exclude their respective gains and to keep the costs for relays as low as possible.

As described in D3.1 the HL-2 PHY modes were selected as the basic physical layer operating in the 5GHz band, with standard 20MHz HL-2 bandwidth. The scenario described in D3.1 was studied meanwhile and found to be too small to have the necessary flexibility in order to meet the requirements mentioned above. It was therefore replaced by a 16x16 blocks grid, with basic parameters based on UMTS 30.03 [62].

4.2.2.2 Single-Hop Deployment

In order to define the deployment for the Single-Hop case an estimation on possible cell sizes was performed first which can be found in annex 10.1.2.1. A number of different resource allocation schemes has been investigated aiming on the selection of a deployment utilizing the fewest resources to cover the selected area. The best pattern found, offering sufficient C/I at the cell boarder, is a reuse 4 pattern as shown in Figure 4-4. Please note that reuse is used in this context to denote a reuse in the time domain since a pure TDMA concept is assumed.

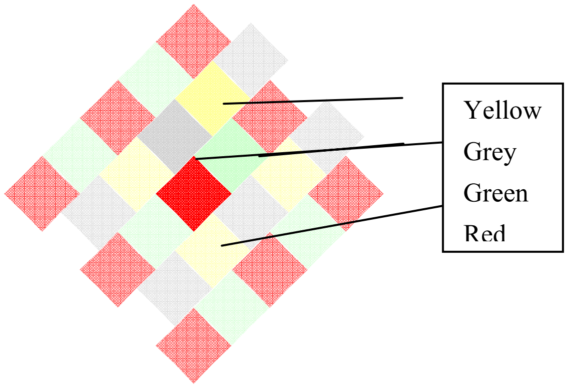


Figure 4-4 Single-Hop Deployment

In order to determine the throughput for each cell, the resulting cell size needs to be calculated first. Figure 4-5 shows the resulting cell size based on the C/I that is observed for the red and yellow resource for a UT traveling from the AP of the central cell one block east and than to the south towards the light-yellow cell. The ideal handover point is at the C/I crossover point between both resources after traveling around 380m. Due to the regular structure of the scenario, this calculation is also valid for estimating the handover point between the central cell and the light-grey cell to its right. The cell size for the central cell is shown in Figure 4-5.

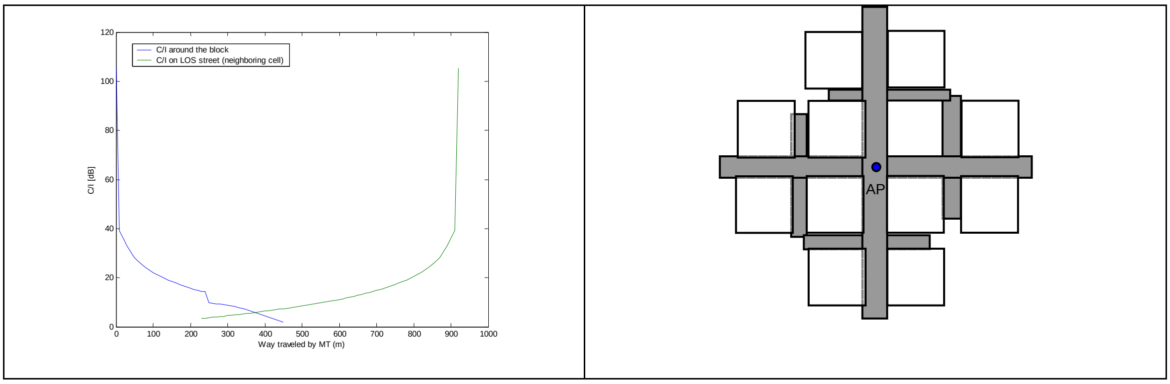


Figure 4-5 Single-Hop cell size for selected deployment

The C/I distribution along the way of the UT can be translated into a throughput vs. distance function by applying the PER vs. C/I functions of the best suited PHY mode for a given AP to UT distance[11]. The relationship between PER and C/I for each PHY mode was derived from link-level simulations with the ETSI-C channel model and HL-2 PHY mode parameters. Resulting throughput curves can be found in annex 10.1.2.2. The results for downlink LOS and non-LOS connections (i.e. connection around the block) are given in Figure 4-6 for the cell size found. The average downlink throughput was calculated over all possible UT locations for the cell size found considering the throughput at the relevant UT positions as given in Figure 4-6. The average throughput for the cell was calculated to be 22.2Mbit/s if the whole resource could be used and fair scheduling is assumed. Considering a reuse factor of 4.the average value for the cell is 5.55MBit/s.

[11] This strategy implies perfect link adaptation, i.e. always the best suited PHY mode for a given C/I is used.

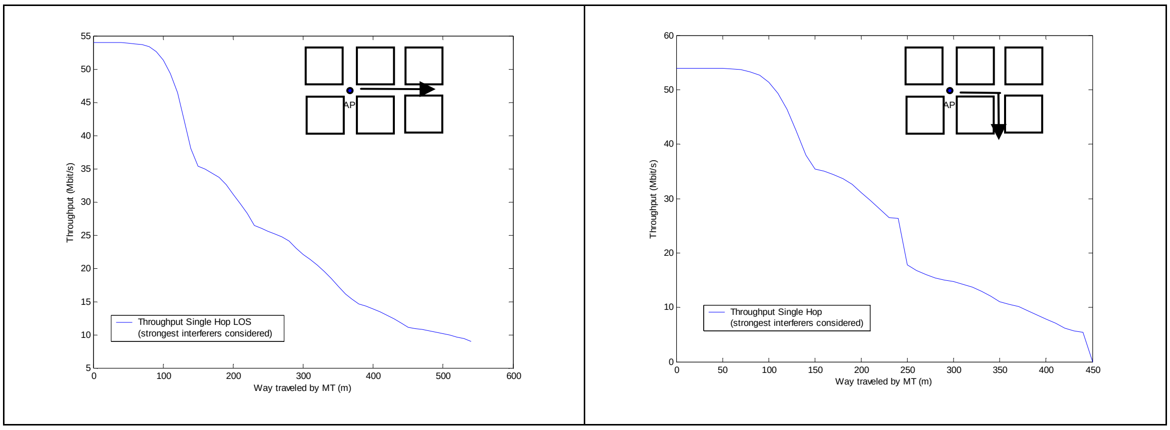
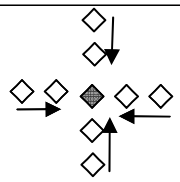


Figure 4-6 Single-Hop throughput

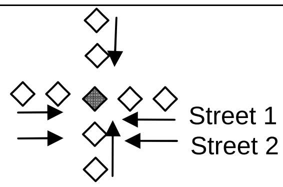
4.2.2.3 Multi-Hop Deployment

In order to ensure a fair comparison between both concepts - Multi-Hop and Single-Hop deployment - , the underlying assumption was to design both deployment concepts such, that each of it achieves its best performance in terms of capacity. Due to the properties of the investigated Manhattan deployment the following constrains where taken into account.

- Since the performance for all connections involving the relay is limited by the AP-to-relay link due to traffic aggregation, this link needs to provide a higher throughput, than the relay-to-UT links. This means in turn, that the C/I-situation on this link needs to allow the use of a respective PHY mode.
- Unlike Single-Hop transmission, where separation of the co-channel cells needs to be mainly considered for the LOS path, in Multi-Hop transmission the specific situation at the relay needs to be taken into account as well when planning the deployment (cf. Figure 4-7). In the considered concept only the specific resource, which is assigned to a node, is used for transmission while receiving nodes listen to the resources of the other entities. In order to obtain high C/I values on the relay links it is necessary to carefully observe the interference situation at the receiver for all resources it may need to receive. This requires also the consideration of other interference sources if placement of the relays is done on street crossings as shown in Figure 4-7.



In case of Single-Hop deployment, the main co-channel interference reaches the center cell through those streets providing a LOS towards the co-channel cells (LOS-street).



In case of Multi-Hop deployment, co-channel interference needs to be taken into account stemming from cells in LOS to the center cell and from those cells in LOS to the relays (NLOS street for the AP).

Figure 4-7 Single-Hop vs. Multi-Hop reuse separation

In order to meet the constrains given above and to deal with the interference situation at the relay, the reuse cluster scheme shown in Figure 4-8 was used to construct the Multi-Hop deployment. Construction was done by placing the reuse cluster in the scenario such that every street crossing is equipped with a node (either AP or relay). The Multi-Hop clusters where constructed from an AP in the center and 4 RNs grouped around the AP based on the assumption that AP density should be identical to that in the Single-Hop case. Nodes not belonging to a Multi-Hop cluster where deleted from the deployment.

Figure 4-8 shows the relevant part[12] of the Multi-Hop deployment. Since nodes are placed on street crossings a LOS path between all possible locations of a UT and a relay or AP can always be ensured. It can be seen by comparing the reuse cluster scheme (cf. Figure 4-8) with the Multi-Hop deployment (cf.

[12] Regular extension over the whole area is not shown here for simplicity reasons

Figure 4-8) that some resources are never used throughout the scenario. The reuse factor 12 shown in Figure 4-8 is therefore effectively reduced to a reuse factor of 9.

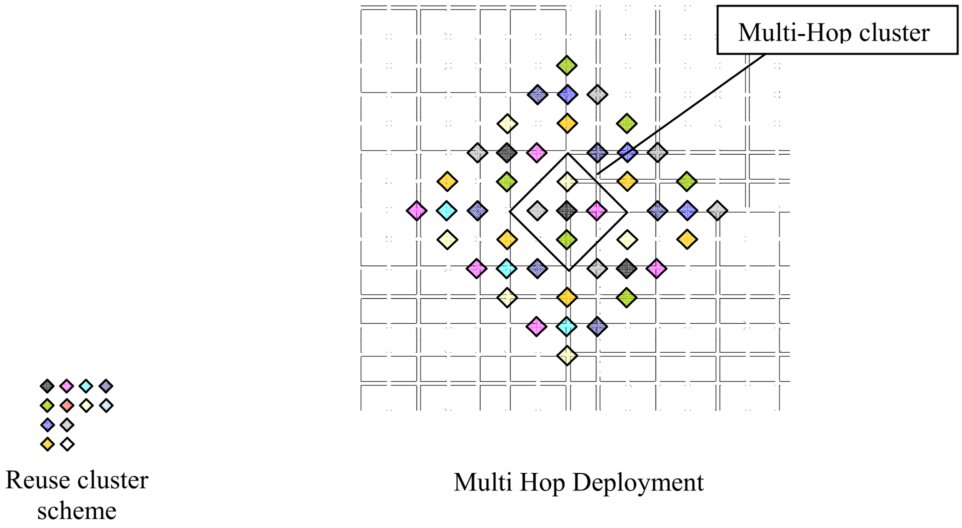


Figure 4-8 Multi-Hop deployment

Similar to Single-Hop, the area covered by a Multi-Hop cluster needs to be determined and can be easily obtained due to the symmetric structure of the deployment as shown in Figure 4-9.

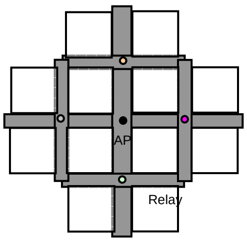


Figure 4-9 Area covered by a Multi-Hop cluster

In order to calculate a mean throughput of a Multi-Hop cluster, which is comparable to that in the Single-Hop case, it is required to determine the time interval reserved for the transmission of AP and Relays belonging to a Multi-Hop cluster with its associated UTs in relation to the superframe size. The whole calculation can be found in annex 10.1.2.3 and only relevant parts are given herein.

Equation (3) shows the total superframe size depending on the transmission times for the AP-to-UT and Relay-to-UT communication based on (40) to (45) and the parameters given in Table 10-2 and Table 10-3.

In order to determine the MH-cluster throughput, the split between $t_{AP<->MT}$ and $t_{Rel<->MT}$ needs to be defined for which two solutions have been found.

$$t_{Superframe} \approx 3 \cdot t_{AP<->MT} + 16 \cdot t_{Rel<->MT} \tag{3}$$

The first solution is based on using an amount of resource per area similar to that used in the Single-Hop case. When taking the observation into account that the AP offers better throughput on the LOS streets, it can be seen in e.g. Figure 4-9 that each AP covers about the same area on the LOS streets as the four relays together. If all Multi-Hop clusters are considered, it can be seen that the APs cover about the same area with 3 resources as the relays with 6.(cf. Figure 4-8). If the users are distributed uniformly across the deployment area it is fair to assume that the APs should use twice the amount of resource (time) than the relays ($t_{AP<->MT} = 2 \cdot t_{Rel<->MT}$).

The second solution is based on alignment of the average throughputs archived in the Multi-Hop cluster. Since $\frac{T_{AP<->MT}}{T_{Rel<->MT}} \approx 1.11$ we can align the throughputs by selecting $t_{AP<->MT} \approx 0.9 \cdot t_{Rel<->MT}$

With the approaches mentioned and

$$T_{MHCluster} = \frac{t_{AP<->MT}}{t_{Superframe}} \cdot T_{AP<->MT} + \frac{R_{RelCluster} \cdot t_{\mathrm{Re}l<->MT}}{t_{Superframe}} \cdot T_{Rel<->MT} \tag{4}$$

the MH-cluster throughput can be calculated to be 11.58MBit/s for $t_{AP<->MT} = 2 \cdot t_{Rel<->MT}$ and 10.85MBit/s for $t_{AP<->MT} = 0.9 \cdot t_{Rel<->MT}$.

4.2.2.4 Conclusions

It has been shown that Multi-Hop offers about twice the throughput compared to a Single-Hop deployment with the stated assumptions under consideration of a standard interference situation. To be fair it needs to be mentioned that the total number of network elements is 5 times higher for the Multi-Hop deployment compared to the Single-Hop case. Even though the costs of relays are likely remarkably lower than those for APs, there will be some cost increase due to site acquisition and additional maintenance necessary for the Multi-Hop deployment although these costs are quite hard to predict[13]. Additional overhead introduced by the higher number of network elements and overhead by increased number of handover situations was not studied and will affect the Multi-Hop performance as well. On the other hand the performance of Multi-Hop can be increased by use of directional antennas for the AP to Relay and connections to the UT. Additional gains for Single-Hop can be expected by use of smart antennas and by moving to a roof-top deployment which may or may not be selected because of its cost/performance ratio. Since the exact gains of these options are not clear at this point in time, it can be concluded that Multi-Hop remains an interesting candidate technology for the WINNER air interface and further studies are necessary to understand the full gain of it.

4.2.3 Performance comparison of PMP and Mesh Mode

One of the promising standards that defines interesting and future-oriented protocol elements and that should be considered when designing a new 4G air-interface is the standard IEEE 802.16a. It specifies two operation modes, a PMP and Mesh mode. The PMP mode has been specified for single-hop communication. Different to that, the Mesh mode supports multi-hop communication, which is one key feature of the WINNER air-interface. Both modes are based on the same physical layer and very similar burst structures. In the following section it is briefly explained how the PMP mode can be extended to support multi-hop communication. This has been already presented in the previous deliverable D3.1 [1]. The main focus of this section is a comparison of the two modes with respect to overhead, and thus, the maximum achievable throughput. The insight gained from performance comparisons should provide some guidelines for the design of an appropriate protocols for forwarding.

4.2.3.1 Multi-hop approach for PMP mode

In the first deliverable D3.1 [1] the concept for the multi-hop extension for the PMP mode has been described. Here, it is briefly reviewed to get a better understanding of the following performance comparison between the Mesh and PMP mode.

Correspondingly to the sub-scriber stations (SS) in the Mesh mode that have routing capabilities, the PMP AP is introduced. The PMP AP is a new instance in the case of the PMP mode that performs the data scheduling and the relaying. .The PMP AP can be interpreted as the Central Controller (CC) in HIPERLAN/2 [9]. Multi-hop transmission is made possible with a new frame structure. The idea is to join *n* frames for *n* - 1 connections between *n* PMP APs, respectively hops, obtaining a unique joint frame, which is composed by *n* sub-frames. In the following Figure 4-10 a simple 2-hop topology is assumed with two PMP AP and n_c final SS, which are served by the last PMP AP.

[13] An investigation in WP 7 is currently carried out on this issue

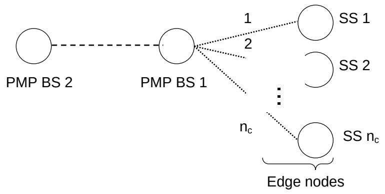


Figure 4-10 Exmpale topology with two PMP APs

The resulting frame structure for a two-hop topology (n = 2) is depicted in the next Figure 4-11.

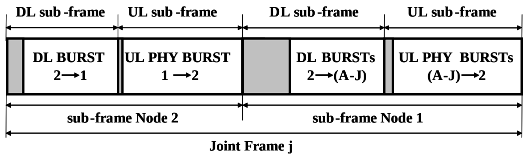


Figure 4-11 Joint frame in PMP mode composed of two sub-frames in order to support multi-hop communications

The joint frame is composed by two DL sub-frames respectively for the data connection between PMP AP 2 and PMP AP 1 and data connections among PMP AP 1 and n_c edge SSs. Every MAC PDU transmitted on the link between PMP AP 2 and PMP AP 1 is extended by the Mesh sub-header to carry information about the destination, respectively source SS. In the example. within the sub-frame handled by PMP AP 2 only one Bandwidth Request opportunity is used by PMP AP 1 for requiring resources from PMP AP 2, since there is only one connection served by PMP AP 2.

4.2.3.1.1 Performance results

The equations for the evaluation of the efficiency on the MAC layer for the PMP and Mesh mode has been described already in deliverable D3.1[1].In this section the results for the example multi-hop scenario depicted in Figure 4-12 are derived. The scenario is composed of n PMP APs / resp. 1 Mesh AP and n-1 Mesh SSs deployed in a linear topology. The node n acts like a AP in the PMP / resp. Mesh network. The node 1 (PMP AP / Mesh SS) directly serves n_c SSs deployed in a tree topology, where n_c denotes the number of bidirectional connections. Hence, every n-hop bidirectional connection is established between a node in the tree topology and the node n in the linear topology.

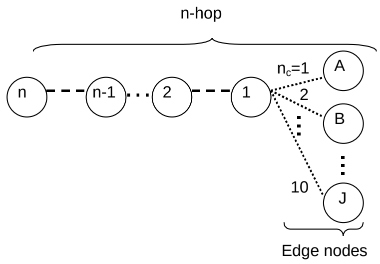


Figure 4-12 Example network topology for performance comparison of PMP and Mesh mode

The analytical evaluation is performed under the assumption that n_c bidirectional connections are already established in the network. The same assumption holds for the Mesh air-interface.

The parameters for the physical layer used in the performance evaluations are presented in the following Table 4-1.

Table 4-1 OFDM parameters

N_{FFT}	256
N_{SD}	192
BW	20 MHz
Frame duration	10 ms
K	¼
Q	7/6
T_{symbol}	13.7 µs
N_{symbol}	729

The parameters that affect the maximum efficiency on the MAC layer for the different modes are presented in the following tables, for the PMP mode in Table 4-2, the Mesh centralized scheduling (CS) in

Table 4-3 and for the Mesh distributed scheduling (DS) mode in Table 4-4, respectively.

Table 4-2 PMP parameters

n_c	Variable
N_{RNG}	0
$N_{BW\text{-}REQ}$	10 for last hop 1 for other hops

Table 4-3 Mesh CS parameters

n_c	Variable
N_{CH}	1
N_{NODE}	n_c+n
N_c	$2n_c$
#SS*	N

Table 4-4 Mesh DS parameters

n_c	Variable
N_{SCHED}	0
$N_{REQUEST}$	0
$N_{AVAILABILITY}$	0
N_{GRANT}	$n_c+(n-1)$
N_c	$2n_c$
#SS*	n

Since the network entry procedure has only a minor impact on the performance of the achievable throughput, it is neglected and, thus, in the PMP mode the number of initial ranging opportunities N_{RNG} is set equal to 0. In the two-hop scenario the joint frame is composed by two sub-frames, which are different in length. The sub-frame handled by PMP AP 2 contains only one BW-REQ opportunity since only one link is setup between node 2 and node 1. The edge nodes are allowed to send bandwidth request within one of 10 BW-REQ opportunities that are reserved for this purpose in the sub-frame handled by PMP AP 1.

***N*$_{\mathrm{CH}}$ in**

Table 4-3 is the number of available channels in the band assigned to the Mesh AP, N_{NODE} the total number of nodes in the wireless network and N_C the effective number of active connections; e.g. n_c bi-directional connections corresponds to $N_c=2n_c$ effective connections.

For the Mesh DS mode, N_{SCHED} denotes the number neighbours of which the distributed scheduling information is carried in the message; N_{REQUEST}, $N_{\mathrm{AVAILABILITY}}$ and N_{GRANT} are respectively the number of allocation requests, availabilities and grants in the message.

In Figure 4-13 and Figure 4-14 analytical results of the Mesh mode and PMP mode are compared for one-hop and two-hop connections, respectively. The efficiency is shown as function of the number of active bi-directional connections. Consistent with the results presented in deliverable D3.1 [1], it is indicated that the centralized scheduling (CS) outperforms the distributed (DS) when the Mesh air-interface is adopted. Results for Mesh CS are obtained without considering the overhead introduced by the MSH-CSCF message since it is transmitted when the network topology or available channels are changed.

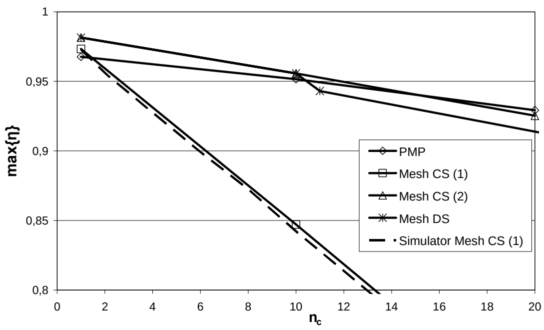


Figure 4-13 One connection: comparison between PMP and Mesh mode for different number of connections

The Mesh air-interface has been analyzed considering two different system parameter tunings. In the first approach the edge SSs (A-J) have a transmission opportunity to send a request message in every Control sub-frame (Mesh CS (1)) in the uplink (from SSs to Mesh AP). The second approach consists of not providing a transmission opportunity for every node in each frame (Mesh CS (2)). Within the Control sub-frame there exists only a number of transmission opportunities for request/grant messages equals to the number of hops *n*. Nodes in the linear topology fill these opportunities with grant messages in the downlink (Mesh AP to SSs). Since these messages are transmitted periodically, edge nodes can use free opportunities for transmitting resource requests in the uplink. The network reacts quicker to a new resource request when the first approach is adopted. However, as shown both in Figure 4-13 and Figure 4-14, a higher overhead is introduced in this case by the MAC protocol.

The maximum efficiency for the adapted PMP air-interface in order to support the multi-hop scenario is presented in Figure 4-14 for 2-hop connections. To take into account the new joint frame structure the number of MAC PDUs in the joint frame is obtained as sum of the number of available symbols in the two sub-frames and T_{FRAME} corresponds to the joint frame duration. Moreover, a short preamble is required for every uplink connection, hence a short preamble is used for every UL PHY BURST in the UL sub-frame for data transmission among edge nodes and node 1. On the contrary, the node 1 transmits only one UL PHY BURST, thus only one short preamble is introduced in the overhead calculation.

The Mesh CS (1) approach has been validated by means of event-driven simulations of the MAC protocol implemented in ns-2 [10]. The small gap among curve profiles of analytical and simulation results is due to the different estimation of the overhead introduced by MAC headers, Mesh sub-headers and CRC fields. In the analytical evaluation it is assumed that the overhead for all bursts is grouped together and is preceded by a pure data part (sum of all payload). This is different to the simulation, where each individual data part comes along with the overhead, and hence, a different recursive calculation is performed in the simulator and the analytical approach to determine the max. number of MAC PDUs that fit in one frame, i.e., the truncation is performed at different steps in the calculation.

Performance results in Figure 4-13 and Figure 4-14 indicate that the Mesh mode outperforms the PMP for low number of connections.

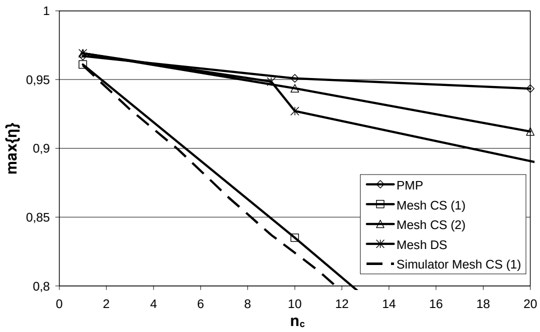


Figure 4-14 One-hop connections: comparison between PMP and Mesh mode for different number of connections

The Mesh air-interface can be applied to the backbone of a wireless network with a limited number of connections. Several SSs shall be served by one node of the backbone network. For this purpose, the PMP air interface can be applied on the last hop towards the SS since it performs quite well for many single-hop connections. In addition, it has been shown that even the PMP mode with some proposed modifications can be used for a multi-hop connection, but the current version of the standard does not support this mode of operation. However, it is worth to investigate this opportunity by the combination of different protocol elements of the PMP and Mesh mode, and improve the performance of the Mesh mode in the multi-hop topology via respective modifications.

In summary, the PMP and Mesh mode has been compared in single- and multi-hop topologies. For multi-hop support the PMP mode has to be modified. From performance results it can be seen that the Mesh mode can be advantageously applied for a low number of connections. Different to that PMP is better for large number of active connections. This leads us to the conclusion that a combination of the protocol elements of the PMP and Mesh mode in multi-hop topologies tend to be a promising approach, which will be investigated in the framework of the MAC design in the future taking into account the physical layer of the WINNER air-inteface.

4.3 Radio access network functionalities in multi-hop deployment concepts

4.3.1 Time Division Based Relaying

The principal purpose of the relay node in WINNER is the extension of a single AP's coverage. In WINNER system, this issue is very crucial since the propagation loss is very high in assumed high-frequency bands, and as a result, high transmission power is required for user terminals. However, this is not feasible solution e.g. due to power consumption. Thus relay nodes or repeaters might be the answer to the problem. The service coverage extension without installing new access point has been an important issue even in 2G and 3G systems. The solution for 2G and 3G is to use a repeater. However since WINNER wideband is targeting for low frequency reuse the interference management is a challenging problem. Introduction of repeaters to increase the low cost coverage is not seem feasible as a general solution (in some special cases they might be usable), since repeaters amplify all the noise/interference in the network. Moreover relays using L2 forwarding capability, the unnecessary transmission from RNs can be avoided and because each RN can have signal processing ability, the noise and interference can be removed every time packet goes through RN.

4.3.1.1 Time Division Based Relaying in TDD System

The concept considered in this section is based on work presented in [46]. The time division for relay communications is realized by defining a subframe, reserved for communication between relays and user terminals. The relay-AP communication is arranged as the relay would be a user terminal.

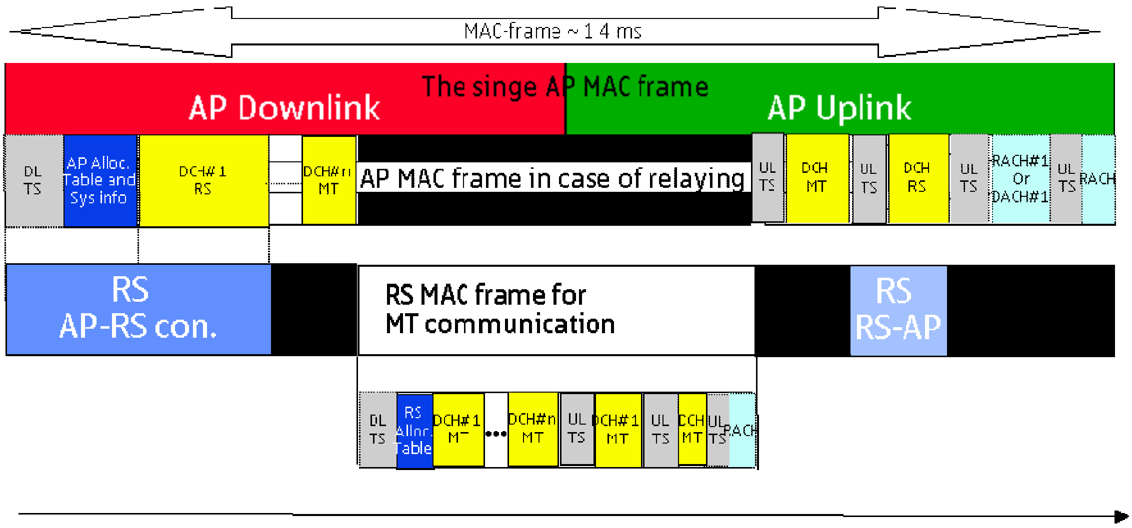


Figure 4-15 AP and RN MAC frames. Black color represent the time when AP or RN is not sending or receiving in current concept.

In our default deployment scenario an AP is always one-hop away from each FRN. The total length of the FRN subframe is dynamically configurable, enabling AP to effectively adapt to varying traffic conditions in own and different cells. In other words, AP allocates a constraint to FRN-UT communication. Each FRN then independently allocates parts of its subframe to TX-to-terminal and RX-from-terminals time. Thus, each relay can independently optimize its hardware utilization based on its current uplink and downlink needs. However, if relay maximum transmission power is significantly larger than UT transmission power, the TX/RX turn-around moment has to be kept constant, and the constraint provided by AP determines also the downlink-uplink proportions of the FRN-UT communication. The maximum subframe length and the subframe position is decided by AP and allocated around fixed TX/RX turn around time.

The time frame from reception to transmission is quite tight in proposed MAC framed based relaying concept. There might be practical problems in implementing this kind of system. If needed, an additional one frame delay in FRN communication could be introduced. This would of course add further communication delay to system.

The AP is currently assumed to be inactive during the relay MAC frame. However, it might be possible to utilize this period for communications with cells close to the access point. In addition, if beam forming is used, the DOA information could be used to schedule traffic to UTs in a different direction. Whether the required power coordination and interference reduction techniques can be developed is left for further study.

It is clear that proposed scheme try to achieve low-cost coverage at the expense of delay and throughput performance. However the relative diminution of performance is under study.

4.3.1.2 Range of Coverage Area

An advantage of fixed relays over a dense access point deployment is cost reduction. The major part of the savings is assumed to come from the installation-cost (including placement rents). Nevertheless, hardware costs should also be considered, and the receiver structure of a relay node should be kept as simple as possible. The concept presented here and in [46] avoids having to transmit simultaneously to UT and AP, and therefore it requires only a single transceiver per bandwidth for a relays station.

The complexity of the relay implementation should be kept in reasonable level. The cost pressure for the relay is not as severe as for a terminal, but naturally, the cost should be taken into account. The complexity goes hand-in-hand with power consumption and price. Thus the complexity should be kept as low as possible. The actual implementation of hardware can be done in different ways, and can be optimized only after detailed requirements are present. The most important requirements affecting the complexity are modulation, duplex operation, output power and spectral purity of the transmission, operation frequency and frequency bandwidth, sensitivity requirement, selectivity requirement, dynamic operation requirements and finally the form factor of the device. As a reference, one can take the complexity analysis of the terminal [47]. If the RF complexity of the relay is decided to equal the terminal complexity the main parameters of the operation has to be kept the same. If, for example, more output power is needed, then the complexity is increased.

The cell ranges for different WINNER radio modes affect the way of inter-working of the modes. Therefore, we provide estimates for cell ranges with wide band and narrow band mode using simple link budget calculations. Since it is possible that narrow band spectrum allocation is located in a lower frequency band, for narrow band mode both 2 GHz and 5 GHz center frequencies are considered.

This simple link budget calculation shows that with the current measured macro cell urban environment path-loss model [47], the average size of the cell can't be more than 200 m. The path loss model is shown below and the standard deviation is 5.7 [dB].

$$PL(d) = 28.3 * \log_{10}(d) + 53.5 \text{ [dB]} \tag{5}$$

In case 200-meter terminal-AP separation the average uplink wideband SNR is about 4.8 dB (target 1 dB and fading margin 3.8 dB, which is less than slow fading STD) and this does not take the interference into account. More detailed description parameters and assumptions used in these link budget calculations can be found from chapter 11. The link budgets for wideband are shown in Figure 4-16.

5GHz Downlink (WB ~100MHz)		
Parameters	**Value**	**Unit**
TX power @ PCB output	43.0	dBm
Noise bandwidth	8E+07	Hz
TX antenna gain	10.0	dBi
Range	**700.0**	m
Path loss @ range	134.0	dB
RX antenna gain	0.0	dBi
Received power (no fading) @ PCB input	-81.0	dBm
Noise floor	-174.0	dBm/Hz
integrated noise	-9.5E+01	dBm
RX noise figure	9.0	dB
Receiver degradation	0.0	dB
SNR @ BER = ?	1.0	dB
Sensitivity	-84.8	dBm
Fading margin	**3.8**	dB

5GHz Uplink (WB ~100MHz)		
Parameters	**Value**	**Unit**
TX power @ PCB output	23.0	dBm
Noise bandwidth	8E+07	Hz
TX antenna gain	0.0	dBi
Range	**175.0**	m
Path loss @ range	117.0	dB
RX antenna gain	10.0	dBi
Received power (no fading) @ PCB input	-84.0	dBm
Noise floor	-174.0	dBm/Hz
integrated noise	-9.5E+01	dBm
RX noise figure	6.0	dB
Receiver degradation	0.0	dB
SNR @ BER = ?	1.0	dB
Sensitivity	-87.8	dBm
Fading margin	**3.8**	dB

Figure 4-16 Simple link budget for WB ~ 100MHz, 5GHz carrier frequency DL and UL connection.

The downlink-uplink imbalance is due to different transmission powers.

In 3GPP macro model [47], which is made for 2GHz centre frequency, urban environment and assumes the AP antennas 15 meters above rooftops, the path loss equation is shown below and the STD of the slow fading is 10 dB, which is however assumed to be too big.

$$PL(d) = 37.6 * \log_{10}(d/1000) + 128.1 \text{ [dB]} \tag{6}$$

Despite the different slow fading STD the same fading margin is used to make the comparison easier. Thus using these path loss models, the narrowband range is more than doubled in 2GHz centre frequency. Even though the comparison should not be based results with same fading margin, this emphasizes the problem of 5GHz network deployment.

5GHz Uplink (NB ~12MHz)		
Parameters	**Value**	**Unit**
TX power @ PCB output	23.0	dBm
Noise bandwidth	1E+07	Hz
TX antenna gain	0.0	dBi
Range	**360.0**	m
Path loss @ range	125.8	dB
RX antenna gain	10.0	dBi
Received power (no fading) @ PCB input	-92.8	dBm
Noise floor	-174.0	dBm/Hz
integrated noise	-1.0E+02	dBm
RX noise figure	6.0	dB
Receiver degradation	0.0	dB
SNR @ BER = ?	1.0	dB
Sensitivity	-96.8	dBm
Fading margin	**4.0**	dB

2GHz Uplink (NB ~12MHz)		
Parameters	**Value**	**Unit**
TX power @ PCB output	23.0	dBm
Noise bandwidth	1E+07	Hz
TX antenna gain	0.0	dBi
Range	**870.0**	m
Path loss @ range	125.8	dB
RX antenna gain	10.0	dBi
Received power (no fading) @ PCB input	-92.8	dBm
Noise floor	-174.0	dBm/Hz
integrated noise	-1.0E+02	dBm
RX noise figure	6.0	dB
Receiver degradation	0.0	dB
SNR @ BER = ?	1.0	dB
Sensitivity	-96.8	dBm
Fading margin	**4.0**	dB

Figure 4-17 Simple UL link budget for two different central frequency 5GHz and 2GHz

As can be seen from wide band results, the uplink range is significantly smaller than the downlink range. Based on these results, the narrow band range is roughly twice the wide band range, if the frequency allocations are roughly at the same frequency band. If narrow band allocation is in 2 GHz band, the range is about four times the wide band range. The discussion above points out that the cost of relay nodes increase if the performance requirements are stretched. Therefore, we should at least initially aim at keeping the relay range roughly on the same level with user terminal, although power constraints are not as stringent to the FRN as for mobiles.

4.3.1.3 Time Division Based Relaying for Dual Band Concept

The WINNER air interface concept proposal presents a dual bandwidth approach. The effects and requirements of dual band concept for the relaying are identified. We discuss the deployment schemes where the wide band relay nodes co-exist and co-operate with the narrow band mode. We then continue by considering possibilities and challenges of relay nodes using both wide narrow bands.

The narrow band frequency allocation might be with a centre frequency around 2GHz. The link budget calculation above shows that the narrow band coverage area is much larger than wide band coverage area in such frequency allocation scheme. For such a scenario one should study ways to improve wide band coverage within an AP service area, in order to provide an even capacity through out the service area, or at least make sure that the capacity can be increased in locations where the high traffic is expected.

Co-operation of narrow band and wide band mode provides some interesting features for the systems. It is easier to provide better frequency re-use scheme for the narrow band mode, hence the narrow band mode can be used isolating the wideband coverage areas. That kind of setup would reduce the effects of inter-cell interference. Moreover, narrowband could be effectively utilized for transmitting control information and small packet traffic. In particular, if narrow band mode does not have delays caused by multi-hop communication, the flows with very tight delay requirements should be transmitted by narrow band mode.

If modes are operating on different frequency bands, the two-hop deployment for wide band relays is not quite straight-forward – one has to assume that relays have longer ranges than user terminals or arrange multi-hop communication. One possible solution would be using a special feeder mode for FRN-AP communications. It still however unclear, how such feeder mode should work, and what kind of benefits it will offer.

The relaying using both wide and narrow band modes increases the hardware complexity of a relay node. As discussed above, the need for relaying in narrow band mode is less compelling. However, there could be situations or possible system concepts, where a unified dual band relaying concept should be considered. The dual band relaying is of interest in environments where neither radio mode can be received. This situation is likely to appear in indoor environments like garages, offices and homes. Dense urban regions are also of interest, although the capacity needs in urban areas might be high enough for justifying a dense AP network. Moreover, for 5GHz bandwidth the cell range can be rather limited also for narrow band mode; thus range extension might be of interest.

A major factor when considering relaying concepts for different radio modes concurrently is the differences of the propagation properties of the radio modes. If the coverage areas and penetration properties are very different, there is hardly a need for a joint dual band relay. If narrow band relaying is seen necessary, it should be implemented separately from wide band relaying, most likely using the same ideas and principles as in wide band relaying. However, if the propagation properties are reasonably similar, that is both mode use similar centre frequencies, dual band relaying could be beneficial.

The need for a dual band relaying depends on the degree of co-operation between the different radio modes. If narrow and wide band modes operate independently the gains from a dual band capable relay node come from combining two relay nodes to one. However, if the existence of narrow band mode is assumed in anyway for the wide band operation, the narrow band relaying would become almost mandatory feature. This need arises from bad shadow fading conditions, like indoor coverage. Moreover, if narrow and wide band modes can co-operate tightly, dual band relay could provide possibilities for optimizing the relaying scheme even though the both modes could operate independently.

Relaying can effectively combat the loss of connection due to shadowing by buildings. The relaying concept based on one presented in section 4.3.1.1 can be extended to dual band operation. The layer 2, however, will need more functionality, since GLL needs to be implemented - at least to a degree.

Extending the cell range by a dual band approach poses some additional challenges. The ranges of different modes are different, even though similar centre frequency would be in use. The narrow band mode has a longer range due to reduced noise levels, and possibly different link technologies used. At the same time, narrow band mode offers lower throughputs. Deployment of dual band relays within wide band range can cause problems with terminal association due to overlapping narrow band services. Moreover, providing seamless or almost seamless coverage for the wide band will require artificially reducing the narrow band range by power control. On the other hand, larger throughputs from wide band mode are mostly wasted, if RN-AP communication is done through narrow band mode. Heterogeneous approaches have fewer challenges than homogeneous relays for this kind of deployment scenarios.

4.3.2 Scheduling issues in wireless multi-hop networks

4.3.2.1 Introduction: scheduling for Fixed Relay Network

As explained in paragraph 4.2.2.2.2 of deliverable D3.1 [1], a promising new radio access network architecture envisioned is the Fixed Relay Network (FRN), where many advantages are expected in terms of coverage, flexibility, throughput and QoS provisioning [12]. A FRN (see Figure 4-18) is constituted by fixed transceivers, called relays, that establish a wireless mesh topology connected through APs (AP) to a core network. Relays offer connectivity to user terminals almost like APs in cellular architectures; for this reason FRN are also referred to as multi-hop cellular extensions [13].
In such networks support of multi-hop is mandatory and efficient routing and scheduling algorithms are required to exploit at best network resources and to efficiently manage internal connections and load from and to the core network resources.
In paragraph 4.2.5.4 of deliverable [1] we addressed the problem of optimal routing for FRNs [14], here we start discussing the key issues of transmission scheduling.

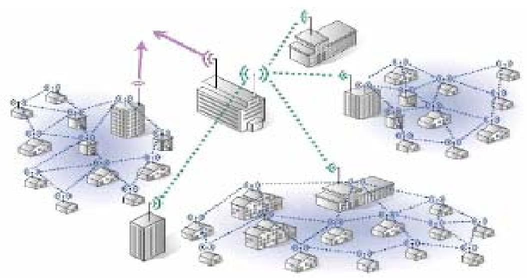


Figure 4-18 Fixed Relay Network (FRN)

Generally speaking, FRNs are based on transmissions over a shared medium. Therefore, the first issue to be considered for managing transmissions is the multiple access problem. Multiple access schemes for multi-hop wireless networks can be classified as random access schemes or reservation schemes. The access scheme adopted by IEEE 802.11 networks is a random access scheme based on CSMA/CA and RTS/CTS. Reservation schemes divide the radio resources into time slots and assign slots to nodes for transmissions according to a scheduling algorithm. Random access schemes have been recognized not adequate for multi-hop wireless networks supporting QoS. For this reason we focus here on reservation schemes and scheduling algorithms.

Scheduling algorithms can be used to plan a transmission scheme for a quite long period of time based on traffic statistics (static scheduling), or executed run-time to adapt to traffic dynamics (dynamic scheduling). Centralized scheduling algorithms require a central controller that gets all information and provide a solution, distributed algorithms can be run by all nodes in the networks and use local information only.

4.3.2.2 Centralized scheduling

A centralized controller must provide a scheduling scheme based on traffic requirements of network nodes and on interference compatibility constraints.

Obviously, traffic requirements depend on traffic relations (source-destination pairs), traffic intensity, and the routing scheme. As in [14] we assume that traffic statistics are known and that the routing scheme is able to provide a path for each source-destination pair without considering the transmission scheduling. Based on this information, we can calculate the number of slots needed for each node. Therefore, the scheduling scheme must assign the required number of slots to all nodes satisfying interference constraints.

Thus, to fully define the scheduling problem we need to model interference. A simple interference model assumes that parallel transmissions of a generic pair of nodes (i,j) are allowed or not according to the topological characteristic of an interference graph G(V,E). In particular, if $(i,j) \in E$, i and j are not compatible and they cannot transmit at the same time. A similar model with a different graph can be obtained considering compatibility between links. This model is that implicitly adopted by IEEE 802.11 access scheme that prevents transmissions from neighbors of i and j if i is transmitting/receiving to/from j.

Traffic requirements can be modeled considering a vector W, where w(v) is the number of slots required by node $v \in V$. Given the channel rate we can also define the set of slots available C.

Formally a scheduling solution defines a subset $f(v) \subseteq C$ for each node $v \in V$ such that:

- $\forall v, |f(v)| = w(v)$: each node gets $w(v)$ distinct slots;
- $\forall (u, v) \in E, f(u) \cap f(v) = 0$: two neighbouring nodes get disjoint sets of slots.

It can be easily observed that the problem of finding a feasible scheduling is equivalent to the well known graph multi-colouring problem (Figure 4-19) which has been extensively studied for the channel assignment problem in cellular systems [18][19].

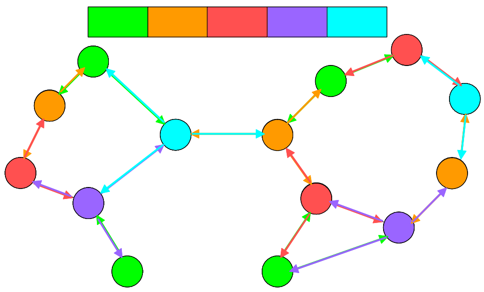


Figure 4-19 Graph multi-colouring

The graph colouring and multi-colouring problems are known to be NP-hard and are usually solved with heuristics. More complex interference models can be also designed considering explicitly the SIR (Signal-to-Interference Ratio) constraints, similarly to what proposed in [20] for the channel assignment problem.

New models and algorithms for the centralized scheduling problem is therefore currently under study, in more detailed manner.

4.3.2.3 Ditributed scheduling

Distributed scheduling algorithms require that each node is able to compute the slots that be used based on local information. The concept of local information is somehow misleading. In multi-hop wireless networks it is commonly assumed that each node can exchange information with its neighbours. Therefore, we can assume that a distributed scheduling algorithm run by a node can use the information on the slots used by the same node and its neighbours.

It is evident that distributed scheduling algorithms based on the compatibility model presented in previous section can be defined if the interference graph is also the network connectivity graph.

Of course, such distributed schemes must be supported by a MAC layer that is able to provide local connectivity information. In [21] a new MAC scheme that provides such information and signalling message to support slot reservation is presented. We are currently working on defining distributed scheduling schemes based on such MAC protocol.

4.4 Assessment of mobile relays deployment concepts

4.4.1 Introduction

The concept of mobile relays has been introduced from previous work in [1]. A number of areas have been identified, mainly related to deployment concepts, types of mobile relays, ser/Usage Scenarios and deployment concepts, list of issues that need to be investigated. As stated in [1], the main issue with mobile relays, with reference to the fixed relays, is the mobility and as a result, the non-uniform reception power levels also resulting in the non-guaranteed coverage area and the non-continuity of service provision. Thus, in this section we intend to address some of those issues by selecting some specific deployment concepts and investigating some aspects related to them. Additional work on mobile relays with reference also to the cooperative relaying concept can be found in Section 2.6.3.

4.4.2 High level description of specific deployment concepts for mobile relays

As it has been pointed out in [1], due to the large number of degrees of freedom e.g. types of mobile relays, usage cases, mobility scenarios etc, a large number of "cases"/deployment concepts can be identified. Each one of these concepts needs specific investigation and although some parameters/characteristics are generic for all of them, a number of other characteristic are different and as such each deployment concept needs to be investigated on its "own". Thus, due to the large number of deployment concepts, in this section we will select two of them in order to perform further investigations. The criteria for this selection were how realistic they are, probability of adoption for future network deployment, complexity and ease of initial evaluation. Those scenarios are the following

Scenario 1→ ("Park scenario-MRtB"). We assume that mobile relays are fitted on buses, which provide coverage in a park. (MRtB→Mobile Relay type B). An example for Scenario 1 is the one depicted in Figure 4-20. Hyde Park in London is shown and around it all the public transport buses (bus numbers) are depicted. Thus, mobile relays could be fitted on those buses to provide coverage to the park. The coverage is shown by the circles.

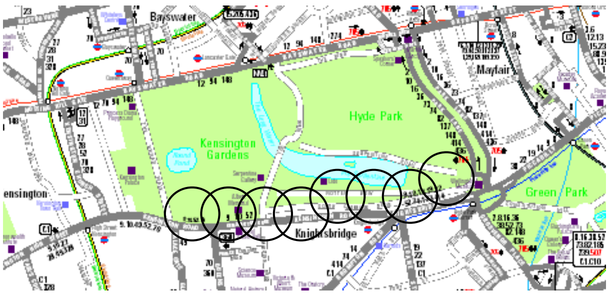


Figure 4-20 Hyde Park coverage from mobile relays (Black circles)

In order to evaluate this deployment concept, we use the "model" depicted in Figure 4-21. Specifically, we assume that the Mobile Relay (MR) is moving with a velocity of V on the horizontal axis (X), vertically to the AP – User Equipment (AP-UT) connection line. Z is the distance the relay is covering. d(AP_Relay) and d(Relay_UT) are the AP-Relay and Relay-UT distances on the (Y) axis. R1, R2,..Rn are the different positions of the relay with a distance equal to the step we will use in the simulation. UE1, UE2, UEi are the different positions of the UT on the vertical axis. For each case (multiple positions of a Relay) the UT will be assumed to be stationary.

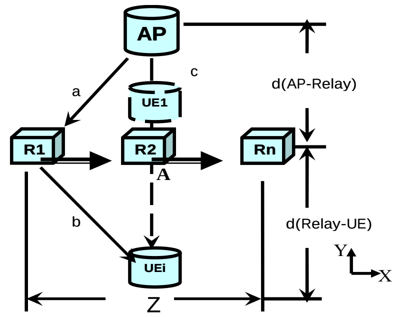


Figure 4-21 Model for Scenario 1

In this scenario the intention is to evaluate the quality of the reception at the UT for multiple positions of the UT and the relay, for a number of degrees of freedom. At the same time, other issues like continuity of services will be analyzed and discussion will take place.

Scenario 2→ ("User-terminal relay - MRtA"). We assume that in a hotspot/urban area, e.g. public square / park area, a number of UTs (or just one) are (is) elected to provide relaying functionalities to other terminals. (MRtA→Mobile Relay type A). Scenario 2 is more applicable (and also characterized) for cases of a large concentration of UTs with low or no - mobility. The model used for Scenario 2 is shown in the Figure 4-22. What is shown in Figure 4-22 is a coverage area from an AP and groups of different types of terminals. Each of those terminals could potentially be used as a mobile relay, depending, of course, on some kind of selection mechanism. We assume that each of them, as depicted with different shapes, have different capabilities, characteristic etc. For instance, Type5 terminals (user phone) have high mobility, low power availability and limited functionalities, whereas Type1 terminals (e.g. laptop) are characterized by very low /no mobility, high power availability and high processing capabilities.

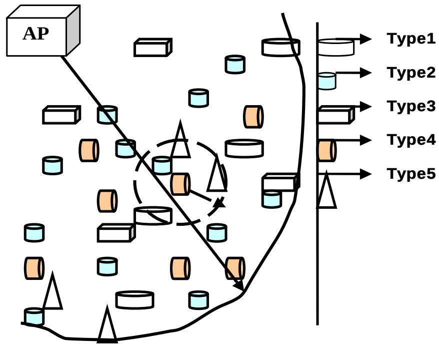


Figure 4-22 Model for Scenario 2

The high level process for such a model could be the following

- Definition of terminal groups based on their capabilities
- 1st level evaluation of all terminals and selection of the best ones e.g. use those of low mobility
- 2nd level evaluation of specific parameters for those terminals selected to provide relaying capabilities e.g. connectivity issues
- Cyclic process of re-evaluation of terminals/deployment parameters based on triggers/periodic events etc.

This process can be done either at the AP or at a UT identified as "Target_UT". In our case we will assume the latter. The dotted-line circle is assumed to be the area of interest around a "Target_UT" where we will look for potential mobile relays.

- The above two Scenarios were selected to address two different issues: Scenario 1 →Evaluation of the coverage / cooperative mobile relaying schemes and Scenario 2 → Connectivity issues for cooperative mobile relaying

4.4.3 Specific Description of two deployment concepts/Requirements

Scenario 1

For Scenario 1, some general degrees of freedom with reference to the Figure 4-21 are

- AP→Tx power: Pt(AP), AP-Relay distance, AP-Relay channel characteristics, Antenna characteristics,
- Relay→Tx power: Pt(Relay), Velocity, Trajectory, Relay-UT distance, Relay-UT channel characteristics, types of relays
- UT→AP-UT distance, AP-UT channel characteristics, trajectory, velocity
- Other→Number of Relays, snapshot period for relay movement, relay scheduling (inter arrival time), number of receivers in the UT etc

The values chosen for the simulation will try to reflect different cases of typical deployment concepts of small/large cells, urban/ bad urban area e.g. Pt(AP)=5/10/20W, Pt(Relay)=1/5/10W. What we will evaluate for Scenario 1 is coverage/reception levels of direct and indirect path from the AP and Relay respectively, SIR measurements in the UT and cooperation mechanisms. The main degrees of freedom under which this Scenario will be evaluated will be the (AP – UT) distance, (AP – Relay) distance, Pt(AP), Pt(Relay), Path loss models, Number of combined paths at the UT (under the cooperative relaying scheme)

What Figure 4-23 shows is the received power from the direct path (from the AP) and the indirect path (from the Relay) for the Case 1 parameters from Table 10-4.

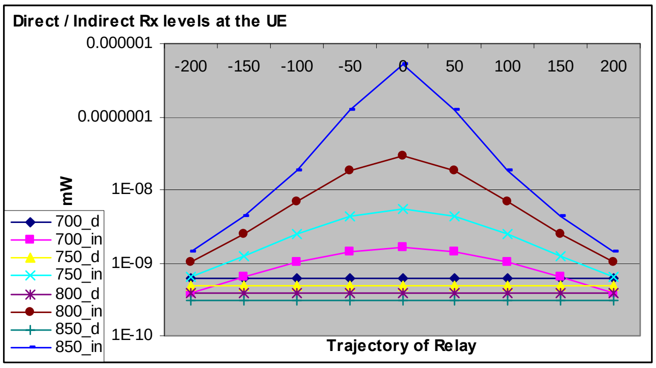


Figure 4-23 Received power levels for direct/indirect path for Case1

The main issues related to this graph are the following

- Reception power from the AP is always constant for each UT position. (as expected)
- The received power at the UT from the indirect path follows a "Gaussian-like" distribution.
- The more the relay moves towards the minimum distance of the UT-AP (Point A in Figure 4-21), the higher the gain from the indirect (relayed) path is
- The highest gain is witnessed in the area of [-100m, 100m] on the X axis for the relay trajectory
- On the horizontal axis for the Relay, even at the points of +/- 150 from Point A, the gain of the indirect path at the UT is reasonable, although for a realistic scenario we would rather "operate" the relay when it is in the area defined at +/- 100m, as stated in the previous point
- This graph is quite basic and will be used to initiate some discussion on other areas like power control, cooperative relaying, scheduling of MRN etc.

Scenario 2

Scenario 2 will be mainly investigated under the perspective of cooperative mobile relaying. Some of the issues to be analyzed will be the instantaneous /average connections, number of connections per device, and distance profiles.

In this Scenario we assume that a "Target_UT" (as part of large population of UEs) will be trying to select the best mobile relay(s) to establish communication with as shown in Figure 4-22. Mobility of terminals follows Brownian motion / random walk. The above will be evaluated for a number of degrees of freedom including the Cell max range, number of potential mobile relays, max distance from the target UT in which to look for MRN, MR/target_UT velocity.

4.4.4 Simulation results regarding coverage investigation

The main results presented here, similarly to [3], show the difference of the received power levels at the UT, for different cases for the direct and the indirect path. With that, we effectively define those areas that the UT would be receiving the relayed rather than the direct path. We denote *deltaPr* the difference (in dB) between the indirect and direct path at the UT. A number of cases are identified. The parameters related to those cases are presented in the tables, presented in the ANNEX III specifically Table 10-4, Table 10-5 and Table 10-6. Similar results are also included in the Annex III.

Figure 4-24 shows the deltaPr for a Relay position at 900m and the UT at distances of [700m, 1100m] from the AP, with a step of 50 meters. For almost all the cases, deltaPr is positive which means that for almost the whole area of 400x500 meters for all positions of the relay, the UT would rather be "locked" to the path through the relay. It is also shown, as we will better see further down, that for symmetrical places around the Relay, the reception levels at the UT are better for the AP→Relay→UT configuration rather than the AP→UT→Relay. This should be used in planning processes, that for a fixed point/trajectory of a relay, the target should be to provide coverage to the UEs which are on the AP→Relay→UT topology configuration. Of course, under other circumstances e.g. cooperative relaying, that might not be the case. For the position of 900m meters, the gain is very high because the relay is effectively passing by just next to the UT. Figure 4-24 is a quite informative 3-d graph that portrays in the x-y-z domain the gain in terms of received power levels. So, it would be interesting if for future work for more complicated models this type of graph could be used to portray reception levels for an area, always with reference to the time domain as long as for mobile relays, location and, as a result, power will always be changing.

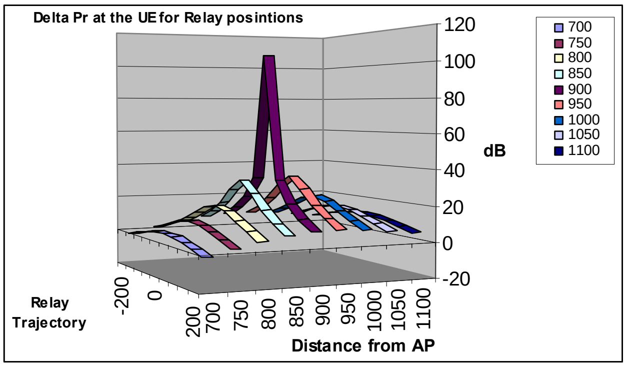


Figure 4-24 3-d Graph for DeltaPr for case 1

We have to state that for all the cases we will be investigating, we assume that the Mobile Relay transmits at a fixed power, which is a percentage of the AP power. Figure 4-25 (a) is a 2-d representation of Figure 4-24, for Pt(Relay)=50%Pt(AP). It portrays in a more visible way the crossings of the lines with reference to the zero-point axis (X-axis) and shows the positive or negative gain of the indirect over the direct path. It is also visibly easier to see when comparing different cases. Specifically, from Figure 4-25 (a) we see that

- A "Gaussian-like" distribution of the deltaPr is shown for some of the cases.
- For positions of the UT close to the Relay e.g. 850/950m, the gain is not linear with reference to the distance the relay is "travelling". There is an area, that of [-100, 100m], where the gain is

increasing/decreasing substantially. For instance, for the 950m case, the difference in dB between DeltaPr(200m) and DeltaPr(100m) is around 19-8=11 dB , whereas the difference for DeltaPr(100m) and DeltaPr(0m) is around 34-19=15 dB.

- For symmetrical positions of the UT to the relay, the closer the UT is to the relay, the less the difference is. For instance, DeltaPr(850m)-DeltaPr(950m)=34-32=2 dB, whereas DeltaPr(1050m)-DeltaPr(750m)=16-9=6 dB.
- The difference between the min/max points (relay positions of 200m and 0m) for the case of 950 meters is that of 34-8=26 dB, whereas for the 1100m case is only 11-5=6 dB.

Most of the above calculations are taken (where applicable) for the relay position of "0".

Figure 4-25 (b) is the same as Figure 4-25 (a), but power of the MR is changed and is 10% of the Pt(AP). What we see is that the same type/trend of graphs is witnessed. The difference between min and max values for the UT at 950m from the AP is that of 27-2=25dB, same as in the previous analysis. (Applicable for the relay at the positions of 0m and 200m). There is roughly a 5-7 dB difference between most values of Figure 4-25 (a) and (b). However, even with Pt(Relay)=10%Pt(AP) we still get acceptable gain of the indirect over the direct paths at distances close to the Relay.

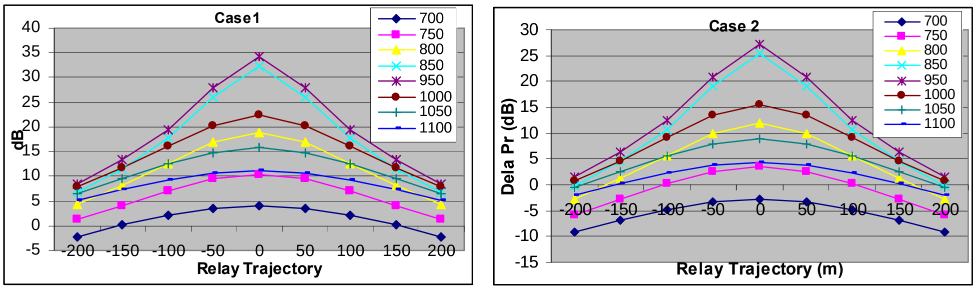


Figure 4-25 DeltaPr for different power levels for Case1 (a) and Case2 (b)

All SIR values will be calculated for all the positions of the Relay [-200, 200] with step 50m.

4.4.5 Conclusions

What has been shown so far is that even for relative small distances from the AP e.g. 400m, mobile relays can offer quite substantial gain compared to the direct path from the AP. Of course a number of issues need to be studied into more detail e.g. presence of mobile relays. However, if some of those issues are solved e.g. either indirectly by scheduling more mobile relays or indirectly by using mobile relays when they are close to the target area and for fast "applications" , then the system could gain from the presence of the mobile relays. In this section we evaluated some simple coverage results. However, as it was evident in the analysis a number of other issues surfaced, some of which we plan to investigate in the future. Thus, future work could include more realistic scenarios taking into account a non- uniform interarrival time of buses, distribution in time, traffic models (in streets), etc, different capabilities for those mobile relays e.g. different power allocation techniques

4.5 Routing and Forwarding in Multi-hop cellular networks

4.5.1 Routing and Forwarding in Multi-hop cellular networks

It has been accepted that employing relays in a cellular network has the potential to solve the coverage and capacity problems for high data rates in macro-cells and hot-spots. A number of parameters affect the performance of a multi-hop cellular system including the number of relays, their location, and radio resource coordination strategy between the users, relays and access points as well as routing and forwarding method. It is considered routing in both fixed and mobile relays however the main emphasis is on the fixed relays. We also concentrate on the non-contention-based MAC access in cellular frequency band. The developments in the contention-based multi-hop communication e.g. IEEE 802.11s is also closely followed.

Hereby "routing" is considered as a mechanism that establishes and maintains a route between source and destination. Note that in a multi-hop cellular network "source" and "destination" are defined regarding to the access network. Therefore, for the uplink, by "destination" we mean access point and similarly for the downlink by "source" we mean access point. Moreover, "forwarding" means the process of data transmission from a node to its participating neighbouring node along a route. In Subsection 10.3.1 of Annex III we have presented a brief overview on the routing and forwarding methods in ad-hoc networks and their appropriateness for multi-hop networks.

Routing and Forwarding (R&F) methods for multi-hop cellular networks would try to utilize and exploit the specific inherent characteristics of multi-hop cellular networks, these characteristics include:

- *Small number of hops*: There is a small number of participant relays in a route in an actual cell of a full scale multi-hop cellular network compare to a full scale ad-hoc network.
- *Central processing*: In multi-hop cellular networks there is an infrastructure that is in contrast with the case of ad-hoc networks. Therefore implementation of central processing and decision-making is promising. This feature not only simplifies the route selection and forwarding process but also provides the opportunity of utilizing the routing and forwarding mechanisms to adaptively manage network resources. That can be performed jointly with other layers functionalities (e.g. joint route selection and packet scheduling).
- *Location information and data flow direction*: The location information and flow direction in both uplink and downlink are available, especially for the case of using fixed-relays.
- *Multi-user diversity:* The availability of diverse routes to different access points provides a very rich diversity, which can be exploited using appropriate access point assignment, routing, forwarding and scheduling.
- *Less energy constraint*: In contrast with the case of ad-hoc networks, for the case of using fixed-relays the power constraint and energy efficiency are not the main objective. Note that this is
- *Cooperation incentive:* This characteristic is basically valid for mobile relays. Due to the fact that there is an infrastructure which can deal with charging issues, consequently there could be a framework that encourages the users to cooperate. For the case of utilizing fixed-relay, relays are a part of network infrastructure that is provided by the network provider.

We should add that the scalability of routing algorithm here is not as critical as it is in ad-hoc networks. Because the routing and forwarding mechanism should be applicable for one or more cell(s)/sector(s). In the followings we discuss the main challenges and fundamental trade-offs in the designing of appropriate routing and forwarding methods for multi-hop cellular networks.

4.5.1.1 Challenges and trade-offs

The main challenge of designing a routing and forwarding method for multi-hop cellular network is to consider inherent characteristics of multi-hop cellular network (as was presented in Subsection 4.5.1) and to exploit the knowledge in the problem to provide the required QoS to the users. In other words, an appropriate routing/forwarding method may regard the each "route' as an actual network resource that should be managed and utilized opportunistically to improve system efficiency utilizing the most available knowledge. Accordingly, there are a number of challenges that includes the followings:

- Complexity: Computation and communication complexities are the basic challenges. However in this specific case, the communication complexity is more critical. There is a fundamental trade-off between the communication complexity and capacity in multi-hop networks that should be specially considered for designing the forwarding methods. Communication complexity can be defined as the actual communication overhead required for transmission of a unit of data from the source to the destination. Usually the computational complexity is a function of the number of parameters involved into making a decision or performing an action. It is also a function of the complexity of the decision procedure. However, there is an opportunity of performing central processing in the access point; both forwarding and routing methods should be optimized for minimum computational complexity. Note that in some circumstances the communication complexity can be replaced by computational complexity through more complex decision procedures.
- Exploiting user level diversity: As it is mentioned earlier there are different kinds of diversities in user level that can be exploited using scheduling and forwarding methods. Diversity exists in different levels: diversity of appropriate links for forwarding to the next hop, diversity of routes in a cell, and diversity in routes in a network. The diversity can be exploited through packet forwarding, route selection and access point assignment respectively.
- Unavailability of perfect knowledge: Most of the methods are based on the assumption of the availability of perfect knowledge in appropriate time (e.g. channel state, upstream queue length, etc.). It may not be practical in actual cases.

- However, there are fundamental differences between ad-hoc and multi-hop cellular networks, it is observed in the literature that ad-hoc routing algorithms are usually customized and improved to fit into multi-hop cellular networks. In the next paragraph we provide a brief survey and provide some general observations on ad-hoc routing methods.

4.5.1.2 Routing and forwarding for different scenarios

Since the scenarios in multi-hop networks cover a range of situations including mobile and fixed relays, homogenous and heterogeneous relays as well as contention based an non-contention based MAC, it would be inefficient (and most likely impossible) to conclude on a generic multi-purpose routing forwarding mechanism. Here we emphasis again that our main focus is fixed relays with non contention-based access in cellular band. Therefore, that would be essential to 1) define different relaying modes, then 2) selecting a set of forwarding and routing method(s) suitable for each mode 3) harmonizing these methods to be considered as a part of future standards. In this regard the special consideration of IEEE 802.11s would be a matter of concern.

We note that inter-system interconnection and routing in a ubiquities communication environment can be treated as the separate problems. The former can also be considered as an upper layer routing functionality in a hierarchical routing framework for seamless inter-system inter-connection.

In the future reports we will provide a specific set of key parameters and factors for performance evaluation and effectiveness evaluation of routing and forwarding methods for different scenarios base on the output of WP2 on the physical layer mechanisms. We will also examine the proposed methods in WP3.2#3.

4.5.2 The 802.11s approach to the homogeneous relaying deployment concept

In the contest of the WINNER's short range scenarios employing homogeneous relaying deployment concepts, the work currently performed within the IEEE 802.11s standard committee [63] must be mentioned and taken into account. The reasons that leaded to the creation of a specific amendment to the original specification came from some commercial as well as technical considerations:

- Multiple vendors of mesh networking products already exist in the marketplace, serving different customer needs and providing solutions for different deployment environments
- Specification of a standard way for mesh products from different vendors to interconnect is likely to fuel large-scale adoption of such systems
- Interconnectivity across domain boundaries is likely to emerge as an important market requirement

This means that the approach considered is the same adopted within the WINNER project with respect to the definition of the homogeneous relaying deployment concepts.

This Task Group (TG) has as scope to develop an IEEE 802.11 Extended Service Set (ESS) Mesh with an IEEE 802.11 Wireless Distribution System (WDS) using the IEEE 802.11 MAC/PHY layers. The project aims to provide a protocol for auto-configuring paths between APs over self-configuring multi-hop topologies in a WDS to support both broadcast/multicast and unicast traffic in an ESS Mesh using the four-address frame format or an extension. The above need has risen because of the observation that original specification of the 802.11 standard provided a four address format for exchanging data packets between APs for the purpose of creating a WDS, but did not define how to configure or use the latter.

An IEEE 802.11 Extended Service Set (ESS) Mesh is a collection of APs interconnected with wireless links that enable automatic topology learning and dynamic path configuration. It is functionally equivalent to a wired ESS, with respect to the Stations (STAs) relationship with the Basic Service Set (BSS) and ESS.

One of the crucial features that the future standard requires is the interoperable formation and operation of an ESS Mesh. Further, it shall be extensible to allow for alternative path selection metrics and/or protocols based on application requirements.

A target configuration is up to 32 devices participating as AP forwarders in the ESS Mesh. However, larger configurations may also be contemplated by the standard. From the WINNER point of view, it is important to notice that the intended architecture shall allow an ESS Mesh to interface with higher layers and to connect with other networks using higher layer protocols. More precisely the amendment is supposed to allow the use of one or more IEEE 802.11 radios on each AP in the ESS Mesh.

The technical aspects that are going to be carefully investigated during the standardisation process concern mainly three points or criteria (as they have been called) that will make certainly raise the most challenging issues. It is worth taking a deeper look considering that they will likely affect more the technical contributions. Further, they made up the framework on which the investigation and the technical contribution have been based so far.

1. Compatibility
   In principle all the standards belonging to the 802 family are supposed to be in conformance with the IEEE 802.1 Architecture, Management and Interworking documents as follows: 802.1 Overview and Architecture, 802.1D, 802.1Q and parts of 802.1F. In this respect the purpose of the project has been edited with the clear aim of ensuring the mentioned compatibility. More precisely the formulation of the criterion states that ESS Mesh specifies one possible Wireless Distribution System (WDS) that behaves in every respect as an IEEE 802.11 Infrastructure Mode network. As such, it is entirely compatible with the IEEE 802.11 architecture and, by inference, compatible with the IEEE 802 architecture, including IEEE 802.1D, IEEE 802.1Q, and IEEE 802.1F.

   A similar approach is also adopted in the design of the WINNER system where the definition of new deployment concepts and new air interfaces has to take into account the current existing solutions that the market already uses. Especially in short range scenarios such as hot spots or in building, the extraordinary success that the IEEE 802.11 is witnessing can not be neglected.

2. Distinct Identity
   Before the creation of the TGs, it was needed to clearly identify the peculiar aspects of the future standard with respect to other standards effort ongoing currently and the issues to be addressed and not already treated in other groups/committees.

   It has been stated:

   - The original 802.11 standard does not define how to configure or use a WDS. The project to be developed in the future task group has to define this missing functionality by providing a mechanism to produce an ESS Mesh in an auto-configuring manner.
   - The 802.1D has been designed for wired environments and treat mobility as extraordinary events
   - The IETF MANET group is focused on the design of Layer 3 routing protocols to set, as well as to exploit, peer-to-peer communications between STAs. Moreover instantaneous radio awareness is out of scope of MANET unlike ESS mesh should provide tight integration between MAC and the multi-hop WDS
   - The TGf might appear as addressing similar issues but it investigates the interoperability between APs from different vendors focusing on fast handoff support for radius problems while the aim of TGs primarily concern itself with topology discovery, delivery, and mobility management for a wireless DS implementation

   With respect to the existing 802.11 standard family the ESS Mesh is the only one aiming to define a wireless distribution system.

3. Technical Feasibility
   As probe of the technical feasibility of the ESS Mesh project, a project carried on in the U.S Naval Research Lab (NRL) was provided. It was based on the Dynamic Backbone Algorithm (DBA) and was finalized with implementation and field test in a real world environment. Besides, some tests using mobile vehicles were conducted and measures of throughput, delay and packet loss were performed. Those tests validated the use of DBA as a possible framework for the future standard and proved the existence of scientific basis for it.

Following the creation of the Task Group, several groups seem to have been created in order to deal with different aspects of the project. More precisely two groups seem to be currently active and have issued some documents: the TGs ESS Mesh Network Architecture Framework Ad Hoc team, and the Mesh Media Access Coordination Ad Hoc team.

Given the recent start of the standardization work many contributions have been proposed, but still no clear direction for the definition of the system have appeared. It is worth to mention anyway that some interesting and revolutionary concepts have been presented ([66], [64], [65], [67]) and a clear differentiation with the IETF MANET work has been claimed.

4.5.3 Flexible Extension of the Infrastructure-based Network via Mobile Relays

Infrastructure-based wireless networks can be extended in their coverage by mobile nodes with relay-capabilities spontaneously acting as forwarder for other nodes outside the range of components belonging to the infrastructure, specifically APs. In such networks nodes can communicate without the need of an infrastructure with each other by using intermediate nodes as relays, resulting in multi-hop communication. Thus, data from / to the Internet have to traverse several hops between the AP and those stations over a respective path. One basic challenge in such kind of networks is the routing. One promising approach, which is called location-based ad-hoc on-demand distance vector routing (L-AODV) has been described in deliverable D3.1 [1]. It is a hybrid routing scheme, which combines AODV as an on-demand topology-based routing scheme with the location-based stateless routing. The location-based routing, which is also referred to as geographical routing is introduced in the route-repair process of AODV, since it does not require route maintenance and determines the next hop towards the destination on-the-fly. To select the next neighbour towards the destination, greedy forwarding is chosen since this is a simple and efficient method with knowing only the location of the neighbours and the final destination. Especially in highly mobile environments frequent route breaks happen and can be repaired in an efficient way, improving the overall performance. But even for low mobility and almost static environments AODV provides quite low overhead and good performance, which makes the hybrid routing scheme a promising approach for a large range of mobilities. In the following section the performance is shown by means of event-driven simulations.

4.5.3.1 Performance Evaluation of L-AODV

The L-AODV routing protocol has been implemented in ns-2 [10]. The air-interface model is similar to that of IEEE 802.11, with a radio transmission range of 250 meters and a transmission rate of 2 Mbps. The packet size is fixed to 512 byte. 20 sources deliver packets at a rate of 4 packets/s to randomly selected destinations. A different number of stations are uniformly distributed over a rectangular area of dimension 1000 m × 1000 m. Stations move according to the random waypoint mobility model. They randomly choose their destination with a random speed between 0 and 20 m/s. Once the destination is reached, a new destination is randomly chosen after 30 s pause time. The simulated time is 200 s.

In order to compare the performance of the two routing protocols, end user perceived performance metrics have been measured, i.e. throughput resp. packet delivery ratio and transmission delay. In addition the overhead has been determined.

4.5.3.2 Throughput Performance of L-AODV

In Figure 4-26 the throughput as function of the number of stations is shown.

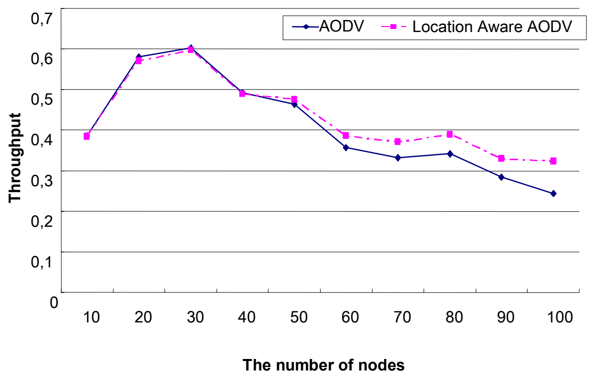


Figure 4-26 Throughput for AODV and L-AODV

Higher throughout can be achieved for L-AODV for high station densities. However, when the number of stations is below 40 conventional AODV and L-AODV have almost the same performance. L-AODV benefits from the fact that with a higher number of stations the probability increases to find a neighbor that is closer to the destination.

4.5.3.3 Delay Performance of L-AODV

In addition to more packets delivered, L-AODV also delivers packets significantly faster, cf. Figure 4-27.

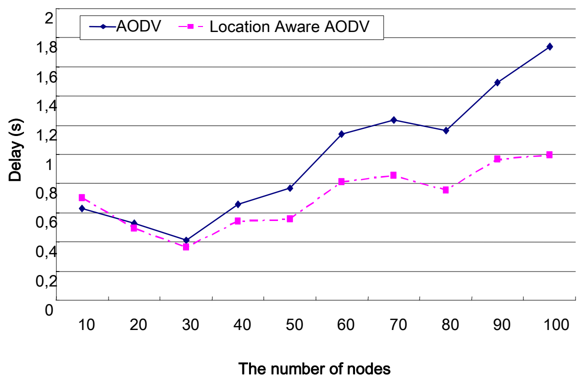


Figure 4-27 End-to-end packet delay

Similar to the comparison of the throughput, the performance gain of L-AODV increases with increasing density of stations. L-AODV saves the time in route recovery process, especially in the network with a number of stations exceeding 40. This behaviour can be explained by the immediate forwarding of the packets after the next-hop neighbour has been found. No timely route discovery and setup is needed, only the beacon packet has to be broadcast and at least one responds from a station closer to the destination has to be received.

4.5.3.4 Overhead Performance of L-AODV

In the following Figure 4-28 the overhead introduced by the routing protocol for AODV and L-AODV is depicted.

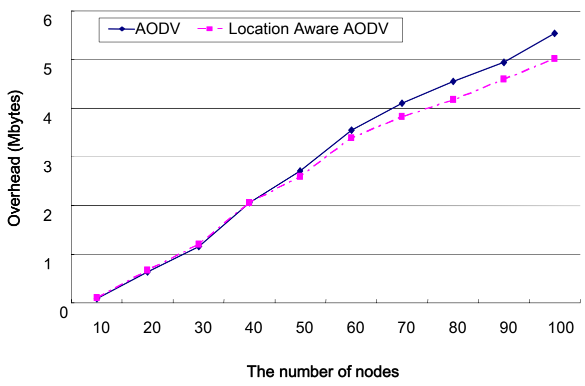


Figure 4-28 Routing protocol overhead

It can be recognized that L-AODV needs slightly less overhead than AODV, especially for high number of stations.

From this and the previous results it becomes obvious that the new Location-aware AODV protocol outperforms the conventional AODV routing protocol. The combination of on-demand topology-based routing and the geographical forwarding copes well with different network topologies and operation conditions. In case of a low number of stations the performance of the new scheme shows as good performance as the conventional AODV. But when the number of stations increases it benefits from its feature of greedy-forwarding. Hence, depending on the network condition it always switches to the mode of operation that achieves a good performance with respect to throughput, delay, and overhead. L-AODV combines the advantageous of the on-demand routing protocol AODV with the good performance of location-based routing in rapidly changing network topologies with high station densities. It is expected that the benefits of L-AODV can be enriched when introduce Location-Aided Routing (LAR), since the required information needed for LAR are discovered by L-AODV. This considerable reduces the overhead during route discovery. Furthermore, LAR also can be introduced for the local-repair process in AODV as alternative to greedy forwarding. This, however, introduces slightly higher delays but most probably improve AODV, resp. decrease the overhead. Another approach instead of greedy forwarding is directional flooding or “distance-aware” forwarding where the node closest to the destination will forward the packet as the first one. A relay having received the packet will therefore be in contention with other relays having received the packet, too. Intelligent selection strategies for the relay to transmit next have to be developed.

4.5.4 Performance Evaluation of the Multi-Constrained QoS Routing Algorithm

4.5.4.1 Introduction

Due to the need for Quality of Service provisioning in WINNER, the investigation of a joint approach to the Multi-Constrained Routing algorithm and the Optimized Link State Routing protocol was proposed in [1]. Since then the main emphasis has been laid on evaluating the applicability of the OLSR protocol to the specific WINNER network architecture along with verifying the efficiency of the Dijkstra routing algorithm, and especially its generalized version, intended to be the basis for the multi-constrained approach. This solution makes it possible to find k shortest paths between a pair of source and destination nodes. As a result an application may select such a route from the resulting set which best meets the criteria given. So not necessarily the shortest path must be chosen, but for example the longer one with more bandwidth available. For the investigation purposes the ns-2 network simulator [42] equipped with the OOLSR plug-in [43] was exploited. Significant effort in the initial phase was also put on verifying the compliance of the aforementioned OOLSR extension with the RFC 3626 [44] and on identifying the necessary updates to the source codes of the entire simulator.

4.5.4.2 The assumptions for the simulation research

According to the initial assumptions for this phase of the WINNER project, no mobility issues were taken into account. As a result a mesh network of 49 fixed nodes was considered, which where distributed regularly, however a bit unevenly so that a better evaluation of the investigated solution could be possible (Figure 4-29).

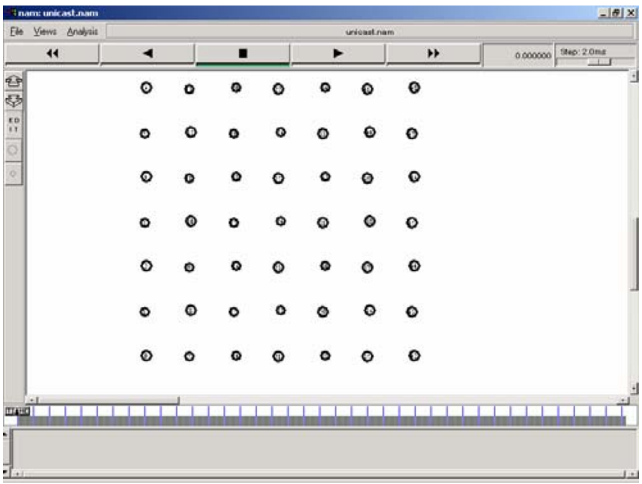


Figure 4-29 Topology of the considered network

To this end the position *(X, Y)* of a node was assumed to be a function of its number *i* and was defined in the following way:

$$\begin{aligned} Y &= \left(i \bmod \sqrt{N}\right) * 100 + 1.5 * (-1)^i \\ X &= \left\lfloor i / \sqrt{N} \right\rfloor * 100 + 2.5 * (-1)^i \end{aligned} \tag{7}$$

where *N* represents the number of nodes and is an integer having an integer square root.

At this stage of the research there were always one transmit and one receive node selected. A *UDP Agent* connected to the transmit node was generating a constant bit rate *CBR* traffic with the packet size of 128 bytes and the interval of 100 ms. Among other important parameters, the *WirelessChannel* with the *TwoWayGround* propagation model was used, the *802_11* medium access control and the *DropTail/PriQueue* queueing mechanism. The overview of the *ns-2* simulator equipped with the *OOLSR plug-in* and the current state of its further modifications along with the results are presented in the subsequent sections.

4.5.4.3 An overview of the ns-2 simulator equipped with the OOLSR plug-in

The *OOLSR plug-in* is implemented in accordance with the *RFC 3626*. The implementation is very precise especially in that part of the source code, which contains the *Breadth First Search* routing algorithm [45]. There is also a simplified version of this algorithm provided, which is optimized for highly complex simulations and therefore does not support *multiple interfaces* as well as *MID* and *HNA* messages [46]. *MID* messages are sent by *OLSR* nodes in order to declare their *multiple interfaces* over the network. As *multiple interfaces* are not supported, there is also no need for processing this type of messages. *HNA* messages are connected with the *Host and Network Association Information Base*, which contains the information about the nodes that may act as gateways to the associated hosts and networks. These features are not of the most importance and that is why it was decided to modify the simplified version of the algorithm at first.

4.5.4.4 Modifications to the ns-2 simulator and the OOLSR plug-in

According to the *RFC 3626* [44], the routing table is updated in the following cases:

- a change in the link set is detected,
- a change in the neighbour set is detected,
- a change in the 2-hop neighbour set is detected,
- a change in the *Multiple Interface Association Information Base* is detected (as it was mentioned before, this case is beyond the scope of this research at least for the time being).

It is crucial to note that the process of making updates to the routing table does not result in any additional exchange of messages either in the *1-hop neighbourhood* or in the remaining part of the network. The routing table is formatted in the following way:

```
1.  R_dest_addr      R_next_addr      R_dist      R_iface_addr
2.       …                …               …             …
```
(8)

At the beginning all the entries form the routing table are removed before the calculation of the new ones and then the information concerning the topology is collected. Finally the *Breadth First Search* algorithm is executed, which is intended to find the shortest paths to the specific destination nodes, each *R_dist* hops away. Those nodes are identified with the *R_dest_addr* and are accessible via the interface identified with the *R_iface_addr*. The problem is that the total metric *R_dist* is always incremented by 1, which is obviously against the *Quality of Service* provisioning. Nevertheless, for the purposes of the first approximation of the proposed solution, the *Breadth First Search* algorithm was replaced with the *Classical Dijkstra* one with unitary link costs. After that the more appropriate case with variable link cost was investigated, where the cost was based on the distance between each pair of nodes, which was calculated with the use of the coordinates. This case should be then perceived as the first evaluation of the multi-constrained approach. For the sake of clarity an example operation of the *Classical Dijkstra* routing algorithm with variable link costs, running on an *OLSR* node, along with a part of the debug report generated by this algorithm, is presented in the annex in the subsection entitled: Performance Evaluation of the Multi-Constrained QoS Routing Algorithm.

The following parameters were evaluated during the research: *NRTE* (no route available) and *IFQ* (no buffer space) for both the *Breadth First Search* algorithm, provided with the *OOLSR plug-in*, and its modified version in the form of the *Classical Dijkstra* algorithm. The results are presented in Table 4-5.

Table 4-5 NRTE and IFQ parameters evaluation

	Breadth First Search (original plug-in, unitary link cost)	Classical Dijkstra (modified plug-in, unitary link cost)	Classical Dijkstra (modified plug-in, variable link cost)
IFQ [%]	0.81	0.81	0.81
NRTE [%]	0.04	0.04	0.04

The final extension of the *Classical Dijkstra* routing algorithm to its *Generalized* version seems to be straightforward however it caused a lot of unpredictable problems in the final stage of the research. Those problems were connected with the necessity of very complicated reorganization of the *ns-2* source codes and the lack of documentation for the *OOLSR plug-in*. Finally it was decided to start the development of a stand alone simulator for the purposes of the validation of the target system. The results are intended to be presented soon.

4.5.4.5 Conclusions

The conclusions and results obtained with the aid of the *ns-2* network simulator with the *OOLSR plug-in* prove that there is no penalty for replacing the *Breadth First Search* routing algorithm with the *Dijkstra* one in case of the *OLSR* network. It is then very likely that an improvement will be observed after the final evaluation of the *Generalized Dijkstra* algorithm, which will allow an application to select, under the given constraints, the best route out of the *k* possible. However neither the *ns-2* network simulator nor the *OOLSR plug-in* will be used any longer in order to complete this task. The reason is that although those tools were ideal for obtaining the first approximation of the prospected results for the simplified version of the target system, the effort necessary to reorganize and update their source codes seems to be more time and resource consuming than the development of an optimized stand alone solution, better suited to the investigated problem.

4.5.5 Impact of Smart Antennas on WiFR Routing for multi-hop networks

4.5.5.1 Introduction

In D3.1[1] a QoS routing for multi-hop wireless networks called Wireless Fixed Relay (WiFR) routing has been presented [14]. In that proposal, a new model for the QoS routing problem in multi-hop wireless networks with bandwidth constraints and an algorithm for its solution suitable for Fixed Relay Networks (FRNs) is proposed. The model is an extension of the well known multi-commodity flow problem [15] where link capacity constraints are replaced with new ones that take into account interference constraints among different radio links. The model guarantees that the rates of routed flows are compatible with radio channel capacity, but does not require to explicitly solve the scheduling problem. Since the characteristics of FRNs allow to control the path selection of each flow, in order to solve the proposed problem, a new routing algorithm based on a heuristic with some simulation results is presented in D3.1.

In that proposal, relay nodes were assumed to use omni-directional antennas both in transmission and in reception. Use of omni-directional antennas implies that all transmissions, even unicast ones, are de facto made in physical broadcast and effect of homogenous radio interface is that transmissions that overlap on a receiving relay will result in collision and hence in loss of packets. In other words, when node i transmits to node z (see Figure 4-30), the radio signal reaches not only node z but also all the other nodes in its radio coverage (u and v) wasting in this way portion of their bandwidth, in fact other nodes don't receive useful signal but anyway they see channel as busy, i.e. they cannot transmit or receive useful signal in the meanwhile.

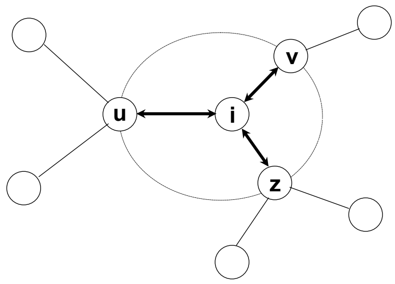


Figure 4-30 One hop constraints of node i

Moreover in a Fixed Relay Network (FRN), to compute how much bandwidth is consumed by a transmission, it is necessary to consider not only links adjacent to transmitting relay, i.e. links of its one hop cluster (called neighbourhood), but also two-hops-away links in order to avoid to set up route that will generate collision or overload due to the exposed problem.

The introduction of smart antennas reduces considerably the use of network resources for every single transmission because their "directivity" allows covering a smaller area and allows to waste less network resources in relays not belonging to the path selected, so additional available network resources are expected to route new requested connections. In conclusion, the use of smart antennas is expected to enhance the network throughput, that is the number of bytes routed with respect to the number of requested bytes by relays.

4.5.5.2 Smart antenna model

In addition to the omni directional ones, antennas in general may be classified as directional, phased array, adaptive, and optimal. The implementation of all of them on the relay nodes, instead of using omni directional antennas, has been taken into consideration. The adaptive and optimal antennas have been rejected because of the high cost to equip relays with these antennas, while the directive antennas have been rejected due to the difficulties to use such antennas in FRNs. In fact they give little flexibility in mesh network scenarios and in the future it has been expected to settle plug & play and self organizing relay nodes.

Therefore our attention has been focused on phased array antennas that don't allow the spatial separation of different flows which can be simultaneously received or transmitted on the same radio channel (SDMA scheme), but permit to select each time the direction where the maximum gain would appear. This is useful in order to reduce the number of relays which receive the useless signal. In particular after several studies (see details on paragraph 10.3.3 of Annex III) the simplified model of coverage shown in Figure 4-31 has been selected. It can be recognized that only the main lobe and the first secondary lobes give an important contribution.

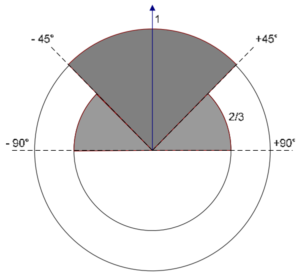


Figure 4-31 Simplified model of antenna array coverage

This model takes in consideration the worst conditions and it is suitable for each transmission power because the relations between coverage areas remain unchanged. The model has the following features:

- Maximum coverage from -45 to 45 degrees due to the principal lobe (0 degree is the direction of maximum radiation, i.e. the direction toward which we want to transmit). This value has been selected in order to have a safety margin against lobes spreading when the beam is pointed at angles higher than 30 degrees with respect to the normal. In fact in this condition the pattern begins to visibly deform and the lateral lobes increase their width resulting in a higher coverage, thus it's better to have a safety margin.
- 2/3 of maximum coverage from 45 to 90 degrees and from -90 to -45 degrees due to the secondary lobe. Even if the results of coverage area of the second lobe always covers less than half of maximum coverage, a value of 2/3 has been selected, as said before, in order to work in worst conditions taking into account a safety margin against lobes spreading when the beam is pointed at angles higher that 30 degrees with respect to the normal.
- No coverage outside these angles because a strong attenuation is expected.

4.5.5.3 Simulation results

To evaluate the impact of the antenna model explained above on the WiFR routing [14], a new interference model has been added into the event-driven network simulator ns-2 [10]in addition to the WiFR algorithm and the optimized TDMA MAC layer already developed [1]. The simulations presented here have been conducted using two ray (ground reflection) channel, a provided bandwidth of 2 Mbit/sec, packets of 1Kbyte, given traffic matrix with a number of sources which is the 20% of number of relays and Constant Bit Rate traffic sources with different random data rates which sometimes require up to 30% of provided bandwidth. Exponential sources should have been more suitable for the FRN purpose but our attention has been focused only on WiFR performance, in terms of network throughput, with or without smart antennas implementation. Different network topologies have been used with different number of relays (30,60,90,120), random distributed over a rectangular area of dimension 1000 m × 1000. Radio range utilized is from 150 to 550 meters, which is the region expected for relay deployment [14]. In addition simulations with omni-directional antennas have been performed using favorable conditions due to the low complexity of such antennas with respect to the smart ones. In fact, given the traffic matrix, routing has been defined trying 30 times to route the given connections selecting them in random order and maintaining the best attempts as final routing. For smart antennas the order of picking up connections is given at the beginning and only an attempt is done.

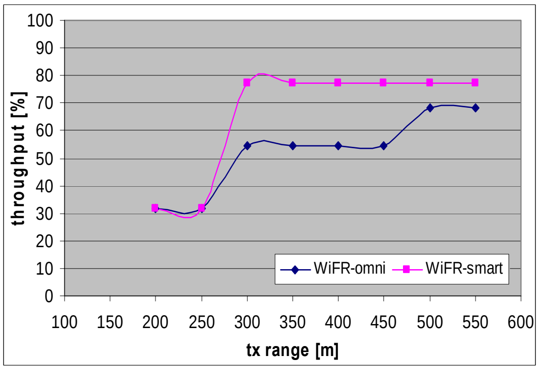


Figure 4-32 WiFR-omni vs WiFR-smart, random topologies, 30 relays, high traffic

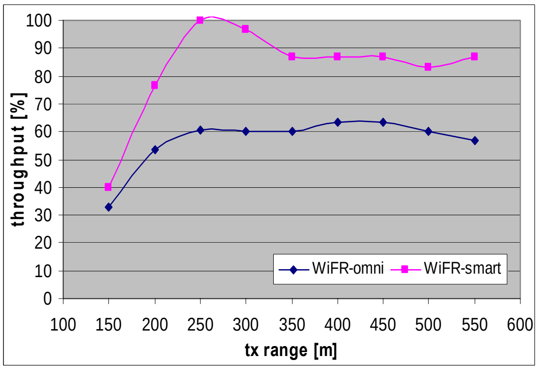


Figure 4-33 WiFR-omni vs WiFR-smart, random topologies, 60 relays, high traffic

It can be recognized in Figure 4-32 that for 30 relays and lowest values of radio range, the network is partially connected and the number of paths for each couple source-destination is very limited. For this reason blocking effect occurs and algorithm gives the same low throughput in both case. When network reaches low/medium connectivity, throughput obtained equipping relays with smart antennas is increased of about 23% thanks to the lower waste of resources due to the particular coverage of these antennas. As the radio range increases beyond 450 meters, throughput with WiFR-omni increases because of the diminishing of route length and hence of re-transmission, while throughput with smart antennas remains always higher (77%) but constant because the number of connections routed is already so high there is no more bandwidth available.

Figure 4-33, obtained with 60 relays, confirms that WiFR-omni is outperformed by WiFR-smart. This gain is already visible at a radio range of 150 meters because the high number of relays allows the network to be connected even with lowest value of radio range, exploiting the advantages of smart antennas implementation. In addition, also for these simulations WiFR-smart quickly reaches its maximum value and then keeps itself at value of throughput higher than 30/40% with respect to WiFR-omni. The only difference with the case with 30 relays is that, reached the maximum (in this case performance of 100 % has been obtained), network throughput starts to decrease, according to the already known WiFR algorithm performances due to the increased terminal density and the consequent blocking effect[14].

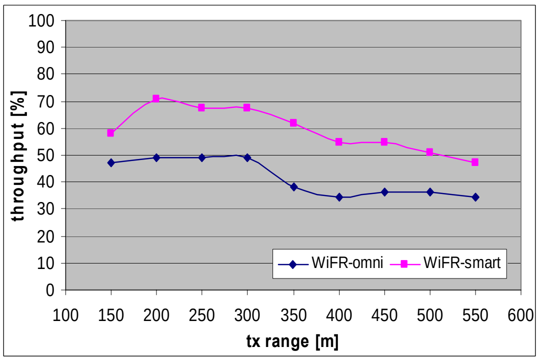


Figure 4-34 WiFR-omni vs WiFR-smart, random topologies, 90 relays, high traffic

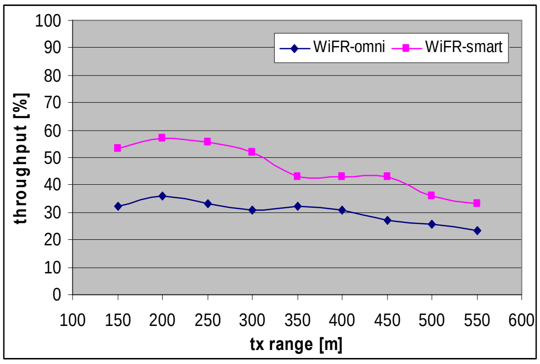


Figure 4-35 WiFR-omni vs WiFR-smart, random topologies, 120 relays, high traffic

Figure 4-34 and Figure 4-35 show that WiFR-smart outperforms WiFR-omni even in FRNs with high number of relays (90 and 120). It can be noticed that curve obtained with smart antennas follows the profile of curve obtained with omni-directional ones but the gain is always about 10-18%, except for radio range of 150 meters, due to the high number of relays used in these simulations. In fact, as we said, the new model of antenna allows a small number of neighbors to be reached by the useless signal, but the higher the terminal density, the higher the number of neighbors involved which waste their bandwidth.

4.5.5.4 Conclusion

From the set of simulations conducted to evaluate the impact of smart antennas on WiFR QoS routing for multi-hop wireless networks, the following conclusions can be taken out.

About random generated topology, WiFR routing in presence of smart antennas outperforms WiFR in presence of omni-directional antennas in network of medium, large dimensions (in terms of number of relays).

The trend of WiFR-smart curves is the same of WiFR-omni one; in fact it achieves first a local maximum in throughput for a radio range of 200-300 meters, then it starts to decrease (especially in the network with a high number of relays).

WiFR routing with smart antennas has the best gain in networks with a number of relays from 30 to 60. In this range the high cost to equip relays with smart antennas, may be compensated by the really high gain in network throughput.

When the number of relays exceeds 60 a significant gain can be still recognized, but the higher terminal density allows that during a transmission the number of relays which receive the useless signal is high despite of using smart antennas. It results in a lower gain with respect to networks of 30 or 60 relays.

Finally it can be recognized that with the increasing of number of relays a decreasing of transmission range is envisaged in order to have the best performance results. This effect underlines how an efficient power control mechanism that acts as topology control is crucial and suitable for Fixed Relay Networks (FRNs) because can adapt the terminals transmission power according to network needs [16].

4.6 Cooperative Relaying

In this section, we offer a more detailed investigation of the concepts cooperative relaying. Recall from [1] that cooperative relaying differs from conventional store-and-forward relaying in that the source's transmission is taken into account at the destination. By doing so, the inherent diversity of the relay channel can be exploited - in addition to the benefits of reduced end-to-end path losses provided by conventional relaying.

We start in section 4.6.1 with a discussion of multi-antenna aspects in relation to cooperative relaying. Related system connectivity models are examined in section 4.6.2. Further, in section 4.6.3 a look at qualitative aspects on mobility as related to cooperative relaying is taken. In section 4.6.4 we discuss the impact of the number of hops on system performance for various scenarios. Concluding remarks are offered in section 4.6.5.

It is worth noting that some of the aspects discussed in this subsection are also applicable to and important for conventional relaying; this holds in particular for the system connectivity models (4.6.2) and the discussion of the number of hops in relay chains (4.6.4).

4.6.1 Multi-antenna aspects

4.6.1.1 Further development and assessment of multi-antenna parallel fixed relays deployment concepts (CU)

4.6.1.1.1 Introduction

Many contributions on this topic have suggested that for a significant diversity gain or order, participating relays should not fully decode the received signal [39]. This is due to the complication caused by error propagation, which impacts any cooperation schemes in a tremendous way. In the same way, the classical work of Sendonaris, Erkip and Aazhang [36][37] stressed the importance of reliable decoding of the inter user channel. This interuser channel facilitates the exchange of cooperative information. Schemes have been proposed to combat the error propagation. For instance, threshold decoding of relay signal was shown to improve the overall system performance [38], a protocol now known as adaptive decode and forward (AdDF) scheme.

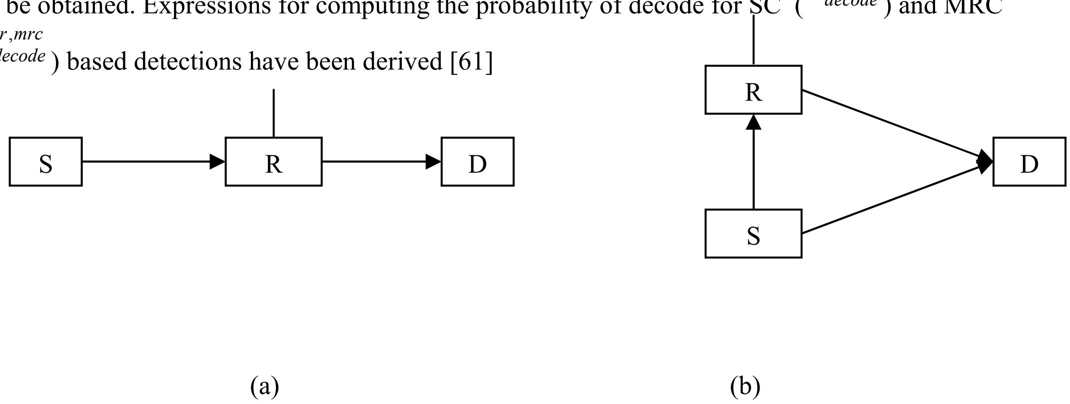
In AdDF, the relay decodes and forwards only when the SNR is above a certain preset threshold. In this scheme the process is divided into two stages or phases. In the first phase, the source broadcasts its information. The relay node (RN) and destination received possibly faded versions. The DS stores the signal it received for future processing. In the second phase, the RN either does or does not forward a newly pre-processed signal to DS. If the RN did forward, the DS combines its delayed signal (stored) with this new version from the relay using MRC. Otherwise, the relay will be silent (DS senses this by lack of sufficient received signal strength), the DS will then resorts to only the stored sample for decoding. Note that a little complexity can be kicked into this second phase that could improve system performance. The DS could ask the source to re-transmit in the event that the relay is silent. This complex AdDF has yielded no substantial performance advantage over the AdDF [27]. Thus, Simple AdDF is commonly used to demonstrate the benefit of this scheme. To derive further diversity order from multiple relays network, the authors in [39] proposed using Hurwitz Radon code (HRC). In this approach, the received signals at the relays are arranged in the order of magnitude and phases according to the HRC and the modified sequence of symbols are forwarded in parallel to destination. With this, an R/2 order of diversity is obtained from R parallel relays.

Fixed wireless relays may have the capability to carry multiple antennas contrary to the common assumption of relays with single antennas. We now investigate the scenario where each relay is equipped with L antennas. Again, we consider the decode-and-forward relaying strategy; i.e., the relays are of the regenerative (digital) type. Since by design, the relay is expected to be cost-efficient and of low complexity, the relay may use a subset of these L antennas; in the limiting (worst) case scenario a relay uses only 1 antenna through selection combining of its L antennas. In this case, only one signal detection chain is required at each relay similar to the relays that do not use multiple antennas. Therefore, in terms of cost, this selection diversity-based relay system incurs no significant penalty (other than the cost of the extra antennas and selection mechanism). If the relays can utilize signals from more than one antenna, then the classical generalized selective combining (GSC) scheme can be employed. For forwarding, however, each relay uses only one antenna. Our earlier work, which includes the study of hybrid macro/micro-diversity in wireless networks [30], could be viewed as a foundation on which to build the analysis of parallel (distributed) relays.

4.6.1.1.2 System of multi-antenna and parallel fixed relays

We assume homogeneous relaying. At the relays, the threshold decision to forward criterion introduced in [27][38] has been adopted (this scheme is referred to as adaptive decode and forward, AdDF, as well). In this new architecture (multi-antenna relays) the proportion of time the relays forward their signals is higher than that in the one antenna case. Hence, we conjecture (for the time being intuitively, without any diversity order attached) that the diversity order can be enhanced. This conclusion is actually derivable from the work in [35]. In that study two intermediate relays (say, R1 and R2) are considered in amplify-and-relay (analogue/non-regenerative) type multi-hop networks. It is assumed there that R2 listens to R1, and combines the signal via R1 with that from the source S; the resulting signal is denoted as R2 (S, R1). The sources signal and the forwarded signals from R1 and R2 are subsequently utilized at the destination as S + R1 + R2 (S, R1), where + could be MRC (maximal ratio combining). The diversity order of this scheme is calculated to be 3 in [35].

Before we embark on system performance analysis, we present the improvement obtained in decode and forward probability using multiple antennas at the relay node. The architecture is depicted in Figure 4-36. Our efforts will be restricted to first, using single antenna detection (SC) chain to put relay node cost comparable to the conventional case and second, using all the antennas (as in MRC) when the condition is relaxed in such a way that high processing cost at the relay is tolerable. In the case of SC it implies that before the relay decides to decode and forward, it first selects a branch with the largest SNR. i.e., γ = max [α_1^2, …, α_L^2] = max [γ_1, …, γ_L], where L antennas are installed. The joint PDF of selecting n largest from N i.i.d RV is given in a general form in [40] as p(γ_1, …, γ_N) = $n!\binom{N}{n}[F(\gamma)]^{N-n}\prod_{l=1}^{n} p(\gamma_l)$. Our focus is on selecting one from N, and given that the PDFs of γ_i, $i = 1,2,\ldots L$, are assumed to independently and identically distributed, the PDF of the selected branch can be obtained. Expressions for computing the probability of decode for SC ($P_{decode}^{r,sc}$) and MRC ($P_{decode}^{r,mrc}$) based detections have been derived [61]



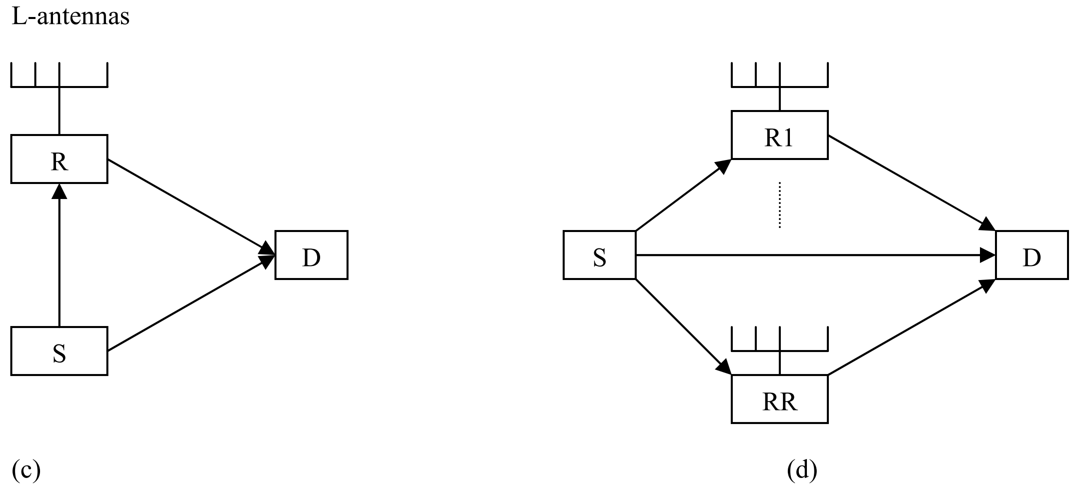


Figure 4-36 Network relay deployment: a). Conventional, b). Single cooperating one antenna relay, c). single cooperating multi-antenna relay d). Multiple cooperating multi-antennas relays

Figure 10-17 (a) shows the improvement in the probability of decoding at the relay when two antennas are deployed but one branch is utilized at a time (SC), two antennas are deployed and utilized (MRC) and the case of no diversity (L = 1). The plot shows that a significant improvement is obtained with the multi-antenna scheme over the single antenna strategy. Results are shown for Nakagami parameter m of 1, 2, 4, and 6.

4.6.1.1.3 MRC-based Threshold DF multi-antenna relays: Performance Analysis

In the previous sections we have shown that deploying multiple antennas at relays shows some improvement in decode and forward probability, which implies that the number of times the destination depends on diversity signal(s) is increased. In the following we present derivation for the probability of a relay making error when a threshold is imposed. The conditional probability of error for equal amplitude modulation schemes can be expressed as

$$P_{\mathrm{mpsk}}(e \mid \gamma) \approx h \,\mathrm{erfc}\left(\sqrt{\gamma g d / 2}\right) \tag{9}$$

where, $d = \sin^2(\pi / M) E_s / N_0$ and for BPSK equality holds in the above equation and *M=2, h*=1/2, g=2. For higher MPSK constellations, *h*=1. For coherent orthogonal BFSK *M=2, g=1, h=1/2,* and coherent BFSK with minimum correlation $g = (3\pi + 2)/6\pi$ and *M=2.* The combined SNR for a receiver that bears *L* antennas is given as $\gamma = \sum_{i=1}^{L_c} \gamma_i$, where *Lc = L= 1* implies that diversity is not employed. When *Lc = 1, L >1,* we have selection combining (SC), for *Lc = L, L > 1* the maximal ratio combining (MRC) and finally 1<*Lc<L, L>1* we have generalized selection combining form of diversity. The error performance at the relay that is equipped with *L* antennas and operating in Nakagami fading channel is derived. This error performance is vital towards deriving the end-to-end error performance of the multi-antenna parallel relays being investigated. Firstly, we considered that the relays implement MRC and later, the case of SC will be considered, since the relay is expected to be simple and cheap, SC may be more attractive strategy. Error performance expressions have been derived for the threshold-based SC and MRC DF relays [61]. These expressions are denoted as $P_e^{r,sc}$ and $P_e^{r,mrc}$, respectively.

Figure 10-17 (b) shows the performance of a relay with L diversity antennas that also puts a threshold on the decodable received SNR. The analytical results are the solid curves and the simulations denoted by star, square, and circle. The Nakagami parameter value is 1 (*m*=1, Rayleigh fading). The curve for no threshold decoding is also shown for comparison purposes. It is observed that deploying multiple antennas at the relay does improve the error performance, which of course, is expected by intuition. What intuition fails to reveal however, is how much of improvement is obtained from this deployment and the impact of the various factors involved. With this piece of information, a system designer could be guided on the best choice of antenna deployment strategies, for instance, number of relays and antennas to deploy. Figure 10-17 (c) also shows the error performance for the cases where Nakagami parameter *m* is set to 2 or 6. Finally, a preliminary result on the end-to-end error performance of the multi-antenna parallel relays is given in Figure 10-17 (d), where we have compared the analytical and simulations results and strong agreement is observed.

Based on the results presented thus far, deploying multiple antennas at a fixed relay appears to be a promising strategy. We have observed that the probability of reliable decoding and forwarding at the relay is significantly enhanced leading to tremendous improvement in the end-to-end error performance of the network. It has been assumed that the fading distributions between the tripartite (source to relay, source to destination and relay to source) are the same. To obtain performance results for non-identical fading distribution scenarios is straightforward. In fact, this is one of the main motivations for adopting Nakagami distribution in the analysis. Let us illustrate with an example. Take for instance, a fixed relay that is positioned in such a way to have LOS to the source and NLOS to destination. Since, it would most likely be that the source faces NLOS to destination, the source-relay channel can be modelled with a

$$K = \frac{\sqrt{m^2 - m}}{m - \sqrt{m^2 - m}} \tag{10}$$

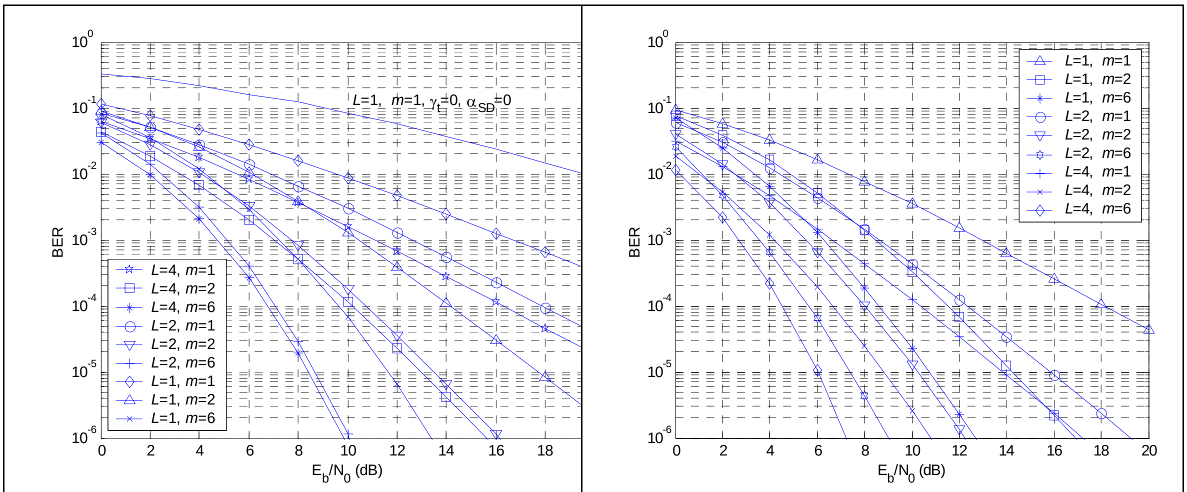
suitable m (using the transformation), and the relay-destination and source-destination channels will both be modelled using the Rayleigh distribution.

4.6.1.1.4 SC-based Threshold DF multi-antenna relays: Performance Analysis

This scheme has been analysed and some discussions and results are presented in Annex 10.4.1.

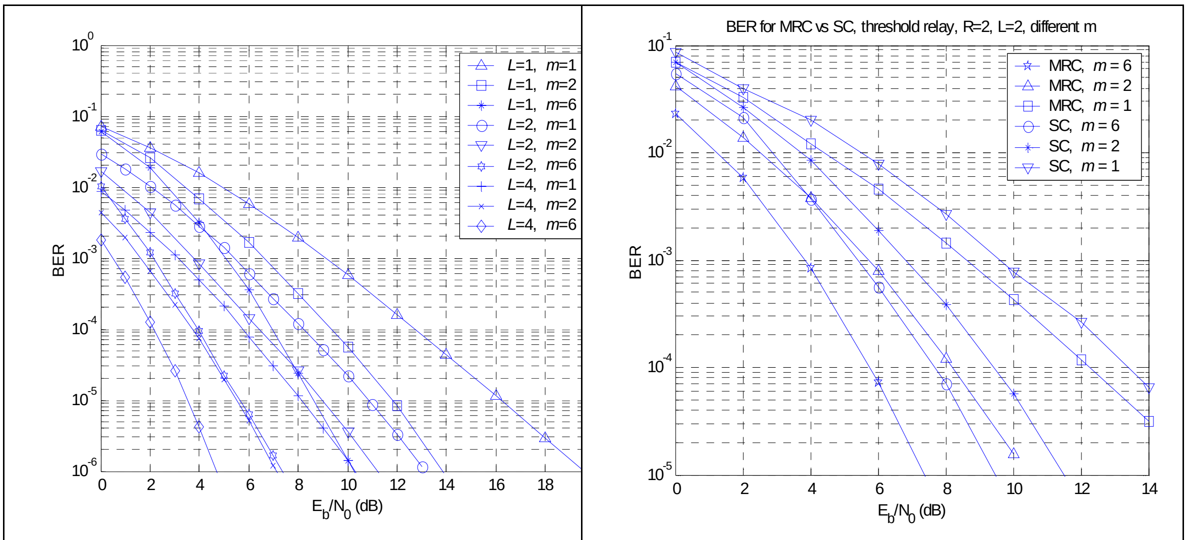
4.6.1.1.5 Numerical Results and Discussions

The end-to-end system performance has been investigated using the analytical expressions as well as through simulation campaigns. The performance of the multi-antenna parallel relays in symmetrical channels is shown Figure 4-37(a), (b), and (c), respectively, for R = 1, R = 2, and R = 4. -relay cooperating with the source. Figure 4-37 (a) shows the performance of conventional relaying (L3DF) as well.



(a). R = 1

(b). R = 2



c). R = 4

d). MRC-based vs SC-based

Figure 4-37 Performance of threshold-based multi-antenna parallel relays in Nakagami fading channels

At low BER regime, the multi-antenna relay systems yield tremendous gains over the conventional form of relaying. Let us compare the SNR requirements at an error rate of 10^{-2}. Looking at Figure 4-37, the following approximate gains are obtained over the conventional relaying, 10.5 dB (L = 1), 13 dB (L = 2), and 14.5 dB (L = 4) for m = 1 channel condition. These gains represent significant improvement.

Furthermore, at BER $= 10^{-3}$, the dual antenna case is about 4 dB superior to the single antenna case, and when four antennas are deployed, this gain is about 5 dB. For m=2, however, the gain of dual antennas over the single antenna is about 1.5 dB while that of four antennas is 2.5 dB. From these cases, it can be deduced that more diversity gain is obtained for multi-antenna system in Rayleigh (m = 1) than in less scattering channels (m > 1), which again confirms the notion of the benefit of rich scattering environment. Furthermore, it seems that with dual antenna significant improvement can be obtained. In this sense, dual antenna at the relay may be strongly recommended without complexity and cost of relays being unreasonable.

The Figure 4-37 Figure 4-37(b) show the performance of 2-relay network. Let us compare the different system configurations at BER $= 10^{-4}$. For the Rayleigh fading situation, it is observed that a gain of 6 dB is achieved for R = 2, L = 2 over R = 2, L = 1 and when the number of antenna is increased to 4 (i.e., R = 2, L = 4) this gain increases only marginally to 7.5 dB. For the less fading cases (m > 1), comparing these architectures, we observe that the gains are reduced. For an example, when m = 2, a gain of 3.5 dB is obtained for (R = 2, L = 2), and 4.7 dB for (R = 2, L = 4) over (R=2, L=1). The trends observed in Figure 4-37 (a) and (b) are generally observed for the multi-antenna 4-relay network. The performance of this system is shown in Figure 4-37 (c).

Let us now make some cross-system comparison, where Table 4-6 has been given for this purpose. We consider the SNR required for an error rate 10^{-4} for different parameters, L and m.

Table 4-6 MRC-based relay detection: System comparison at BER = 10^{-4}

	R=2		R=4	
# of Antennas (L)	M=*1*	m =*2*	m=*1*	m=*2*
1	*18.0 dB*	*11.5 dB*	*13.0 dB*	*9.2 dB*
2	*12.0 dB*	*8.0 dB*	*8.0 dB*	*6.4 dB*
4	*10.4 dB*	*6.8 dB*	*5.8 dB*	*3.8 dB*

It can be observed in Table 4-6 that for the channel condition m = 1, R = 2, L = 2 outperforms R = 4, L=1, although, the number of detection chains required in both configurations are the same. However, the case of R=2, L=2 has an additional advantage of system deployment cost over R=4, L=1, where it can be argued that it will be a lot cheaper to install additional antenna at the relay than deploying extra relays all things being equal. When dual antenna is deployed with the R=4, we observe that R=4, L=2 outperforms R=2, L=4 by about 1.6 dB for m=1 and about 0.4 dB for (m=2). The gain of R=4, L=2 over R=2, L=4 has to be viewed carefully. If we consider the cost of these two systems, we can quickly dismiss this gain as inconsequential, i.e., is too small in comparison to the concomitant cost and complexity over the R=2, L=4. In fact, as can be seen, this gain disappears at less fading (m > 1). In conclusion, the system configuration R=2, L=4 should be preferred to R=4, L=2 for deployment, similar such conclusions can be derived in other scenarios.

Let us also compare the two modes (SC-based and MRC-based relay detection) for Rayleigh fading channel environment (m = 1). We found by observing Figure 4-37 (d) that MRC-based detection could yield no more than 2 dB gain over the SC-based. Furthermore, we observe that though MRC-based is about 2 dB superior to SC-based relay detection this is at expense of huge system cost in the R = 4, L=4 system architecture. For instance, while the SC-based requires only 4 detection chains (one at each relay node), the MRC requires 16 (four at each relay node). The small SNR advantage cannot offset the cost disadvantage incurred in using MRC. Therefore, SC offers a good trade-off between cost and performance. Finally, for higher values of Nakagami parameter, though MRC still exhibits superiority over SC-based these gains are so minimal to justify for the cost involved. This further tilts the balance in favour of SC-based detection at relays. A table similar to Table 4-6 has been used to compare the SC-based detection for different system configurations (See Table 10-7 of the Appendix and the discussions following it). In conclusion, deploying microdiversity at relays may end up in considerable savings in the number of relays to be deployed in a given area and selection combining strategy at the relay could represent excellent trade-off between performance and cost.

4.6.1.1.6 System Aspect of Alamouti Based Relaying

The Alamouti diversity relay based concept was suggested and evaluated on a link level in [1]. A natural way forward is to investigate the performance of the concept on a radio network perspective. The system model of the concept is shown Figure 4-38 where a ring of relays (RN) is placed on a circle of radius R and has as a centre the cell AP. Naturally the ring radius is smaller than the cell radius. Each set of two relays serves as two separate antennas for applying the Alamouti's space-time coding diversity scheme.

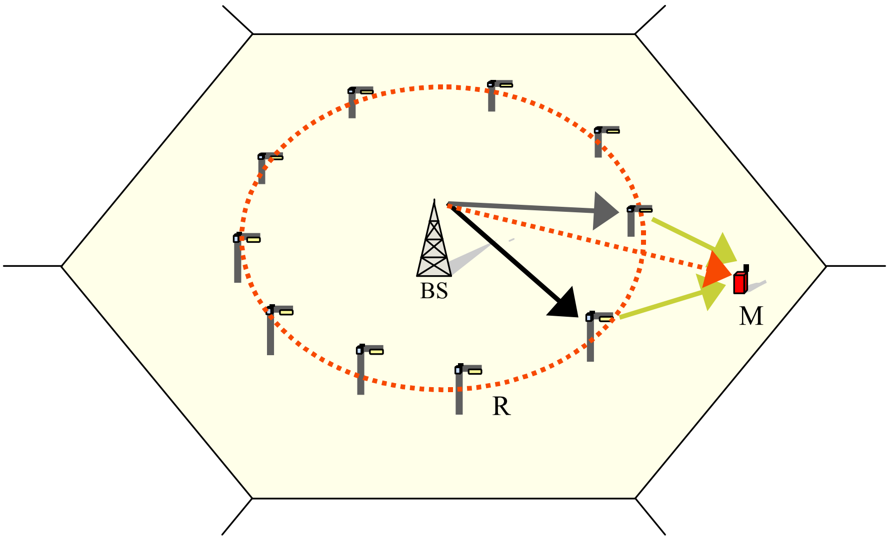


Figure 4-38 Alamouti diversity based relaying--- System model

A simple static simulator was used to evaluate the Alamouti based concept. The simulator flowchart is shown in Figure 4-39. The system cell radius was fixed to 1 km and the default RN ring was set to 0.5 km. A frequency reuse of 1 was assumed. The number of relays per cell was limited to 10. The AP and RN power, respectively Ptx and Prs, were set either to 0.01 W or to 10W. The shadow fading standard deviation was assumed to be 10 dB. Finally the following parameters were assumed:

- Bandwidth: 20 W (used in noise power calculation)
- Number of users in the studied cell: 1000
- Attenuation factor from AP to RN: -3
- Attenuation factor from AP to mobile: -4

- Attenuation factor from RN to mobile: -2.5

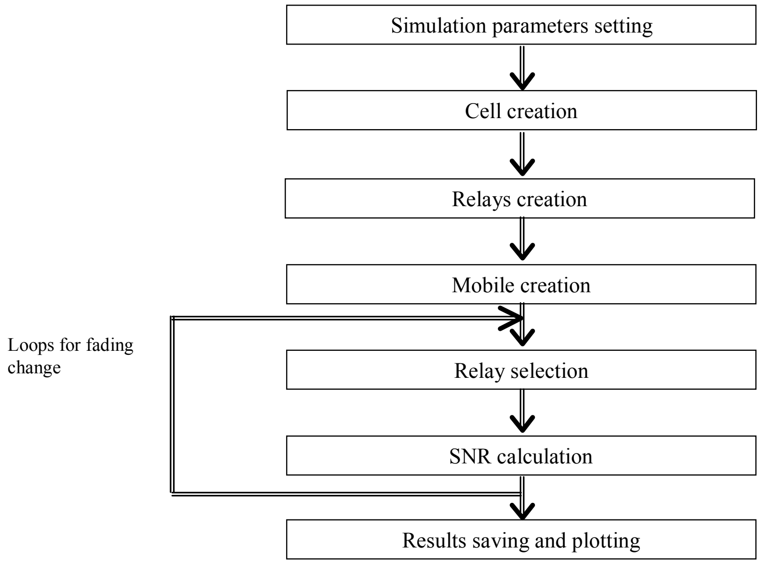


Figure 4-39 Simulator Flow Chart.

Note that only the non-regenerative case, i.e., amplify-and-forward, was considered. The selection of the RN was determined by two of the relays that provide the highest received SNR.

The capacity was investigated for two extreme interference cases:

➢Line Of Sight (LOS) interference that means that:

Attenuation factor from an interfering AP to a target RN: -3

Attenuation factor from an interfering RN to a target mobile: -2.5

➢Non-Line Of Sight interference where the attenuation factor from an interfering AP to a target RN and from an interfering RN to a target mobile is equal to -4

Performance:

The deployment of a relay based Alamouti concept provides a substantial capacity and SNR gain over a traditional system as it is shown in Figure 4-40 and Figure 4-41 except for very high SNR. Note that the CDF for both SNR and capacity is more centred on the median CDF value for the relaying system which indicates a good quality for most users. In fact almost 90% of the users will have higher capacity when relaying is deployed. Finally it is interesting to notice that the increase of the power of the RN from 0.01 to 10 did not translate in a substantial capacity or SNR increase.

In Figure 4-41, although the power level AP and RN were chosen to be low, the capacity with relay is still quite stable and high. The area that relays can provide capacity gain expands. Also when using relays, the fact of adding a direct path does not improve the performances.

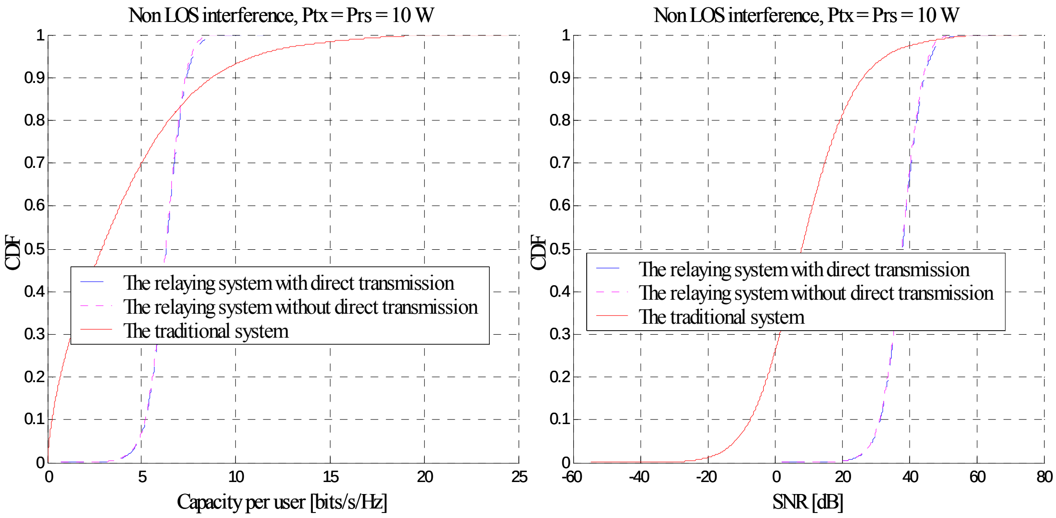


Figure 4-40 Non LOS Non-LOS interference, reuse 1, Ptx=Prs= 10 W --- capacity & SNR.

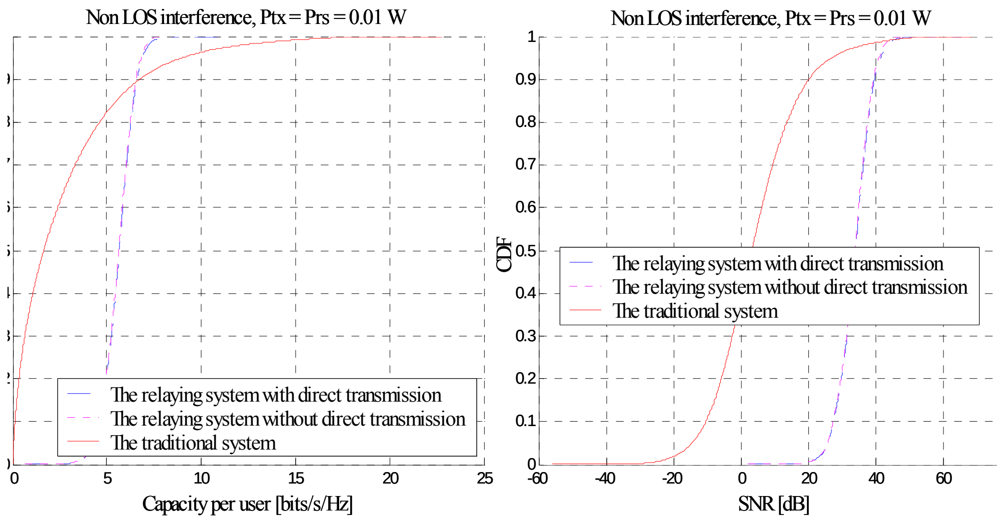


Figure 4-41 Non-LOS interference, reuse 1, Ptx=Prs= 0.01 W --- capacity & SNR.

When a LOS interference was considered, the SNR of the 2nd hop is lower than that of the 1st hop and no gain in terms of SNR was observed by using relaying. In that case the fact of adding a direct path helped to improve the SNR and the capacity but not substantially to justify the deployment of a relay system.

For larger cell radius a relay system can provide a substantial capacity gain as it is shown in Figure 4-42. It was assumed that the RN ring radius is at half distance of the cell radius. It can be observed from the same figure that by increasing the cell radius, the capacity gain of a relaying system over a traditional system increases. This is very clear from the bottom right of Figure 4-42 where the relay system ensures an effective coverage for all users.

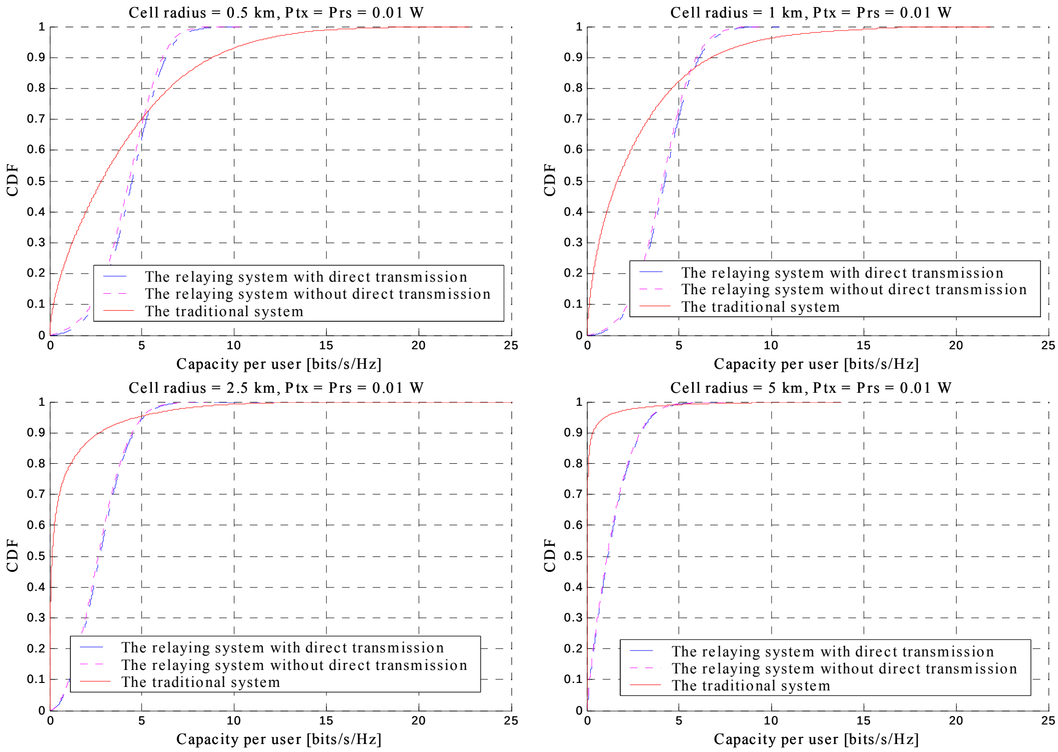


Figure 4-42 Impact of the cell radius on the capacity.

<u>Conclusions under the stated assumptions</u>

The following observations can be drawn from using Alamouti's based relaying on a radio network:

- A visible capacity benefit with the Non LOS interference, even with high AP and RN powers
- There is no capacity (or even SNR) benefits with the LOS interference scenario, regardless the AP and RN powers.
- A capacity benefit only at low SNR region with high AP and RN power was observed, while visible capacity benefit with almost 90% probability for low AP and RN powers was noted.
- Relay can provide effective coverage for large cell radius.

4.6.2 System connectivity analysis

4.6.2.1 Introduction

Recent findings in the literature have shown that the performance of wireless relaying networks can be increased through the application of distributed spatial diversity techniques that rely on the mesh connectivity between wireless user terminals [50][51][52][53][54][36][37][38][57][58]. Multi-user diversity [50][54] achieves spatial diversity by relaying the signal along multiple routes in parallel. Multi-hop diversity [51][52] achieves spatial diversity from the concurrent reception of signals that have been transmitted by multiple relays in serial along a single primary route. Cooperative diversity [36][37][38][57][58] achieves spatial diversity by sharing information between the source terminal and cooperating relay terminals such that each user of the cooperation group sends information to the destination using all of the cooperating terminals. The symbol error probability of parallel combinations of serial relaying channels is derived in [59]. The aggregate SNR of arbitrarily connected amplified relaying channels is analyzed in [53].

Each of these distributed spatial diversity techniques places different requirements on system resources due to a reliance on mesh connectivity between terminals. Therefore, system resource constraints that limit the terminal connectivity constrain the distributed spatial diversity techniques that can be applied. The system resource constraints considered are the available number of orthogonal relaying channels, the ability of relays to diversity combine incident signals, the ability of the destination to combine incident signals, the ability of receivers to combine signals on multiple orthogonal channels, the ability of transmitters to transmit signals on multiple orthogonal channels, and the ability of receivers to cancel the effects of interhop interference [52].

This section extends the previous work with the incorporation of interhop interference cancellation as a system resource constraint, expanded discussion on the interrelationship between system resource constraints, expanded discussion on the relationship of system resource constraints and RN placement, improved simulation of system connectivity models with varying RN placement, and results related to optimal connectivity algorithms for the various system connectivity models.

4.6.2.2 Incorporation of Interhop Interference Cancellation (IC) as a System Resource Constraint

The previous work is extended to include interhop interference cancellation as a system resource constraint.

Interhop Interference Cancellation (IC): This constraint defines the ability of receivers to cancel the effects of interhop interference created by the retransmission of signals on the same channel at different hops along a multi-hop transmission path [52]. Interhop interference is a special case of intersymbol interference (ISI) that affects wireless relaying networks where channels are reused, and can be mitigated through traditional equalization techniques. Use of interhop interference cancellation increases the system cost since more complex equalization hardware is required for received signals.

- Interhop Interference Cancellation (IC): Receivers are able to cancel the effects of interhop interference. More complex equalization hardware is required. There is no connectivity constraint.
- No Interhop Interference Cancellation (NIC): Receivers are not able to cancel the effects of interhop interference. Less complex equalization hardware is required. The connectivity impact is that networks with two channels available (2CA) have a maximum hop depth of two. Note that this constraint is simplistic in that it does not consider that in practice channel reuse may be possible in some circumstances due to sufficient spatial separation between terminals.

The constraint combinations and resultant system connectivity models are extended to include interhop interference cancellation.

The following terminology is used when describing the resultant system connectivity models:

- Single Relay (1R): Relays connected to one transmitter.
- 2Chnl Relay (2R): Relays connected to the subset of transmitters on one channel.
- Full Relay (FR): Relays connected to all transmitters previous along the transmission path.
- Single Destination (1D): Destination connected to one transmitter.
- 2Chnl Destination (2D): Destination connected to the subset of transmitters on one channel.
- Full Destination (FD): Destination connected to all transmitters.
- 2Chnl Source (2S): Source connected to a subset of receivers on both channels.
- Full Source (FS): Source connected to all receivers.
- Two Hop (2H): Network has maximum hop depth of two.

The majority of system connectivity models result from constraint combinations with 2 channels available. Only the system connectivity models with full relay connectivity, FR1D and FRFD, are exclusive to constraint combinations with N channels available. The impact of the other system resource constraints on the connectivity of relaying channels is much more diverse when there are 2 channels available.

Annex 10 shows the constraint combinations and resultant system connectivity models.

The following figure shows the transitions between the different system connectivity models for various constraint changes.

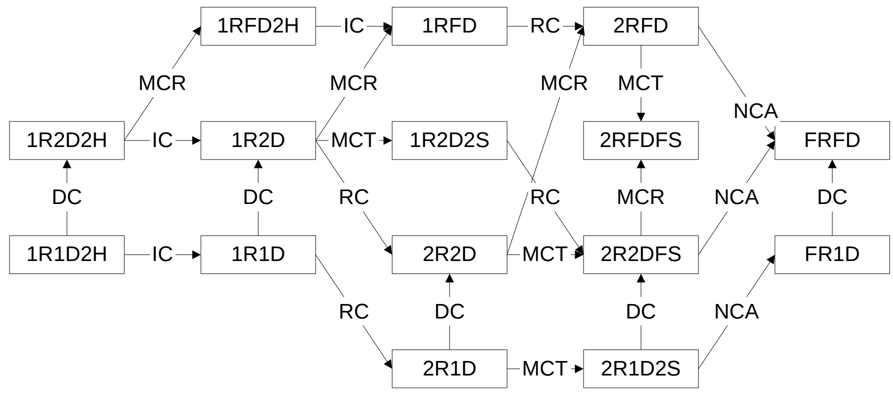


Figure 4-43 Transitions between System Connectivity Models

Transitions are in the direction of decreased system resource constraints. Transitions that decrease system resource constraints without improving the system connectivity are not shown.

Annex 10 shows examples of all the system connectivity models.

4.6.2.3 Interrelationship Between System Resource Constraints

The interrelationships between the system resource constraints with respect to their impact on the connectivity of relaying channels are described in this section. The following set of interrelationships can be extracted from the exhaustive list of constraint combinations:

- Destination combination always has an impact on connectivity, independent of all other constraints.
- Interhop interference cancellation always has an impact on connectivity for constraint combinations with 2 channels available. Interhop interference cancellation never has an impact on connectivity for constraint combinations with N channel available.
- The number of channels available only has an impact on connectivity for constraint combinations with relay combination and either multiple channel reception or multiple channel transmission.
- Multiple channel reception and multiple channel transmission only have an impact on connectivity for constraint combinations with either relay combination or destination combination.
- Multiple channel reception and multiple channel transmission have equivalent and exclusive impact on connectivity for constraint combinations with N channels available. Multiple channel reception and multiple channel transmission have independent impact on connectivity for constraint combinations with 2 channels available.
- Relay combination and multiple channel transmission only have an impact on connectivity for constraint combinations with either interhop interference cancellation or N channels available.

4.6.2.4 Minimum Cost Constraint Sets

The sets of constraints that result in each system connectivity model while minimized the system cost (the minimum cost constraint set) are derived in this section. The minimum connectivity constraints sets for each model are derived using the following implicit ordering, with increasing system cost, of the system resource constraints:

- Destination Combination: Combination hardware is required on the destination.
- Multiple Channel Reception: Multiple channel combination hardware is required on receivers. The incremental system cost is considered to be greater than destination combination because it involves more complex multiple channel combination hardware.
- Multiple Channel Transmission: Multiple channel transmission hardware is required on transmitters. The incremental system cost is considered to be greater than multiple channel reception because it involves comparatively more power and interference generation.
- Interhop Interference Cancellation: Equalization hardware is required on every relay. The incremental system cost is considered to be greater than multiple channel reception because it

involves equalization hardware on every relay instead of combination hardware only on the destination.

- Relay Combination: Combination hardware is required on every relay. The incremental system cost is considered to be greater than interhop interference cancellation because it involves more expensive combination hardware.
- N Channels Available: N orthogonal channels are available. The incremental system cost is considered to be greater than relay combination because it involves N-2 more channels being provided by the system for every active source-destination pair.

The following figure summarizes the minimum cost constraint set for each system connectivity model.

Connectivity Model	Minimum Cost Constraint Set
1R1D2H	{2CA, NRC, NDC, SCR, SCT, NIC}
1R1D	{2CA, NRC, NDC, SCR, SCT, IC}
1R2D2H	{2CA, NRC, DC, SCR, SCT, NIC}
1R2D	{2CA, NRC, DC, SCR, SCT, IC}
1R2D2S	{2CA, NRC, DC, SCR, MCT, IC}
1RFD2H	{2CA, NRC, DC, MCR, SCT, NIC}
1RFD	{2CA, NRC, DC, MCR, SCT, IC}
2R1D	{2CA, RC, NDC, SCR, SCT, IC}
2R1D2S	{2CA, RC, NDC, SCR, MCT, IC}
2R2D	{2CA, RC, DC, SCR, SCT, IC}
2R2DFS	{2CA, RC, DC, SCR, MCT, IC}
2RFD	{2CA, RC, DC, MCR, SCT, IC}
2RFDFS	{2CA, RC, DC, MCR, MCT, IC}
FR1D	{NCA, RC, NDC, MCR, SCT, NIC}
FRFD	{NCA, RC, DC, MCR, SCT, NIC}

Figure 4-44 Minimum Cost Constraint Sets

4.6.2.5 Simulation of System Connectivity Models with Varying Relay Terminal Placement

The system connectivity models are applied in a series of simulations that provide a comparison with respect to probability of error for various network topologies and allow the performance impact of the individual system resource constraints to be isolated. The single hop channel is shown for reference. A BPSK modulation scheme is used for simplicity of exposition. The simulations use the equations derived in [52][53] with a propagation exponent of 4 and equal power allocation for all transmitting terminals such that the total transmit power is constrained to the transmit power of the single hop reference channel. Maximal ratio combining is assumed.

Simulations are presented for three relaying methods and four network topologies. The first relaying method is amplified relaying, also known as non-regenerative or amplify-and-forward relaying. For amplified relaying each intermediate terminal simply combines and amplifies the received signals from preceding terminals before retransmission. The second relaying method is decoded relaying with error propagation, also known as regenerative or decode-and-forward relaying. For decoded relaying with error propagation each intermediate terminal combines, digitally decodes, and re-encodes the received signals from preceding terminals before retransmission and decoding errors at relays propagate as decoding errors at the destination. The third relaying method is decoded relaying without error propagation, also known as adaptive decode-and-forward relaying. For decoded relaying without error propagation each intermediate terminal combines, digitally decodes, and re-encodes the received signals from preceding terminals before retransmission and decoding errors at relays propagate as decoding errors at the destination. Decoded relaying without error propagation upper bounds the performance of decoded relaying and corresponds to an error-free version of the adaptive decode-and-forward protocols presented in [38][56][57]. A more detailed specification of these protocols is provided in [38][52][53][56][57].

Annex 10 describes the simulated network topologies in more detail and provides the corresponding simulation results for the system connectivity models.

4.6.2.6 Relationship of System Resource Constraints and Relay Terminal Placement

The following figures respectively summarize the impact of the system resource constraints on each relaying method for the three network topologies where network connectivity has been optimized. The suboptimal example network topology is not of interest because the chosen network connectivity does not fully leverage the available system resources.

Relaying Method	NCA	RC	DC	MCR	MCT	IC
Amp	Small	Small	Large	Large	Small	Large
Dec w Pr	Small	Med	Med	Small	Med	Large
Dec w/o Pr	Small	Med	Large	Large	Small	Med

Figure 4-45 Impact of System Resource Constraints for Example Network Topology

Relaying Method	NCA	RC	DC	MCR	MCT	IC
Amp	Small	Small	Large	Large	Small	Large
Dec w Pr	Small	Med	Med	Small	Med	Large
Dec w/o Pr	Small	Med	Large	Large	Small	Med

Figure 4-46 Impact of System Resource Constraints for Linear Network Topology

Relaying Method	NCA	RC	DC	MCR	MCT	IC
Amp	Small	Small	Large	Med	Small	Med
Dec w Pr	Small	Small	Med	Small	Med	Small
Dec w/o Pr	Small	Small	Large	Med	Small	Small

Figure 4-47 Impact of System Resource Constraints for Central Network Topology

The impact of the constraints is different for amplified relaying, decoded relaying with error propagation, and decoded relaying without error propagation. The results indicate that for amplified relaying and decoded relaying without error propagation the priority is to maximize the connectivity of the destination terminal, while for decoded relaying with error propagation the priority is to equalize the connectivity of the destination and relay terminals. For amplified relaying the diversity order of the system is dependent on the connectivity of the destination. For decoded relaying with error propagation the diversity order of the system is constrained by the connectivity of the minimally connected relay and therefore limited to one. For decoded relaying without error propagation the diversity order of the system is dependent on the connectivity of both the destination and relays.

The impact of the system resource constraints is different depending on the network topology. The impact of relay combination, multiple channel reception, and interhop interference cancellation is larger for network topologies with more linear node distributions (where the relay terminals are distributed with dissimilar respective distances to the source and destination terminals). These network topologies are therefore more conducive to relay terminals connected in serial. The impact of destination combination is larger for network topologies with more central node distributions (where the relay terminals are distributed with similar respective distances to the source and destination terminals). These network topologies are therefore more conducive to relay terminals connected in parallel. The impact of the number of channels available and multiple channel transmission is generally independent of the network topology.

The impact of the system resource constraints relative to each other is different for amplified relaying, decoded relaying with error propagation, and decoded relaying without error propagation. However, for a given relaying method the relative impact of the constraints is independent of the network topology. For network topologies with more linear node distributions there may be a performance benefit at low SNRs to connecting relay terminals in serial instead of in parallel. This occurs when the attenuation improvements gained from relay terminals connected in serial outweigh the diversity improvements gained from relay terminals connected in parallel. For network topologies with more central node distributions relay terminals connected in parallel always achieve the best performance.

Comparison of the results for the suboptimal example network topology and optimal network topology yields some interesting insight with respect to the relative importance of optimizing the network connectivity for the various relay methods and system connectivity models. Optimizing the network connectivity for amplified relaying has less impact than optimizing the network connectivity for decoded relaying (with and without error propagation) because it does not suffer from bottleneck relay terminals. For amplified relaying and decoded relaying without error propagation the optimal network connectivity tends to use all available relay terminals in order to maximize diversity improvements. For decoded relaying with error propagation the optimal network connectivity tends to use only a subset of available relay terminals to maximize attenuation improvements while minimizing the number of relay terminals that can propagate decoding errors. Optimizing the network connectivity for system connectivity models with full or single destination connectivity has less impact than optimizing the network connectivity for system models with moderate (2Chnl) destination connectivity. Optimizing the network connectivity for system connectivity models with full or single relay connectivity has less impact than optimizing the network connectivity for system models with moderate (2Chnl) relay connectivity.

4.6.3 Mobility Aspects

Cooperative relaying, as has been extensively described in [1], is a means of enhancing the quality of the signal at the receiver (in our case the UT) by using not only one indirect path (that through the relay) but also either additional indirect paths from other relays and/or the direct path from the AP, if , of course, this is “available”. In this section we will analyse some issues for cooperative mobile relaying, as identified in [1], with reference to the two deployment concepts described in section 2.3.3. Nevertheless, whatever the scenario, there are some basic, underlying issues that are generic for both of the scenarios which can be investigated. In general, for mobile cooperative relaying it is important to develop a mechanism for evaluation and selection of the best available terminals (potential mobile relays) to provide relaying functionalities.

4.6.3.1 Cooperative mobile relaying for Scenario 1

A number of buses might be available during the course of time, so that paths from a number of them can be “used” by the terminal. The model is shown in the next Figure 4-48.

In this case we assume uniform distribution of relays with equal distances from each other. A more non-uniform/statistical model can be anticipated for future work.

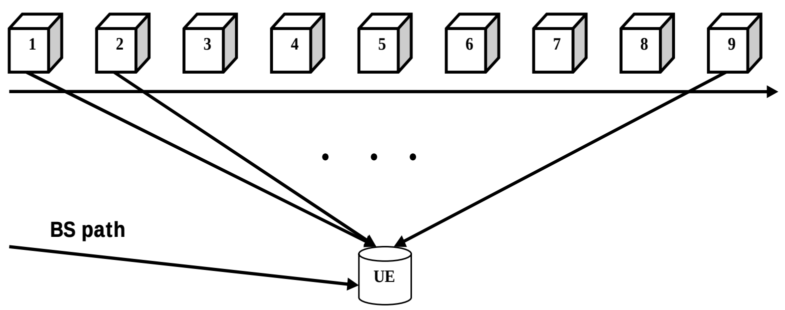


Figure 4-48 Cooperative mobile relaying for Scenario 1

The main degrees of freedom are the (a) number of available relays denoted as M, (b) Number of relays to take into account denoted as N, (c) Multiple Access scheme, number of receivers at the UT

As some of the above pause some limitations e.g. number of receivers at the UT, it is evident that not all available relays can be used. Only a sub-set of those M relays will be used. For reasons of substantial increase in complexity and delays Vs small additional gains, a trade off has to be done. Thus, in order to find the optimum solution some selection criteria need to be used to select the best relay(s) to be used.

The intention is to show how SIR values are altered (e.g. enhanced) as we take into account different combining possibilities. Some kind of initial criteria could be extracted, even at such a simple scenario.

What has been identified in the section for mobile relays (2.3.3) is the issue of non-continuity of services. Thus, if we take a look at Figure 4-23 Received power levels for direct/indirect path for Case1, we see that due to this "Gaussian-like" distribution not all the relay positions "offer" proper power levels at the UT. Moreover, in some cases those power levels might not be at all adequate for the UT. In these cases we should develop the strategy for some kind of relay selection/combining mechanism, under the umbrella of "cooperative mechanisms", to provide a more uniform-like distribution of received power levels/services.

Another issue we should take into consideration is that of the availability of carriers (buses in our case) at every time instant. If we look at the model of Scenario 1, the higher the number of buses available, the larger the group which we can select the best relays from. This means that issues related to scheduling, inter-arrival time of buses should be taken into account so that at any instance an adequate number of mobile relays is present to provide in the course of time uniform-like services in the target area/target UT. Following this, statistical results could also be used to extract information or to aid a better "planning" and deployment of mobile relays. For instance, if we could use "historic" information combined with the types of relays, measurements, maps and Kalman (prediction) filters the trajectory of the mobile relays could be predicted and as such a better prediction of cooperative "availability" could be made.

4.6.3.2 Cooperative mobile relaying for Scenario 2

For Scenario 2, similar issues apply but due to the more "random-stochastic" form of terminal distribution, velocity, mobility etc, limitations exist which should be taken into account. For instance, for Scenario 1, maps can be used to show potential trajectories of mobile relays, which cannot be applied for Scenario 2. This adds possibly additional functionalities for the selection mechanism e.g. more frequent updates, need for statistical/historic (=past) results.

Characteristics of UTs that can act as Mobile relays (MRtA), compared to "purpose-built" mobile relays (MRtB) are different. Some main differences are

- Advantages: lower velocities→ Longer availability within a cell; smaller distances to travel; larger in numbers (distribution in a cell); Large granularity of terminal classes/capabilities
- Disadvantages: Limited battery power; lower antenna height; lower capabilities; non-availability of terminals [Users not permitting the network to use their terminals as relays]

This means that MRtB inevitably target different deployment concepts.

Of course we have to bear in mind that depending on the types of UT acting as relays different levels of complexity exist. For instance, a laptop is considered to be a high-end mobile relay and as such has higher capabilities e.g. lower to zero velocity, possibly fixed power, more antennas (MIMO) etc compared to a simple mobile phone (low-end terminal).

Thus, a number of issues related to the MRN that should/could be taken into account (applicable not only for Scenario 2 but also for Scenario 1) are the following: trajectory; velocity; power available; "Historic" data (e.g. number of times that a MR is in the area to cover (depending on e.g. velocity), length of time during which the MR stays within the vicinity of the target_UT) ; RSS (Received Signal Strength); location; link budget/propagation issues; geographical area to cover; number of UEs to cover; capabilities of the terminal.

Additionally, combinations of the above for a number of relays could be taken into account. For instance, instead of having one relay transmitting in the full power, we could use three of them transmitting in 1/3 of their power, thus providing more uniformity. Although not simulated at this point in time, the selection algorithm should, in general, include a process of identifying needs, taking into account a number of parameters/measurements/static information, finding the UT population/area to serve, finding the relay-capable terminals, setting the criteria and selecting the best relay(s) to be used.

Thus, as an initial evaluation, the results presented will have to do with the investigation of some connectivity issues and statistics that could be used in cooperative mobile relaying mechanisms, based on the model description that was presented in Figure 4-48.

4.6.3.3 Simulation results for Scenario 1

For Scenario 1, we have to consider the following degrees of freedom such as multiple access technique, receivers in the UT, number of available relays etc. Specifically, in our case we assume that we have a number of relays present in equally spaced distances. We assume that interarrival time of each relay is the same and that they travel with same velocity. At this point, this is a reasonable assumption, although for future work models that are more realistic can be applied. As stated in the previous section we have: M the total number of available relays, N the number of relays to use for combining and K the number of relays, which are assumed to be interference. We assume a single frequency network. A number of combinations of paths can be used to “enhance” the signal reception at the UT. In order to investigate that, we will use SIR values. K out of those M relays will be modeled as interference. Effectively, we will have M=N + K. The SIR values will be calculated based on the following formula

$$SIR = \frac{\sum_{i=1}^{N} \Pr_i}{\sum_{i=1, j \neq i}^{K} \Pr j}, with(N+K) = M \quad (11)$$

We will assume 3 cases by varying the number N, M and both at the same time. (a) N=1, for one receiver and no combining, (b) Combining of two paths (N=2) and (c) Combining of 3 paths (N=3). Figure 4-49 **(a)** shows results where no combining takes place. Effectively M=9, N=1 and all other K=8 paths are used as interference. The results are based on Case 1 from Table 10-8. They are for a distance of 700m of the UT form the AP. Figure 4-49 (b) shows results where we use two-path combinations (the odd paths) out of all nine paths. As interference we use all other 7 paths. As we see the SIR value profile changes and tends to become more uniform while the average values almost doubles.

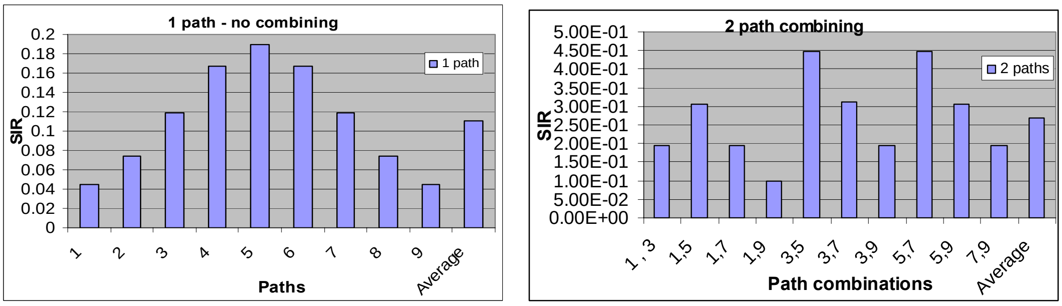


Figure 4-49 (a) and (b) Results for (a) M=9 and N=1 and (b) M=9 and N=2

4.6.3.4 Simulation results for Scenario 2

For Scenario 2 a number of results related to connectivity issues will be presented. The results will be based on Table 10-8 which lists the parameter used for the simulations. The rough model is described as follows: A terminal “target_UT” with a certain movement is present in a cell area defined as “Cell max”. Its initial position is random in the cell. Other terminals (potential mobile relays) are present in the same area and their movement is modelled based on the Brownian movement to represent a random walk. We define an area around the “target_UT” with radius “Max_Distance”. We also define the ratio of the velocity of the “target_UT” over the velocity of any other potential mobile relay as “UT/MR ratio”. Their ratio effectively represents how fast the target UT is moving with reference to the other terminals. All those values are listed in the Table 10-8. What will be presented will be (a)the Cell area, target_UT and potential mobile relays, (b) the actual and average number of connections within the target area, (c) the min, max and average distance of the potential Mobile Relays from the target_UT, (d) the paths of all relays in the x-y domain and (e) the number of connections for each Relay. Figure 4-50 shows the first 3 types of results. The first graph (Graph 1) is just a x-y representation of the target_UT and the mobile relays movement within the cell borders. Graph 2 shows the number of the total connections within the “Max Distance” area. The straight line shows the average number of connections over all the snapshots. Finally, for Graph 3 we calculate the minimum and the maximum distance of all the potential mobile relays within the “max_distance” for each snapshot and we present that. The average distance of all the potential mobile relays in the area defined by the “Max_Distance” value is also shown.

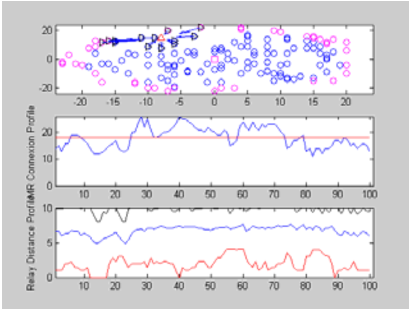


Figure 4-50 Results for Case1

For Case 1/2/3 we vary the number of Mobile Relays. This number is set to 50/150 and 300. What Figure 4-51 shows are the coverage area and the routes of all the relays for those three cases. What is shown is that for those specific parameters for the 300 MRN case, the total area is covered adequately by all mobile relays. If we calculate the number of connections (related to Figure 4-50 – Graph 2) we find out that those are Case 1 → 9, Case 2 → 18 and Case 3 → 38. What we see is that the values almost double as the number of mobile relays doubles. However, it might be the case that after certain values for the MR number, the number of connections becomes saturated and is not following the linear/proportional increase. What is more, that high number of possible connections (20-40), might not be practically usable for a "target_UT".

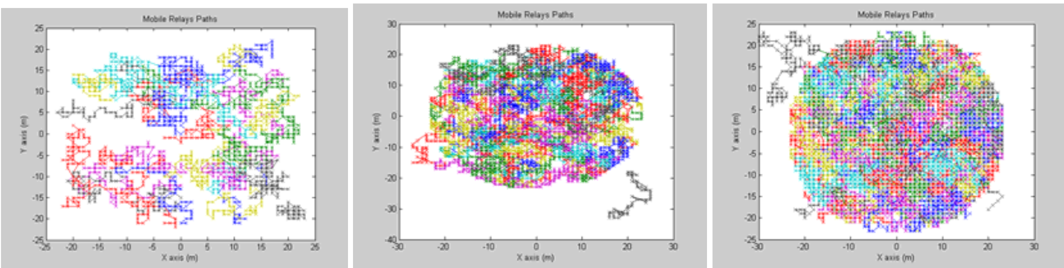


Figure 4-51 Case 1/2/3 – Coverage area for 50/150/300 MRN

For a complete list of the results, please refer to 10.4.5 in Annex 10.

4.6.3.5 Conclusion on cooperative mobile relaying

The intention of this section was to present results with reference to issues that might affect cooperative mobile relaying, in comparison to fixed cooperative relaying. It was also the intention to highlight other issues that should be taken into account in the design of a cooperative mobile relaying scheme. We showed that statistical results could be used extract information for mobile relays. However, due to the non-deterministic system, more careful analysis should be made to extract more "deterministic" results and make predictions for the "status of those mobile relays" and , thus, their availability. The previous analysis was made with reference to the two scenarios we have analysed in previous sections, which (scenarios) try to represent realistic and of high possibility to be implemented deployment concepts. A number of assumptions were made, for a simplified initial model of those scenarios. However, for future work, a more realistic model will be designed and a more holistic approach will be implemented, to take into account multiple parameters at the same time.

4.6.4 Multi-hopping and the optimum number of hops

4.6.4.1 Multi-hop Concepts

So far, our considerations of cooperative relaying have mainly focused on the case of two-hop communications. We generalize the discussion to the case of multi-hop transmission, where conventional relaying is included in the investigation. Our aim is eventually to determine the optimum number of hops for certain scenarios.

To this end, we assume that a path ("relay chain") has been established by a higher-layer routing protocol, and that a certain resource is available for the end-to-end communication from the path's source to the destination. Before analyzing multi-hop protocols, we discuss various options for assigning resources to the individual transmissions in a multi-hop chain.

Resource Usage

The orthogonality constraint[14] calls for assigning orthogonal resources, *i.e.*, different time slots or frequencies, for reception and transmission at a relay. For the two-hop scenarios considered so far, this has inherently led to an interference-free protocol. In multi-hop scenarios, there are two different general approaches:

- **No resource reuse (NRRU):** The available resource can be subdivided into k resources, one for each of the k hops. Compared to direct transmission with rate R, links must operate at rate kR, but interferences between the individual transmissions in the chain are avoided. This corresponds to the NCA case discussed in section 4.6.2.
- **Resource reuse (RRU):** The available resource is subdivided into $k'<k$ resources for the k hops. The required rate of the individual links is only $k'R<kR$, but *feedforward* and *feedback* interferences are incurred from *reusing* the resources in the chain. Figure 4-52 shows an example. Note that the NRRU scheme is a special case of the RRU case with $k'=k$. For $k'=2$, this corresponds to the 2CA case discussed in section 4.6.2.

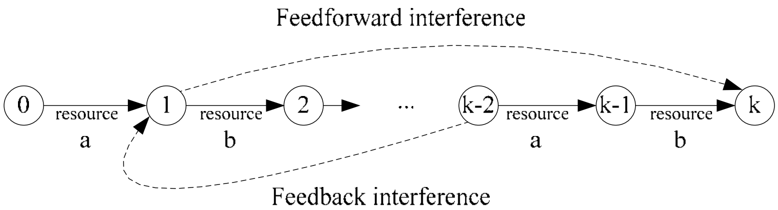


Figure 4-52 Example multi-hop network with resource reuse (RRU). Node 1 transmits at resource b, thereby interfering reception at node *k*. Likewise, the transmission of node *k-2* at resource a interferers with reception at node 1.

As a result of possible interferences, the instantaneous signal-to-*interference*-and-noise ratios do not just depend on a single channel realization (e.g. fading coefficient), but on all interfering transmissions and their fading contributions within the chain.

Cooperative Cascaded Multi-hop Relaying

The basic idea of cooperative relaying in the context of multi-hopping is to combine previous transmissions of the relay chain at the respective receivers. For long relay chains, the implications of the orthogonality constraint would call for assigning a large number of resources and appropriate combining as illustrated in Figure 4-53(b).

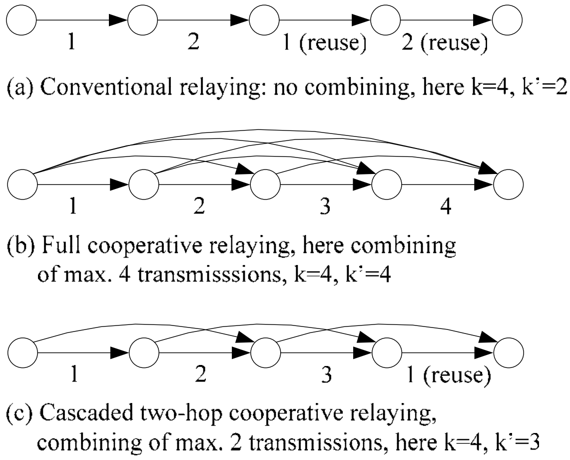


Figure 4-53 Reuse of resources in various *k*=4 RRU relaying schemes. In conventional relaying (top), the third hop can reuse the resource of the first hop, eventually requiring just two different resources to obey the hard orthogonality constraint. By contrast, *full* cooperative relaying (middle)

[14] The orthogonality constraint is the inability of smaller terminals to receive and transmit simultaneously at a single resource (time period and carrier frequency).

calls for the use of four resources. Cooperative *cascaded two-hop cooperative relaying* (bottom) represents a tradeoff: it requires three resources.

To save resources and to simplify the combining process, we introduce a *cascaded two-hop cooperative relaying scheme*. The idea is simple: each node in a relay chain combines just the preceding two transmissions. Other transmissions are discarded, as they contribute to a lower extent to the combined SINR. This is illustrated in Figure 4-53(c) and in more detail in Figure 4-54. The latter exemplarily depicts the mode of operation with k'=3 phases. Each of the RNs[15] receives two previous signals before retransmitting. Note that this scheme is the concatenation of the basic two-hop building block that we have analyzed in deliverable [1].

It becomes evident that such cascaded cooperative relaying constitutes a tradeoff between the advantages of cooperative relaying on the one hand and the challenges of combining complexity, resource allocation, and scheduling on the other hand.

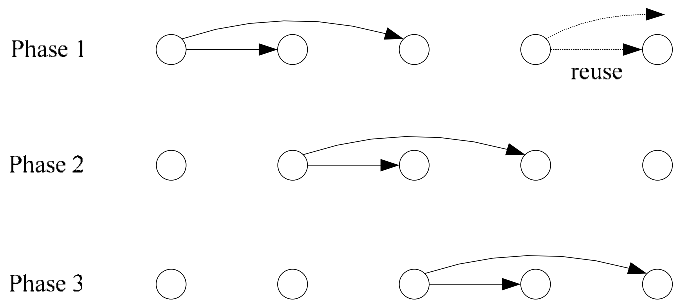


Figure 4-54 Operation of the cascaded cooperative two-hop relaying scheme. It requires three phases ($k'=3$). In this example, the first phase is reused. RNs combine the transmissions from two phases before retransmission.

Example Scenario

For purposes of simple exposition, we shall exemplarily rely on a k-hop scenario as depicted in Figure 4-53, where $k-1$ RNs are equidistantly positioned on the straight line connecting source (node 0) and destination (node k). The path gain between any two neighboring nodes is equal.

4.6.4.2 Analysis

A complete analysis of the above listed protocols, namely conventional and cooperative relaying with and without resource reuse is provided in [68]. There, the two most contrary fading channel models AWGN and Rayleigh, respectively, are examined; the results allow for deriving general conclusions of the performance in LOS and NLOS conditions. For purposes of compact presentation we focus on essential results in this deliverable.

4.6.4.3 Discussion of Multi-hop Concepts

From the discussion in [68], one can conclude that (i) in connection with interference cancellation (denoted here by $\eta << 1$)[16], the resource reuse scheme (RRU) schemes outperform the interference-free NRRU scheme, as the latter suffer from a per-link rate kR instead of $k'R$ as the RRU schemes ($k'<k$), and (ii) that a minimum interference cancellation η is necessary to sufficiently lower the outage probability floor that results from mutual interferences.

``What is the optimum number of hops?"

It is further discussed in [68] that for sufficiently strong interference cancellation, the SNR gain monotonically increases in the number of hops. Hence, provided such strong interference cancellation that ensures that the desired outage probability is reached, the optimum number of hops for the resource reuse scheme (RRU) is infinity. In general, however, the optimum number of hops decreases as the interference cancellation efficiency reduces.

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[15] except the first relay

[16] Note that $\eta << 1$ implies that weak signals can be decoded with high probability.

For the NRRU scheme, which assigns orthogonal resources for each of the hops, there is a tradeoff between the pathloss reductions from shorter hops on the one hand and repetition coding and increased per-link rates on the other hand. Figure 4-55 illustrates the resulting optimum number of hops for Rayleigh and AWGN scenarios, respectively, as a function of the targeted end-to-end spectral efficiency R.

For Rayleigh channels, the results in Figure 4-55 suggest that the *two-hop* protocol is optimal for a wide range of scenarios; it is applicable up to rates $R \leq \sim 5$ bit/s/Hz, while conventional relaying provides benefits only for $R \leq \sim 3$ bit/s/Hz (red lines). For AWGN channels, conventional and cooperative relaying exhibit the same optimum number of hops for almost all spectral efficiencies. The minor broadcast advantages that cooperative relaying realizes from the optimized power fraction in the two-hop case leads to the shown differences. For rates $R \geq 4$ bit/s/Hz, direct transmission remains favourable in either case for α=4.0.

More generally, for both channel models and relay schemes, a low number of hops is beneficial under our assumptions. Finally, recall that the example scenario constitutes a certain best case for relaying: nodes are equidistantly places on a straight line. For realistic - and hence more irregular - scenarios, one would expect even lower number of hops to be most beneficial.

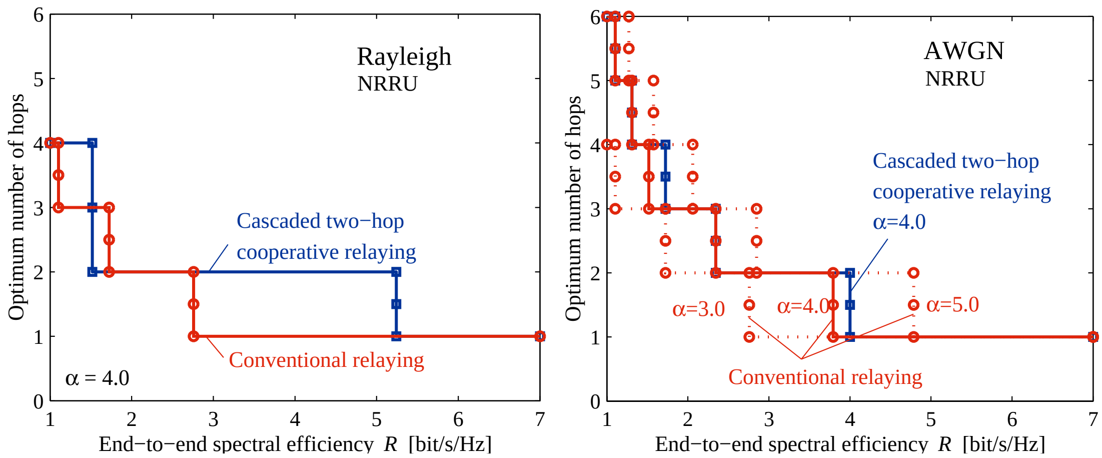


Figure 4-55 Optimum number of hops in the example scenario vs. targeted end-to-end spectral efficiency in Rayleigh fading (left) and AWGN channels (right). Using two hops is beneficial for the cooperative AdDF protocol for a wide range of end-to-end rates, while conventional relaying falls back to direct transmission already at lower rates. Parameters: target outage probability 10^{-2}, various path loss exponents α.

4.6.5 Approach Towards Harmonization of Concepts

With respect to cooperative relaying, and even more generally, a number of concise conclusions can be drawn based on sections 4.6.1 to 4.6.4:

- Various cooperative concepts have been proposed and studied. They differ in protocol nature (static vs. adaptive), forwarding strategy (amplify-and-forward vs. decode-and-forward), and other parameters.
- Adaptive decode-and-forward protocols are promising. Their adaptive nature limits error propagation by forwarding only when a certain quality level (SNR, CRC check) can be guaranteed. Their decode-and-forward property makes them similar to conventional store-and-forward relaying, which is attractive from an implementation point of view.
- The system connectivity analysis has shown that for amplified relaying and decoded relaying without error propagation the priority is to maximize the connectivity of the destination terminal, while for decoded relaying with error propagation the priority is to equalize the connectivity of the destination and relay terminals.
- In this respect, using multiple antennas at relays can significantly enhance the performance. Implementations using selection combining require only a single RF chain (receive or transmit); therefore, they offer a good trade-off between cost and performance.
- Using just two hops is well-known for its simplicity with respect to routing and resource allocation. In addition, even under ideal conditions with respect to combing and capacity exploitation, the two-

hop approach is optimal for a wide range of targeted rates and path loss exponents in fading channels as it provides a trade-off between diversity gains and path loss reduction on the one hand and rate increase and repetition coding on the other hand.

- Mobility puts a number of additional challenges.

It is proposed that adaptive decode-and-forward protocols are investigated in more detail in cooperation with T3.3 and T3.5 to identify white spots and to pursue their integration into the WINNER system concept.

4.7 Deployment Concepts Based on Initial Proposal for WINNER modes

This section will present the consolidated and agreed definition of new deployment concepts concerning the Wireless Feeder System as well as the heterogeneous relay node (HERN), in particular taking into account the initial concept for the WINNER air interface modes proposed by WP2.

4.7.1 Background and Definitions

Regarding the wireless feeder concept in the context of the WINNER project, is defined as the wireless feeder system used to connect the WINNER APs to its transport network by means of point to multipoint connections (PTM topology). Of course, from a general point of view the transmission technology used by the system for feeding the WINNER APs could be based on wireless or wire line solutions, but at present we are interested on the wireless solution, like an important alternative to the wire line solution, in order to benefit of wireless option characteristics just as the fast, less expensive and more flexible deployment, reachable with the wireless solution. Also, it is important to note that regardless the solution adopted in each particular case, the transmission technology to use would be exclusively for connecting the APs with its transport network, and therefore the WINNER users could not connect to this network, which is transparent for final users. In the initial assumptions for deployment scenarios, collected in the internal report IR3.1 [113], the wireless feeder concept was included as a complementary scenario to the four basic scenarios (In-building, Hot spot, Urban/Suburban and Rural), assuming some optimistic figures in particular for range and throughput parameters, under different conditions (LOS and NLOS), and for different environments (hot spot, urban and rural). Likewise, in deliverable D7.2 [104],the wireless feeder sub-scenario has been refined in order to provide a clear definition to WP5 (Channel Modelling), in which is currently developing models with the goal to obtain more realistic values of the maximum reachable ranges for the wireless feeder system. One of the prioritised deployment scenario, decided also in [104], for Phase I of the project, was the B.5 defined as the wireless feeder for WINNER APs in a hot spot area with LOS propagation conditions and then with the antennas of the master station and peripheral stations located on the building rooftop. Concerning the frequency band to use in the model, which WP5 has to develop for B.5 scenario, though initially was thinking in frequencies below 6 GHz for NLOS conditions and above 6 GHz for LOS conditions, and in spite of B.5 has been defined with line of sight between antennas, for deriving the channel model to use in this scenario, it seems more reasonable to utilize frequencies in the interest band of WINNER, that is, around 5 GHz. Anyway in the annex it is included the preliminary results of range estimations based on simple propagation models (interference and multipath effects not considered), for one of the transmission technology used for this purpose such the IEEE 802.16 or WMAN. We have selected the IEEE 802.16 because is one of the most promising technology supported by some important Standardization Organism, and currently is being used for the link between the RNC and Nodes-B (interface Iub) in the UMTS system. The goal of these calculations was to compare the initial ranges established for this kind of system with the maximum estimated ranges for current technologies such IEEE 802.16. The main conclusion of this theoretical estimation was that the proposed ranges for Wireless Feeder Systems in the IR3.1 are much larger than the estimated.

With regard to heterogeneous relay node, this is a new network element wirelessly fed by another relay or AP using a given radio technology, and serves to another relay or final users using in this case a different radio technology. In general terms, the involved radio interface technologies might be, different modes of the same radio access technology (RAT), or completely different RATs, that is, different standards like WLAN and WMAN (for example in STRIKE project was investigated the particular case of HiperMAN and HiperLAN/2 interworking [114]), UMTS and WMAN, or even any legacy system and WINNER system. Nevertheless, in WINNER, the investigations concerning the HERN concept, are focussed on the case of different modes of the same RAT, and in particular for the WINNER air interface modes. Besides, other important aspect to take into account is the fact that the same physical node can integrate both concepts, homogeneous and heterogeneous relaying as we will see in the next section. Therefore, in other words we can say that in the context of WINNER project, the HERN as logical node, is a wireless network element belonging to the WINNER Radio Access Network (RAN) and connected to the transport network through an AP, and it uses two different WINNER radio interface modes for holding a communication in a given instant of time.

4.7.2 Different Approaches for Feeding RN

With the current definitions of WINNER air interface modes, the difference between heterogeneous and homogeneous is not that black and white, since both the AP and RN may employ several common modes, say mode 1 and mode 2 (e.g. the wide area and the short range modes). To illustrate the problem and assuming two modes at stake, Figure 4-56.. shows the five different alternatives in terms of the modes used by each of the elements involved in a given multi-hop communication (AP, RN and UT). Out of these five possibilities the first one is a homogeneous relay node, the second one is a heterogeneous relay node, whereas the rest may be both. We now have two different possibilities to define heterogeneous relay nodes:

- The second case shown in Figure 4-56, denotes always the pure heterogeneous relaying concept, whereas the first case where the RN appears like a UT toward the AP, is only a heterogeneous relay node if this has a possibility to use different modes when communicating with the AP (i.e. cases 3, 4 and 5 of Figure 4-56).
- The second case is always a heterogeneous relay node whereas the first case is only a heterogeneous relay node if one mode is always used to feed the RN and another mode is always used to communicate with the UT (case 3 of Figure 4-56 if one assumes that mode 2 is only used for feeding the RNs).

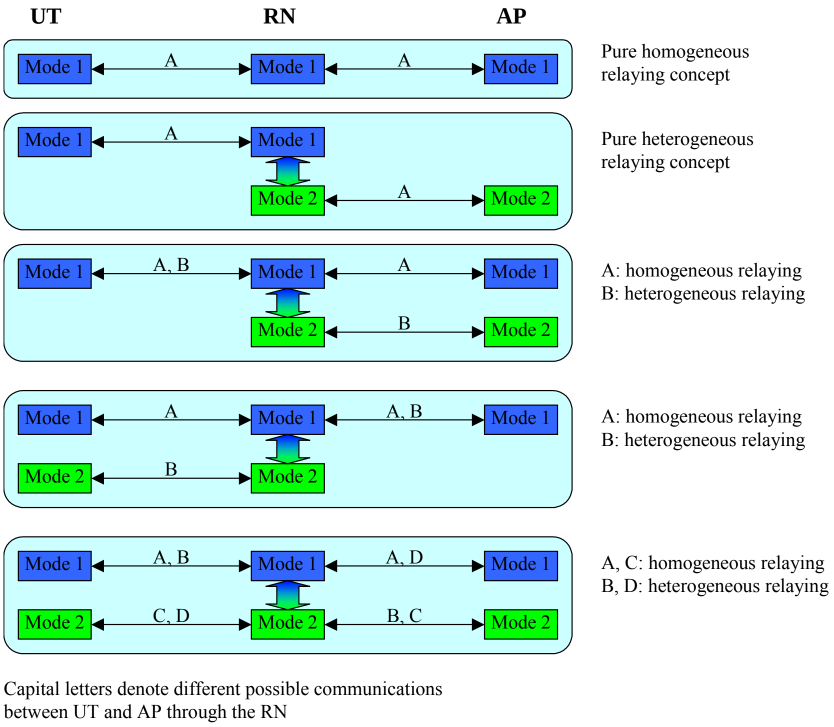


Figure 4-56 Five different cases explaining the homogeneous and heterogeneous relaying concepts

On the other hand and taking into account the initial WINNER air interface modes, we can also think in two different kind of HERNs depending on the mode used by the AP for feeding the HERN. It should be considered the fact that so far, the proposed modes from WP2 are based on two different duplex schemes, that is, pure TDD and half duplex FDD (H-FDD), and each one of them will have its own frequency band. Besides from the four basic modes, only the “Wide area – Cellular” mode will utilize the H.FDD duplex scheme, whereas the other three modes (“Wide area – Feeder links”, Short range – Cellular” and “Short range – Peer to peer”) will utilize the TDD duplex scheme but using the same frequency band. Besides the feeder links mode has been specially though for feeding but not for giving service to final users. Therefore, Figure 4-57 shows the two possible kind of HERNs in terms of the mode used for feeding the RN. The top part corresponds to the first type where the HERN is seen by the AP like a set of users working in the same mode used by the AP in its coverage area, and the bottom part shows the second type where a specific mode for feeding the HERN is used. It is important to mention, that in the current deliverable and from a heterogeneous relaying perspective, the second case will be the contemplated in the further sections, where we will focus on the development of new deployment concepts based on the utilization of RNs, using a particular mode specifically thought for relaying purposes, and different to the modes used for serving to final users.

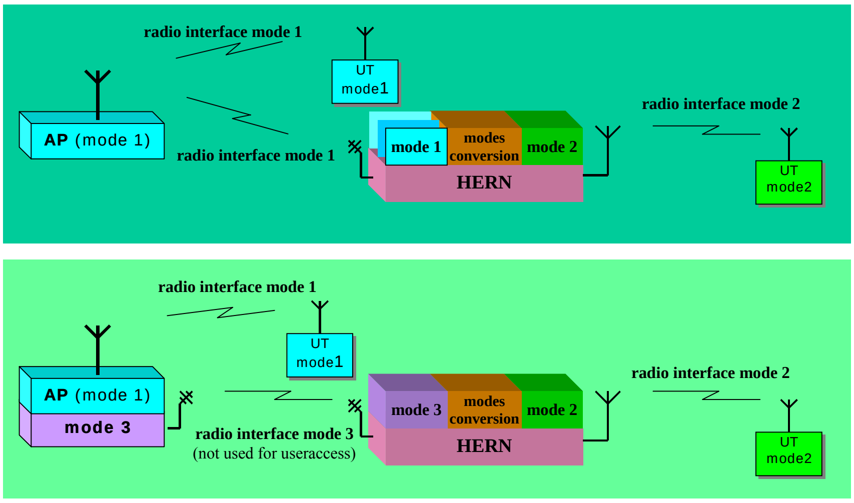


Figure 4-57 Different kind of HERNs depending on the mode used in the AP-RN link

Of course, the development of the new relay concept depends strongly on the characteristics of the radio interface involved in the operation of this element. Although the new WINNER air interface is not yet definite, and there are yet some important aspects to solve in the definition of the different modes to contemplate in this new radio interface, at this moment there is a firm and clear proposal coming from the activities carried out in WP2 [115], about the first outline of the WINNER air interface concept, as well as the different basic modes of this interface initially contemplated. So, we will use in our subsequent study of the modes involved in a relay node, the initial conclusions and reflections about the four basic modes identified in WP2, that is, "Wide area – Cellular", "Wide area – Feeder links", "Short range – Cellular" and "Short range – Peer to peer" modes.

4.7.3 Deployment Examples using a specific mode for AP-RN link

The decision of what duplexing methods to use in the WINNER radio interface is at this moment practically definitive, and the most promising schemes from the different alternatives analyzed in D2.5 v0.1 [115] are pure TDD and half duplex FDD. In the same way according to the initial modes identified in WP2 for WINNER radio interface, we will analyze and compare the different combinations of these modes in a RN in order to present the most appropriate alternatives from different viewpoints such technical complexity, economical or spectral efficiency. It is important to remark that these modes are provisional, and on the basis of our requirements (deployment point of view) we should participate in the final definition of these modes. In general terms we can distinguish two cases: when the involved modes in the RN are using the same duplexing methods, or when they are using different duplex schemes. Also we perform the analysis for different scenarios attempting to find the best option for each of the WINNER scenarios.

It is important to note that the selection of duplexing method in any communication system is always an important aspect for reaching an efficient deployment network from capacity and radio resources utilization points of view. In the case of relay based system, more than duplex scheme we could talk about triplex scheme, since in any connection end to end from AP to UT through a relay, there are four phases, the downlink from AP to relay, the forwarding phase comprising itself of DL (from relay to UT) and UL (from UT to relay) phases, and the uplink phase from relay to AP. In the particular case of heterogeneous relay nodes, the temporisation of these four phases could be more complex than in the case of homogeneous relays, since when there are two different modes involved in the communication, before forwarding the data (either downlink or uplink) it is necessary to perform in the RN the conversion from one mode to another one, and so we have to consider in the time distribution between the different elements, the extra time needed for this process.

So the present section deals with the study of RNs working in two different modes of the initial proposal for WINNER air interface. It is convenient to mention that all the contemplated cases assume the use of a specific mode for feeding the RN (feeder link), that is, the AP is employing a mode specifically thought for feeding, different to the mode used by the AP for serving UTs in its coverage area.

On the other hand, it is clear that the coexistence of different radio systems, in the same geographical area or even in the same location, gives rise to potential problems of electro magnetic compatibility (EMC), including both immunity and emission concepts. Besides the equipment has to fulfil with the fixed emission specifications, in order to avoid the wrong operation of another ones. So in this section we outline also the coexistence problem of involved WINNER air interface modes in a HERN, trying to identify the potential interference problems which could arise as a result to use in the same location, that is the RN, two different radio interface modes from the four basic modes proposed by WP2. It should be noted that the WINNER modes are not yet definitive and besides from an interference analysis point of view there are many important aspects such precise frequency bands, output power, and spectrum emission mask with the maximum levels of allowed out-of-band and spurious emissions, which would be necessary to know in order to perform an exhaustive and complete interference analysis. In other words, for an useful interference study of the new WINNER air interface modes, is required to know perfectly the specifications of the radio signals involved in those modes. Anyway, the preliminary considerations should take into account for the design of the radio interface, as well as for the selection of the frequency bands to use in those modes. Of course the fact to consider the potential interference problems during the specification process, will aid the coexistence of different radio interface modes in the same location.

4.7.3.1 Overview of proposed Modes for WINNER Air Interface

From WINNER initial assumptions some system requirements were established [113]. For example, in short range scenarios with low mobility support, has been estimated an aggregated throughput of up to 1 Gbps, as long as for wide area scenarios with high mobility support a total throughput about 100 Mbps has been considered enough. So, it is clear that a very wide band in order to achieve this throughput will be necessary. However, for signalling and low throughput services, a narrower band would be more suitable since it requires less processor power. Currently there is a proposal coming from WP2 [115], as working assumptions, for using two bandwidth allocations, one of 2x20 MHz and another one of 1x100 MHz. Besides the bandwidth of the former allocation will most likely change to cope with the wide area requirements . Summarizing WINNER could use two different bandwidth channels (although not necessarily) to cope different traffic rates. Regarding this aspect, the initial assumptions are:

- The wideband channel (~80-100 MHz) for supporting an aggregate traffic of 1 Gbps.
- The narrowband channel (~5-20 MHz) for having a larger coverage.

According to [116], Half Duplex FDD is considered for the wide area and Pure TDD for the short range. It is required a throughput of 1Gbps in the short range for what a 100 MHz band channel is needed for each operator. In the wide area, low throughput is admissible, so few channels of around 20 MHz for operator should be enough.

One of the most important assumption for WINNER air interface is the flexibility that this interface has to provide in order to be able to adapt at different user requirements and scenarios. Of course, the necessary number of modes for reaching an ubiquitous radio system from a user perspective, is very large. Anyway from WP2 [117], four different physical layer modes have been identified which somehow cover up all the user requirements and scenarios, envisioned for future wireless communications. Of course depending for instance on the different deployment scenarios more modes may be defined in the future. In any case at present the four basic modes are:

1. Wide area – Cellular: **mode A1**. Paired narrow-band channels (H-FDD, 20 MHz). Multiplexing and multiple access method: OFDM / (F+T)DMA for DL and SC / (F+T)DMA for UL.
2. Wide area – Feeder links: **mode F1**. Unpaired wide-band channel (TDD, 100 MHz). Multiplexing and multiple access method: OFDM / (F+T)DMA for DL and OFDM / (F+T)DMA for UL.
3. Short range – Cellular: **mode A2**. Unpaired wide-band channel (TDD, 100 MHz). Multiplexing and multiple access method: OFDM / (F+T)DMA for DL and OFDM / (F+T)DMA for UL.
4. Short range – Peer to peer: **mode A3**. Unpaired wide-band channel (TDD, 100 MHz). Multiplexing and multiple access method: OFDM / (F+T)DMA for both, DL and UL.

The frequency band of operation for all of these modes is 0.175 GHz < f < 6 GHz. The mode F1 here is understood like the mode used for connecting AP to AP, AP to RN, or RN to RN. The nodes involved in the links are stationary, the range is about 1 km, and the peak data rate per bandwidth is around 5 Mbps / MHz. Regarding the multiplexing method, except for the uplink in the Wide area – Cellular mode that SC (Single Carrier) has been considered, OFDM-based single frequency cellular network using spread spectrum techniques (for minimizing inter cell interference problem) is proposed for the rest of WINNER modes. The technical challenge for this proposal of course is the synchronization accuracy in the cellular environment, as well as the channel estimation or prediction.

Henceforth the studied cases, included in the next sections, are assuming always the use of a particular mode for the AP-RN link ("Wide area – Feeder links" mode), and therefore we may distinguish two different situations, one using the same duplex scheme (TDD) and the other one using different duplex schemes (TDD and FDD). Also in the examples included in the next sections, we have considered that the AP implemented only two WINNER air interface modes, the "Wide area – Feeder links" (mode F1), existent in all the cases, and depending on the particular case, the "Wide area – Cellular" (mode A1) or the "Short range – Cellular" (mode A2).

4.7.3.2 Same Duplexing Methods

For the case of RN using the same duplex scheme, that is pure TDD (mode F1 for AP-RN link and mode A2 for RN-UTs links), depending on the mode used by the AP in its coverage area, we can distinguish two different scenarios, as Figure 4-58 shows. In the first scenario, the AP and RN use the same mode when communicating with the UTs, e.g. mode A2 (short range). In the second scenario, the AP and RN use different modes when communicating with the UT, e.g. the AP uses mode A1 (wide area) whereas the RN is using mode A2 (short range).

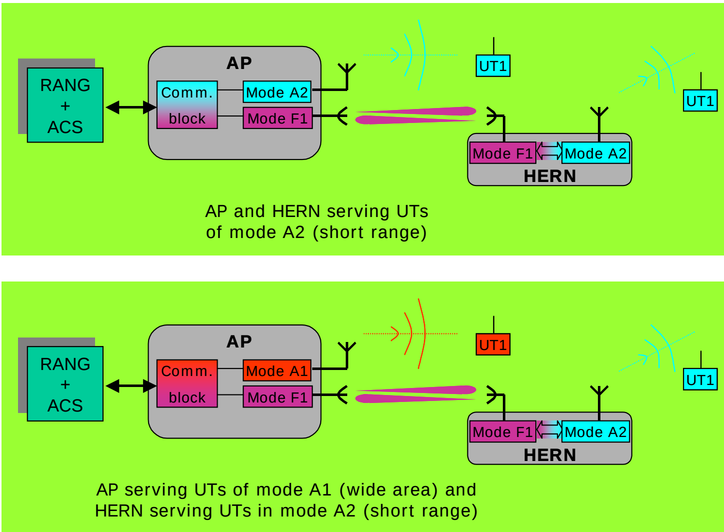


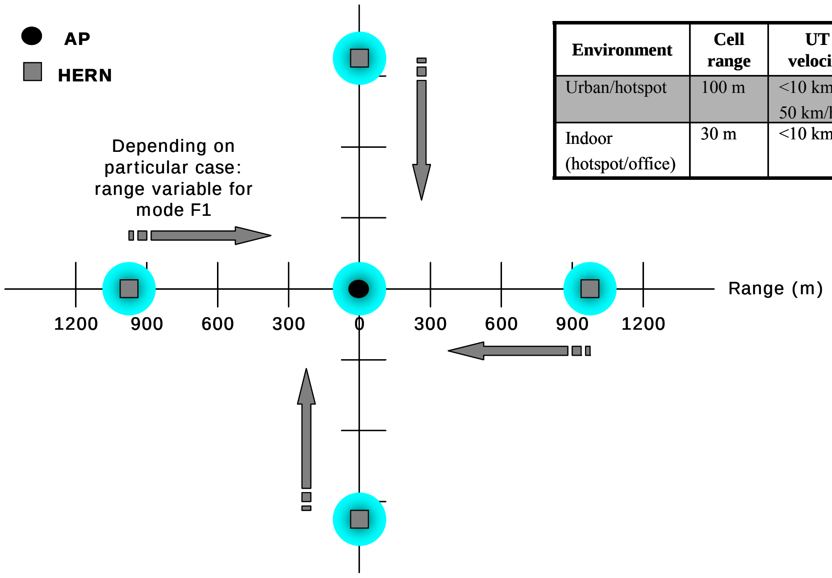
Figure 4-58 Two different scenarios for RN using modes with the same duplex scheme

Concerning the AP design, it should be noted in all the cases, the presence of a functional block referred to as "Communication and Signal Process Block". The purpose of this part, in general terms, is to process the data coming from the transport network (RANG + ACS) through its corresponding interface, and deliver this processed data to the radio frequency part of the AP. Obviously, in the other direction this block has to perform the opposite process. In other words, this block performs the process relative to the communications between the AP and its controller. Though there are different possibilities to distribute the functionality of an AP between its own elements, to follow we list some of the functions to implement within the communication block.

- Multiplexing and de-multiplexing of the communication flows.
- Process of the signalling for the allocation of the RF channels to the traffic channels.
- Process of the information concerning to the Operation and Maintenance of the radio network.
- Ciphered and deciphering of the information. Also it is possible to implement these functions in the radio frequency block.

Of course, the final design of this block as well as its functionality ought to be harmonised with the current protocol architecture proposed in D3.1/D3.2 and the current scheduling architecture proposed in the XWP-MAC document.

Taking into account the initial requirements fixed by WP2 for mode F1, that is, a cell range around 1 km and a peak data rate of 5 Mbps per MHz, the radio network deployment possibilities are very large. For instance, modifying the radio power of the mode F1, we can separate more or less the RN from the AP, and hence we might utilize different power levels to cover different kind of scenarios. In any case, for the particular instance of same duplex scheme and assuming always the use of mode F1 for the AP-RN link, we can distinguish two different types of deployment concepts. These are when the AP is using the mode A2 for serving at the final users in its own coverage area, and when the AP is using the mode A1 for this same aim. In this way, Figure 4-59 illustrates an example of the first type of deployment concept, where both AP and RNs are using the mode A2 for giving service at final users. For clarity the AP-RNs links using the mode F1, are not shown in this figure. It is convenient to remember that due to the stationary characteristics of these links, it is envisioned the use of high directive antennas and more efficient coding and modulation schemes. Of course depending on the location of the UTs, they will be directly served by the AP or by means of the nearest RN. This example shows a particular case of radio network deployment, one AP and four RNs, in a urban/hotspot environment with low mobility support, but for users demanding high data rate services. In terms of the range of mode F1 (adjustable by means of the radio power used by this mode), of the amount and distribution of RNs, as well as of the mode A2 ranges for both, AP and RNs, different kind of scenarios could be covered with this deployment concept. Assuming the same frequency band for modes A2 and F1, it should be noted that the total throughput in a given time instant, which obviously will depend on the particular conditions of proximity to the AP or RN, and velocity of each user (i.e. an average of 500 Mbps for a wideband channel of 100 MHz), will have to be dynamically shared between the AP-RNs links, and the established communications by the AP and the RNs with the final users. On the other hand, the aggregated data rate of a certain RN in mode A2, would be limited by the throughput assigned by the AP to this RN.



Environment	Cell range	UT velocity	Peak data rates / Bandwidth
Urban/hotspot	100 m	<10 km/h 50 km/h	10 Mbps / MHz 1 Mbps / MHz
Indoor (hotspot/office)	30 m	<10 km/h	10 Mbps / MHz

Figure 4-59 Example of urban/hotspot environment for AP and RN serving in mode A2

Figure 4-60 describes one possible frame structure, from AP and RN perspectives, for this class of radio network deployment, where all the involved air interface modes are using the same duplexing method (TDD) and the same frequency band. As it can be seen, the frame is split in TX/RX transition gaps (black boxes) and in two sub-frames, one for AP downlink connections, and another one for AP uplink connections. Likewise, for each of these phases and from AP point of view, there is an idle period of time, which is used by the RNs to broadcast frame control and transmit user data to the UTs connected at the respective RN, in the DL sub-frame, and to receive the requests (contention phase) and user data from the UTs camped on the RN coverage area, in the UL sub-frame. In the same way, from the AP perspective, the useful part of the DL sub-frame is distributed between the frame control and user data for both, UTs directly served by the AP and RNs. Alike, the useful part of the UL sub-frame is distributed between the requests and user data for both, UTs and RNs connected to the AP. Concerning the radio resources distribution between the UTs directly served by the AP and the RNs, although in Figure 4-60 is shown an arrangement based on the use of a variable group of sub-carriers, it seems more favourable to use a distribution in time, since for the uplink direction in order to avoid overlapping and interference problems, mainly between different UTs (the RNs have a fixed position and so it is not needed any sub-carriers group as guard band), it is necessary to leave a certain number of sub-carriers as guard band, being thereby an alternative with less spectral efficiency than the distribution in time because in this case the AP is able to determine more exactly the time where the UT has to transmit, shortening the guard-time between UTs.

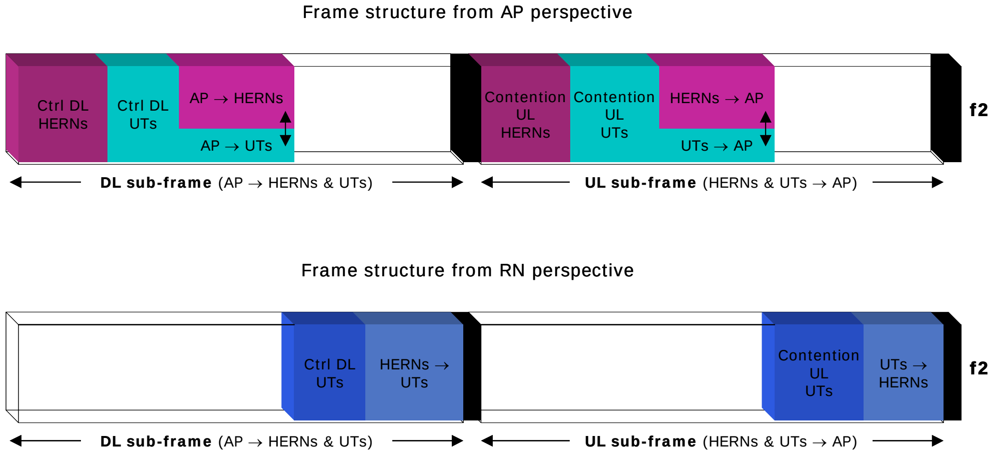


Figure 4-60 Frame structure for AP and RN serving in mode A2

Regarding the second type of deployment concept, Figure 4-61 shows an example of mixed environment, in particular urban – urban/hotspot areas, where the AP is using the mode A1 (Wide area – Cellular with FDD duplex scheme) for serving at the UTs in its coverage area, as long as the RNs are working in short range mode, that is mode A2, for giving service at its UTs. Like in the previous example the AP-RNs links using the mode F1, specially though for this purpose, are not shown. In this concrete example, the ranges for modes F1 and A1 are 300 m and 400 m respectively, although obviously there are other possibilities, as long as the ranges do not exceed the maximum values established for each of these modes (1 km for mode F1 and 2 km for mode A1). Regardless of the ranges used by each of the modes, in this kind of deployment due to the AP for serving final UTs, is using one frequency band (FDD duplexing method with paired channels of 20 MHz) different to the used by the modes F1 and A2, the total throughput provided by the TDD channel (bandwidth of 100 MHz), is only shared between the users served by the RNs and the AP-RNs links.

On the other hand, one important characteristic of this type of deployment, is the possibility to exploit the umbrella capability, since with a suitable adjustment of the ranges for feeder links (mode F1) and cellular wide area mode (mode A1) (i.e. adjusting the radio power used by these modes), the users could be served in a continuous way, sometimes directly by the AP in mode A1, other times through the RNs in mode A2.

The frame structure for the TDD channel and from the RN point of view, would be similar to the shown in Figure 4-60, but in this case the distribution of times or sub-carriers would be only between the AP-RN links and the UTs served by the RNs, since the UTs directly served by the AP is using other frequency band.

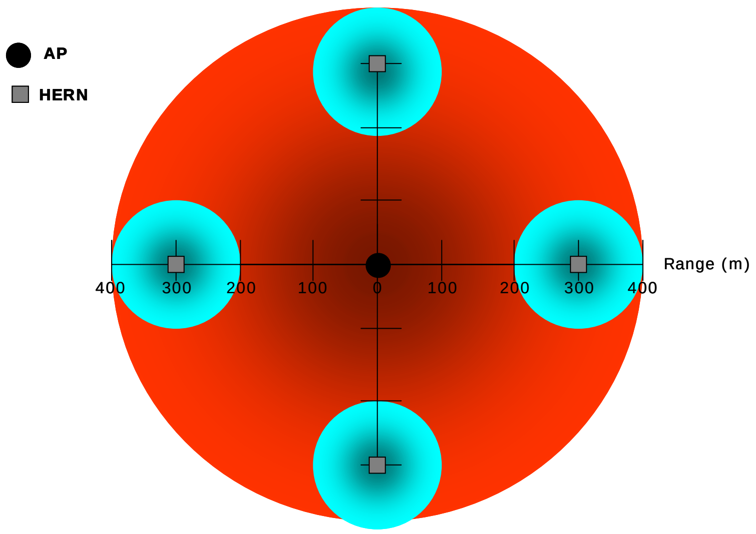


Figure 4-61 Example of mixed environment urban – urban/hotspot, for AP serving in mode A1 and RN serving in mode A2

Finally for the case of relays using the same duplex scheme (TDD), it is convenient to note that from the RN point of view, the transmission and reception of the communications between the RN and the other elements (AP and UTs) are distributed in different periods of time, and so it should not appear any interference problem. Anyway it would be possible to think also in other ways for sharing the radio resources, that is, splitting the same band in different frequency or code sub-groups, or even re-using the same frequency band in all the RNs connected to the AP, utilizing high directive antennas. The spectral efficiency along with the interference problem are two important aspects to consider when in a particular case, we have to make a decision about the best way to distribute the radio resources (time, frequency, code). For example in a system based on OFDM modulation, as we have mentioned before, the partition of the communications in different groups of sub-carriers, may cause a low spectral efficiency, at least in the UL direction, due to the guard bands necessary to separate the different groups (between 1 and 10 % of the total number of carriers of the OFDM symbols). Regarding the possible interference caused by neighbouring cells, it is important to note that for the feeder links mode (RX, reception intervals) the problem may be practically solved by means of the use of high directive antennas. On the other hand, for the reception intervals of the other mode ("Short range – Cellular"), which is using sectorial or omni-directional antennas, the UTs of the neighbour cells using the same band and close to the border of the RN coverage area may provoke some interference problem, although if all the cells are synchronized the problem could be avoided using, for example, different codes for neighbouring cells. If synchronization is not assumed, the worse case of interference will be generated between UTs of different operators working in TDD mode, and so the guard band between the TDD channels of each operator must be lengthened. Of course, in general terms, proper separation between FDD and TDD modes of the same or different operators, should have distant frequency allocation in order to avoid interference. For example TDD bandwidth can be placed in the middle of the paired bands of FDD in order to have a continuous spectrum for WINNER as it is proposed in IR6.2 [118], or in another frequency location if WINNER can not have contiguous spectrum allocation.

4.7.3.3 Different Duplexing Methods

For this case we have pure TDD for AP-RN links and half duplex FDD (H-FDD) for RN-UTs links. H-FDD here should be understood as the communication where the RN can transmit simultaneously in DL and UL, while the UTs served by RN have only the possibility to transmit or receive in a given interval of time. 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 perpective, the fact to have to implement two different frequency bands, one for the TDD mode and the 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.

Similar to the previous section, for the case of RN using different duplex schemes (pure TDD and H-FDD), depending on the mode used by the AP in its coverage area, we can distinguish two different scenarios, as Figure 4-62 shows. In the first one, the AP is working in the same mode used by the RN to give service at its final users, that is mode A1 (wide area). And in the second one, the AP is working in a different mode of the used by the RN, that is mode A2 (short range). Of course these are just examples and more scenarios may be envisioned.

As in the previous section, the block referred to as communication block, is the part of the AP in charge of the communications between the transport network (RANG + ACS) and the radio frequency part of the own AP. The design of this block, as we have mentioned before, will have

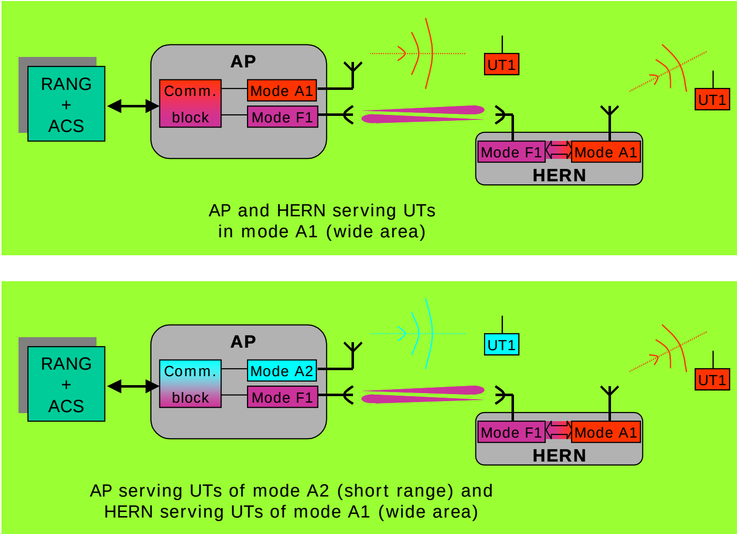


Figure 4-62 Two different scenarios for RN using modes with different duplex scheme

An example of a deployment concept for the first scenario where the AP and RN are using the mode A1 (FDD wide area) for serving UTs is given in Figure 4-63 that illustrates the particular case of a rural environment with an AP and four RNs. The connections between the AP and the RNs using the feeder links mode (mode F1), are not included. The dotted circle represents the coverage area of the AP working in mode A1 and using omni-directional antennas. The overlap between the coverage areas of the AP and the RNs, could be reduced well decreasing the radio power of the AP, well increasing the range of the mode F1, so that the deployment be similar to a cellular layout. Besides the RNs in this example, are using sector antennas and so we may have a reuse distance of one, obtaining thus a higher spectral efficiency. Assuming a bandwidth of 20 MHz for the mode A1 channels and a maximum data rate of 1 Mbps / MHz (for stationary conditions), each RN could support till 40 Mbps of DL and UL aggregated data rate. On the other hand, the mode F1 is using a TDD channel of 100 MHz of bandwidth with a foreseen data rate around the 5 Mbps / MHz, thanks to the use of high directive antennas and the static conditions of the AP-RNs links. Therefore to be able to accommodate the total traffic of the RNs, the maximum number of these nodes should be lower than the ratio between the total data rate of mode F1 (i.e. 500 Mbps) and the peak data rate of mode A1 (i.e. 40 Mbps). Nevertheless, we could increase the number of RNs (perhaps five or ten times more) due to multiplexing gains and taking into account that 40 Mbps is only the peak value. On the other hand, if we are able to improve the AP-RNs links so that the ranges of mode F1 be larger, the coverage area of this kind of deployment could be increased.

Another example of this type of deployment (AP and RN serving in mode A1), for a mixed environment suburban – rural is shown in Figure 4-64. In this particular case, the RNs are using omni-directional antennas and the distances between the different elements involved in the deployment are such that there are large areas of overlapping, and so it is not possible that the opposite RNs re-use the same radio resources. However on the other hand, the requirements for the handover between coverage areas would be less demanding (there is more time for handover between areas).

Environment	Cell range	UT velocity	Peak data rates / Bandwidth
Rural	2 km	0 km/h 250 km/h	1 Mbps / MHz 0.5 Mbps / MHz
Suburban	1 km	0 km/h 90 km/h	1 Mbps / MHz 0.5 Mbps / MHz
Typical urban	500 m	0 km/h 70 km/h	1 Mbps / MHz 0.5 Mbps / MHz
Bad urban	400 m	0 km/h 50 km/h	1 Mbps / MHz 0.5 Mbps / MHz
(Outdoor to)	400 m	5 km/h	1 Mbps / MHz

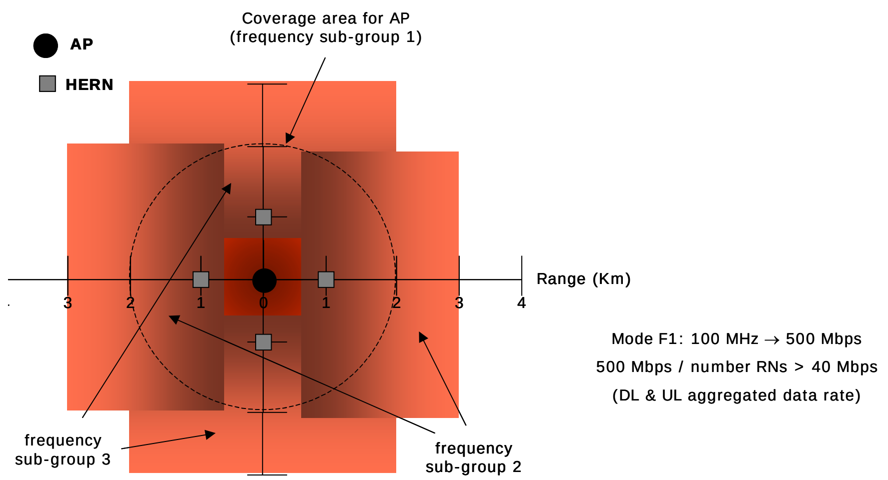


Figure 4-63 Example of rural environment for AP and RN serving in mode A1

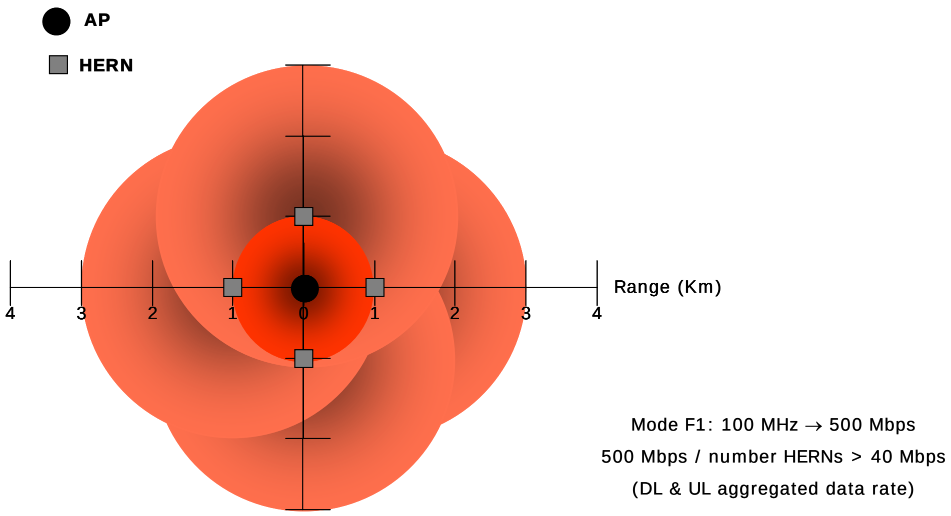


Figure 4-64 Example of mixed environment suburban – rural, for AP and RN serving in mode A1

One possible frame structure for this kind of deployment, where the AP and RNs are using the mode A1 for serving the final users and the AP-RNs links are implemented by means of the mode F1, is shown in Figure 4-65. Due to the TDD channel, in this case, is used only for the AP-RNs links, the total bandwidth of this channel (100 MHz) is distributed between the RNs. The frame is split in downlink and uplink sub-frames, and in TX/RX transition gaps (black boxes). As we have mentioned before, the RNs located in opposite sites, could re-use the same radio resources, and so in the Figure 4-65, RN3 is using the same resources of RN1, and RN4 the same resources of RN2. In the same way, the mode A1 used by both, AP and RNs, for serving at UTs should be shared by means of different frequency groups, or like in this particular example by means of different code group.

Concerning the frame structure, one important consideration to take into account is that the length of the frame should be equal in both modes and start in the same time, although of course the frame's length could be variable.

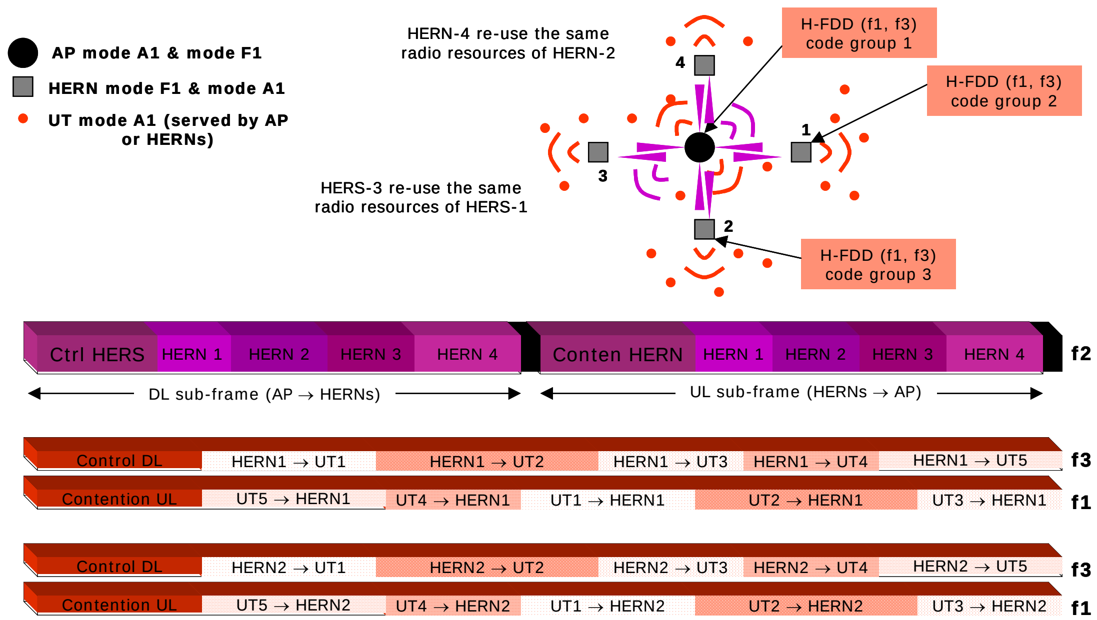


Figure 4-65 Frame structure for the example of rural environment with AP and RN serving in mode A1

With regard to the second type of scenario for the case of RN using different duplexing methods and the AP using mode A2, Figure 4-66 illustrates a deployment example for a mixed environment urban/hotspot – typical urban. The only difference with the previous example where the AP was using mode A1 for giving service at UTs, is that in this occasion the bandwidth of the TDD channel has to be shared between the AP-RNs links and the communications with UTs of the own AP.

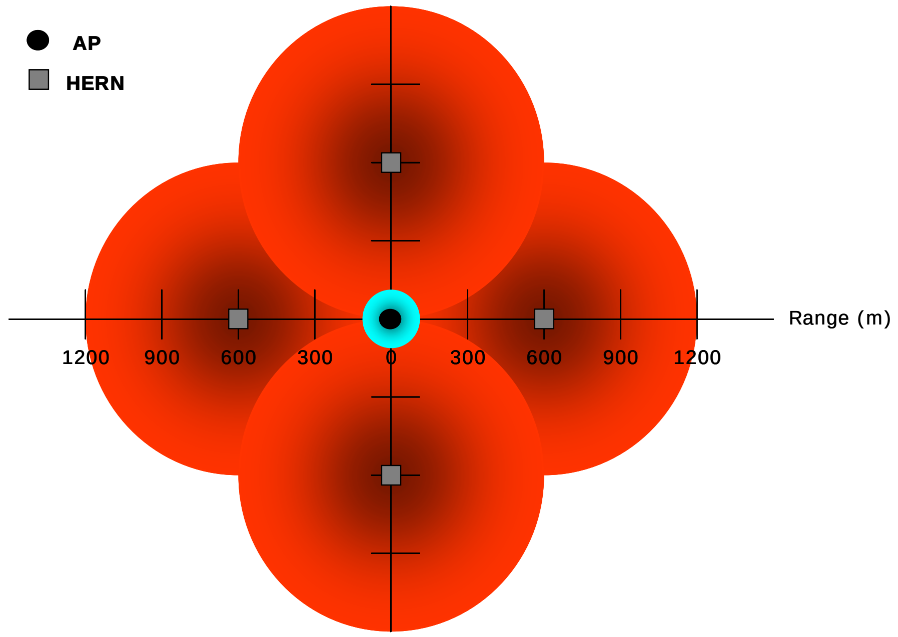


Figure 4-66 Example of mixed environment urban/hotspot – typical urban, for AP serving in mode A2 and RN serving in mode A1

In the case of an RN using different duplex schemes for its connections, that is, the feeder links mode for the AP-RN link (TDD) and wide area cellular mode for serving to UTs in its coverage area (FDD), at least in a first approach, the most important aspects to consider from an interference point of view, are on the one hand the separation between the two bands (TDD and FDD) used by the involved modes in the RN, and on the other hand the distance between the antennas used by each of the modes in order to avoid coupling effects each other. Due to the dimension requirements and the physical constraints imposed by urban and office installation locations of the RN, having to be installed sometimes in lamp posts, it will be necessary to get the required isolation with a much closer spacing between antennas, and then must be separated by a substantial distance to avoid cross-coupling. In order to prevent possible blocks in the reception (RX) of some of the modes used in the RN, provoke by the output power of the transmission (TX) of the other mode, it would be suitable ensure isolation levels of around 80 dBm between the reception and transmission antennas of the modes involved in the RN.

4.7.3.4 Common Considerations for previous Deployment Examples

Although in the previous cases, we have generally used the time as parameter for distributing the total radio resources between the UTs and RNs, of course there are more possibilities like for example in a system based on OFDM-CDMA, the use of sub-carriers or codes group, or even a combination of these three parameters (time, frequency and code), for allocating the available resources. It is clear that every solution has its own advantages and disadvantages, which should be analyzed in terms of spectral efficiency. For instance, in an OFDM system the radio resources for the downlink could be distributed between all the UTs and RNs connected to the AP, splitting in a continuous way the total useful carriers in different sub-carriers groups. However, for the uplink it would be necessary to leave a certain number of sub-carriers as guard band between every communication, in order to avoid overlapping in the reception. In any case, there will be to do evaluations and simulations for each alternative and under different conditions, in order to determine the best way to distribute the radio resources.

The big handicap to solve in any kind of deployment based on relaying, is the control of the delay produced in the multi-hop communication, since in downlink direction the data transmit from the AP to the RN in mode F1, has to be translated to the another mode, A1 or A2. Likewise in uplink direction, the data transmit from the UTs to the RN in mode A1 or A2, has to be translated to the mode F1. Hence the data received in mode F1 by the RN from the AP in the frame n, will be forwarded in mode A1 or A2 to the respective UT in the following frame, as well the data received in mode A1 or A2 by the RN from the UT in the frame n, will be forwarded in mode F1 to the AP in the following frame. Anyway, the delay in a RN working in F1 (TDD) and A1 (FDD) compared to a pure TDD case, could be actually shortened as it should be possible to perform that translation very fast due to the fact that the RN may transmit and receive at the same time.

On the other hand, there will be to analyze the operation of RN in extreme cases, for example what should be the behaviour of RN when there is not traffic in the mode used by the RN for serving final users. In this situation one possibility would be to transmit long preamble and control frame in DL subframe, and contention slot (both for initial ranging and BW requests, or would be enough to transmit only the contention slot period for initial ranging) in UL subframe, whereas the rest of time not power transmission is produced till the next frame. Other aspects to analyze and discuss could be:

- The MAC frame duration: fixed or variable. Minimum and maximum lengths for frames.
- Lengths of sub-frames DL and UL in TDD channels, as well as frames DL and UL in FDD channels (maximum and minimum).
- Chunks and bins design.
- Primitive parameters.
- Gaps and guard intervals.
- Interference analysis.

4.7.3.5 Particular Case of Moving Networks (Train Scenario)

The problem for providing Internet services inside vehicles at high speed could be divided in two parts; the external segment and the internal segment. The external segment deals with the difficulties for vehicles in motion at very high speeds (higher than 200 km/h), to receive any radio-frequency signal. On the other hand the internal segment deals with the problem of the distribution inside the wagons of the train, in order to provide access to the passengers.

During the last years, some initiatives have appeared in order to supply Internet services in mobile vehicles at high speeds. Most of these initiatives, according to [119], are based on the use of 802.11 with its different versions for the communication inside the vehicle (internal segment), and the use of some cellular technology combined with satellite system for the external connection of the vehicle (external segment). Furthermore, to maintain the access and service inside the tunnels along the train path, currently some solution based on radiating cable (leaky feeders or slotted cables) is usually adopted, although the problem at present is the high cost of the deployment of this technology. It should be noticed that one of the most important advantages of this technology is the absence of Doppler Effect in the area covered by the radiating cable, and so it is very convenient for the coverage of high velocity vehicles inside tunnels. Some important works about this technology can be found in the references [120], [121] and [122].

From a WINNER point of view, the problem of the train scenario, at least at first look, could be solved by means of relay nodes installed on the top of the train. Initially the deployment for solving the external segment, could vary from one per train to one per wagon, changing the distribution inside the train in terms of the selected option. Of course, it seems more appropriate to use only one RN for the external connection of the train (for example installed in the central wagon), and then to perform the distribution inside the train to the final users, by means of other simpler and cheaper RNs.

On the other hand, it is clear that the handicap of the train scenario, in addition to the common problems of any radio communication (i.e. interferences, attenuation, fading and coverage), is the negative effect of the high velocity of the train. As a result of this velocity, two effects should take into account. Those are the Doppler Effect and the handover time between cells.

The handover problem is mainly related to the required time for connecting with the next cell, since depending on the complexity of the handover algorithms, could happen that the time inverted in this operation provokes the breaking of the link between the train and the new cell, giving rise to a reduction in the quality of the service inside the train. So, in order to minimize the handover times, those algorithms should be optimised, making use for example of the fact that the train goes always along the same sequence of cells, which will be fixed by the path of the train. In the same way, regarding the efficient and optimised development of the power control algorithms for this particular case, we should make use the information about the exact distance between the AP, installed on land, and the RN, installed on the train , since the train enters in a cell till it leaves the same cell. Obviously, the distribution of the APs along the path of the train, is other important deployment issue that we should consider in the development of the train scenario, and so the WINNER air interface mode selected, for example, in the external segment, should take into account the possibility to handle a higher power, in order to increase the distance between APs and then decrease the deployment costs. Besides we could increase this distance if we utilize very high directive and smart antennas with line of sight conditions. Another deployment proposal specifically targeted for this scenario could be also to place the APs right next to the track with a fixed beam pointing in the direction of the track.

Regarding the negative effects caused by Doppler Effect, hereinafter we include the main comments and conclusions deduced from the analysis performed in [123], about the radio aspects concerning the suitable frequency ranges for systems beyond IMT-2000. In this work, the requirements to future spectrum are discussed, and the maximal frame duration time (t_{frame}) in terms of the coherence time (t_c), according to the below equation, is calculated for different velocities and frequencies. Also there, t_{frame} is considered as a time between two channel estimation points, and it is assumed that available frame duration is 10% of t_c. Taking into account these assumptions, the t_{frame} obtained in this analysis for a speed of 250 Km/h was of 0.11 ms for a carrier frequency of 2 GHz, and 0.036 ms for 6 GHz. Also assuming that the frame overhead is constant, the number of frames per second and hence the total overhead per second, enlarges according as the frequency and velocity increase. In this way, the available integrated payload per second at 250 Km/h speed remains 91% at 4 GHz and only 82% at 6 GHz carrier frequency compared to integrated payload per second at 2 GHz and the same speed. Summarizing, the calculations of this work illustrated that with increasing carrier frequency and velocity the transmission efficiency decreased significantly.

$$t_{frame} << t_c = \frac{1}{v} \cdot \frac{\lambda}{2} = \frac{1}{2 \cdot v} \cdot \frac{c_0}{f} \tag{12}$$

Therefore for the selection of the WINNER air interface mode used in the external segment in train scenario, it is very important we take into account the characteristics of the mode, in order to fit adequately with the requirements from a handover and Doppler Effect point of view. For instance, it could be proposed something similar to soft handover technique, wherein the train during the handover cell, remains a certain time under the two cells, provoking a slight traffic load, but decreasing considerably the handover time.

From the four basic WINNER air interface modes identified in WP2, and in a first approach, we could choose, the “Wide area – Feeder link” mode for the external segment (link between the APs uniformly distributed along the train path and RN inside train), and the “Short range – Cellular” mode for the internal segment (UTs using this mode or another mode thanks to heterogeneous relaying inside the train). Anyway we have to take into account the high mobility support that the used mode for the AP-RN link should require, and then should be contemplated this particular support in the design of the selected mode for the AP-RN link. Besides the mobility support, other important aspect to consider in the design of the air interface mode for using in the external segment, would be the maximal reachable ranges which should be around several Kms, in order to minimize the deployment costs. In respect of these pre-requisites, it should be noted that the basic mode before mentioned ("Wide area – Feeder link"), so far does not contemplate neither high mobility support and large ranges. In fact this basic mode, at least in the initial proposal from WP2, is thought for stationary connections and cell ranges of 1 Km (insufficient to optimize the deployment costs in the train scenario). Therefore, one possible solution for the train scenario, could be the inclusion of a particular mode based on "Wide area – Feeder link" but contemplating a high mobility support and higher ranges. In other words, we will most likely to define more modes on top of the physical layer modes.

In this sense, Figure 4-67 illustrates a deployment example for the train scenario case, where the mode used for the external segment, referred to as F1', is similar to the "Wide area – Feeder link" mode from WP2, but including mobility support and higher ranges (around 5 Kms) than that mode. The modes included in this example, and used for serving final users, referred to as A1 and A2, could be the "Wide area – Cellular" and "Short range – Cellular" modes respectively (from WP2 proposal). Of course, the APs uniformly distributed along the train, would include the mode A1 (H-FDD duplex scheme) for serving final users working in this mode and camped in the coverage area of the corresponding AP (even like Figure 4-67 shows, UTs inside the train and working in this mode, could be served directly by the AP), as well as the mode F1' exclusively used for feeding the RN on the top of the train. On the other side the RN could implement modes A1 and A2 for serving UTs inside the train.

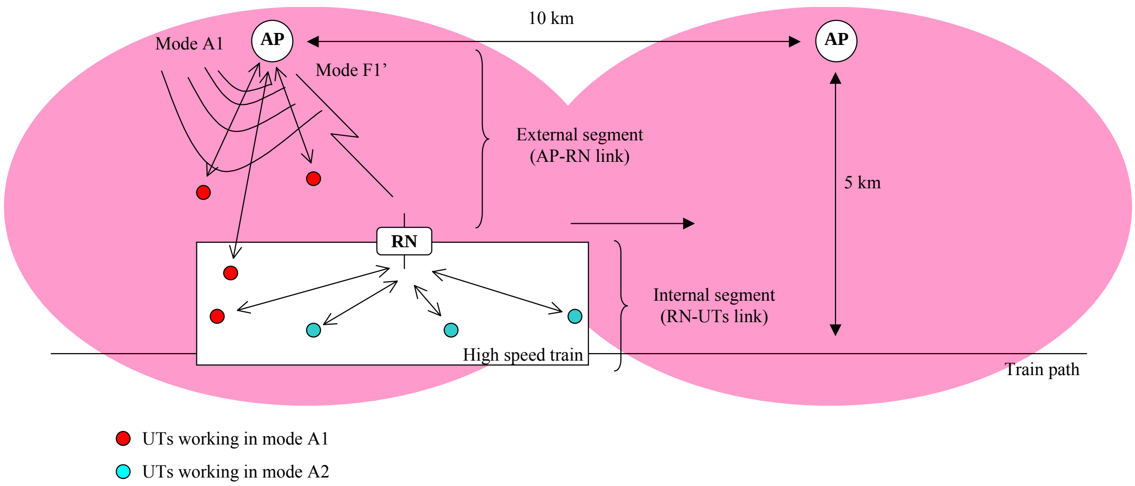


Figure 4-67 Deployment example for the train scenario

Finally and as conclusion we can say that the design of the mode used for feeding the RN over the train (external segment), should take into account the following requirements and considerations, which we will have to discuss also with persons active within WP2:

- High mobility support with fast handover algorithms based on the known train trajectory.
- Optimised power control algorithms based on the known distance of AP-RN link along the train path.
- In order to minimize the negative effect of Doppler Effect, the maximal frame duration time, considered this as the time between two channel estimation points, should be much shorter than the coherence time, which depends on the train speed and the frequency used for AP-RN link.
- The range and the data rate of this mode must adapt to a feasible deployment cost of the train scenario, trying to reach ranges around 5 Km and peak data rates at least of 0.5 Mbps per MHz. These ranges could be obtained using high directive antennas and handling larger radio power, although it must consider the impact over the rolling infrastructure, in particular the aero dynamical problems derived of the antennas location.

4.7.4 Conclusions

The previous deployment examples based on the use of the basic modes initially identified by WP2, have shown the priority to refine those modes in order to be able to chose the best deployment option for each scenario. So, it is clear the necessity to increase the contribution from WP3 in the WP2 activities for defining and clarifying the different modes of the new radio interface. In the same way, we propose to follow up the achievements of WP2, as well as to collaborate in the decisions of this work package about the characteristics of WINNER air interface modes. Somehow, we must influence in the modes design, taking into account the possible implications and repercussions of those modes for the deployment and design of the RNs, of course from different point of views such economical, complexity, interference, spectral efficiency and so on. Besides, we have identified several scenarios such the wireless feeder concept and the train scenario, with some particular requirements, which we will have to discuss with WP2 people in order to define more modes of the initially identified.

It is also proposed for the near future, to use the methodology developed by AU (Weighted Spectral Efficiency) for comparing the deployment concepts presented in this section with other alternatives (i.e. single hop or multi-hop using homogeneous RNs). Therefore, we will have to define particular scenarios with concrete values of some parameters such cost of APs and RN working in different modes, coverage areas, and environment.

Finally, further development of the previous deployment examples would be needed, focussing the study on the problems with relaying for each of the scenarios. Even, we could add another dimension in the two main deployment concepts currently contemplated (RNs using the same and different duplex schemes), as is the overlapping and non-overlapping coverage areas of APs and RNs involved in the deployment examples introduced in this section.

4.8 Conclusions and Future work

In this chapter further investigations on multi-hop deployment concepts within the homogeneous and heterogeneous relays context has been documented and the results of the analyses carried out up to now have been reported.

The combination of the protocol elements of PMP and Mesh modes in multi-hop topologies appears as a very promising approach. Moreover, the presented comparison of a multi-hop network approach against the single-hop one, leans towards using relaying as an essential component for WINNER air interface.

Crucial and challenging mechanisms such as time division based relaying, scheduling, routing and forwarding in multi-hop cellular networks are described and partially assessed. Further, performances of effective routing approaches and the impact of smart antennas on QoS routing for multi-hop wireless networks are studied.

Even if a number of issues still need to be studied into more detail in the future, results show how the system could gain from the presence of the mobile relays since the latter can offer quite substantial gain compared to the direct path from the AP.

Finally, a relevant part of this chapter consists in the investigation of many issues concerning the cooperative relaying deployment concept. Several characteristics are detailed and it is proposed that adaptive decode-and-forward protocols are investigated in more detail to identify white spots as well as to pursue their integration into the WINNER system concept.

The analysis of the deployment examples based on the use of different types of heterogeneous relays, and taking the initial proposal of WINNER air interface modes into account, has shown on the one hand the potential benefits, which may be achieved implementing a mode specifically thought for feeding the RNs from the AP, and on the other hand the necessity to participate tighter in the development of the radio interface, for including the requirements identified during this preliminary analysis, in the design of the different modes for the new air interface.

The aim of the rest following part of the project is to pursue with the analysis of the multi-hop systems since even if some interesting results have been here presented, a number of issues need to be detailed in order to prove the effectiveness of such systems. More precisely, the next deliverables are supposed to shed light on the performance that might be attained with this kind of approach.

5. Advanced single-hop deployment concepts

5.1 Introduction

In this chapter, we present advanced single-hop deployment concepts for cellular wide-area, metropolitan-area and short-range cellular deployment. These promising techniques should be considered as elements in the WINNER system design. First, we present an advanced frequency reuse scheme in cellular adaptive TDMA/OFMDA systems, then we continue with describing proposed enhancements to the 802.16a system (adaptive coding and modulation, SDMA and digital pre-coding), UMTS HSDPA features (AMCS, F-ARQ, FCSS, STTD as well as the combination of some of these features) and finally we present an overview of some studied enhancements for IEEE 802.11n related to multiple antennas and the MAC protocol (SDM and Aggregated-MSDU).

5.2 Advanced frequency reuse in cellular adaptive TDMA/OFMDA systems

5.2.1 Introduction

How can we in a wide-area cellular adaptive TDMA/OFDMA downlink as proposed in [96] chapter 6.1, simultaneously obtain good coverage and a high average spectral efficiency? This chapter outlines a solution to the problem. The proposed method relies on two mechanisms:

- coordinated scheduling among sectors on the same site (AP) to suppress interference and
- partial frequency reuse. One frequency band (with reuse 1) transmits to near users, while another band, with orthogonal resource sharing among clusters of 3 access points, transmits to far-off users in the sectors.

The key idea is to apply a frequency reuse factor >1 only where it is needed most; in the outer part of the sector, where the signal from the serving access point is weak and the interfering access points are relatively strong.

With the above mechanisms, it becomes possible to attain coverage and high average spectral efficiency over the whole sector, without assuming multi-user diversity or multiple receiver antennas. The resulting signal-to-interference ratio (SIR) and spectral efficiency have been evaluated as a function of the position within the sector, by summing over all relevant interferers. The traffic density is assumed to be constant over the area, and hexagonal coverage areas are assumed for the access points. Both triangular sectors and diamond-shaped (30 degree rotated) sectors are considered. Antenna pattern and an exponential path loss are taken into account. The spectral efficiency is calculated for one user per sector (no multi-user diversity) with adaptive modulation and static channels including path loss, and then results are briefly given for flat Rayleigh-fading channels. The estimates neglect shadow fading and noise (i.e. assuming an interference limited system), and represent a first approximation of the more complicated realistic situation. The presented techniques are briefly given in [141] and additional details are available in [108]. In the conclusions we discuss various ways of improving the spectral efficiency even further.

5.2.2 Description of the frequency reuse strategies

We assume AP at fixed locations that all utilize the same spectral band of width s_0. To estimate the spectral efficiency, a deterministic approach with a continuous user distribution is used. A basic simplifying assumption is that *the number of active users per unit area is constant over the considered area*. It is then reasonable to assume that the locations (sites) of the access points are regularly spaced. A conventional hexagonal pattern is used, see Figure 5-1.

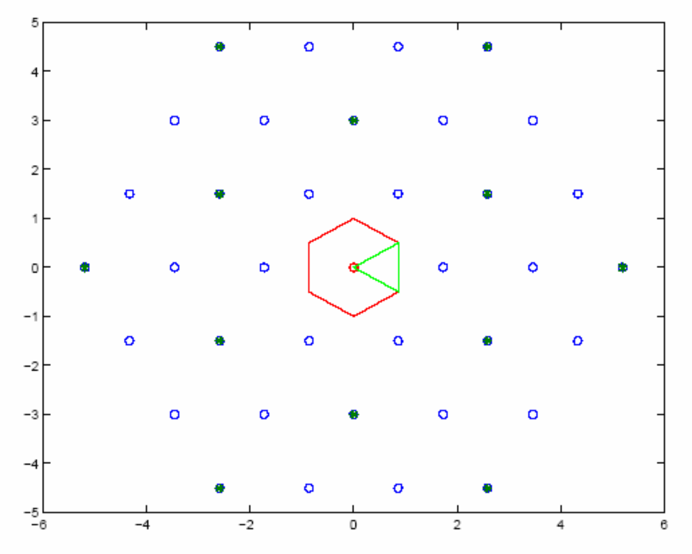


Figure 5-1: The hexagonal site pattern. Access points which have 6 sectors/lobes are located at the indicated positions. There are 37 access points, having 222 sectors (or cells). "Our" access point, with the red hexagonal coverage area, is at the origin. The sector of interest is in this case triangular and directed toward the positive *x*-axis. Rings represent all access points that are taken into account in the analysis of the interference. Noise and interference from outside of this area is neglected. When transmitting to the outer parts of sectors, a reuse 3 pattern of groups of 3 sites is used. The sites that transmit simultaneously with our site, and thus cause interference, are indicated by crossed rings.

The coverage area of a access point is defined as the area where the signal is strongest from that particular station. The resulting hexagonal coverage areas are partitioned into N_s sectors, also called lobes or cells. We may in general assume the beam-widths and lobe directions to be adjusted slowly, to adapt to traffic variations within the coverage area. An equal number of active users within each lobe could be a reasonable criterion for the adaptation.

With the above assumption of constant traffic density, all lobes will then on average have equal width $360/N_s$ degrees. We here assume N_s=6 sectors of equal fixed width 60 degrees and side length R at all sites. The SIR in one of these sectors (the triangle pointing to the right from the origin in Figure 5-1) will be investigated in detail, by taking interference from the 36 first, second and third-tier interferers (within radius 5.5R) into account. More distant interferers and noise will be neglected in the analysis.

We will consider the triangular sector shown by Figure 5-1 and the left part of Figure 5-2. Note that the closest interferer (at x=1.73R) will then also have a triangular sector, beaming along the negative x-axis into our sector. We also consider a diamond shaped sector, obtained by shifting the sector by 30 degrees within the hexagon, see the right part of Figure 5-2.

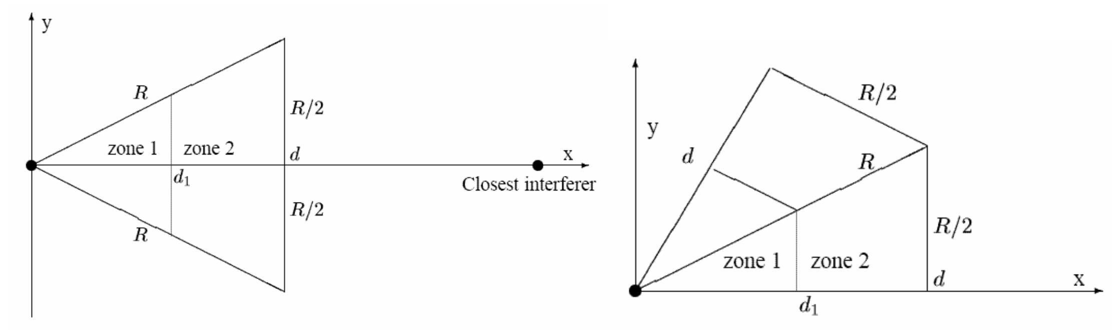


Figure 5-2: Enlargements of the two types of 60 degree sectors considered. A triangular sector is shown in the left figure, and a diamond-shaped sector, with centre direction +30 degrees, is shown in the right figure. We use frequency reuse 1 of the frequency band s_1 in zone 1, while the frequency band s_2 is allocated to the outer zone 2, where it is shared over 3-site groups in a reuse 3 pattern

Interference is reduced in this system by two strategies:

- to reduce interference from adjacent sectors of "our" site, the site scheduler will avoid using time-frequency bins that are allocated to users in our sector, which are close to the site boundary (within an angle β from the boundary). Likewise, our sector will not transmit during bins that are allocated to such border users in neighbouring sectors.
- When transmitting to distant users within the sector (beyond distance in what is denoted zone 2 of the sector, see Figure 5-2), we use a separate part of the total frequency band s_0 that will be denoted s_2. In zone 2, we use reuse partitioning 3, i.e. sites are grouped into clusters where each site is given exclusive access to 1/3 of the resource. The reuse pattern is synchronized over all sites, as in present FDMA/TDMA systems. This strategy radically improves the SIR within zone 2. It also helps to equalize the perceived SIR within the sector, and increase the modulation levels that can be used when transmitting to distant users, close to the outer cell boundary. The switching distance $d_1 <= d$ is the main system parameter. For the extreme values $d_1=0$, we have a frequency reuse 3 system, and for $d_1=1$ the system have frequency reuse 1.

The scheduling between sectors is discussed in more detail in chapter 5.2.3, the zone resource sharing is discussed in more depth in chapter 5.2.4 and the SIR due to path loss and inter-cell interference is derived in chapter 5.2.5.

5.2.3 Coordination of Sector Scheduling

We assume that the azimuth antenna radiation power pattern of the 60 degree sector antenna consists of a scaled sinc function pattern, with side lobes damped by more than 30 dB. The transmit power is normalized to 0 dB at the sector centre and it has been scaled to decrease to -8.1 dB at 30 degree. The 3 dB beam-width is 39 degrees. The lobe creates significant interference in the neighbouring sector, but the power decreases rapidly with increasing angle.

To boost the power close to the cell boundary and reduce the interference in neighbouring sectors, the following strategies are used:

- when transmitting to a border user within β degrees from the cell boundary, the transmit power of that time-frequency bin is boosted by a factor 2 (3 dB). This reduces the variation of the received power as a function of the angle.
- When transmitting to border users, the neighbouring sector will not be allowed by the site scheduler to use the same time-frequency bin. This strategy is more efficient than a soft-handover strategy, in which the same information is transmitted in both sectors and in the same bin with the normal power level.

We thus completely eliminate interference where it is most severe, i.e. close to the sector boundary.

The proposed coordinated adjacent sector scheduling corresponds to a frequency reuse factor 2 for the fraction of resources utilized by border users. The angle β is a system parameter which balances the capacity loss due to exclusive transmission against the remaining interference level generated in the neighbouring sector. It should be straightforward to optimize the angle β for a given antenna pattern. The resulting capacity reduction due to the scheduling constraint is

$$c_1 = 1 - \frac{1}{2}\frac{\beta}{30}. \tag{13}$$

Assuming β=7.5 degrees, which corresponds to 25% of the area of the 60 degree sector, we get a capacity reduction c_1=7/8.

When calculating the interference from the neighbouring sites, the angle with respect to the cells of the interferer is calculated and the angle-dependent total overlapping radiation pattern of the antenna is taken into account. The assumed 3 dB power boost during β/30=25% of the time (for fully loaded cells) is cancelled by the fact that the neighbouring cell is prevented from transmitting. It is therefore neglected in the calculation of the interference power.

5.2.4 Resource Sharing Within Zone 2

When transmitting to users in the outer part of the sector (zone 2), we assume the same constant transmit power as for the inner zone, except for border users where the power is increased by a factor 2, as described above. In zone 2, we use the part s_2 of the total allocated spectral bandwidth s_0, while the remaining part s_1 is allocated to the inner zone 1 of the sector. All the sectors of "our" access point are simultaneously allowed exclusive use of 1/3 of the resource s_2, on average over time. The resource is shared among clusters of three sites, in a classical reuse 3-pattern. This orthogonal resource sharing can be realized by spectral partitioning, time division or a combination of both. The spectrum resource s_1 is used with reuse 1 in zone 1.

Assuming that the same strategy is used at all sites and that the spectral bandwidth s_2 allocated to the distant users is the same at all sites, we may calculate the resulting capacity reduction due to reuse and the corresponding reduction of the interference level in a straightforward way [108]

$$c_2 = \frac{s_1 + s_2/3}{s_0} = \left(\frac{d_1}{d}\right)^2 + \frac{1}{3}\left(1 - \left(\frac{d_1}{d}\right)^2\right), \tag{14}$$

where we have assumed a partitioning of the total spectrum s_0 in proportion to the areas of the two zones (motivated by constant user density). The total equivalent frequency reuse factor becomes $1/(c_1c_2)$. For example, d_1=0.71d gives a frequency reuse factor 1.71.

5.2.5 The SIR due to Path Loss and Inter-cell Interference

We calculate the interference generated by all access points of significance. All interferers within a distance 5.5R from some part of the sector of interest are taken into account. Short-term frequency selective fading and log-normal shadow fading are neglected. Only the path loss is taken into account, and it is assumed to follow a simple power law, the Okamura-Hata model

$$P_R = \frac{g}{r^\alpha} P_T, \tag{15}$$

where P_R is the received power, g is the channel gain, r is the distance from the access point, α is the propagation exponent and P_T is the transmitted power.

The *SIR* encountered by a user located at (x, y) in the inner zone 1 of our sector of interest becomes

$$SIR = \frac{\frac{g}{r^\alpha} P(\theta)}{l\left(\sum_i \frac{g}{r_i^\alpha} P(\varphi) + g I_s(\theta)\right)}, \tag{16}$$

where the summation is taking into account all interferers from the first, second and third-tier interferers (within radius 5.5R), as shown in Figure 5-1. Here,

$r = \sqrt{x^2 + y^2}$ is the distance from the access point to the user

$\theta = \arctan(x/y)$ is the angle to the user, relative to the sector symmetry line.

$P(\theta)$ is the angle-dependent antenna pattern (boosted by factor 2 in the coordinated sector scheduling area)

l is the traffic load factor in interfering cells.

$P(\varphi)$ is the transmit power of the interfering antenna at angle φ.

$r_i = \sqrt{(x - x_i)^2 + (y - y_i)^2}$ is the distance to interferer i located at (x_i, y_i)

$I_s(\theta)$ is the remaining power due to interference from adjacent sectors at the site (from outside of the coordinated sector scheduling area i.e. for -30 degree + $\beta < \theta <$ 30 degree – β).

In the outer zone 2 of the sector, the expression for the SIR is similar to the equation for zone 1, except that we only have to sum over the active cell sites within the neighbouring 3-site clusters. As indicated by Figure 5-1, there are 12 such sites within the distance 5.5R from our sector, denoted by stars within the rings. We have six interferers at distance ≈3R and six at distance ≈5R.

Figure 5-3 (left) shows the *SIR* calculated by (11) along the centre (left) and edge (right) of the triangular sector. The corresponding equivalent reuse factor $1/(c_1c_2)$ is 2.29, 1.71 and 1.31 respectively.

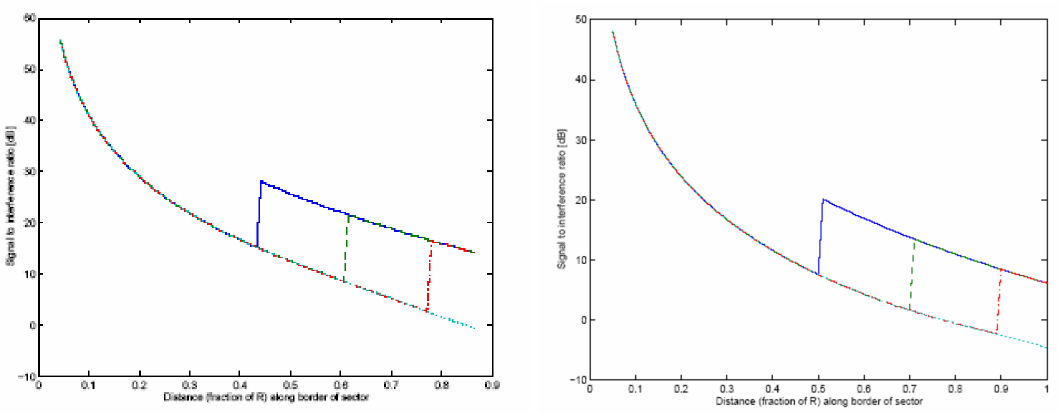


Figure 5-3: The *SIR* in dB along the *centre (left)* and along the *edge (left*) of the triangular sector, as a function of the distance from the access point, measured in units of the triangle side length *R*. The border between zone 1 and zone 2, d_I, 0.50*d* (solid), 0.71*d* (dashed) and 0.90*d* (dash-dotted). The lower outer dotted curve corresponds to d_I=*d*, i.e. frequency reuse one out to the border of the sector. The traffic load is *l*=1 and the propagation exponent *α*=4.

Figure 5-4 shows contour plots of the *SIR* distribution within the sectors for d_I=0.60*d*=0.52*R* (upper and lower left), d_I=0.70*d*=0.61*R* (upper right) and d_I=0.90*d*=0.78*R* (lower right).

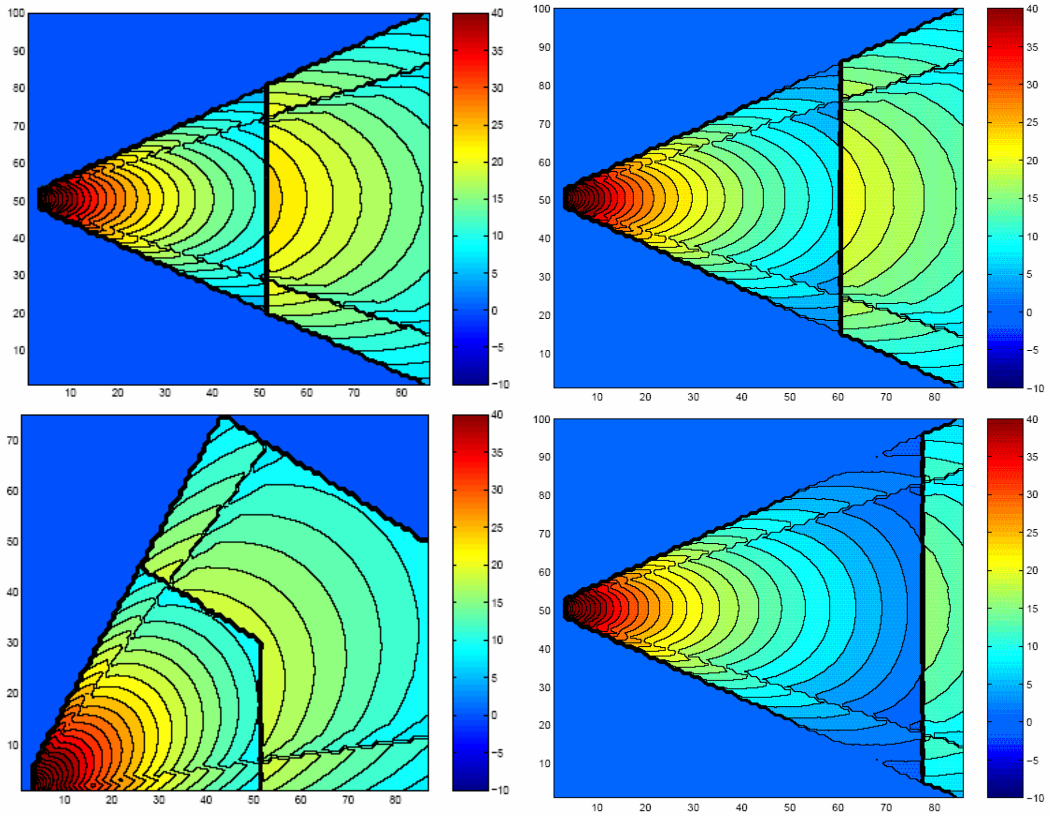


Figure 5-4: Distribution of the *SIR* in dB for the triangular and diamond-shaped sector with d_I=0.60*d*=0.52*R* (upper and lower left), d_I=0.70*d*=0.61*R* (upper right) and d_I=0.90*d*=0.78*R* (lower right). The traffic load factor is *l*=1 and the propagation exponent *α*=4.

5.2.6 Estimate of the Spectral Efficiency

An estimate can be obtained for the spectral efficiency in bit/s/Hz as a function of the position within the sector based on SIR, when the system is interference limited. We estimate the raw spectral efficiency (the number of bits per symbol) under the following assumptions (additional details can be found in [108]):

- Uncoded adaptive modulation using 8 different modulation levels BPSK, 4QAM, 8QAMcross, 16QAM, 32QAMcross, 64QAM, 128QAMcross and 256QAM.

- The user is alone in the sector, and is not moving, with a constant channel gain that is independent of frequency. Thus, multi-user diversity effects is not taken into account, we do not utilize the variability of the channel in time and frequency. In the real system, utilizing these effects will improve the spectral efficiency.
- The path loss is assumed to follow the simple Okamura-Hata model (defined in chapter 5.2.5), but short-term frequency selective fading, and log-normal shadow fading are neglected.
- The number of bits per symbols obtained in the outer zone 2 has to be multiplied by 1/3 to obtain the spectral efficiency for the frequency band s_2.
- In both zones, we have to multiply by the factor c_I=7/8 in order to take the capacity loss due to scheduling between sectors into account.

Figure 5-5 shows the calculated spectral efficiency within the triangular sector for the two values d_I=0.50d=0.433R and d_I=0.71d=0.61R.

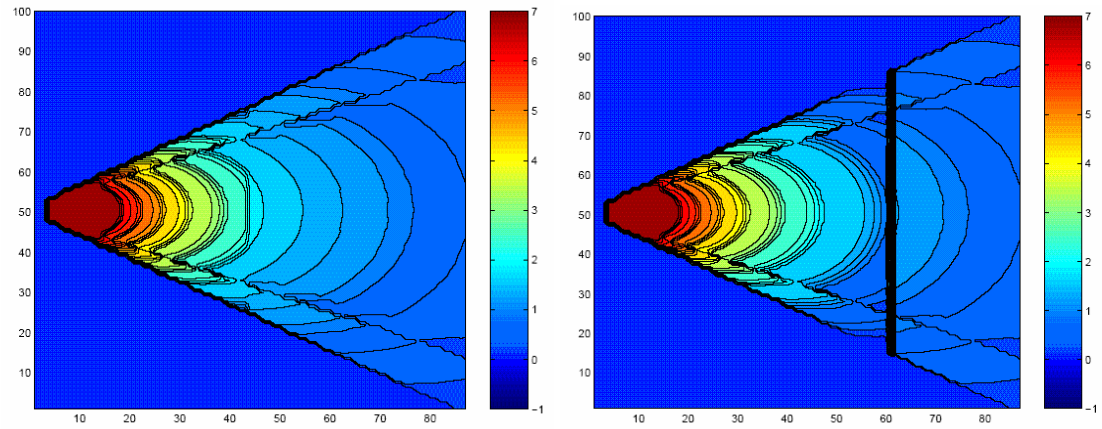


Figure 5-5: Raw spectral efficiency in bit/s/Hz as a function of the position within the triangular sector for d_I=0.50d=0.433R (left) and d_I=0.71d=0.61R (right). The traffic load factor l=1 and path loss attenuation factor α=4.

Note that above we have not optimized the zone border defined by d_I. In [108] we optimize d_I based on the average spectral efficiency. A good trade-off between coverage (based on "goodput", i.e. number of bits per symbol times the probability of correctly received link level packet [108]) and average spectral efficiency seems to be when d_I is chosen to be in the interval 0.6d-0.7d.

Some conclusions can be made:

- Both the overall spectral efficiency and the coverage become somewhat better for diamond-shaped sectors as compared to triangular sectors.
- The spectral efficiency within the inner zone 1 is higher for the diamond-shaped sector. This is because zone 1 is affected by nearby interferers and the diamond geometry shifts the angle of the sectors so that no nearby sectors point directly towards each other. This reduces the interference, due to the assumed antenna pattern. The interference power is reduced by around 5 dB at 30 degree angle compared to a situation where the centre of a sector of an interfering cell points directly towards us.
- On the other hand, the spectral efficiency within the outer zone 2 is worse for the diamond-shaped cell than for the triangular cell. The reason is that one of the nearest of the 12 interfering cells that transmit in s_2 will point directly toward the diamond-shaped sector with direction +30 degrees.

To summarize, diamond-shaped 60 degree sectors in hexagonal coverage areas provide a somewhat higher spectral efficiency than triangular sectors, at the price of reducing the already low capacity offered within zone 2. The triangular sector might be preferred for that reason. With adaptive access point antennas, one could transmit in diamond-shaped sectors (rotated by 30 degrees) to nearby users at the frequencies s_1, while triangular sectors are used at frequencies s_2 for contacting the far users. The result would be a pattern that inherits the strengths of both designs, but not their weaknesses.

In [108], results for one Rayleigh fading user with two antennas and considerations for a not fully loaded system are analysed, and the conclusions are:

- the spectral efficiency for a Rayleigh fading channel becomes somewhat lower than for a static channel with the same SIR. For a fully loaded system (l=1) the optimum of the limit d_I changes little with respect to the static case.

- An assumption of full load, is rather extreme, and is appropriate only for estimating the ultimate total system capacity. In a more realistic case, e.g. where the load factor is l=0.25 and each terminal has two antennas using maximum ratio combining (MRC), the optimal zone boundary is shifted markedly outward, to around d_l=0.9d.

Thus, it is likely that the inner zone 1 should comprise the major area of the sectors in a realistic interference scenario, and the spectral reuse factor becomes close to one.

5.2.7 Conclusions and future work

The scheme for wide-area cellular interference avoidance proposed here assures that the whole sector, not only the area closest to the access point, can be covered without a large loss in spectral efficiency due to the frequency reuse scheme. The key idea is to apply a frequency reuse factor >1 only where it is needed most, in the outer parts and along edges of the sector, where the signal from the access point is weak and the interference is strong.

The discussion above has been simplified and given in a quite general context. Thus, further work should be made:

- Shadow fading is not included in the model. In practice, the resource sharing defined by the parameters β and d_l should be dynamically configured, based not only on the position within the cell, but also taking into account measurements of the actual interference situation.
- Analyses and simulations adapted to user scenarios, traffic models, channel models and antenna patterns within WINNER.
- Assumed system load for the design is important. A fully loaded system is not likely, so a system target load in the WINNER context should be assumed.
- Multiple antennas are important enabling technology for the WINNER system and should be incorporated in the design and analyses.

In addition, there are other promising candidate techniques, which should be studied to further improve the spectral efficiency:

- Apply the proposed coordinated scheduling not only to the sectors within each site, but also to the strongest interferers in other sites, e.g. by coordinating all sectors of a 3-site reuse pattern. However, this coordination requires inter-site coordination capabilities in the network or use of a central inter-site scheduler, e.g. by using a Radio over Fiber solution to connect co-located APs and the remote site antennas.
- When the traffic load varies between sites and sectors, adapt the resource partitioning accordingly for the zones and between the sectors. However, adaptive zone definition and sectorization will increase the interference when the partitioning is no longer the same over the whole system.
- For an interference limited system, as assumed above, use of power control would not improve the spectral efficiency, but if the reception quality is noise-limited in zone 2, we could improve the spectral efficiency by increasing the transmission energy allocated to spectrum band s_2.
- Multiple receiver antennas for interference rejection combining (IRC) should be studied. Good interference suppression by IRC should be possible with relatively few antennas in the UT if local scattering is limited. In [109]and [110] the relative strengths of interferers in downlinks in hexagonal and street-canyon environments were investigated, including reuse 3 patterns. In almost all cases two dominating interferers contributed to the major part of the interference energy.

5.3 Further possible enhancements to IEEE 802.16a

In this section, further possible enhancements to improve the 802.16a system are briefly described. The aim is to provide additional potentials of single-hop deployment in WINNER context and in particular about the IEEE 802.16 standard that defines interesting elements which should be considered when designing a new 4G air-interface.

5.3.1 Advanced IEEE 802.16a PMP MAC scheduler for adaptive array system

The purpose of this section is to analyze the impact on system performance related with the introduction of advanced features at the PHY layer. The PHY features considered will be adaptive coding and modulation, antenna array and digital precoding. The main aim addressed by the introduction of such feature is the possibility to exploit spatial division multiple access (SDMA). The expected results of this work is to increase the global system throughput and to reduce global mean delay adding only an antenna array on the AP transmitter maintaining the OFDM structure with 256 carriers and to introduce an advance scheduling algorithm to manage the adaptive antennas and the SDMA.

In order to do that, an additional functional entity, called X-layer, which will provide information exchange between MAC/PHY layers should be inserted (see Figure 5-1). The X-layer scope will be to synthesize PHY layer information and to render them available to MAC layer in order to optimize air interface usage. The X-layer module is finalized to acquire Channel state informations (CSI), calculate with them the beamforming and finally supply synthesized data to Mac-scheduler. An advanced Mac scheduler should then exploit information coming from X-layer in order to optimize the downlink data frame to be transmitted, considering the actual channel conditions, the environment topology and actual offered traffic to users.

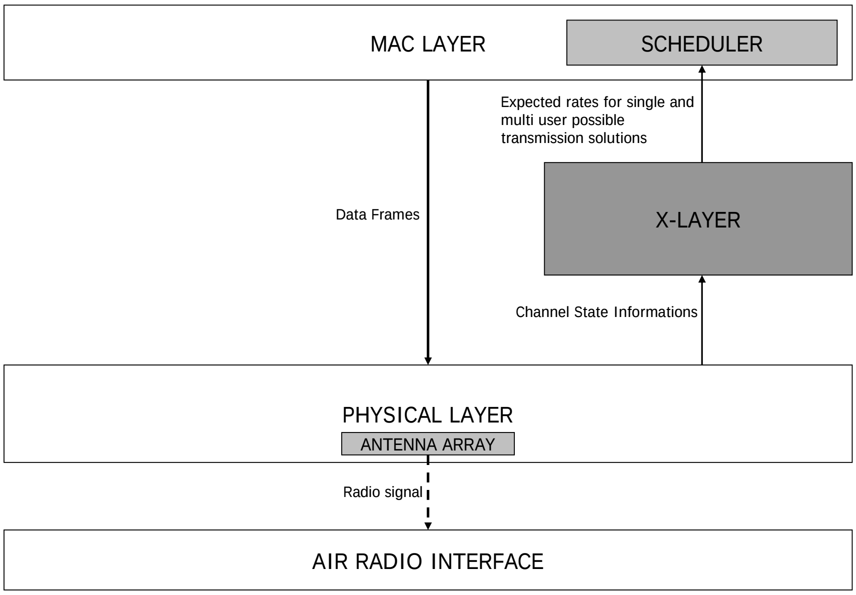


Figure 5-1 Block scheme of the new concept

The realization of SDMA brings a new challenge. Before transmission a careful analysis of environment geometrical parameters is needed in order to avoid (or reduce at least) the interference between flows transmitted to different users. The contemporary transmission to different users involves also spatial precoding that divide the different streams toward the right SS. All elaborations made by "X-layer" are synthesized into aggregated parameters that are submitted to upper layer that will create the transmission frame appropriately using the scheduler inside its structure.

Scheduler policy

The scheduling policy must manage all connected user's traffic requests considering at the same time the informations received from X-layer mudule. The choice of scheduling policy is a crucial task for wireless networks there have more bonds compared to the wired ones: the main difference is that the wireless channel is not always available to all users and it introduces typically bursty errors. The wireless scheduling policy should consider the actual channel state of different users in order to optimize resources exploitation. For example, a user, that momentarily has a bad channel realization, could be served with a robust modulation and coding scheme (with a low date rate) or alternatively could wait for a better realization of the channel in which it can transmit with a higher data rate. Beside this the X-layer provides the way to allocate contemporary transmission to more users through SDMA. Then the scheduling policy has to manage different degrees of freedom like, time, modulation and coding schemes and space.

5.3.2 Conclusion and Future work

The use of the technique explained above allows to reuse same frequency and time intervals in order to serve users separated in space so creating a growth of available bandwidth proportionally to number of spatially separated flows that we have allocated.

All concepts introduced are directed finalized to a more efficient usage of available radio resources in order that Wireless network can support all services characteristic of wired ones like Home/Business

Broadband Internet Access especially with the same QoS. Naturally this will require a larger complexity in the hardware and software architecture with related increase in development costs.

In order to evaluate the system performance of the new proposed concept event-driven simulations will be carried out.

5.3.3 How to use the new concept in the new WINNER air interface

The definition of the new WINNER radio interface is still in progress, however a first draft of the WP2 air interface concepts by Task 2.7 is now available. The new winner air interface foresees many characteristics that refers to our new concept; in particular the “Wide Area – feeder links” mode has the correct parameters of frequency carrier, duplex scheme and modulation/multiplexing method. Moreover the transport channel structure contains all the entities necessary for the complete management of new blocks introduced and the physical channel structure is very similar to the 802.16a one used in our project. Hence in conclusion the new concept could be implemented also in the new WINNER radio interface but a feedback mechanism about the knowledge of actual channel state by the MAC layer must be foreseen.

5.4 Enhanced HSDPA

The introduction of HSDPA (High Speed Downlink Packet Access) enhances beyond 3G mobile systems by offering higher data rates in busty and asymmetrical traffic profile. Future wireless cellular networks will support integrated multimedia applications with various quality of service (QoS) requirements. Providing QoS differentiation in high-speed wireless data networks is considered as a promising solution for the increasing multimedia demands from wireless end users. To support an evolution towards an enhanced network and multimedia services higher that 10Mbps, HSDPA goal is to increase user peak data rates, quality of service, as well as to improve spectral efficiency for downlink asymmetrical and bursty packet data services.

HSDPA uses a special HS-DSCH channel (High Speed Downlink Shared Channel) which is similar to the Release’99 DSCH channel (Downlink Shared Channel), but without fast power control. To increase peak data rates and cell throughput as well as reduce retransmission delay, several techniques have been proposed for HSDPA including higher order modulation and lower-redundancy coding combined with incremental redundancy. The MCS technique (modulation and coding scheme) may be changed dynamically on a per-TTI basis AMC (Adaptive Modulation and Coding). Further enhancements to reduce basic link performance requirements include layer-1-based F-HARQ (Fast Hybrid Automatic Repeat Request) and transmission/reception antenna diversity (TxD/RxD). Parallel channels are facilitated by multi-code as well as MIMO (Multiple Input Multiple Output) antenna techniques. Mobility and link performance may be improved using FCS (Fast Cell Selection/Switching) although the actual value of this feature has not been unanimously proven. AMC offers a link adaptation method that can dynamically adapt modulation-coding scheme to current channel conditions for each user. In a system with AMC, users close to the access point usually have good radio link and are typically assigned higher order modulations and higher code rates (e.g. 64 QAM with R=3/4 turbo codes). The modulation-order and/or code rate will decrease as the distance of a user from BTS increases. H-ARQ provides a retransmission mechanism for the lost or erroneous information. There are many schemes for implementing H-ARQ - Chase combining, Rate compatible Punctured Turbo codes and Incremental Redundancy. New studies are being performed how to control and management and integrate radio resource control schemes with these new technologies. The research focus on the dynamic resource allocation with QoS control in HSDPA system with AMC and H-ARQ mechanisms. New scheduling algorithms such as Delay-sensitive Dynamic Fair Queuing (DSDFQ) [112] , to meet delay requirements of multimedia applications as well as maintain high network efficiency are new further enhancements to HSDPA. The approach can easily adapt to load fluctuations of different traffic classes and varying wireless channel conditions caused by user mobility, fading and shadowing.

The performance of these individual techniques as well as their combined use must be evaluated to develop efficient RRM (radio resource management) algorithms and reduce the complexity of HSDPA to support high performance while facilitating the access point as well as user terminal implementations with reasonable complexity/cost. The performance and cost/complexity issues of further improvements will be considered within future 3GPP standardization framework to further evolve the WCDMA concept.

5.4.1 User tthroughput for different MCS

Several user throughput curves for the foreseen HSDPA MCSs and the maximum obtainable throughput curve are shown in Figure 5-2 [111]. It is interesting to note that the 8PSK 3/4 scheme is always lower than the maximum throughput curve which indicates that it will not become active in the case where link adaptation is based solely on the optimization of throughput per individual link. However, with 8PSK it is possible to alter the power instantaneously since the receiver does not depend on amplitude information. Hence, from a network perspective, it may in fact be better to use 8PSK 3/4 over the 16QAM 1/2 scheme. In terms of link performance, however, the 16QAM scheme remains superior for all scenarios including varied time dispersion and varied UT velocity (up to 250 km/h).

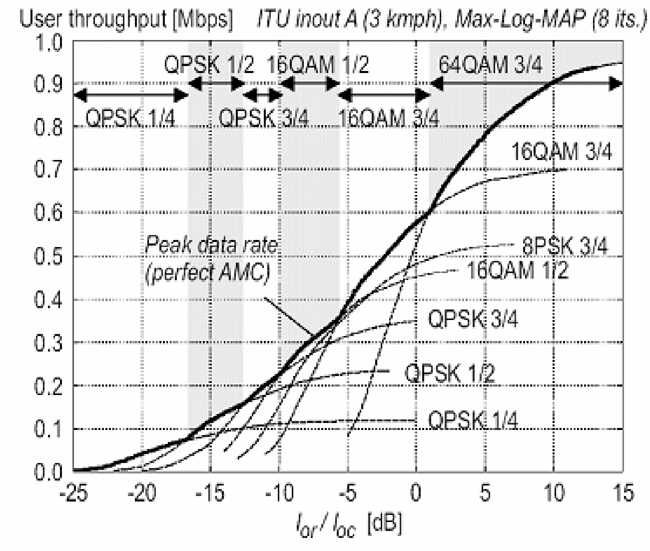


Figure 5-2 Link performance results of HSDPA MCS set

5.4.2 MCS Performance with F-HARQ

The combination of MCS with F-HARQ techniques applied to HSDPA is envisaged to increase the user throughput since F-HARQ has a significant influence on the throughput. The desired link level performance target depends heavily on the use of HARQ. While robust detection without HARQ requires very low BLERs, the use of retransmissions facilitates very high BLER values, e.g. on the order of 50 percent. The actual operation point depends on the slope of the MCS performance curve and, hence, the HARQ adjustment depends significantly on the operating environment. A typical target is to limit the number of required retransmissions to a maximum of 1- 2.

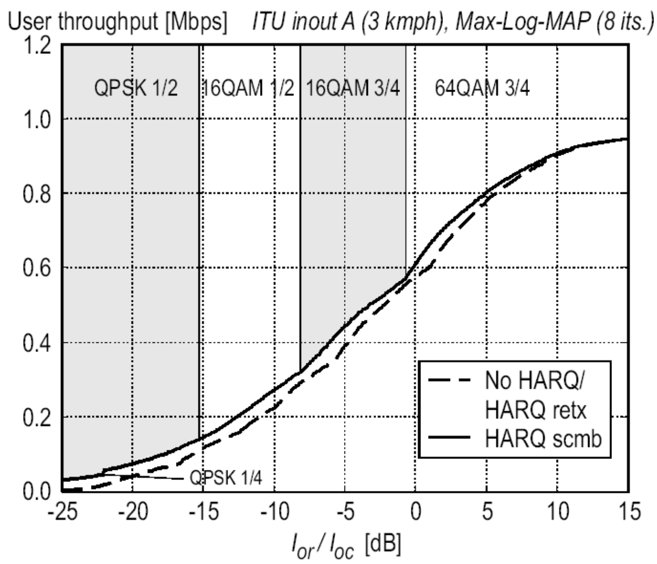


Figure 5-3 User throughput including the effect of MCS with F-HARQ

The results of simple retransmission as well as no HARQ have been plotted with a label “No HARQ/HARQ retx”). The use of soft combining (labeled “HARQ scmb”) adds approximately 1-2 dB of

SIR gain. Note that the gain of soft combining is best for the lower SIRs for each MCS where the BLER is high.

5.5 Possible enhancements for the conventional deployment concepts in short range scenarios

Despite the definition of new air interface for conventional deployment concepts is not yet achieved within the WINNER project, some possible enhancements for those systems must be investigated in order to provide some highlights about the capabilities that in the future might be attained. In the short range contest (in building and hot-spots) this leads to consider the ongoing work performed by the IEEE 802.11n committee.

Currently a few complete (MAC + PHY) proposals are being examined within the .11n standardisation process. It is out of the scope of this document to investigate in the detail the content of each proposal, but some main enhancements proposed are worth being overviewed. They represent an improvement of the most popular WLAN system to date that have to be taken into account in the context of the WINNER design of enhanced conventional deployment concept.

One significant challenge TGn has been dealing with is the efficiency of the MAC protocol. Besides, the need of the backward compatibility imposes to design proposals not disruptive with respect to the legacy 802.11. Both these constraints, along with the explicit purpose of increasing the throughput performance, have led to the conception of interesting technical solutions that are likely to represent the next generation of short-range-targeted systems, including the WINNER one.

Since the improvement of the MAC protocol efficiency coupled with the interoperability with legacy devices resulted to be quite a complex issue, the attainment of great throughput so far relies mainly on the PHY layer. A lot of progresses have been done in this area in the last few years. More precisely, the MIMO transmission techniques as well as new advanced coding algorithms look very promising and are generally adopted. Wider bandwidth, with respect to the original .11 legacy (20 MHz), is also considered as an optional feature to further improve the performance of the system. This latter solution has to be carefully examined since it can shed light on the capabilities of a broadband system and since the WINNER approach for the short range scenarios seems to go toward large bandwidth utilization (100 MHz and 2048 carriers).

The original purpose of the amendment is to provide on top of the MAC layer at least 100 Mbps, but depending on the number of antennas used and consequently on the spatial streams available, a very larger throughput is achievable at the PHY layer (it more than doubles). The proposals mandatory modes are likely to be limited to Spatial Division Multiplexing (SDM) with two streams transmitted on two antennas in a 20MHz bandwidth. However, exploiting up to 4 spatial streams and increasing the band (40 MHz) enables to go beyond 400 Mbps. The mandatory modes are proposed in conjunction with a legacy convolutional coding scheme, but optionally some advanced coding such as Low Density Parity Check (LDPC) is proposed. Closed-loop MIMO with per-stream rate adaptation is also likely to be considered as an option.

Beside the gain in terms of performance, the other evident advantage that multiple antennas can provide to a WLAN system is the coverage increase, even when using only one receiver stage. However, it is not yet sure whether 802.11n will adopt robust modes such as Space-Time Block Coding (STBC) or hybrid SDM-STBC modes in order to increase the range, or whether it will only focus on increasing the bit rate in the vicinity of the AP.

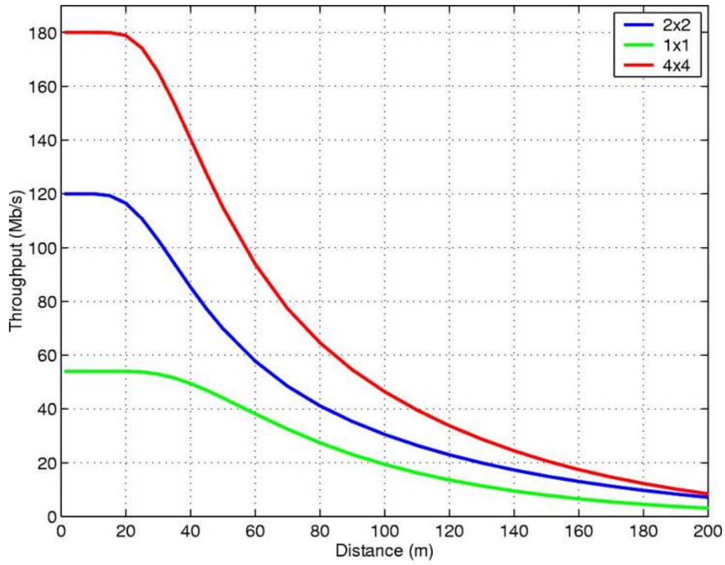


Figure 5-4 Throughput vs. distance for different MIMO techniques

The figure above clearly shows that at the same distance the throughput results dramatically improved at the PHY layer employing multiple antennas, and a combination of SDM-STBC. Obviously, approaching the limits that the propagation conditions impose this difference blurs.

The obvious drawbacks of employing MIMO techniques are the raise of the complexity of the devices and their power consumption. The latter problem is not relevant in the AP deployed in infrastructure based networks but at the receiver side it must be carefully managed and balanced with the adoption of mechanism enabling to save power, especially with the deeper diffusion of multimedia applications. To alleviate this limitation some improvements of the MAC protocol can be particularly effective.

The improvement of the efficiency of the current 802.11 MAC layer results more complex to achieve because of the required interoperability with the existing legacy. It is well known that the CSMA mechanism offers a very low efficiency compared to a controlled access such as the centralized TDMA. This lack becomes all the more evident as the size of the packet sent decreases, since the overhead needed for the transmission approaches or even exceeds the number of useful data bits. That is, the medium is exploited in a wasting manner because of the multiple time intervals that take place for the correct evolution of the protocol.

Four main solutions have been envisaged so far to improve the performance of the .11 MAC protocol. They roughly aim at increasing the maximum number of octets in the MAC Service Data Units (MSDUs) as well as decreasing the number of Inter Frame Spaces (IFSs) transmissions in a frame by means of the aggregation of plural MSDUs and use of reduced IFSs, Finally the work already performed within the TGe has shown that the adoption of a Block ACK (acknowledgment) policy allows a strong reduction of the control overhead with a consequent benefit for the efficiency. Moreover it must be noticed that the introduction of an MSDU aggregation strategy requires to some extent the use of the Block ACK to prevent the loss of the gain that the aggregation allows.

The aggregate exchange sequence consists mainly in the transmission of groups of frames “at a time”. More precisely, one or more MSDUs being sent to the same receiver can be aggregated into a single Aggregated-MSDU.

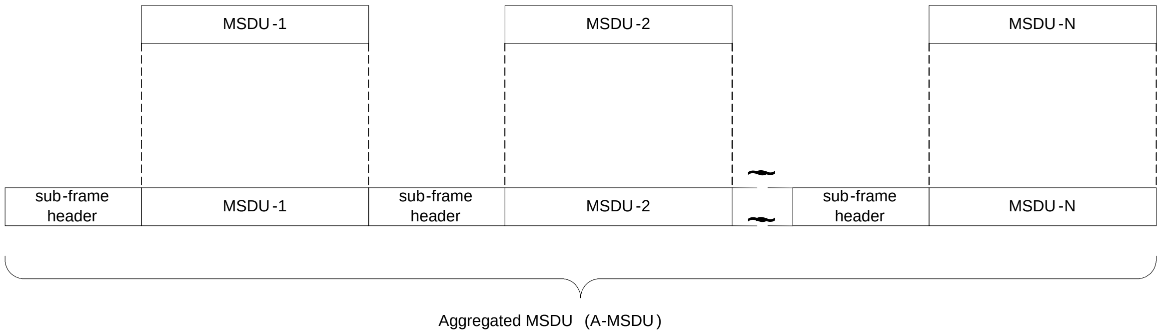


Figure 5-5 Formation of an A-MSDU

This technique permits also to save power since the sensing phase of the medium access is sensibly reduced.

It is evident that all the mentioned MAC improvements have been designed with the intention to preserve the interoperability. This concern should be considered in the WINNER project as well because the modifications presented are intended to be easily adapted to a better performing OFDM based PHY layer: this is one of the most important aim of the project.

6. System Simulations Methodology

System simulations are an essential tool in assessing relaying concepts in WINNER framework. Several partners have taken up the task of building a system simulator, which would be suitable for studying network deployment scenarios. Building a system simulator is a rather demanding task, and many of the features can be implemented in various detail. In order to be able to use the results from different simulators, a reasonable level of comparability of the results needs to be guaranteed. Collaboration between partners has resulted in a set of common system simulator features, and agreements on how to model certain aspects of the system. Moreover, studying WINNER network deployment concepts require modelling a few unconventional technologies in the system simulator. The questions how to model such novel technologies have been jointly discussed in WINNER WP3. The results of the work aiming to provide the common approach for the system simulations done by partners are presented in this chapter.

Section 6.1 summarizes on the general level the system concept developed in WP2 and WP7, concentrating into the aspects that affect the system simulations in WP3. Section 6.2 present the common L2S interface developed in WINNER from WP3 perspective. Also some simplifications to the common interface that have been seen necessary are discussed. Sections 6.3-6.7 describe some of the crucial simulator functionalities and the default principles for modelling them. Section 6.8 highlights the traffic models proposed in XWP and summarize the models relevant (simulated) in WP3. Section 6.9is divided into two parts: The first part describes the grid based environment models (e.g. Manhattan), while the second part describes the statistical environment models (e.g. hexagon approach). Section 6.9 describes the actions and methods used by WP3 to ensure that the system simulation results produced by different partners are comparable.

6.1 General Air Interface

Various air interfaces will be/were proposed [96] and will be mainly investigated on the link level. On the system level the generalized multi carrier transmission will be investigated for wide area and short-range scenarios are summarized in Table 6-1. The Generalized Multicarrier (GMC) transmission, considered for the uplink in the wide area scenario, includes OFDM and single carrier transmission as special cases [101].

Table 6-1 Multiple Access schemes considered in WINNER

Mode	Downlink	Uplink
Wide area	OFDM / (F+T)DMA	GMC / (F+T)DMA
Short range	OFDM / (F+T)DMA	OFDM / (F+T)DMA

The medium access control (MAC) protocols are based on Multiple Access Technologies, which enable the access to the spectrum of resources for multiple users/stations. MAC plays a decisive role especially when Quality of Service is considered. The MAC-protocol needs to possess information about the status of the channel in the sense of knowing the available resources and the requirements of the parties interested in using the resources. MAC protocols can be divided into tow categories: contention-free and contention based protocols[107]. The contention-free protocols ensure that the transmission is successful by ensuring that it is not interfered by other transmissions. On the other hand the contention-based protocols cannot ensure interference-free transmission. This can happen when multiple users try to exploit a common part of the spectrum resource simultaneously.

In the downlink communication the MAC operates as a shared channel and the available resources (chunks) are dynamically shared among different UTs. Downlink data transmission is performed over the downlink scheduled data channel (DSDCH). In the uplink, the MAC allows data to be transmitted either over scheduled or contention-based transport channels. In fact a UT can either transmit directly via the uplink contention-based data channel (UCDCH) or the uplink scheduled data channel (USDCH) that is used to send a scheduling request asking the network for dedicated resources.

6.2 Link to System Interface

This section describes the link to system level interface coming as recommendation from the Cross-Work package and in case of mapping alternatives and simplifications on it in order to make simulations feasible.

The modelling of the interface has to deal with the trade of between modelling accuracy against complexity at system level. The interface should be as simple as possible in terms of computation complexity as well cover all the link properties with sufficient accuracy in order to investigate all the desired system aspects.

Each modelling has its disadvantages in terms of applicability, means some simplifications may or may not accurately model the link performance under certain circumstances. On the other hand the interface should be able to be flexible in the way that changing physical characteristics or channel models during the project phase can be easily adapted and need no further changes on the interface. The complexity of the link to system level interface goes in expense to the modelling accuracy of other algorithms.

6.2.1 Link Quality Estimation

Within system level simulations the link performance estimation is abstracted with lookup tables or equivalent regression polynomials. Recent interfaces use the instantaneous SINR calculated within the system level simulator to make an estimation of the packet error rate (PER). For third generation systems, where DSSS techniques are used, the PER prediction can be applied on the despreaded "narrow" band channel and must not take broadband effects into account. For 4G broadband systems however, where coding along the frequency axis may be applied, this has to be taken into account too. WP2 made a literature study and selected the best-suited PER prediction mapping function. From system level point of view, the mapping is based on the effective SINR, which is the non-linear average based on the mutual information of all the frequency-time tiles involved for one coding block. In case of an OFDM system, the SINR values must be calculated for each (or a group of) sub carrier and for the duration of one symbol (or a group of symbols). In order to come to the PER, this effective SINR will be mapped by a one-dimensional mapping function or a simple lookup table. For more details on the link to system level interface and the latest information see [102].

$$\mathrm{SINR}_{\mathrm{eff}} = \beta \cdot I_{m_{\mathrm{ref}}}^{-1}\left(\frac{1}{P_u}\sum_{p=1}^{P_u} I_{m_p}\left(\frac{SINR}{\beta}\right)\right) \tag{17}$$

$$I_{m_p}(x) = m_p - E_Y\left\{\frac{1}{2^{m_p}}\sum_{i=1}^{m_p}\sum_{b=0}^{1}\sum_{z\in X_b^i}\log\frac{\sum_{\hat{x}\in X}\exp(-|Y-\sqrt{x}(\hat{x}-z)|^2)}{\sum_{\tilde{x}\in X_b^i}\exp(-|Y-\sqrt{x}(\tilde{x}-z)|^2)}\right\} \tag{18}$$

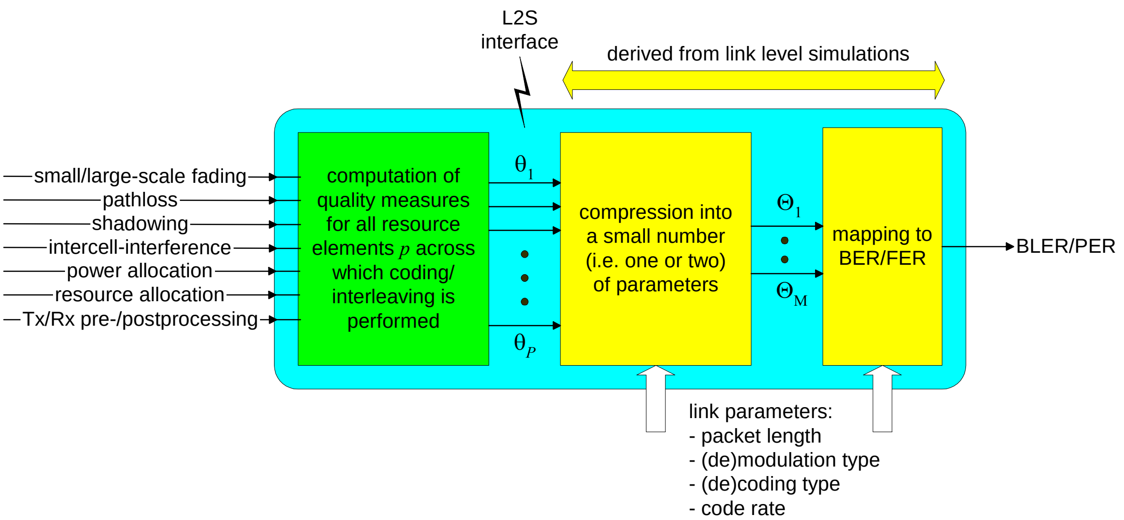


Figure 6-1 Link Quality Model

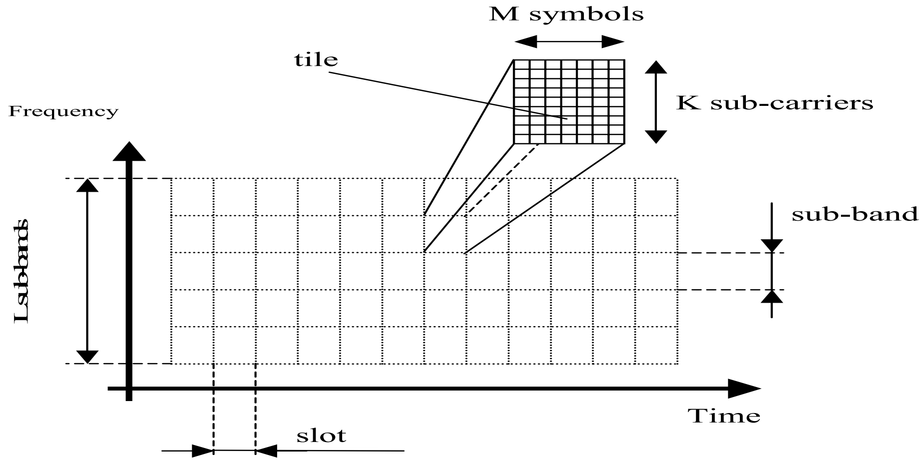


Figure 6-2 Time-Frequency structure

The summarization of the symbols to tiles is possible with negligible failure within the coherence bandwidth, i.e. where the channel can be assumed to be constant. Over the time axis, the tile can be as long as

$$\Delta t_{\min} \approx \frac{\lambda}{v} \tag{19}$$

where λ is the wave-length and v is the speed of the user. Over the frequency axis the tiles can be as brought as

$$\Delta f \approx \frac{1}{\left|\tau_{\max} - \tau_{\min}\right|} \tag{20}$$

The term in the denominator represents the delay-spread, i.e. the coherence time is inversely proportional to the delay spread. The Signal-to noise ratio has to be calculated at least for each tile separately.

6.2.2 Signal Power Calculation

The receive signal power at frequency f (or equivalent sub carrier) and time instant t can be calculated as follows:

$$C_{lm(f,t)} = \frac{TXP_{ml} \cdot g_l(m) \cdot g_m(l)}{L_{lm} \cdot S_{lm}} \left|h_{lm}(f,t)\right|^2 \tag{21}$$

TXP_{ml} .... Transmission power from node l towards terminal m

$g_l(m)$.......... The antenna gain of node l towards node m

$g_m(l)$ .... The antenna gain of node m towards node l

$C_{lm(f,t)}$ .... The receive power at node l transmitted from node m

L_{lm} .... The geometrical path loss between node l and node m

S_{lm} .... The shadow fading between node l and node m

Here l and m denote the sending and receiving nodes in the network respectively. As can be seen from equation (21) the received carrier power can be divided into a highly time dependent term and a “slow” varying term. It can be assumed, that the path loss, antenna gains and the shadowing are determined by location and distance and do not change but remain constant as long as terminal remains almost stationary. Thus the changes in this part of the equation are assumed to change in slow nature (depending on the granularity of mobility/velocity, i.e. how often the UT location is changed). This granularity is much longer than one MAC reservation period. However in the worst case these parameters can be assumed to be constant for one code-block. This yields to

$$C_{lm(f,t)} = C_{lm} \cdot |h_{lm}(f,t)|^2 \tag{22}$$

with introducing the slowly time dependent term

$$C_{lm} = \frac{TXP_{ml} \cdot g_l(m) \cdot g_m(l)}{L_{lm} \cdot S_{lm}} \tag{23}$$

The time variation period of the highly time dependent term can be estimated according the equation (19). The higher the frequency smaller the correlation time length e.g. in 5GHz with 250 km/h and 100 km/h the approximated correlation time is 0.9ms and 2.0ms respectively. Thus it is proposed that if the reservation period for DL and UL is of order 2ms the channel at system level can assumed to be constant. Thus the granularity of the channel updates in the SISO system simulations could be ones at the begging of the DL and UL reservation period. For the MIMO system simulations the correlation properties of MIMO channel should be studied further before this kind of simplification.

6.2.3 Interference Calculation

The interference observed by a user can be calculated as the sum of the receiving powers at the same physical resource at time instant t, e.g. in one tile/OFDM-symbol.

$$I_{lm(f,t)} = \sum_{\substack{k=1 \\ k \neq m}}^{N} \frac{TXP_{mk} \cdot g_k(m) \cdot g_k(l)}{L_{lk} \cdot S_{lk}} |h_{lk}(f,t)|^2 \tag{24}$$

Here again the path loss, the antenna gains and the shadow fading are constant during one coding block and a term-separation is possible

$$I_{lm(f,t)} = \sum_{\substack{k=1 \\ k \neq m}}^{N} I_{lm} \cdot |h_{lk}(f,t)|^2 \tag{25}$$

with

$$I_{lm} = \frac{TXP_{mk} \cdot g_k(m) \cdot g_k(l)}{L_{lk} \cdot S_{lk}} \tag{26}$$

The interference situation might change from OFDM symbol to OFDM symbol, thus even though the received interference from one source might not change during the several OFDM symbols the number of interference sources might change (dependent on the allocation strategy).

With this the signal to noise and interference ratio calculates as

$$SNIR_{lm} = \frac{C_{lm}}{N + I_{lm}} \tag{27}$$

This interference calculation described is used for 3G simulations. Another view on this interference model can be obtained using the following channel for the interferer

$$I(t) = \sum_{i=1}^{N_I} \sum_{j=0}^{N_i} A_{i,j}(t) \cdot \cos(\omega \cdot t + \varphi_{i,j}(t)) \tag{28}$$

$A_{i,j}(t)$ .... Transfer-coefficient of one component of the multi-path channel

$\varphi_{i,j}(t)$ .... Phase-shift for one component of the multi-path channel

N_i .... Number of multi-path components per interferer

N_I .... Number of interferer

and can be derived through

$$P_I = \frac{1}{T_s}\int_{t'=0}^{T_s}\left|\sum_{i=1}^{N_I}\sum_{j=0}^{N_i} A_{i,j}\cdot\cos\left(\omega\cdot t'+\varphi_{i,j}\right)\right|^2\cdot dt'$$

$$= \left\langle\left|\sum_{i=1}^{N_I}\sum_{j=0}^{N_i} A_{i,j}\cdot\cos\left(\omega\cdot t'+\varphi_{i,j}\right)\right|^2\right\rangle \le \left\langle\sum_{i=1}^{N_I}\underbrace{\left|\sum_{j=0}^{N_i} A_{i,j}\cdot\ \cos\left(\omega\cdot t'+\varphi_{i,j}\right)\right|^2}_{\hat{A}_i^2\left|h_i(f,t')\right|^2}\right\rangle = \tag{29}$$

$$= \sum_{i=1}^{N_I}\hat{A}_i^2\cdot\underbrace{\left\langle\left|h_i(f,t')\right|^2\right\rangle}_{ff_i} = \sum_{i=1}^{N_I}\hat{A}_i^2\cdot ff_i \Rightarrow$$

$$P_I \le \sum_{i=1}^{N_I}\hat{A}_i^2\cdot ff_i = \sum_{i=1}^{N_I}\frac{TXP_i\cdot g_{T,i}\cdot g_{R,i}}{L_i\cdot SF_i}ff_i.$$

Here we need to apply the triangle inequality, which yields a too high/pessimistic value for the calculated interference.

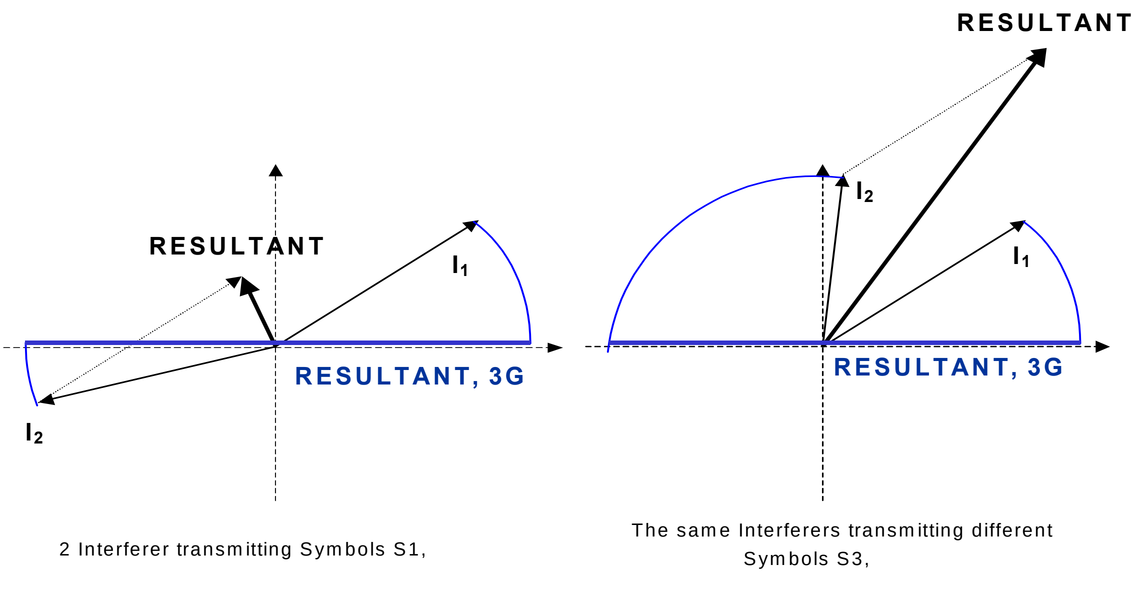


Figure 6-3 Calculated (Blue) and real (resultant) affiliated interference power at the receiver

Figure 6-3 exemplifies the difficulties arising from this model. It shows two interferers approaching at different phase relations to each other at the receiver. In the left phase difference is approx. 180 degree. The projection onto the real axis (blue) is the interference calculated to the described approach and exemplifies the applying of the triangle inequality. The interference seen at the receiver is drawn as the resultant-vector.

6.2.4 Simplified WINNER Interference Model

The L2S interface described above is rather heavy, and simplifying it would be desirable. Thus, this section presents a potential way of reducing the computational complexity, to be further studied in upcoming deliverables. The interference calculation described in the former chapter and the assumption for deriving this model can lead to a simpler model, which may be also applicable for WINNER system-level simulations. The proposed link-to-system level interface (see chapter 6.2.1) allows for some assumptions that enable to define a novel more accurate and efficient model. The $SNIR$ is calculated per tile/symbol. It is assumed, that the number of interferer as well as the channel for all the interferer remains constant. Through the duration of the symbol/tile the interferer will sum up coherently at the receiver, regardless of the used modulation. The phase relation of the interferer (or in other words incident waves) can be assumed to be equally distributed and the amplitude of the incident wave depends on the transmission power, propagation and multi-path losses.

Summarizing the aforementioned assumptions for the new approach of modelling the interference yields:

- Not only dependent on the channel/speed.
- Interferer are assumed to be statistically independent
- Different interferer can be assumed superpose coherently

Taking these assumptions, equation (25) and building up a new interference-model, that models the channel not only as fast-fading, but as function of a environment specific parameters yields to

$$I = \underbrace{\sum_i \frac{TXP_i \cdot g_S \cdot g_R}{L_i \cdot SF_i}}_{const.} + \overline{FF}(E) \tag{30}$$

The introduction of the fast-fading as function allows pealing out the sum, having slow-varying terms only and the fast-varying terms.

$$I = \hat{I} + \overline{FF}(E) \tag{31}$$

The reduced complexity can be seen from equation (27). The frequency of evaluation of the sum can be reduced to a minimum. Taking a code-block into account of N OFDM-symbols and M interferers the number of floating-point operations can be calculated (equation (27)) as $N(2M+1)$ for the conventional interference model and $N(2+C)$ floating point operations for the new model. Here C represents the floating-point operations for the evaluation/lookup of the introduced fast-fading function. Expressed in numbers for a code-block of 1000 OFDM-symbols (the number of interferers cancels out) and $C = 50$ the new interference models needs 40 times less floating point operations.

6.3 ARQ

This chapter deals with models for physical-layer ARQ's. Different types of ARQ exist, thus this chapter introduces models for two types of ARQ that are a guideline for more refinements. In general we have to distinct between ARQ and hybrid-ARQ. The difference between both types is that in case of a hybrid ARQ the resent data will be combined with the prior sent data. As simple ARQ (here as ARQ labelled) only the retransmission of data, without any combining at the receiver is understood.

The modelling of this type of ARQ is straightforward. There is no additional impact on the link-to-system interface it can be specified by a simple protocol, i.e.

A ARQ is requested by the receiving party in case of a non-decode able, or more generally a code-block with unacceptable BER. Upon request the transmitting entity resents the data in the next resource with e.g. increased power. If the retransmission uses a different transport format, this needs to be indicated to the receiving entity e.g. through a TFCI/preamble-style packet-header.

In conjunction to this simple scheme the hybrid ARQ extends the ARQ transmission scheme by a combining of the retransmitted data. This combining gain needs to be considered as well, thus the look-up according the link-to-system level interface needs to take this into account too. For a retransmission of the whole code-block with e.g. the same modulation and coding scheme the mapping to the effective SINR and correspondingly the PER-performance estimation can be done as follows:

$$\mathrm{SINR}_{\mathrm{eff}} = \beta \cdot I_{m_{ref}}^{-1}\left(\frac{P_{u,1}}{P_{u,1}+P_{u,2}} \cdot I_{m_{\mathrm{ref}}}\left(\frac{SINR_{eff,1}}{\beta}\right) + \frac{P_{u,2}}{P_{u,1}+P_{u,2}} \cdot I_{m_{\mathrm{ref}}}\left(\frac{SINR_{eff,2}}{\beta}\right)\right) \tag{32}$$

For combining with different or only partially retransmitted packets the scheme may be applied in the same fashion, as long as a dedicated lookup-table for PER-estimation is used. The lookup-table is the trained one coming from link-level simulations having also implemented the ARQ scheme.

6.4 Relaying

Relaying can be classified as either decode-and-forward or amplify-and-forward systems. In the latter it is possible to derive a close form of the SINR at the receiver. Hence the same table lookup can be used for the link to system interface. In the former case, whether or not cooperative relaying is considered it is much more difficult to derive a close form of the SINR. Consequently, a new table look up based on a new quality models, including demodulation model and decoding needs to be considered. One way around is to compute the SINR at each relay then convert it to PER, in case the packet is erroneous then the relay will not retransmit the information. In that case the same L2S can be used at the mobile. Many issues remain not clear and need further investigation for example:

- In case of soft combing at the receiver, it might be useful to use the same approach as ARQ.
- In case of no linear operations are used at the relay (Space time coding e.g. Alamouti based relaying) either a new table lookup based on link level results is required or the same L2S can be used if only the packets at both relays (in the case of Alamouti based diversity) are correctly received. In case the packet is erroneously received at one of the relay then the packet cannot be combined at the mobile unless a direct connection with the AP is assumed (with the same type of transmission).

6.5 Resource allocation/Link adaptation

Under this section a number of different issues should be addressed. Although they could be seen independently from each other initially, it is at a later stage that we can see clear interactions among them. As such, one influences the other and the more we move from 2G→3G→4G systems, the more holistic approach we need to take. Some of the aspects of resource allocation/LA are power control, fast link adaptation, scheduling. Figure 6-4 shows the interactions among them [96]. For instance, we have to take into account at the same time the power allocation/control issues, in order to adapt to the channel conditions as good as possible, by not inducing e.g. interference, but at the same point providing maximum gain in terms of e.g. throughput.

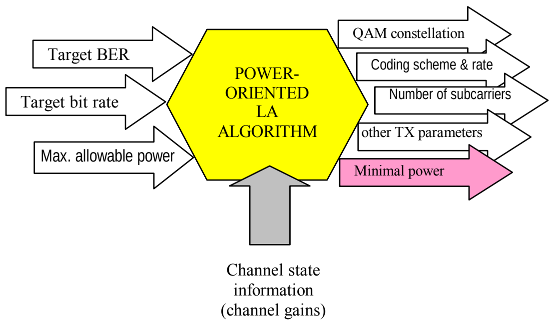


Figure 6-4 Input, output parameters, feedback information and the core of the power-oriented link-adaptation algorithm.

For instance, whereas normally assignment of resources for users is done based on a fixed pattern and link adaptation parameters are then adjusted within these resources, in the future, joint optimisation of scheduling and link adaptation will be required. This cooperative/joint resource allocation can be done under schemes like “water filling”, some kind of multi-user scheduling.

As it has been suggested in [96] it seems that the LA criteria, which are most appropriate, are throughput and power. It is not straightforward to propose one of them. For many applications, maximizing throughput seems to be the most desired. However for energy-aware networks, the battery life will be the crucial element and the power-oriented strategy may be the best candidate. At the same time the use of delay constraints will be important. While adaptation at the link layer will provide the best possible QoS to the application, this QoS may vary with time as channel conditions, network topology and user demands change. For the first evaluation of the link adaptation strategies a throughput-oriented strategy is proposed. For the analysis to be meaningful in a multi-user context, the link adaptation scheme should be combined with an appropriate scheduling strategy.

Regarding fast link adaptation, some outcomes from [96] are the following

- For the WINNER worst case channels, i.e., high speed terminals in severe frequency selective fading environment, full bandwidth fast link adaptation to all users will not be feasible for either FDD or TDD systems if adaptive resource allocation is to be implemented.
- In real applications, there are usually very few high speed terminals within a cell/sector, thus a general system design with fast link adaptation is still well suitable. This is true for both FDD and TDD systems.
- The link adaptation reaction delay is one of the parameters that affect the system design. The shorter the reaction delay, the easier the system design. For FDD this reaction delay could be up to 3-4 times the timeslot duration. For TDD this delay could be up to 4-6 times.
- The TDD system has advantages over FDD under lenient channel conditions where channel reciprocity with respect to SINR can be employed. In this case, systems with link adaptation become simpler and more reliable.

Other issues highlighted by WP2 with reference to link adaptation are the attainable accuracy and quality of CSI estimation/prediction, the effect of the accuracy of CSI estimation/prediction on the system performance and the effect of CSI feedback error on the system performance.

Another typical and popular adjustable parameter within the framework of link adaptation, especially for energy-constrained wireless networks is minimization of the power consumption of a transceiver for a negotiated level of service. Optimisation methods to minimize power consumption may be adjustments of transmission parameters (varying constellation size, code rate, transmission power, antenna beams), i.e. link adaptation, as well as capability to rearrange the system configuration at structural or architectural level according to the changes in operating conditions and/or requirements. Specifically for power control we have the following

- Slow /average power control is used for results in WP2. By slow power control it is assumed that path loss and shadowing effects are fully compensated, while signal strength variations due to fast fading are not compensated. While fast power control might improve its performance, a different approach was taken in [96]. As it was argued, the computational complexity of such systems becomes considerable with the Winner project system parameters due to very high bandwidth, symbol rate and delay spread. Therefore a moderate complexity is targeted, in order to show its potential advantages.
- Use of closed loop power control is avoided (as much as possible) by using the combination of open-loop power control, rate control (ACM) and (H)ARQ techniques.
- Thus, multi-user water-filling may be used in the downlink, and power control may be used in uplinks, e.g. to equalize the received power levels, if desired. The instantaneous peak transmit power is not restricted, as long as the average constraint is fulfilled.

Regarding resource allocation, Chapter 6 in [96] is dedicated in obtaining, preserving, and utilizing orthogonal in the resource allocation to users. To obtain high performance, methods based on adaptation, rather than on averaging, are emphasized. Three different methods are investigated:

- TDMA/OFDMA, or FDMA combined with TDMA using OFDM modulation.
- OFDMA. Here, fixed or slowly time-varying partitions of the sub-carriers are allocated to different users. This method is a special case of TDMA/OFDMA mainly for use for stationary users, with time-invariant channels.
- TDMA, where one or several OFDM symbols are allocated exclusively to users. This is a special case of TDMA/OFDMA of use for mobile users, with a simpler but less flexible resource allocation scheme.

Of the above variants, TDMA/OFDMA is the most flexible, and potentially the most powerful in combination with adaptive resource allocation, based on feedback of channel state information to the transmitters. The feasibility is assessed by theoretical considerations and by the detailed design of resource units in both TDD and FDD systems that take the known constraints into account. The preliminary conclusions from this study are:

- Adaptive TDMA/OFDMA resource allocation that adjusts to the frequency selective fading by link adaptation and attains multi-user scheduling gains is feasible for users at vehicular velocities of 50-70 km/h, at 5 GHz and at reasonable SINR values.
- This principle is feasible in uplink, under the crucial assumption that adequate frequency synchronization can be attained and maintained for all involved terminals..
- Adaptive TDMA/OFDMA is feasible in both TDD and FDD systems. In the example implementation, the worst performance is obtained in TDD uplinks, due to the required long prediction horizons for that case. This is a significant conclusion for the WINNER system design since it implies that the feasibility of adaptive multiple access using SISO links does not place strong constraints on the choice of duplex scheme.

6.6 Imperfections / Real World effects

6.6.1 Synchronization

The perfect synchronisation is practically impossible in the network. The level of the needed synchronisation depends on the system. E.g the TDD requires more synchronisation than FDD. If the network is assumed to be synchronised imperfections in the synchronization and propagation delays causes inter symbol interference (ISI) and inter carrier interference (ICI). The multi-carrier OFDM system has guard band to tolerate ISI and sufficient carrier separations ensure that the ICI is minimised. Thus if not specially studied in the synchronised network the ISI and ICI are neglected in system simulators.

6.6.2 Measurements

Most system level simulation tools do not consider measurement errors, but assume perfect measurements. In reality, measurements are affiliated by measurement errors. Wrong measurements lead to wrong decisions. For example it is well known in today’s 3G systems that the power control mechanism plays a major role and impact the system performance significantly. Usually a fast closed loop power control mechanism is applied. Dependent on the SINR-measurement and the target SINR value a TPC-bit will be signalled to the transmitter. If this decision fails due to measurement errors, transmission power will be adjusted in the opposite direction to the desired one. Simulations have shown, that even in case of very low wrong decision probability the system performance decreases significantly.

Out of this we can conclude, that we need to model measurement errors within the simulators as well. This makes it also possible to assess new technologies regarding the robustness against real world impacts. It could turn out, that a new concept performs best for idealized conditions, but may sensitive to measurement errors.

It is clear, that measurement errors can affect, both link and system level models dependent on the used models. Here we consider only measurement errors from system level point of view that are the basis for algorithmic decisions. For system level these measures are:

- Receive power level
- Interferer power level
- Signal to interference and noise ratio
- Direction estimation error
- Propagation loss
- Transmission power setting error

The first five measures are done within the receiver, thus the calculation and properties of the error will be determined within link level simulations. The sixth error is caused by imperfections of the hardware and mentioned as measurement error only because these errors are caused by errors in the feedback and calibration loops.

6.6.3 Transmission power setting error

This error can be systematic or random. Since there is no estimate about the error available and is further dependent on the specific implementation, a very simplistic model can be applied. The systematic misalignment can be simply modelled by the addition of an offset and the random error via a Gaussian probability distribution with zero mean and a certain standard deviation.

6.6.4 Power level and SINR

These errors depend on the used estimation algorithms on link level and the measurement duration. The estimates may be distributed around the true values with a certain underlying distribution. Additionally a systematic error may occur with decreasing levels (SINR values). Generally the systematic error can be modelled at system level by a shift, e.g. dependent on the SINR, C, etc. . The random error can be modelled by a probability distribution, whose determining distribution parameters (e.g. standard deviation) are dependent on the e.g. SINR, C.

The values passed to the link to system level interface for getting the bit error rate should not be affiliated by an error. Values which are not directly measured, but calculated out of biased measures, like the path loss do not need an additional error.

6.6.5 Direction estimation error

This error will appear on link and system level. On link level this error causes a performance degrade. On system level additionally the interference calculation will be affected. On system level additionally to the LL2SL interface properties of the angular estimation errors are needed. Especially for MIMO transmission this type of measurement error can have significant performance impact.

6.7 Handover

Handover procedure includes two main parts: link layer connection set up and IP route update. It is out of the scope of usual system simulations to fully include them in details. The suggestion is not to model IP route update delay, since it is assumed that most/all of the AP under the same AR.

Four steps of handover need to be simulated:

- **Handover triggers**

This part will define the parameters that will trigger the handover: Link quality, cell load, cell QoS, user locations etc.

- **Handover measurement**

Define the methods of measuring/deriving above triggers. What parameters and how often the measurement will happen in order to work out the triggers need to be defined. This is radio technology dependent and need to work with other partners in Wp2, or partners working in the simulation group.

- **Handover decision algorithm**

This defines the criteria of handover decision and algorithms, which is used to derive the decision indications. This is a function of the user's service profile, current service type, cell condition and overall indications.

- **Signalling and context transfer**

This will define the signalling of handover in link layer and higher layer. In the simulation, the link layer signalling is done in context transfer. The high layer signalling may be modelled as route update signalling delay (the delay in the air interface and network). There are ongoing discussions on the accuracy of hand over models inside Wp3.

6.8 Traffic Models

Traffic models within WINNER have been developed by WP1. These are defined from the perspective of characterising the end to end service level traffic in relationship to the scenarios and the relevant generic applications. The initial models (derived from external sources) are described in detail in [104] [106]. The parameter values in these models have been adjusted to account from traffic levels suited to the WINNER timeframe. These initial models are:

- Internet/browsing
- Conversational voice
- Video streaming
- Audio streaming

Further work is on-going to define traffic models for:

- File transfer
- Interactive activities (e.g. real-time control applications, gaming)

For the calibration case (see section 11 below) a "full-buffer" traffic model is used for ease of comparison.

6.9 Test Environment Models

This section describes two typical test environments that are used for performance evaluation of RAN protocols. Both test environments were defined and described in [104]. Additionally, the path loss models are described in [97].

6.9.1 Models for the Manhattan-like Test Environment

The first test environment features a Manhattan-like structure and is called "Typical Urban Test Environment Model (B.1)". The parameters for this model are described in sections 6.9.1.2 to 6.9.1.6.

6.9.1.1 Purpose

This test environment is used to study effects that are likely to arise from the regular structures known from the big cities of this world where systems like this will be deployed first.

6.9.1.2 Scenario Size

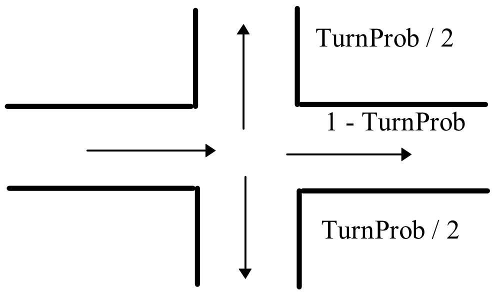
Area	Block size	Street width
6.5 km²	200 m x 200 m	30 m

6.9.1.3 Mobility Model

The mobility model uses a roadmap-based approach. Mobiles move along streets and may turn at cross streets with a given probability. Mobile's position is updated every 5 metres and speed can be changed at each position update according to a given probability. The mobility model is described by the following parameters:

- Mean speed : 3 km/h
- Minimum speed: 0 km/h
- Maximum speed: 5 km/h
- Standard deviation for speed (normal distribution): 0.3 km/h
- Probability to change speed at position update: 0.2
- Probability to turn at cross street: 0.5

The turning probability is illustrated on the figure below:



In addition to the above described geometry based position update approach a time-based position updated is proposed here. The granularity of the position updates (in terms of time) should be an adjustable parameter since this easily allows exchanging accuracy for simulation time.

6.9.1.4 AP Locations

Relay Enhanced Cells (RECs) can have different forms in this test environment, depending on the placement of the AP. In the following picture two examples are shown with Radio APs (either AP or FRN) placed on the intersections. Of course the APs can also be placed in the streets (between the buildings) to exploit the shadowing effects.

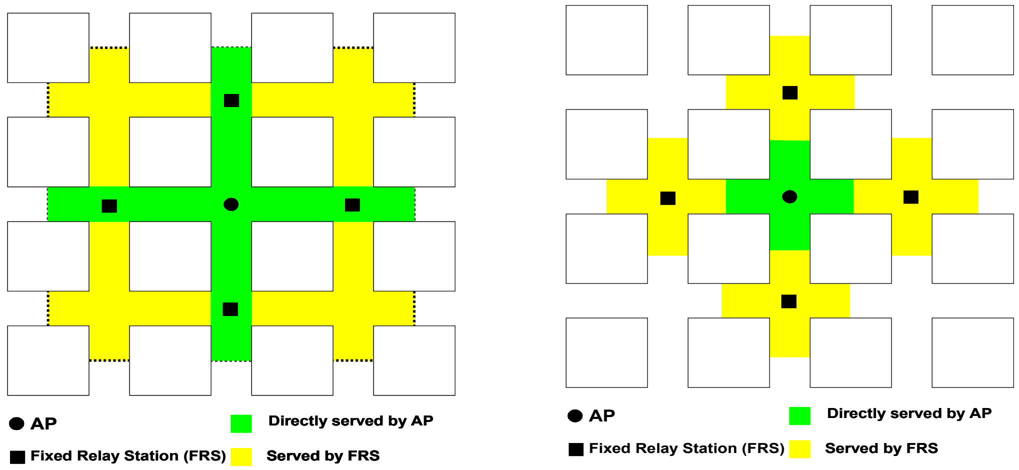


Figure 6-5 Different Types of REC with AP/FRNs on the intersections

6.9.1.5 Path loss Model

Let the path loss PL be:

$$PL(d) = A \log10(d) + B \tag{33}$$

where d is distance in m.

For this test environment different parameters for the NLOS and LOS case are specified:

NLOS: A = 33.7 dB, B = 54.7 dB

LOS: A = 23.4 dB, B = 43.3 dB

The selection between the propagation conditions LOS and NLOS is specified in the original SCM specification [98].

6.9.1.6 Shadowing Model

The shadowing model for this test environment is the well known model from UMTS 30.03 and is described in [97].

For the Manhattan-like test environment the correlation length is 20 m. The distribution of shadowing values is split up into LOS and NLOS case:

NLOS: log-normal, standard deviation 4.7 dB

LOS: log-normal, standard deviation 3 dB

6.9.2 Models for the Hexagonal Grid Test Environment

The second test environment features a classical hexagonal structure and is called “Typical Urban Test Environment Model (C.2)”. The parameters for this model are described in sections 6.9.2.2 to 6.9.2.6.

6.9.2.1 Purpose

Due to its regular structure and wide usage in the domain of performance evaluation of mobile radio networks the effects that arise from this model are very well understood. Therefore, the simplicity of this model suggests using it for simulations that are carried out in order to investigate effects that stem from layer 2 characteristics and should not be covered by effects that stem from the test environment. Consequently, this model should be used for the initial simulations while the Manhattan-like model is for advanced studies.

6.9.2.2 Scenario Size

Area	Cell layout
20 km²	Hexagonal

6.9.2.3 Mobility Model

The mobility model for this test environment is a pseudo random (Brownian movement) mobility model with semi-directed trajectories. Mobiles are uniformly distributed and their direction is randomly chosen at initialisation. Mobile's position is updated according to the decorrelation length, and direction can be changed at each position update according to a given probability. Direction can be changed within a given sector to simulate semi-directed trajectory.

Mobile's speed is constant and the mobility model is defined by the following parameters:

Speed value: constant 60 km/h

Probability to change direction at position update: 0.2

Maximal angle for direction update: 45°

Decorrelation length: 20 metres

6.9.2.4 AP Locations

One general way to model the environment is the statistical one. In this kind of modelling the structure of environment is embedded in the propagation model. This trades the accuracy of environment modelling with the good generality of simulation results. The most common example of this kind of modelling is the hexagon type approach. Example of hexagon layout is shown below in. Figure 6-6

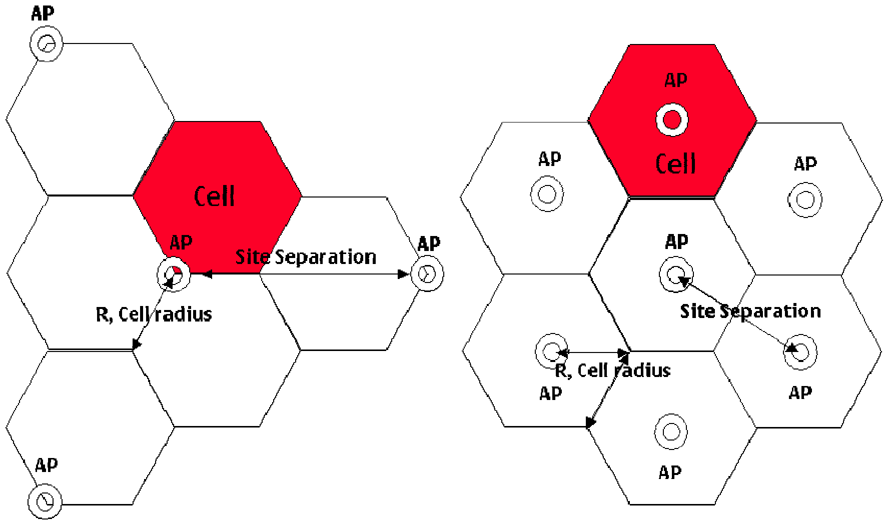


Figure 6-6 Two basic single-hop cellular environment models with hexagon cells on left three sector AP on right omni AP.

6.9.2.5 Path loss Model

According to (33), for this test environment different parameters for the NLOS and LOS case are specified:

NLOS: A = 28.3 dB, B = 53.5 dB

LOS: A = 23.4 dB, B = 43.3 dB

The selection between the propagation conditions LOS and NLOS is specified in the original SCM specification [98].

6.9.2.6 Shadowing Model

See 6.9.1.6.

NLOS: log-normal, standard deviation 5.7 dB

LOS: log-normal, standard deviation 3 dB

6.10 Comparability of Simulator Result

The overall WINNER approach for maintaining comparability of simulation results is described in chapter 2 of the assessment criteria definition [104].

At the moment, there is no common simulation tool within WINNER, which makes ensuring the comparability quite challenging. But on the other hand, existence of several system simulators offers robustness as results of different tools can be compared. It is also desirable to be able to compare WINNER results with those from external sources.

The following actions have been taken to ensure comparability of results:

- An agreed set of traffic models (outlined in section 6.8)
- Usage of the agreed propagation models defined by WP5
- Usage of project-wide test configurations (detailed in [104]).
- A common link-to-system level interface
- A calibration scenario (further detailed below) to ensure fundamental implementations are aligned

By following these project-wide mechanisms with an open description of parameters and techniques, the comparability of simulation results within WP3 will be assured. This will allow comparison between the the simulations on deployment concepts based on the calibration case or WINNER air interface from different partners, as well as benchmarking results reported in [2].

To make sure that the system simulator results of WINNER WP2 and WP3 are comparable at least to some degree, common calibration case has been defined. However, due to different goals of the system simulations in WP2 and WP3, the common calibration case has been modified to accomplish the purposes of both work packages. The WP3 specific calibration case is presented in Annex I (chapter 11).

WP3 specific calibration case assumes that common outcomes from the simulators are e.g. C/I distribution, Rx power distributions, interference distributions, delay distributions and cell throughput. The most relevant of the common outputs (or intermediate outputs) have been chosen for calibration merits (see chapter 11.6). The most common features affecting to the merits should be modelled similarly in each simulator. The calibration case results between different partners are compared. This gives a possibility to identify major problems in the system simulators, thus ensuring more reliable results in the later phase of WINNER.

7. Conclusions and Future Work

The multi-hop concepts in D3.1 showed the potential of relay based deployment concept in terms of increasing radio range and thus coverage of a Broadband Air Interface. First approaches towards RAN- and protocol-architecture have also been provided.

That work has been consequently driven forward towards D3.2, resulting in approaches to integrate both the different WINNER modes and the new network elements to support the several relaying concepts (see Chapter 2, "Architectural Concepts and Generic Functions"). A flexible node architecture covering the requirements of the WINNER RAN has been designed. The presented multi-mode protocol architecture allows to exploit commonalities between different WINNER modes by grouping functions common to the different modes in re-usable generic parts of the protocol, leading to a very flexible protocol and a close integration of the WINNER modes.

The expected key functionalities of the control plane are presented. The integration of relay nodes in the network architecture leads to new requirements on the radio network protocols to allow for example efficient retransmission handling spanning multiple hops. A respective protocol concept has been proposed.

The radio interface architecture studies indicate that the vision of a ubiquitous radio system concept may be realized with relatively simple architectural concepts and building blocks. It is shown, that one single and flexible node architecture model may support multiple deployment scenarios with a manageable number of logical network nodes and interfaces. Moreover, a multi-mode protocol stack together with a unified interface towards upper layers may facilitate seamless interworking between multiple modes and hide the heterogeneity of modes from upper layer protocols and functions. Finally, a simplified channel structure with a relatively small number of channels (and channel types) may handle multiple radio interface modes and deployment scenarios.

Even though the studies so far indicate that the ubiquitous radio system concept can be realized with relatively simple building blocks, there is probably a trade-off between performance and simplicity. An important topic for further work is to study the performance of the concept. Furthermore, the integration of different relaying based concepts into the architecture has not been carried out yet and their impact on the generic functions is moreover unknown. Consequently, some architectural definitions are still rather preliminary and they need to be developed further. In particular, the definition of logical relay node and its relation to other nodes — especially logical access points — is a subject for further studies.

As a result of the first phase concept work presented in D3.1, a large number of proposals for basic concepts were proposed. Since a complete and thorough assessment of all the proposed concepts would not have been feasible with the given resources, a different approach was taken. First, a categorisation was performed with the aim to achieve harmonisation and narrow down the number of concept that have to be subject to an in-depth assessment. The resulting categories of basic concepts identified by WP3 are:

- Single Hop Concepts
- Fixed Homogeneous Multi Hop Concepts
- Fixed Heterogeneous Multi Hop Concepts
- Mobile Multi Hop Concepts
- Cooperative Multi Hop Concepts

Whereas the second and third basic concepts have been identified to bear a sufficient amount of commonalities to further be jointly referred to as "Fixed Relay" Concepts.

A preliminary positioning of deployment concepts vs. scenarios has been proposed.

First examples of "architectural harmonization" have been studied too, having recognized that the "architectural harmonization" is an important part of the overall harmonization process.

As foreseen in WP3, the work on the considered aspects will continue along two directions:

- The finalization of Deployment Concept definition and selection and
- a more specific harmonization on the selected concepts (protocol and L2 functional viewpoints)

will be performed. Further studies on Architectural Harmonization will be carried out.

The mentioned large number of concept proposals and the higher than expected effort for the necessary concept harmonization have led to changes in the WP3 workplan. Nevertheless Chapter 4 has presented first preliminary simulative assessments of multi-hop vs. single-hop concepts.

As a result of the work presented, the combination of the protocol elements of PMP and Mesh modes in multi-hop topologies appears as a very promising approach. Moreover, the presented comparison of multi-hop against single-hop deployment, leans towards using relaying as an essential component for WINNER air interface.

The analysis of the deployment examples based on the use of different types of heterogeneous relays, and taking the initial proposal of WINNER air interface modes into account, has shown on the one hand the potential benefits, which may be achieved implementing a mode specifically thought for feeding the RNs from the AP, and on the other hand the necessity to participate tighter in the development of the radio interface, for including the requirements identified during this preliminary analysis, in the design of the different modes for the new air interface.

The goal for the rest of the project's first phase is to pursue with the analysis of the multi-hop systems since even if some interesting results have been presented here, a number of issues needs to be detailed in order to prove the effectiveness of such systems. More precisely, the next deliverables are supposed to shed more light on the performance that might be attained with this kind of approach.

Depending on the envisaged scenario, also single-hop deployments are in the focus of WINNER and are further developed taking new technologies like adaptive antennas into account. Such enhanced single-hop deployment concepts are shown and assessed in Chapter 5, indicating that these advanced technologies provide an important building block towards achieving the very high data rates envisaged for the high-performance WINNER radio system.

System simulations are an essential tool in assessing the performance of relaying concepts in the WINNER framework. Several partners have taken up the demanding task of building a system simulator. Many of the features of such simulators can be implemented in various levels of detail. A reasonable level of comparability of the results needs to be guaranteed. Collaboration between partners to achieve this has resulted in a set of common system simulator features, and agreements on how to model certain aspects of the system. Moreover, studying WINNER network deployment concepts requires modelling a few unconventional technologies in the system simulator. The modelling of such novel technologies has been jointly discussed in WINNER WP3 and is presented in detail in Chapter 6. Ongoing work is related to the implementation of the specific features of the different WINNER air interface modes as specified by WP2 to enable an assessment of the capabilities of the entire WINNER "System Concept Embryo".

8. Annex I: Technical Details of Architectural Concepts and Generic Functions

8.1 WINNER Architecture

8.1.1 Logical nodes and their relations

User terminal logical node
The user terminal logical node comprises all functionality necessary for an end user to communicate with either another UT_{LN} or a network

The user terminal logical node may be connected to one or several RN_{LN} and/or to one or several AP_{LN}. In addition the UT will be the protocol peer of the $RANG_{LN}$[17], the ACS_{LN} and the AR_{LN}.

Access point logical node
The AP_{LN} is envisioned to terminate and handle the physical layer functions as well as the mode specific link layer functionality. Additionally the AP may act as a proxy between the UT_{LN} and the ACS_{LN} for certain protocol transactions. The AP_{LN} may be connected to one or several $RANG_{LN}$ and one ACS_{LN}. . Moreover, the AP is the node closest to the core (backbone) network that a UT may be (directly) connected to.

Radio access network gateway logical node
The $RANG_{LN}$ terminates the link layer on the network side. It acts as the link layer of the AR towards the UT_{LN}s. One $RANG_{LN}$ can connect to a UT_{LN} via one or several AP_{LN}. As there is a need to be able to change the serving $RANG_{LN}$ of a UT_{LN} by relocating the layer-2-context information, an interface between $RANG_{LN}$s is required. Moreover, the $RANG_{LN}$ is connected to an ACS_{LN}, and to an AR_{LN}.

Access router logical node
The Access Router acts as a normal Access Router per relevant IP layer specifications.

Access control server logical node
The ACS_{LN} controls one or several AP_{LN} and coordinates RRM with corresponding functions in the UT_{LN}. Moreover, it may also coordinate resources between neighbouring AP_{LN}s.

Relay node logical node
The task of an RN_{LN} is to perform relaying between AP_{LN}, other RN_{LN} and UT_{LN}. For some scenarios the RN_{LN} may be equivalent to a combined UT_{LN}, AP_{LN}, ACS_{LN} and $RANG_{LN}$. In these scenarios the RN_{LN} on one side would connect as a UT_{LN} to the AP_{LN}, ACS_{LN} and $RANG_{LN}$, whereas the other side acts as a combination of AP_{LN}, ACS_{LN} and $RANG_{LN}$ towards another RN_{LN} or UT_{LN}. It is for further study if this applies to all relevant scenarios and if a valid consequence is that the RN_{LN} can be removed from the reference architecture.

The RN_{LN} may be connected to one or several RN_{LN} and/or to one or several AP_{LN}, one or several $RANG_{LN}$, one or several ACS_{LN} and one or several AR_{LN}.

Interfaces
The nodes: UT_{LN}, AP_{LN}, ACS_{LN} and $RANG_{LN}$ result in the following logical interfaces that are of special interest from an architecture perspective:

$UT_{LN} \leftrightarrow ACS_{LN}$

$UT_{LN} \leftrightarrow RANG_{LN}$

$UT_{LN} \leftrightarrow RN_{LN}$

$AP_{LN} \leftrightarrow ACS_{LN}$

$AP_{LN} \leftrightarrow RANG_{LN}$

$AP_{LN} \leftrightarrow RN_{LN}$

$RANG_{LN} \leftrightarrow RANG_{LN}$

$RANG_{LN} \leftrightarrow AR_{LN}$

$ACS_{LN} \leftrightarrow RANG_{LN}$

[17] One user flow is always associated with one $RANG_{LN}$, however one $RANG_{LN}$ may hold several flows and hence, one UT_{LN} may be associated with several $RANG_{LN}$.

$ACS_{LN} \leftrightarrow ACS_{LN}$

8.1.2 Physical implementations using logical nodes

In Figure 8-1, several possible realizations that are based on the previously defined logical network nodes are exemplified. The purpose of these examples is to explain how logical nodes can be combined into different physical implementations in order to cope with different operational scenarios envisioned within WINNER. In addition, they also demonstrate how completely different solutions that are aimed for different scenarios are used without introducing an excessive amount of nodes and interfaces.

The logical network nodes can be implemented separately but combinations of some or all of them are also possible. It can be desirable to separate AP_{LN} from the ACS_{LN} and $RANG_{LN}$ in scenarios where the spectrum is the limiting factor and the user mobility is high. In such scenarios the spectrum efficiency calls for centralized ACS_{LN} and the high mobility profits from centralized $RANG_{LN}$. On the other hand, in scenarios with low mobility and vast access to spectrum, it may be preferred to co-locate ACS_{LN} and $RANG_{LN}$ with an AP_{LN}.

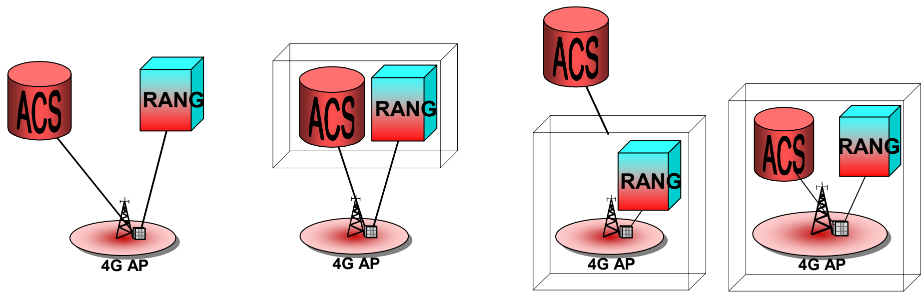


Figure 8-1 Illustration of possible physical node realizations using logical nodes

- In the first example (in Figure 8-1), AP_{LN}, ACS_{LN} and $RANG_{LN}$ are implemented in separate physical nodes. The main benefit is that the functionality is centralized and therefore buffer management and radio resource allocation is efficient. On the other hand, the RLC retransmission protocol round trip time can be negatively influenced by the transport network delays. By locating the user plane functions into the $RANG_{LN}$ and the control plane functions into the ACS, we achieve independent scalability, dimensioning and evolution of the user and control planes.
- In the second example, the ACS_{LN} and $RANG_{LN}$ are placed in the same physical node, which allows more efficient user and control plane information exchange than in the previous example.
- In the third example, $RANG_{LN}$ and AP_{LN} are located in the same physical node whereas the ACS_{LN} is sustained in some central node in the radio access network. The user plane is thus moved closer to the physical layer which can be beneficial with respect to the RLC retransmission protocol round trip time. A drawback is that the buffer management at high user mobility is awkward. In turn the control plane is centralized and therefore the radio resource management can be efficient. However the delay requirements on the transport network can be high.
- In the last example, all logical nodes are implemented in the same physical node. Since both the control and user planes are moved close to the radio interface the buffer management at high user mobility can be difficult and the radio resource management may become inefficient. However the RLC retransmission protocol round trip time is independent of the transport network layer delays.

It is assumed that the AR_{LN} is located in a central network node. In the two first examples (where the $RANG_{LN}$ and AP_{LN} are located in different physical nodes) AR_{LN} can be co-located in the same physical node as the $RANG_{LN}$. In the two last examples (where the $RANG_{LN}$ and AP_{LN} are co-located) AR_{LN} and $RANG_{LN}$ are thus located in different physical nodes.

8.1.3 Multi-mode Protocol Architecture

The WINNER ubiquitous radio system concept provides wireless access from short-range to wide-area, with one single adaptive system for all envisaged radio environments. It will efficiently adapt to multiple scenarios by using different Radio Access Technologies (RATs), i.e., modes, of a common technology basis. The devices of such a heterogeneous wireless infrastructure that use multiple modes (possibly simultaneously) can be referred to as *composite radios* [128]. The composite radio concept implies pre-installed modes complementing each other for optimized network utilization and QoS support. The dynamic selection, installation and adaptation of the devices' modes to the communication environment and changing user demands are introduced with the *reconfigurable radio* concept [129] originating from software defined radios [130].

The integrated projects WINNER and E^2R of the 6th framework research funding program (FP6) of the European Union, belonging to the Wireless World Research Initiative (WWI) [131], concentrate each on complementing aspects of the vision introduced above: Relay-based wireless mobile broadband systems [132],[133] as promising candidates for 4G networks are developed in WINNER. Corresponding to the definition above, such a relay-based system can be regarded as a *composite network*, which is outlined in Annex 8.1.3 as one motivation for the proposed protocol architecture. In contrast, reconfiguration management/functions and over-the-air download of software components are considered in E^2R and are beyond the scope of WINNER.

8.1.3.1 Radios at the Example of Relay-based 4G Networks

Relay-based wireless mobile broadband systems that provide a patchy coverage in densely populated urban areas can serve as a prototypical example for 4G systems. Depending on the multi-mode capability of the network elements, different deployment scenarios can be characterized as shown in Figure 8-2. The network elements related to the air-interface of a relay-based system are namely the UT, the RN and the AP. RNs have the advantage of distributing the high capacity available at the AP into a larger region. They cost-efficiently extend the coverage of a single AP to areas originally not covered by this AP. For the sake of simplicity we limit in our illustrations of Figure 8-2 the available modes to two (mode 1 and 2). In the case of single-mode network elements the relay-based system has a classical multi-hop architecture as depicted in scenario (I.) of Figure 8-2. In a wireless feeder scenario (II.), the RN uses simultaneously different modes of the air-interface for the RN-AP link and the UT-RN link respectively. These modes are preinstalled and used in a complementary way. Thus, corresponding to the definition from above, the RN can be regarded as a composite radio. Scenario (III.) additionally introduces a composite AP: The AP uses one mode for the relay link and a different mode for the link to the UT. An additional third mode can exist to provide a highly reliable low bit-rate link between the AP and RN for signaling purposes (as for instance Radio Resource Management (RRM)). The same stands for the AP-RN link in the composite radio network scenario (V.). There, all network elements are multi-mode capable enabling an efficient fulfillment of capacity and QoS requirements. This implies a joint RRM, inter-mode scheduling and self organization as outlined underneath. The reconfigurable network scenario (VI.) of dynamic configurable network elements with over-the-air protocol (re-)configuration and software downloads leaves the scope of WINNER but is nevertheless enabled through the introduced multi-mode protocol reference model.

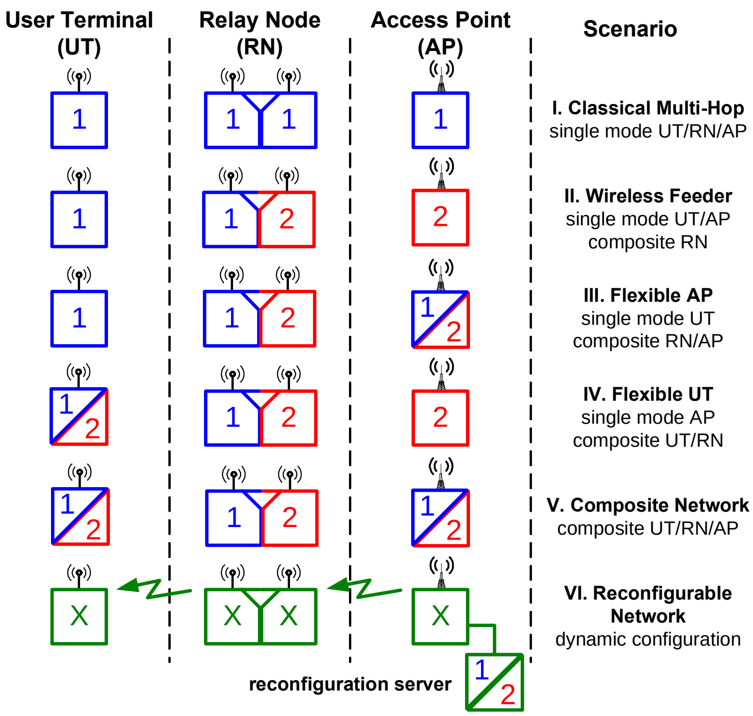


Figure 8-2: Deployment scenarios of relay-based 4G wireless networks characterized through the different usage of two modes (1 and 2).

8.1.3.2 Generic Protocol Stack for Efficient Multi-Mode Protocols

This section introduces the idea of a generic protocol stack for efficiently realizing multi-mode capability in communication protocol software on conceptual as well as implementation level.

The basic idea of a generic protocol stack is that all communication protocols share much functional commonality, which can be exploited to build an efficient multi-mode capable wireless system. The term “generic” can be substituted in the following by “common” and “general”. The aim is to gather these common parts in a single generic stack and specialize this generic part following particular requirements of the targeted mode. The targeted advantages of this concept are: code/resource sharing and protocol development acceleration through reusability and maintainability.

A key issue of the later introduced reference model for multi-mode protocols is the separation of a layer into specific and generic parts. The term generic is used by multiple authors with different knowledge backgrounds leading to dissimilar or even contradictory understandings of genericity. In taking the realization of a protocol stack out of generic and specific parts into account this becomes a software engineering problem of generic programming. Generic programming can be defined as “programming with concepts”. A concept is defined as a family of abstractions that are all related by a common set of requirements. A large part of the efforts in generic programming, especially in the design of generic software components, consist of concept development, i.e., identifying sets of requirements that are general enough to be met by a large family of abstractions but still restrictive enough that programs can be written that operate efficiently [134].

The balancing of the trade-off between general usability and implementation effort is crucial for the success of the separation of complex protocol software into generic and specific parts.

In general, generic protocol software may be realized through parameterizable modules and/or inheritance of system specific behavior. Well known programming patterns from computer science provide thereby a suitability-proven fundamental approach to the efficient realization of reconfigurable multi-mode protocol software.

As depicted in Figure 8-3, generic parts can be identified on different levels in the context of communication protocols:

- Architecture and composition of a protocol stack, introduced in Section 2.2.4.3
- Functions fulfilled by a layer that imply a certain behavior, outlined in Section 2.2.4.3
- Data structures, i.e., protocol data units or information structures, used for communication between peer-entities of a layer
- Protocol framework: Common rules for communication, as for instance the structure of a Medium Access Control (MAC)-frame (sequence and duration of broadcast, downlink and uplink phase)
- Management of a layer and protocol stack

The commonalities form, together with mode specific parts, a system specific protocol stack. An efficient multi-mode capable stack is realized in adding modes convergence management and related functions. This is introduced in the next section in taking up the composite radio paradigm. The introduction of additional reconfiguration management and functions leads to a dynamic reconfigurable protocol stack but leaves the scope of this section.

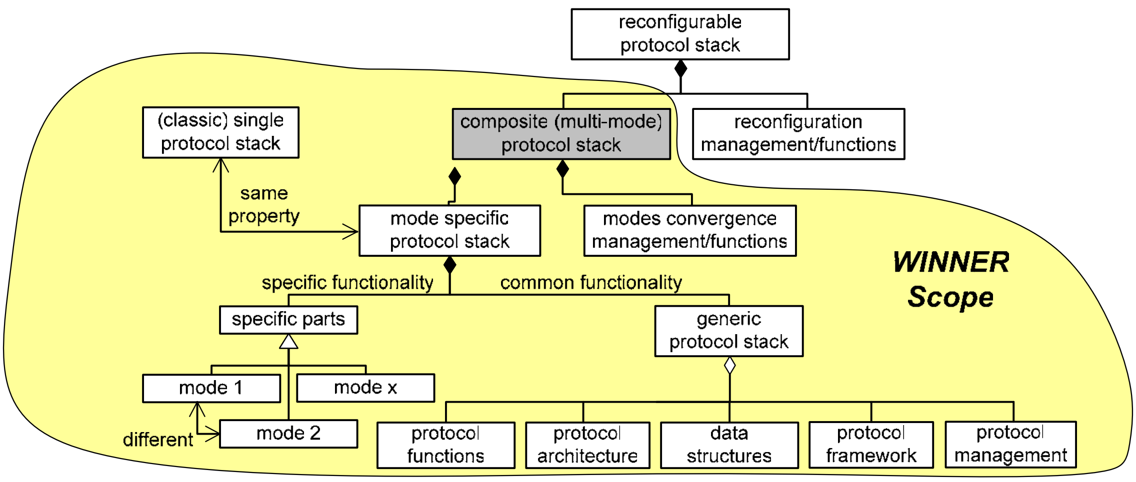


Figure 8-3 UML diagram of the context of protocol multi-mode capability and reconfigurablility (outside WINNER scope, which is marked yellow)

8.1.3.3 Multi-Mode Protocol Reference Model

The parameterization / configuration of a layer is performed by the (N)-Layer Modes Convergence Manager ((N)-MCM). The RRC on the control-plane and the RLC on the user-plane are generic to the layers located above. A mode specific protocol stack has an individual management-plane. RRM, the Connection Management (CM) and the Mobility Management (MM) are located in the Radio Resource Control (RRC) layer. The cross-stack management of different modes completes the reference model for multi-mode protocols in connecting the management-planes of the device's modes with the help of the Stack Mode Convergence Manager (Stack-MCM). Stack-MCM and (N)-MCM exchange in a hierarchical order data between two modes. The switching between modes and the coexistence of several modes is performed by the Stack-MCM. The (N)-MCM enables composing a layer out of different parts as depicted in Figure 2-2 and introduced in the previous section.

The (N)-MCM is the intermediator (cf. Figure 2-3) between the generic and specific parts of the multi-mode protocol stack's layers: All Service Access Points (SAPs), i.e., interfaces, touched during the mode transition of a layer are administrated by the (N)-MCM. In the classical view of protocols as state machines, the (N)-MCM transfers all state variables of the protocol layer between two modes with the help of the Stack-MCM. This could for instance imply the state transfer of being connected from one mode to the other together with a data transfer of received but unconfirmed data frames.

Concrete, the (N)-MCM manages a single layer and has the following tasks and responsibilities which are introduced later in this section:

- Layer composition and parameterization, considering all interfaces related to the transition between two modes
- Protocol convergence: (i.) horizontally – between "generic" and "specific" parts and (ii.) vertically – the mapping of higher layer user data flows in the DLL as known for instance from ATM
- Data preservation and context transfer

Furthermore, the convergence between mode specific protocol stacks is realized through the Stack-MCM implying implicitly the following functions:

- Joint Radio Resource Management (Radio resource coordination) between different modes
- Inter-mode scheduling
- Self-organization (frequency allocation of adjacent relays and APs, user data flow routing)

8.1.3.4 Functions of the Stack Modes Convergence Manager (Stack-MCM)

8.1.3.4.1 Joint Radio Resource Management

The functions of the user and control-plane are administrated by the RRM which may be coordinated centralized or decentralized and the RRM decisions are executed by the RRC of the corresponding modes. The RRM may assign multiple modes to one specific data flow. The RRC provides status information about the mode specific protocol stack in a generic structure to the RRM of the multi-mode protocol. In case of a (semi-)centralized coordination of the radio resource allocation, this generic information structure about the status of the different modes of the protocol stack can be transmitted to enable an adequate decision.

The RRM of a single multi-mode device may also support the coordination across neighboring operating devices as for instance the coordination across access points.

8.1.3.4.2 Inter-Mode Scheduling

The Stack-MCM as intermediator between modes can perform the scheduling among different modes, or delegate the scheduling to a mode-independent scheduler in the generic part of the protocol layer in question. Contrary, the scheduling inside a mode across logical/transport channels is done in MAC-g or RLC-g of the specific mode's protocol stack. The inter-mode scheduling considers the dynamic scheduling of different user data flows over multiple modes. The scheduling strategy may for instance be based on the modes' interference situation which requires a provision of necessary information directly from the PHY if the decision is done in the MAC independent from RRC/RRM. This information about the quality of the radio link is again provided in a generic information structure.

8.1.3.4.3 Self-Organization

The envisaged communication system is able to autonomously decide about its radio resource allocations in taking the environment into account. This implies the for instance the adequate selection of frequencies used for transmission or the routing of user data packets. The radio resource is selected under consideration of interference avoidance with other radio systems. Further, the optimized spectrum utilization coordinated with neighboring radio systems of the same technology is taken into account, which may also be related to efficient multi-hop relaying depending on the selected deployment scenario.

The self-organization comprises scenarios of breaking down and installation of additional devices in an operating communication system. The Stack-MCM has to support the addressed functionalities in activating for instance different modes to provide information about the interference situation or the role of devices (if it is acting as relay or AP) in reception range.

8.1.3.5 Functions of the (N)-Layer Modes Convergence Manager ((N)-MCM)

8.1.3.5.1 Protocol Convergence

The convergence of multi-mode protocol stacks has two dimensions: First the convergence between two adjacent layers, in the following referred to as *vertical convergence* as it is known from the user-plane of H/2 protocol stack. Second the convergence between layers located in the different modes of the protocol stack which have the same functions: In the following referred to as *horizontal convergence*. The generic protocol stack, managed by the (N)-MCM as introduced above, enables both the horizontal as well as vertical protocol convergence.

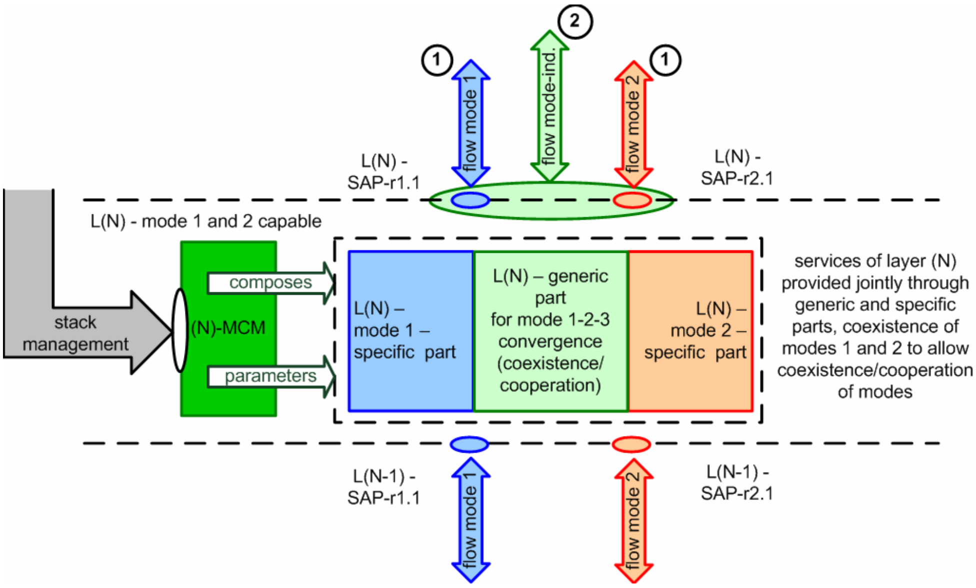
From the perspective of higher layer protocols the multi-mode protocol stack is transparent on the user- as well as on the control-plane, i.e., generic parts terminate the stack to the layers above, as depicted in Figure 2-2. The vertical conversion of the (N)-MCM implies the adaptation of the on the multi-mode protocol stack working packet data protocols to the specific mode. This may be for instance the conversion of an IP datagram in compressing the IP-Header.

8.1.3.5.2 Layer Composition and Parameterization

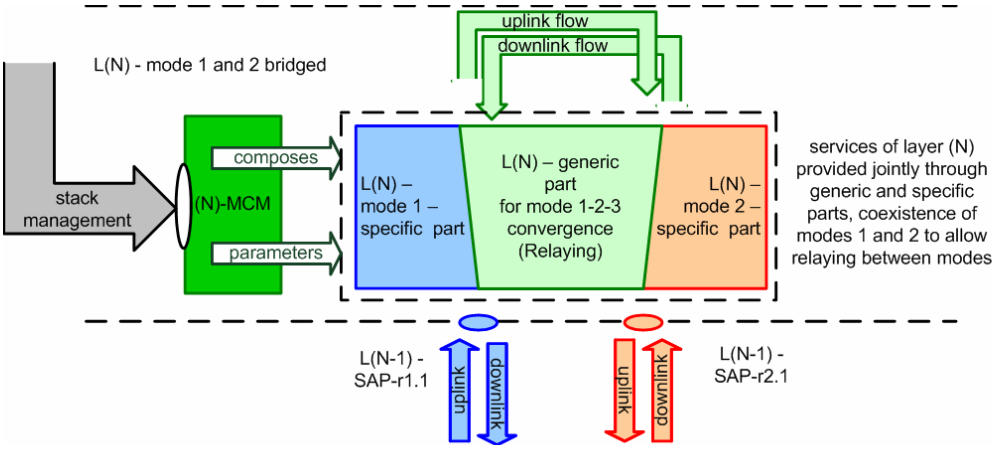
The separated approach of generic and specific parts requires an administration when taking the transition between modes into account: The common generic parts of the old mode need to be adopted for being reused in the new mode of the protocol stack. We assume that the generic parts of a layer exist permanently and are to be recomposed by the (N)-MCM corresponding to the characteristics of the targeted new mode. This assumption may imply a module-based composition concept of the generic parts as introduced in [135]. The composition of the layer out of generic and specific parts is done by the (N)-MCM.

8.1.3.5.3 Data Preservation and Context Transfer

The communication between two modes and the mapping between generic and specific parts of a mode is done by the (N)-MCM (inside layer) and the Stack-MCM (transfer between modes). The transition between two modes can be optimized in using the data from the old mode to the new mode. The ability of a user-plane protocol to reuse status information in the generic part and protocol data after transition to another mode requires an extension of the protocol into the control-plane though it performs only user-plane tasks. Depending on the status of the related protocol parts, the data transfer is referred to as “data preservation” or “context transfer” as illustrated in Figure 2-2: If the generic part is adjusted to the new mode the data needs to be preserved, i.e., adopted, to the new mode. In the case of a deletion of the old specific/generic part the data transfer is named “context transfer” which implies preservation for the new mode.



(a) Multi-mode operation. Either modes coexistence ① or modes cooperation ② are possible



(b) Realization of a multi-mode layer (N) relay. The common (generic) part enables the bridging between layer (N-1) traffic flows in different modes

Figure 8-4 Example configurations of a certain protocol layer or sublayer (N)

8.1.3.5.4 Example 1: Multi-Mode Capable Layer-(N)

Using the basic building blocks introduced in Figure 2-3, the realization of a protocol layer or sub-layer with multi-mode capabilities becomes possible. Figure 8-4(a) shows that the (N)-MCM configures the layer (the set of common and specific functions inside the dashed box) so that simultaneous operation in mode 1 and mode 2 is enabled. We distinguish two ways of operation which we call *modes coexistence* (①) and *modes cooperation* (②). In the first case, the layer can be regarded as "polymorphic", i.e., it can provide services of two (or more) different modes towards higher layers and use services of two (or more) different modes provided by lower layers. In the second case, the layer supports the cooperation of different modes, because it is able to provide mode-independent services towards higher layers, while using services of two (or more) modes provided by lower layers.

8.1.3.5.5 Example 2: Multi-Mode Layer-(N) Relay

This second example in Figure 8-4(b) shows how the basic building block introduced in Figure 2-3 can represent a multi-mode relay. The different functionality of the layer as compared to Example 1 is achieved through configuration by the (N)-MCM. In this case, the layer bridges mode 1 and mode 2 and does not provide services towards higher layers. This way, a (heterogeneous) relay connecting two different air-interface modes can be efficiently implemented, because the functionality in the common part provides an inherent interface for the back-to-back interconnection of the different modes.

8.2 Control-plane functionalities

8.2.1 Winner Handover Procedure

Handover is the process to support terminal mobility, which occurs when a terminal changes the access terminal through which it is communicating. Various solutions have been proposed, but seamless handover – low loss and low latency handover, is still a challenge in mobile telecommunication. The handover process comprises two parts: first, the terminal set up a link layer connection to a new radio station; and second, the communicating terminals update their communication routes. As WINNER might be an IP based mobile system, the mobile IP schemes are briefed in section 8.2.1.1. In section 8.2.1.2, link layer handover procedure is described. Finally the conclusion and future work are identified in section 8.2.1.3.

8.2.1.1 IP Mobility Scheme

IP mobility investigates the IP packets routing updates due to user terminal's mobility. It is generally agreed that IP terminal mobility can be separated into two complementary parts – macro-mobility and micro-mobility and they need two different solutions. The macro-mobility refers to the mobility in a large area, more accurately, handover between different administrative domains (AD). The micro-mobility means that mobility in a small area, or handover within the same AD.

8.2.1.1.1 Mobile IP for Macro-mobility

The Mobile IP (MIP) is well known as a proposal for handling macro-mobility handover. It has been developed over several years at IETF, initially for IPv4 and now for IPv6 as well. In MIP a mobile host is always identified by its home address, regardless of its current point of attachment to the internet. Whilst situated away from its home, a mobile also has another address, called a 'Care-of-Address' (CoA), which is associated with the mobile's current location. MIP solves the mobility problem by storing a dynamic mapping between the home IP address, which act as a permanent identifier, and the CoA, which acts as a temporary locator. The key functional entity in MIP is the Home Agent, which is a specialised router that maintains the mapping between a mobile's home address and CoA. Each time the mobile moves onto a new AP (subnet), it obtains a new CoA and registers it with the Home Agent (HA). Mobile IP means that a correspondent host can always send packet to the mobile: the correspondent addresses them to the mobile's home address – so the packets are routed to the home link – where the home agent intercepts them and tunnel them to the mobile's CoA.

8.2.1.1.2 Micro-mobility Schemes

Micro-mobility assumes the mobility happen within the same admission domain, and the packet has been delivered to the access network's gateway. There are huge amount of work on the micro-mobility problem with many different ideals and protocols suggested. Broadly speaking, there are two ways of dealing with the micro-mobility: mobile IP based schemes; per-host forwarding schemes.

- Mobile IP based schemes. They are characterised by the use of tunnelling, and in general by the mobile acquiring a new CoA each time it moves. Two complementary threads of improvement have been taking place:

  1) Local Mobility Agents

  This scheme assume that mobile IP's problem arise only from the potentially long distance signalling back to the home agent when a mobile moves. The solution to this problem is to introduce a local proxy mobility agent. In this way, when the mobile changes its CoA, the registration request does not have to travel up to the home agent but remains 'regionalised'.

  2) Fast and Smooth Mobile IP based Schemes

  This scheme aims to make the handover seamless, reduction of signalling is not a particularly concern. The most important idea is to use supplementary information to work out that a handover is probably imminent, for example the link layer measurement results (this will be the main focus of Section 4.4.2). Some proactive actions will be taken on behalf of the mobile. The main steps are to acquire a new CoA that the mobile can use as soon as it moves on to the new AR, and to build a temporary tunnel between the old and new ARs, which stops any packets being lost whilst the binding update messages are being sent. These proposals are:

  <u>Two CoAs</u>: it assumes the mobile can listen on two links at once (make before break). Providing a mobile is allowed to hold on to its old CoA for a short period of time after the handover, it can accept packets arriving at the old or new link. This means that when a mobile hands over, packets that are sent before the binding update reaches the home agent will still reach the mobile and be accepted.

  <u>Bi-casting</u>: The bi-casting idea let MH send bi-casting request to HA to bi-cast packets both to the new and old CoA. This has been extended by performing the duplication locally, e.g. at the foreign agent. Additionally, it has been proposed to buffer packets locally during a handover.

  <u>Temporary Tunnel</u>: The basic idea is to establish a temporary tunnel from the previous CoA to the new CoA. Hence, packets coming from correspondent nodes that have not yet been told the new CoA will be forwarded to the mobile.

- Per-host forwarding, which introduce a dynamic layer 3 routing protocol in the access network. In general the mobile keeps its CoA whilst it remains in the access network. These protocols use a new specialised scheme to install per-host forwarding. The general idea is that information is stored in various routers spread through the access network. A downstream packet enters the access network at the gateway; the gateway looks up which of its output ports is the best to use for the particular mobile in question. It then forwards the packet on the selected port towards the 'next hop router'. At that router, the process is repeated. There are two broad techniques for per-host forwarding schemes that have been explored. These schemes are generally concerned with both reducing the signalling load and speeding up handovers.

  1) Cellular IP and Hawaii (Handover-Aware Wireless Access Internet Infrastructure)

Initialisation of the forwarding information is done using reverse path forwarding: when a mobile turns on or enters the access network it sends a packet on the default route to the gateway; each router caches an entry mapping the mobile's identifier (its home address) to the neighbour from which the packet arrived at the node. Thus the downstream packets can be delivered to the mobile simply by following the series of cached mappings associated with the mobile.

In Hawaii, mobile to mobile calls are routed on the most direct route available, i.e. not necessarily via gateway. This is different from what is done in Cellular IP. Also route updates only travel as far as the cross-over router. There are two different route update schemes in case of handover. The 'forwarding scheme' is appropriate when the mobile can be connected to only one AR at a time. It results in downstream data packets being first forwarded from the old AR to the new AR before they are diverted at the cross-over router. However, the 'non-forwarding scheme' is appropriate when the mobile can be connected to both ARs simultaneously. Downstream data packets are diverted at the cross-over router as soon as the path update message reaches it, and so there is no forwarding of packets from the old AR. Because the route updates are directed from the new AR towards the old AR, this means that after several handovers, the path taken by the downstream packets may not be the most direct available.

2) MANET-based Protocol – MER-TORA

MER-TORA (Mobility Enhanced Routing Temporally Ordered Routing Algorithm) builds on the TORA ad hoc routing protocol. In TORA each host and router has a 'height' associated with it. A packet is routed downhill from a source to its destination. The TORA protocol assigns all nodes an appropriate height and then reacts appropriately to any changes in routing topology to ensure that there is still a downhill route to the destination. The height is assigned with respect to a particular destination, i.e. a separate version of the protocol is run for each destination. The 'height' has five elements to it, but in the case of a static network, a node's height is essentially its hop count to destination. A node can have several down hill links to the destination, which means that TORA copes elegantly with meshed network.

8.2.1.1.3 Summary of IP Mobility

It is generally accepted that MIP will be used in macro-mobility, which will use the home agent and tunnel to locate the user when it is not at home network. For above mentioned micro-mobility schemes they are quite similar to each other, the main difference is the terminologies. They all come to the similar point that path updates are localised which only travel between the cross-over router and the old and new ARs. This will reduce the signalling load and also ensures that the path update process is quick.

The requirement of seamless handover is very critical for real time applications, like VoIP. In order to reduce the latency the IP handover procedure should be taken place at the same time as the link layer handover – proactive IP handover. This requires the user terminal to be able to detect the new AR and configure the new CoA while the current link connection is still working.

Soft handover is the best way to do seamless handover, but it requires user terminal to connect with two APs simultaneously and requires very tight timing alignment of traffics between two APs, which is not always possible. Some semi-soft handover scheme has been proposed in mobile IP which requires the user terminal to connect to two APs at the same time, but allow the misalignment of the traffics between them. This can be achieved by bi-casting. Also in order to reduce the packet loss a tunnel between the old AR and new AR should be set up which will forward the packets in-flight to new AR after the old connection has been cut.

8.2.1.2 Link Layer Procedure in Handover

Traditionally, handovers are required within cellular networks to support (seamless) service continuation during terminal mobility. However, the degradation of the link quality to stationary terminal due to increased interference or a traffic congestion-based network decision could also trigger a change of the terminal's wireless AP. Furthermore, a varying customer demand might require switching to a different air-interface with a different radio technique able to provide the envisaged service. This means both mobile and network should be able to initiate the handover.

In conventional cellular mobile system like UMTS, the mobile can make a measurement of radio link signal to noise ratio and make a decision of handover by using handover algorithm based on the measurements. This is adequate because this handover decision is only based on the coverage condition. Furthermore, in a heterogeneous network environment, there are different radio interface modes available and the measured link level signal matrix need to be mapped correspondently in order to make a right decision of handover among them.

Handover in link layer can be separated into 4 steps:

- Link layer triggers.
- Link layer measurement and other information collection.
- Decision making based on different criteria and optimised final decision making.
- Signalling and contexts transfer between different APs (N.B. the necessity of signalling and context transfer depends on the deployment scenario but will always be employed if IP mobility is used).

8.2.1.2.1 Link Layer Triggers

Handover in conventional cellular system, like UMTS, usually comes from two reasons:

1) The radio link quality is worse than minimum requirement or better link connection is available, this could initiate the handover to the better cell.
2) The radio resource in one cell is almost running out but the neighbouring cell has much more free radio resource available, this can trigger a handover to the light load cell.

But in the WINNER system it is foreseen that multiple modes will exist. All these modes could overlap in the same area thus the mobile users have much more choices in terms of radio resources, mobile services, different prices from different operators and different QoS from different AP and networks. All these different requirements can be a trigger to start a handover. Some triggers may happen together. How to prioritise or compromise these triggers is very important in the handover decision making. This will be the main challenge of link layer handover in Winner system.

8.2.1.2.2 Link Layer Measurement and Other Information Collection

Once the trigger happens, the link layer should measure the parameters which define the quality of current link and target link. These parameters could be:

Signal to noise ratio; BER, FER; Signal strength; QoS; Offered data rate of current cell and candidate cell;

Other information concerning the link connection includes:

Price of per bit in the link; User mobility parameters, i.e. speed, direction; Security set up and encryption used in the link; ARQ status of the link; …

The link quality measurement is technology specific, which requires specific process based on the used radio technology. The purpose is to improve the measurement accuracy against the noise and fading effects. Within the same mode the measured parameters are comparable. But the measured parameters from systems operating in different modes may be difficult to relate each other directly. So some mapping function should be developed in order to make these measurements comparable in terms of link quality, QoS, available capacity et. al, which is necessary in the decision making.

Other parameters of the target cell, i.e. the cell load condition, service availability, data rate support capability, cost per bit, are also very important during the handover procedure. The target cell should be able to offer the same service to the handed over terminal which has been offered before; the offered QoS of the target cell should meet the minimum requirement of the user's service profile. The data rate offered from different cells is also important, especially when the terminal is handed over from a cell offering a higher data rate to a cell offering a lower data rate. In this case, the resource reservation and signalling to slow down the packets flow from the sending part to the new AP in advance will speed up the handover procedure and avoid the unnecessary buffer overflow in the new AP.

8.2.1.2.3 Decision Algorithms

Because the triggers of handover are different, the algorithm to make a decision for the handover will of cause be different. Coverage triggered handover will focus on the link connection quality, i.e. signal to noise ratio; QoS initiated handover will focus on the traffic delay and delay variation; network congestion triggered handover will balance the radio resource usage of the neighbouring cells in order to offer better service for the whole area. Price initiated handover will look for cheaper cost per bit. But any of these triggers can not work independently. Link connection quality will be the basic requirement for handover – as people looking for a cheaper service can not survive on a very poor link connection, but may live with a slight worse link connection. A user looking for a wideband service will not accept a low data rate connection, even if it has a better link quality. So location based service availability information, coverage information are very important for decision making.

As a mobile radio link generally exhibits a highly variable environment each change in the radio connection set-up includes the risk of unexpected quality degradation. A common problem is the 'Ping Pong' effect, which comes from the signal fading and relative signal strength variation at the edge of the two cells. The fading effect is not easily to be filtered out especially at low terminal speed.

Location aided handover is assumed to be an effective way to reduce the 'Ping Pong' effects. The coverage map is built gradually through cell operation by using terminals as sensors to measure the radio signal strength with respect to the locations. The measured maps could be for different cells of different modes overlapped at the same geographical area. This information can be used as a handover trigger according to the user's service requirement when users get into a certain area; also this map can aid to the decision making algorithm to reduce the 'Ping Pong' effect caused by the fading, as the map is produced over a long time period and spatially averaged which significantly removes the fading effect. But the location error may still be a problem which could trigger a wrong handover.

To minimise the impact of a handover on service quality the respective algorithm should maintain the minimum QoS of the user profile after the handover. For QoS triggered handover, the improvement of QoS should be obvious after handover. Handover will always cause extra signalling in the network and cause delay in the traffic. Unnecessary handover should be avoided, i.e. those stimulated through the existing radio link and overall network conditions are still adequate or those, resulting in a change to another AP with only negligible better service provision. Therefore, each proposed solution should also cover fall-back mechanisms to ensure service continuation in case of handover failure.

The handover algorithms can be based on different triggers, i.e.

QoS based handover algorithm; Link quality based algorithm; Service availability based algorithm; Price based algorithm.

These algorithms should be supported by the location based information and need to cooperate with each other. As mentioned above, the individual trigger initiated handover can not make a decision based on one respective handover algorithm, as the user's requirements are more complex and need to be considered jointly. It is necessary that on top of these algorithms there should be an algorithm to prioritise and compromise with all decision results derived from the individual algorithms.

8.2.1.2.4 Signalling and Context Transfer

As mentioned previously, if we are suppossed to support IP mobility, fast handover requires layer 3 handover happens at the same time as layer 2 handover, which require layer 3 signalling during layer 2 set up. This requires that while user terminal measures the link layer parameters, it should signal the potential AR to prepare the IP handover, i.e. configuring the new CoA. After the layer 2 handover decision has been made, the link layer context of current AP, like QoS parameters, security parameters, ARQ status information should be transferred to the new AP (N.B. the necessity of signalling and context transfer depends on the deployment scenario but will always be employed if IP mobility is used). These parameters will be used to set up the new radio link without impact on the service quality. The end to end flow control signalling is also necessary to avoid the buffer overflow when the target AP offers a lower data rate. The route update message should be passed to the respective RNs and/or the correspondent node. This should be done before the user terminal detach the current AP.

8.2.1.2.5 WINNER Linker Layer Handover

The WINNER system will operate in different modes, same area might be covered by different modes offering different data rate, QoS, service and prices. Wireless relay will be a significant part in the WINNER wireless network. The RN could also be a point of attachment for a user terminal. The differences between a RN and an AP in terms of measured link quality, QoS, security, et. al. should be studied carefully when making a decision to handover from a AP to a RN.

In order to apply above mentioned steps in link layer handover, the parameter matrix used to describe the link quality should be investigated carefully for different WINNER mode. The measurement algorithms and procedure for different modes should be studied as well. The parameters which are used to define the quality of service of different mode should be defined and mapped with each other. The location based information should be used during the handover procedure. An arbitration algorithm on top of individual handover decision algorithm should be used to prioritise the individual algorithm and optimise the final handover decision. The overall link layer handover architecture is given in Figure 2-4.

8.2.1.3 Conclusion and Future Work

Handover in Winner system is complicated by the multiple handover criterias and multiple handover choices. It is essential to design the algorithm which can derive the indications of link connection quality, QoS, cell load, et al from measurable parameters and compare the parameters among different modes. An arbitration algorithm is needed on top of individual algorithm to prioritise and optimise the decision results. Network delay estimation and traffic synchronisation are fundamental to finally resolve the handover interruption.

As the handover involves many aspects of RRM, i.e. routing, load control, ARQ status handling, flow control, the close cooperation is needed with other partners.

For link layer handover, the indications of link connection quality, QoS, cell load for different Winner modes need to be studied. The measurable parameters from which the indications can be derived should be defined. Finally the decision arbitration algorithm should be developed to prioritise and optimise the decision results.

For IP handover, the latest progress in mobile IP should be followed closely. System delay estimation and traffic synchronisation is essential to reduce the interruption. As the IP handover is very much related to the access network architecture, it is important to work with other partner to define the Winner access network architecture.

8.2.2 A case study for the integration of routing and resource allocation

As a part of our investigation, we studied a preliminary example: Joint Routing and Radio Resource Allocation (JRRRA) for enhanced uplink UTRA-FDD. The main objective of this case study is to investigate whether it is benifical to explore an integrated strategy for routing and resource allocation in a packet-based MCN. The system level simulation results show that the integrated strategy can potentially improve performance in terms of cell throughput.

Noteworthy, this case study does not imply that the routing and resource allocation in WINNER systems should be integrated or centralised. Nevertheless, the conclusions drawn from this case study should certainly be taken into account when defining the interactions/interfaces of the routing and the resource allocation in WINNER systems.

It is worth mentioning that in this study perfect signalling exchange is assumed, i.e., no transmission delays and errors etc. If signalling transmission delays and errors etc. are taken into account, the throughput gains of proposed algorithm given in this section will be affected. Moreover, the centralised architecture of the proposed strategy will inevitably lead to signalling overheads.

8.2.2.1 System Scenario: Enhanced Uplink UTRA-FDD with Fixed relay nodes

The main concerns of this case study are the interactions and coordinations between routing and resource allocation, hence we base our investigation on enhanced uplink UTRA-FDD, whoes routing and resource allocation algorithms are relatively simple: 1) “time slot” selection does not need to be considered during route determination due to the FDD deplex scheme; 2) only a two-dimension resource allocation needs to be considered for “transmission time” and “transmission rate” due to the fact that each user has a uniquely assigned scrambling sequence, thus, normally the spreading code resources occupied by each user do not affect those available for others.

The cell layout is hexagonal grid with 3 tiers (totally 19 omni-directional cells). Fixed relay nodes are assumed to be symmetrically located on the perimeter of a circle centred on the AP. It is assumed that from a user to the AP, maximally two hops could be used, and both hops use WCDMA/FDD, but different carrier frequencies are adopted to avoid the self-interference of RN. Note that most network operators have licenses for more than one carrier frequency, hence above assumption is implementation feasible.

In enhanced uplink UTRA-FDD, the radio resource allocation is managed by NodeB packet scheduling [80][81], which is one of the major enhancements against the uplink of other UMTS versions (which use RNC packet scheduler) for the purpose of faster resource scheduling.

8.2.2.2 Joint Routing and Radio Resource Allocation

A. Introduction

After introducing relay nodes into enhanced uplink UTRA-FDD, a three-dimension resource space is envisioned: “time”, “rate”, and “transmission route”. Based on this, a Joint Routing and Radio Resource

Allocation (JRRRA) algorithm is designed, which consists of two entities: Load Based Routing (LBR) and Transmission Mode Aware Packet Scheduling (TMAPS). These two entities operate synchronously, periodically and with coordinated interactions. The basic procedure at each operation is: first LBR carries out routing to determine user transmission routes, and then TMAPS carries out packet scheduling based on the outcome of LBR to determine user transmission rates and time. Noteworthy, within this algorithm, the decisions on routing affect packet scheduling immediately, whereas the effects of paket scheduling are fed back to routing at the next operation instant. For simplicity, we do not feed back the decisions of packet scheduling back to routing and perform routing again immediately, and so on, and so forth, however this may easily be incorporated in the proposed scheme.

B. Load Based Routing (LBR)

Considering the capacity of uplink UTRA-FDD is interference-limited, thus the most beneficial transmission mode or route for a user is the one inducing the least system interference/load, therefore, we introduce a criterion, namely Load Cost Indicator (LCI), for the assessment of user transmission modes and routes

$$\zeta_{MS_i - RS_x} = I_{RS_x} * L_{MS_i - RS_x} \tag{34}$$

Where $\zeta_{UTi\text{-}RNx}$ is the LCI of the route from user *i* to RN *x*, I_{RNx} is the interference at RN *x*, and $L_{UTi\text{-}RNx}$ is the end-to-end transmission loss from user *i* to RN *x*. Noteworthy, end-to-end transmission loss is the summation of the gains and losses along the path from the transmitter to the receiver, including channel losses, antenna gains, cable losses etc. The LCI of a direct transmission route could be calculated likewise.

The reason for choosing above equation as LCI is shown in the following derivations:

$$\begin{aligned} \zeta_{UT_i - RN_x} &= I_{total,RN_x} * L_{UT_i - RN_x} \\ &= \frac{P_{tx,UT_i}}{\eta_{UT_i - RN_x}} \approx P_{tx,UT_i} * \frac{1 + SIR_{t\,a}}{SIR_{t\arg}} \end{aligned} \tag{35}$$

$\eta_{UTi\text{-}RNx}$ is the load factor of user *i* at RN *x*. $P_{rx,UTi\text{-}RNx}$ is the received power of user *i* at RN *x*. $P_{tx,UTi}$ is the transmission power of user *i*. $SIR_{target,UTi\text{-}RNx}$ is the power control SIR target of the connection between user *i* and RN *x*.

Note that the SIR target of a user normally is fixed given a certain user transmission rate, hence equation (5) indicates that the proposed LCI is proportional to the transmission power, i.e., the bigger the LCI is, the higher the transmission power is needed for delivering same amount of traffic, and thus the more interference is induced to the system. Therefore, the proposed LCI well reflects the load costs of user transmission routes.

Based on (1), for each user, LBR could easily find out and choose the route with the least resulting load at the AP. However, to avoid the ping-pong effect causing unnecessary signalling overhead, a LCI hysteresis (e.g. 2dB) could be applied when updating user routes.

C. Transmission Mode Aware Packet Scheduling (TMAPS)

TMAPS has a similar procedure as the NodeB scheduling algorithm in [81]: firstly, the available AP/cell capacity is estimated; then, users are prioritized based on a certain priority function, e.g. proportional fairness; finally, transmission rate and time are allocated to individual users according to their priorities and needs. Noteworthy, in the final step, before a rate is allocated to a user, load estimation needs to be carried out to see how much AP/cell capacity will be taken if the rate is given to this user.

The major difference between TMAPS and the NodeB scheduling algorithm in [81] is: in TMAPS, the load estimation is enhanced: for users in direct transmission, the conventional load estimation approaches in [81] could still be adopted. Whereas for multi-hop users, the load estimation should be

$$\eta_{UT_i-AP}^{RN_x} = \frac{P_{rx,UT_i-AP}}{I_{total,AP}} = \frac{\eta_{UT_i-RN_x} * I_{total,RN_x} * L}{I_{total,AP} * L_{UT_i-A.}} \qquad (36)$$

$\eta_{UT_i-AP}^{RN_x}$ is the load factor of user *i* at the AP when the user is relayed via RN *x*. $\eta_{\text{UTi-RNx}}$ is the load factor of user *i* at RN *x*. $P_{\text{rx,UTi-AP}}$ is the received power of user i at the AP. $I_{\text{total,RNx}}$ is the total interference at RN x. $I_{\text{total,AP}}$ is the total interference at the AP. $L_{\text{UTi-RNx}}$ is the end-to-end transmission loss from user i to RN x. $L_{\text{UTi-AP}}$ is the end-to-end transmission loss from user i to the AP. $\eta_{\text{UTi-RNx}}$ could still be estimated by conventional approaches in [81] since it is direct transmission from user *i* to RN *x*.

8.2.2.3 Simulation Results

By system level simulations, JRRRA is evaluated and compared with two other cases: non-relaying, relaying with a benchmark algorithm. The benchmark algorithm adopts distance-based scheme to determine user transmission routes [82], and employs conventional scheduling [81] for resource allocation.

Table 8-1 simulation parameters

Parameter	Explanation
Cellular layout	Hexagonal grid, omni-directional sites, 3 tiers (19 cells)
Cell radius *R*	1.8 km
Propagation model	L = 128.1 + 37.6 Log10(R)
Channel Type	3GPP Pedenstral A 3 Km/h
Std. deviation of slow fading	8.0 dB
Correlation distance of slow fading	50 m
User TFCS (kbit/s)	8,16,32,64,128,256,384
RN TFCS (kbit/s)	8,16,32,64,128,256,384,768,1000
TTI	10 ms
Scheduling period / JRPS period	100 ms
Priority	Proportional fairness
Traffic model	Modified Gaming, value set 1 [80]
Number of FRNs	6 per cell
RN-to-AP distance *r*	0.65**R*, 0.75**R*, 0.85**R*
Route update threshold τ	2 dB
Load threshold of packe scheduling	70%
Multi-hop region for benchmark relaying approach	UT-to-AP distance > 0.5**R*

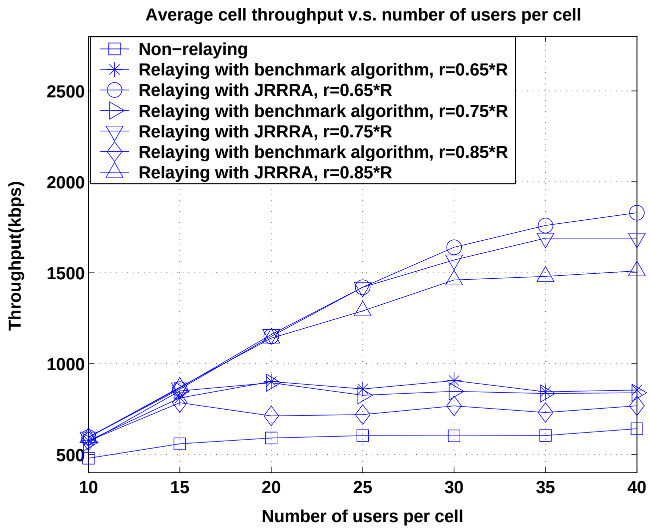


Figure 8-5 average cell throughput v.s. numbers of users per cell

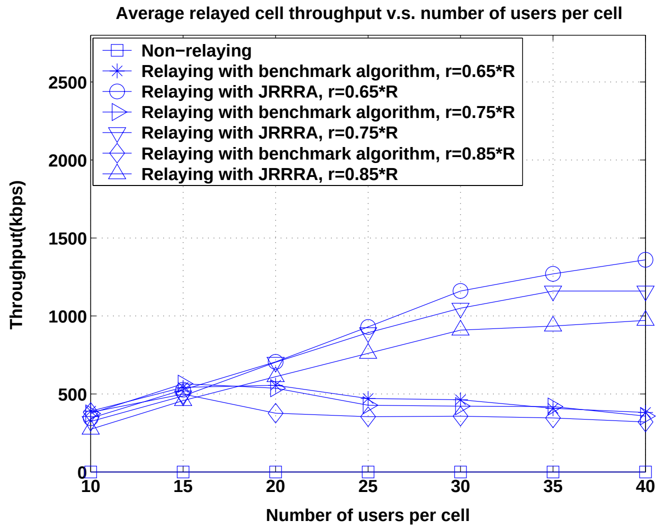


Figure 8-6 Average relayed cell throughput v.s. numbers of users per cell

It could be observed from above figures that when the system is slightly loaded, without the help of relaying, traffic could still be delivered properly. However, when the system load gets heavy, systems with relaying perform significantly better. Moreover relaying with JRRRA could achieve much higher cell throughput than the benchmark approach. This is because: firstly, the benchmark algorithm uses UT-to-AP distance to determine whether a user is eligible to be relayed or not, but distance doesn't exactly indicate whether it is beneficial (in terms of uplink load) or not to relay a user, which implies that alternative routing metrics are needed; Moreover, the interdependency between routing, scheduling and the resulting transmission modes have not been accounted for, and as a result, multi-hop gains could not be effectively captured and translated into throughput improvements. We could also see that the gain of the proposed JRRRA vary with the AP-to-RN distance r, and the maximum cell throughput gain is about 185% obtained when r equals to 0.65*R. This is because currently, we assume users could only be relayed by the same cell RNs, therefore, due to the hexagonal cell shape, when the AP-to-RN distance gets bigger, the number of users around RNs will be less, hence the number of eligible users will be reduced, as indicated by the figures above, when r gets bigger, the relayed traffic reduces.

8.2.2.4 Conclusion and Future Work

From the study above, we could conclude: 1) it could be beneficial to regard user transmission route as one extra dimension of the resource space and carry out routing and resource allocation synchronically; 2) it may be beneficial if resource allocation is aware of user transmission modes (direct or multi-hop) and routes; 3) for a relay-based interference-limited system, the route selection should be based on load/interference costs of individual routes.

8.2.3 Scheduling architecture

8.2.3.1 Motivation for partitioning packet flow control

There are two basic motivations for subdividing packet flow control into service level control followed by a resource scheduler:

1. The total allocation problem is thereby simplified considerably. The service level controller works on the typical time-scale of the packet arrival rates, while the resource schedulers need to work on the much faster timescales of channel resource variations and physical channel slot lengths. It is a well-known principle of process control that control problems that involve significantly different timescales can be structured into several nested simpler control loops, with the inner ones working on a faster timescale than the outer ones. This architecture is called cascade control.
2. The service level controller is foreseen to be located at some distance from the radio APs, such as at the RANG. The service level controller works on a slower timescale and may allocate resources belonging to several modes/APs. The resource schedulers and the resource scheduling queues should always be located close to APs to minimize delays.

8.2.3.2 Generalised scheduling architecture

Figure 8-7 shows the same architecture as previously proposed, but for the case were the control-plane (upper part of the figure) and user-plane (lower part of the figure) functionalities have been separated. This serves the purpose of making it easier to also visualise the uplink and relaying case as much of the control plane functionalities may appear in the same way as for the downlink case, however this is not the case for the user-plane functionalities. One additional difference is that tokens instead of packets are forwarded by the service level controller to the resource scheduler. Each token represents a number of bits, which defines the granularity of the outflow from the upper buffers. The purpose for this modification is twofold: i) the use of a representation by tokens enables the service level controller to be designed independently of the choices at lower levels of e.g. the size of the scheduling unit; and ii) it provides one common approach for both the uplink and downlink direction. This section will briefly outline the proposed scheduling architecture for the uplink and relaying cases.

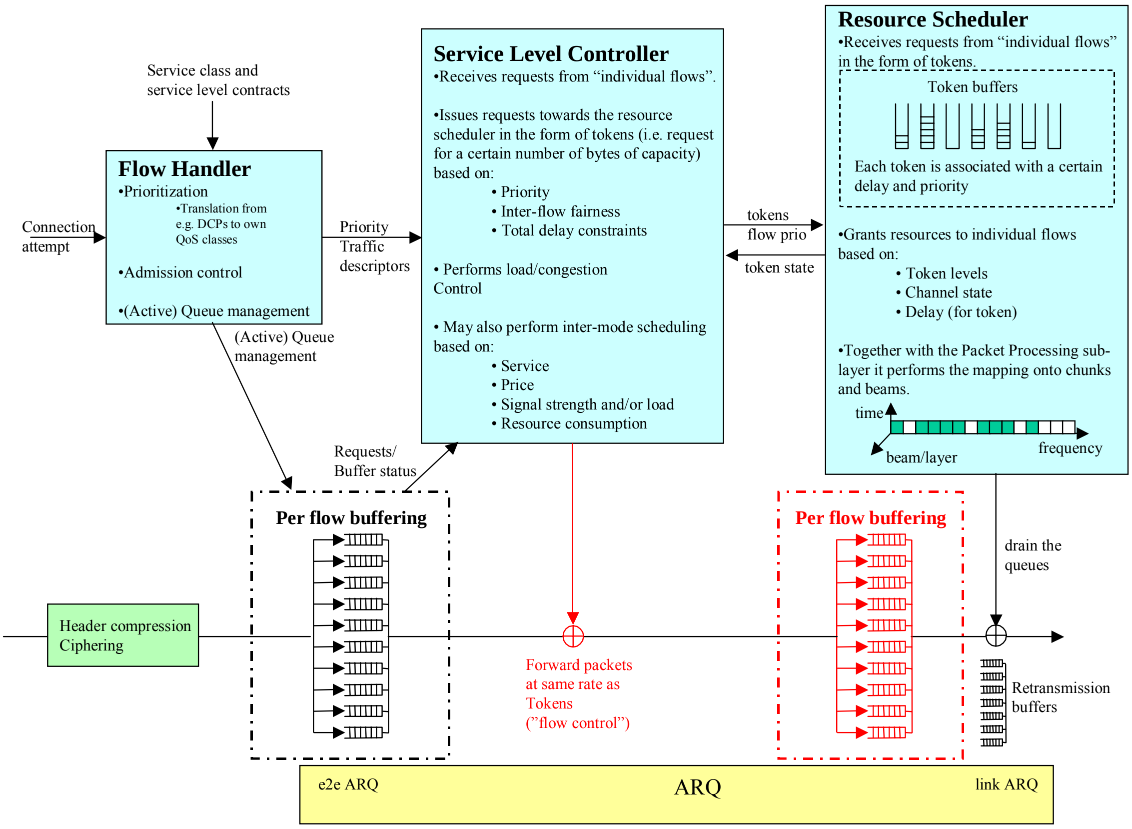


Figure 8-7 Generalised scheduling architecture; Functions marked red may be removed if the service level controller and the resource scheduler are located within the same node

8.2.3.2.1 Scheduling of uplink traffic

The only additions needed to Figure 8-7 to accommodate the uplink case is to provide a division of functionalities in between the network nodes and UT. Two different cases may be foreseen that will result in two different allocations of functionalities – the case were the resource allocation is performed per flow and the case were the resource allocation is performed per UT. Both cases share the fact that all user plane functionality are located in the UT and all control-plane functionality are located somewhere in the network. Nevertheless, in case resources are allocated per UT, the UT also need to implement a service level controller to allocate the given resources to the individual flows. The resulting architecture is given in Figure 8-8.

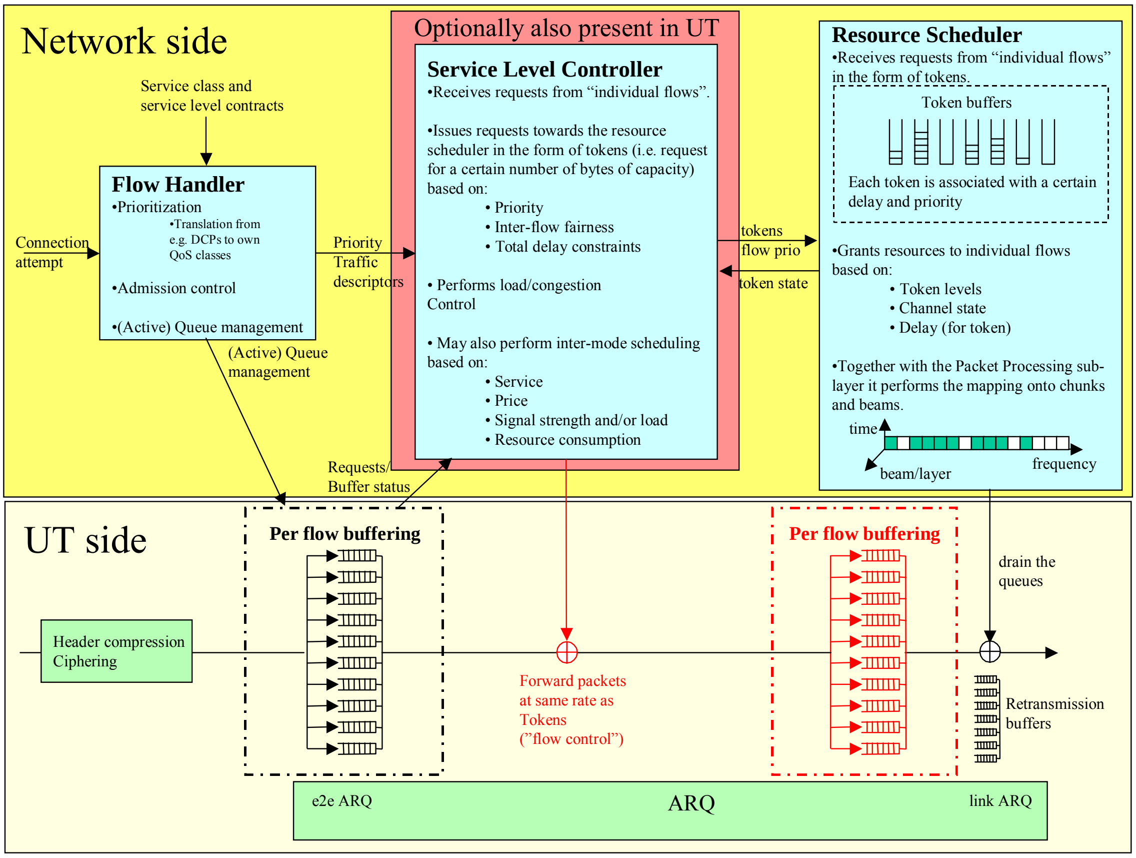


Figure 8-8 Generalised scheduling architecture for the uplink case

8.2.3.2.2 Scheduling of multi-hop/relayed streams

Whether traffic is forwarded using relays or not will not make any difference on the scheduling architecture proposed for the AP and the UT and they will hence not be altered. When performing resource scheduling in a cellular relaying scenario one needs to make a distinction between the different kinds of possible links on the path from the AP to the UT: i) the link between the AP and RN for which the RN appears like a UT to the AP (at least from a scheduling architecture point of view) and hence the RN should perform the actions of a UT as explained above (this link will hence not be discussed further); ii) the link between different RNs; and iii) the link between the RN and UT. For these links several different options on how to perform the scheduling may be envisioned. Two extreme cases are those of completely centralized and distributed scheduling:

1. In the centralized approach all scheduling decisions are made by the AP (as for the uplink case). As for the uplink case one may also here distinguish in between two different cases – the case were the resources are allocated per flow or per node. Consequently, the RN needs only to implement the same functionality as previously described for the two cases for the UT, for both uplink and downlink transmissions.
2. In a fully distributed solution, the uplink node will appear as an AP whereas the downlink node will appear as a UT. This would mean that the RN will have to implement the same functionality as the UT for uplink traffic, whereas for downlink traffic, it would have to implement also the network side in Figure 8-8 except the flow classifier (it would receive this information from the network).

The two approaches naturally has their different pros and cons. For instance, the centralized approach will encounter added feedback delay due to relaying which might complicate adaptive transmission at vehicular velocities since the channel state information may become outdated. This might actually make such transmission impossible. On the other hand, the distributed approach would require more complicated RNs, whose functionality would resemble those of regular APs.

This leads us to believe that a good strategy for scheduling for multi-hop networks is a combination of central and distributed scheduling, that work on different timescales. Nevertheless any such combination may be traced back to the two extremes given above and hence to the proposed scheduling architecture.

One potential idea would be to let the AP partition the resources in the network in between AP, RNs and UTs (over a longer time-scale). To reduce the overhead on the radio links, the final resource scheduling is performed by the RNs (i.e. the RNs schedule the resources they have been assigned by the AP on a shorter time scale, which includes selection of time slots, power control etc[18]).

8.2.3.3 Overall partitioning of time-frequency-spatial resources

The total resources, in the form of chunks and beams/spatial channels within one slot are suggested to be partitioned into three non-overlapping sets:

1. One set is shared by non-adaptive and adaptive resource schedulers for scheduled data channels.
2. One set is used for the other transport channels.
3. One final set of resources within a cell is off-limit to the schedulers. It is used for reuse partitioning between cells, for interference avoidance scheduling between cells, and for interference avoidance between sectors/beams of the same access point.

Within set one and two one also has to reserve resources for relay links (however it is still an open issue whether these resources may be (partly) overlapping, not only with respect to other relay links, but also to resources used by the AP). The parameters that may influence how this partitioning is performed are e.g. duplexing method, multiple access method, deployment scenario etc.

It is envisioned that the allocations of resources to these sets may be changed on a slow time scale (fractions of seconds) according to the traffic conditions and mustbe reported to the terminals. The principles for this partitioning, and the time-scales over which it is readjusted, are important for the overall efficiency of the WINNER system.

8.3 User-plane functionalities

8.3.1 Retransmission Protocols

8.3.1.1 WINNER ARQ Framework

8.3.1.1.1 Example of messages exchanged using a layered solution

Figure 8-9 outlines an example that highlights the drawbacks when using a layered ARQ solution for error recovery (here it is assumed that a L2’ PDU fits exactly into one L2 PDU, if this is not the case the overhead is even greater). The detailed description of this figure is given below.

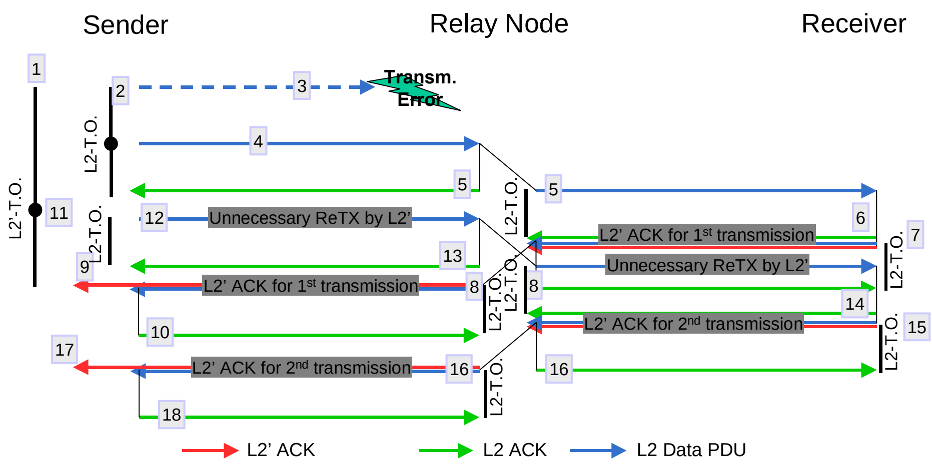


Figure 8-9 Example of messages exchanged using a layered solution with independent ARQ timers

[18] Central scheduling by the AP of these parameters is in principle possible if very low complex RNs are important, but each additional hop generates additional overhead. Especially for long MH-chains this might be quite significant. Due to delays, response time to varying channel conditions would be long, especially in long MH-chains.

1. The L2’ transmitter on the sender side forwards one packet (L2’ PDU) to L2 and starts the timeout timer L2’-T.O.
2. The L2 transmitter sends the packet (as a L2 PDU) to the RN and starts a timeout timer as well L2-T.O.
3. The packet is lost on its way to the RN.
4. Since no acknowledgement arrives at the L2-ARQ transmitter, the L2-T.O. times-out and the L2 PDU is retransmitted.
5. The L2 PDU arrives at the RN and triggers a L2 ACK. The L2 PDU is delivered to the second L2 ARQ and sent to the receiver as a L2 PDU.
6. The L2 ARQ at the receiver successfully receives the L2 PDU and replies with a L2 ACK.
7. The L2' ARQ receiver creates the L2' ACK and sends it to the RN (encapsulated in a L2 PDU)
8. The RN L2 ARQ receiver acknowledge the PDU containing the L2' ARQ ACK and forwards it to the sender (in another L2 PDU)
9. The sender L2 ARQ receiver successfully receives the L2 PDU and forwards the L2' ARQ ACK to the L2' transmitter which is happy now!
10. The L2 PDU is acknowledged.

In the meantime, the L2' retransmission was sent in a similar way (although not necessary, the L2 ARQ just recognizes a new L2' PDU that it has to transmit).

11. The L2' timer at the sender times out as the L2' ACK did not arrive in time. So, the L2' transmitter retransmits its L2' PDU and restarts the timer.
12. The retransmitted L2' PDU is sent as a new L2 PDU to the RN.
13. The L2 PDU arrives at the RN and triggers a L2 ACK. The L2 PDU is delivered to the second L2 ARQ and sent to the receiver as a L2 PDU.
14. The L2 ARQ at the receiver successfully receives the L2 PDU and replies with a L2 ACK.
15. The L2' ARQ receiver creates the L2' ACK and sends it to the RN (encapsulated in a L2 PDU)
16. The RN L2 ARQ receiver acknowledge the PDU containing this L2' ARQ ACK and forwards it to the sender (in another L2 PDU)
17. The sender L2 ARQ receiver successfully receives the L2 PDU and forwards the L2' ARQ ACK to the L2' transmitter which recognizes it as a duplicate ACK!
18. The L2 PDU is acknowledged.

8.3.1.1.2 Examples of messages exchanged using a layered solution with coupled ARQ timers

Figure 8-10 shows an example of messages exchanged when using a layered ARQ solution with coupled L2 and L2’ timers (here it is assumed that a L2’ PDU fits exactly into one L2 PDU, if this is not the case the overhead is even greater). The detailed description of this figure is given below.

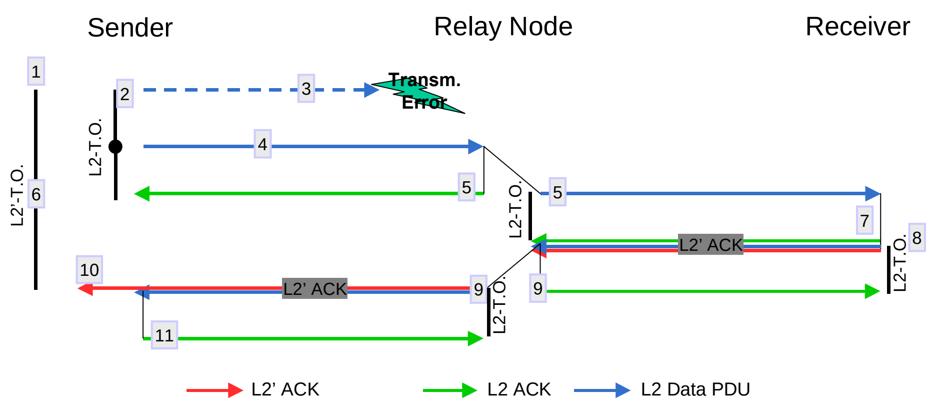


Figure 8-10 Example of messages exchanged using a layered solution with coupled ARQ timers.

1. The L2’ transmitter on the sender side forwards one packet (L2’ PDU) to L2 and starts the timeout timer L2’-T.O.
2. The L2 transmitter sends the packet (as a L2 PDU) to the RN and starts a timeout timer as well L2-T.O.
3. The packet is lost on its way to the RN.
4. Since no acknowledgement arrives at the L2-ARQ transmitter, the L2-T.O. times-out and the L2 PDU is retransmitted.

5. The L2 PDU arrives at the RN and triggers a L2 ACK. The L2 PDU is delivered to the second L2 ARQ and sent to the receiver as a L2 PDU.
6. The sender receives the L2 ACK and restarts the L2' timeout timer.
7. The L2 ARQ at the receiver successfully receives the L2 PDU and replies with a L2 ACK.
8. The L2' ARQ receiver creates the L2' ACK and sends it to the RN (encapsulated in a L2 PDU)
9. The RN L2 ARQ receiver acknowledge the PDU containing the L2' ARQ ACK and forwards it to the sender (in another L2 PDU)
10. The sender L2 ARQ receiver successfully receives the L2 PDU and forwards the L2' ARQ ACK to the L2' transmitter which is happy now!
11. The L2 PDU is acknowledged.

8.3.1.1.3 Details of Relay-ARQ for error recovery

The various states in the transmitter- and receiver- windows as well as in the status messages are explained in Table 8-2, Table 8-3 and Table 8-4.

Table 8-2 The 3 states in a status message.

Value	Description
NACK	Not received by the next node
RACK	Received by the next node but not yet by the Receiver
ACK	Received by the Receiver

Table 8-3 The states in a transmitter window.

Value	Description
ok	PDU acknowledged by receiver
rok	PDU acknowledged by peer (not the receiver)
err	PDU negatively acknowledged by peer (or receiver)
ns	PDU not yet sent

Table 8-4 The states in a receiver window.

Value	Description
ok	PDU received locally / PDU acknowledged by receiver
rok	PDU received locally / PDU acknowledged by peer
err	PDU not received correctly locally / PDU nack'ed by peer

As for the previous cases we will now give an example (see Figure 8-11) where a L2PDU is lost on the first hop link. Nevertheless, before we may give this example we need to specify how the L2 timer should be used as we now only have one timer that should reflect both the link and end-to-end connection. To be able to do this we need two initial timer values, a short and a long one. The short value specifies the interval within which we expect any feedback from a peer. The long value describes the time within which we expect the final ACK.

1. The sender transmits one packet to the RN and starts the L2 timeout timer (short value).
2. The packet is lost on its way to the RN.
3. No (R)ACK arrives at the ARQ transmitter; the timer times-out and the PDU is retransmitted. At the same time the L2 timeout timer is reset to the short value.
4. The PDU arrives at the RN and triggers a RACK. The PDU is delivered to the receiver and another timer (short value) is started locally in the RN (in case of a packet loss on the next link).
5. The sender receives the RACK and restarts the L2 timeout timer (long value).
6. The receiver successfully receives the PDU and replies with an ACK (here the acknowledgement is just a normal control message, not payload).

7. The RN receives the ACK and forwards it to the sender.
8. The SenderNode gets the final ACK and is happy.
9. The sender replies with an ACK for the ACK to ensure that the RN node knows that the sender got the final ACK and that it may finally release the PDU from its queue.

From the figure one may conclude that this approach generates significantly less messages than with layered ARQ.

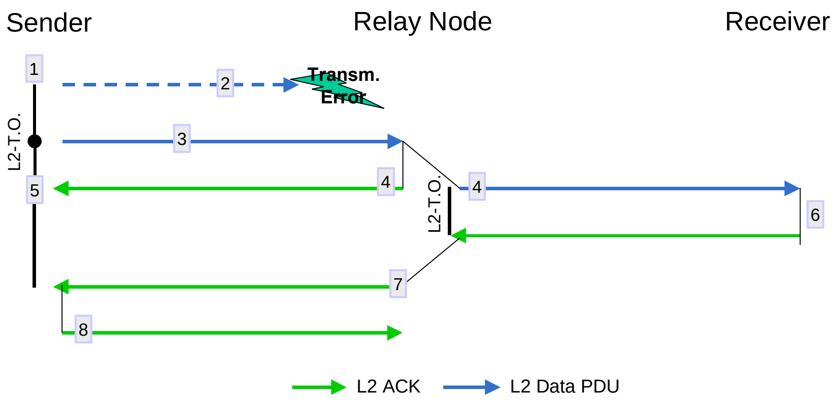


Figure 8-11 Example of messages exchanged using Relay-ARQ.

In Table 8-5 and Table 8-6 the simple rules that the sender and RNs must apply when they receive a status message from a peer node are summarized. Please note that only rows one and two and five to seven are needed in a two-hop scenario.

Table 8-5 Sender action for the different status combinations, upon reception of a status message

Value	Description
NACK	Retransmit to peer/receiver if PDU was received locally
RACK	Do nothing but keep element in window
ACK	Mark as acknowledged and remove from window

Table 8-6 Relay action for the different status combinations, upon reception of a status message (static N-hop scenario).

Incoming Status	RX-window	Outgoing Status	Description / Action
NACK	Not received (ERR)	NACK	Peer has sent a NACK for a PDU that this node has not received yet. => Forward the NACK and transmit the PDU as soon as it has been received locally.
NACK	ROK	RACK	Peer has sent a NACK. PDU is available locally. => Transmit the PDU to the peer
RACK	ROK	RACK	Peer has sent a RACK. Forward that RACK in an outgoing status message.
RACK	Not received (ERR)	RACK	The Peer has received (e.g. over a different path) this PDU, which this node does not have in its local buffer. Forward the RACK (and do not expect to get this PDU locally)
ACK	ROK	ACK	Final ACK by Receiver. Set local tx-window to ACK and remove PDU from buffer.
ACK	ROK / ACK	ACK	Consecutive Final ACK. Forward "Final ACK" and do nothing else
ACK	Not received (ERR)	ACK	The Receiver has received (e.g. over a different path) this PDU, which this node does not have in its local buffer. Forward the ACK (and do not expect to get this PDU locally)

This section will be concluded by outlining the steps taken when changing routes (e.g. the UT moves to another RN and a new RN is added/removed from the path).

8.3.1.1.4 Changing RNs

Independent of whether the UT decides to use another relay, or the decision is taken centrally by the AP, the communication should seamlessly continue with a minimum loss of data. For ease of presentation this section will assume that there are only two hops in between the sender and receiver, however the procedure would be the same for the case were more than two hops are used in between the sender and receiver.

As soon as the RN receives the first packet from the sender, the RN starts to build up an ARQ-state (i.e. the proposed retransmission protocol relies on soft-states) and forwards the L2 PDU to the receiver. The receiver in response sends a status describing what it has received according to the rules described above. The receiver always sends the status to the RN from which it has received the last data packet.

The RN subsequently up-dates the state based on the status from the receiver. Furthermore, it generates its own status report representing the state of the receiver and the receiver-window in the RN (as outlined above). At this point in time the RN informs the sender about data which is missing in the terminal, but which has not been received by the relay before, because the old relay was in charge of those PDUs. The sender consequently responds with the appropriate retransmissions.

In summary, these steps lead to a seamless handover from one RN to the other by exploiting the ARQ state of the other two involved nodes. Effectively, the sender delegates the responsibility for retransmissions to the RN as soon as the RN has sent a RACK for a particular packet. If the RN becomes obsolete, the sender takes the responsibility back.

As an extension of what has been outlined above, one may also envision the case where the data transfer may switch back and forth between two RNs or that two or more RNs are used in parallel to increase the available data rate. Nevertheless, the basic concept can be applied to this scenario as well. Preferably, status messages are sent to all RNs a receiving node is directly connected to. This increases the probability that the status information can be passed on to subsequent RN or the sender node.

8.3.1.1.5 Adding/Removing RNs from the path

Similar to the previous case, a RN might be added in the link chain to increase the overall performance, e.g. due to mobility. When a RN is added the procedure is very similar to the procedure described in the previous section. The added node starts to build up the ARQ-state by interpreting the acknowledgements coming from the subsequent node and by receiving packets from the previous node.

When a RN is removed from the path, the previous node on the path takes over the responsibilities of the dropped node. As for the previous cases, the first status messages on the new path are used to synchronize the nodes that were adjacent to the removed relaying node.

8.3.1.2 ARQ protocol design parameters

The link layer functionality could be placed in the protocol stack as illustrated in Figure 8-12. The ARQ function provided by RLC-g and MAC-r depends on many factors, such as the physical layer, the application requirements, and the ARQ parameterization. In this section, some aspects on ARQ parameters are discussed in relation to the ARQ approaches.

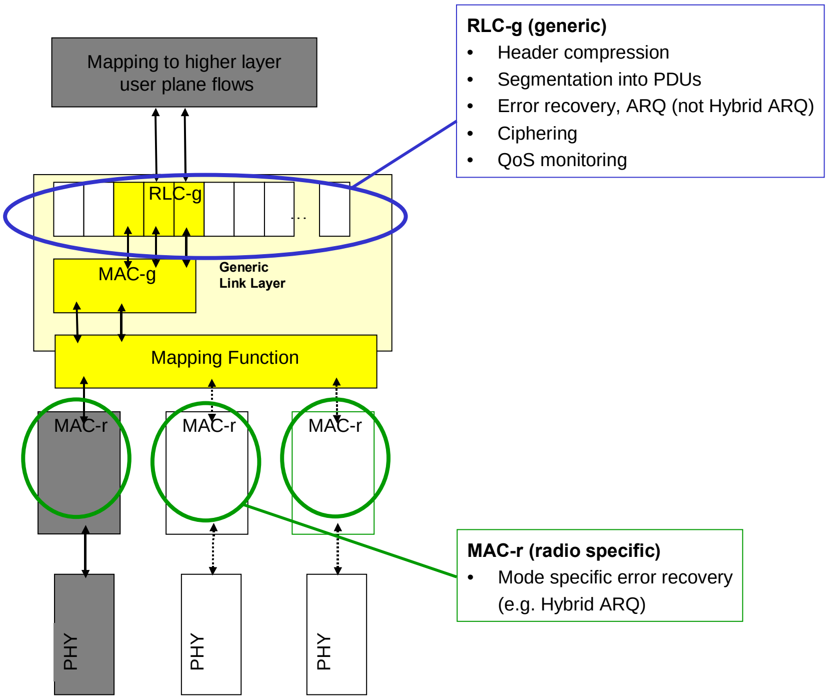


Figure 8-12 Mapping of user plane functionality

8.3.1.2.1 Segmentation and Reassembly

In the RLC-g layer (end-to-end ARQ level) the unit of retransmission should correspond to the submitted data, e.g. an IP packet in most cases. The data submitted to RLC-g in the UT and in the RANG are likely to be divided further into mode specific MAC-r blocks, that are optimized for the channel allocation in the PHY layer, before transmission over the wireless link.

In the Relay-ARQ approach, the retransmission unit is typically an IP-packet both hop-by-hop and end-to-end. By this, no extra delay and buffering are introduced due to segmentation and reassembly. One problem that remains to be solved, is how to handle the variable sizes of IP packets and also the overall efficiency of the protocol when large IP packets are transmitted.

In the layered ARQ approach, as compared to the Relay-ARQ approach, segmentation and reassembly in the MAC-r layer introduce delay and place requirements on buffering. Furthermore, sequence numbering must be managed on two layers, by RLC-g and MAC-r.

8.3.1.2.2 Retrasmission timer

In the layered ARQ approach, both layers must manage a retransmission timer. The timer settings are dependent on the modes used over the hops. The RLC-g timer should not expire before the MAC-r timer. The setting of the MAC-r timer could have a default value for each mode and the RLC-g timer default values depending on the possible combinations of modes over the links involved. The timer settings may be hard to adjust, since large values may result in long delays and short values in unnecessary retransmissions due to timeout. Therefore, estimation of round-trip times on each link, should be possible to use as side information for adaptive timer settings.

In the Relay-ARQ approach, the ARQ functionality of RLC-g and MAC-r is performed with an integrated interlayer communication. This implies that it might be easier to avoid unnecessary retransmissions due to timeout, since only one retransmission timer is required. The timer is set to different values depending on the type of the next expected ACK (RACK or ACK).

8.3.1.2.3 Level of persistence

The WINNER system is intended to be a flexible multi-service access network, including delay-sensitive real-time services. In [94] the authors notice that in order to efficiently support TCP flows, the delay has to be kept small especially for short-lived flows due to the rate-adaptation in TCP. However, there will always be applications with high requirements on reliability for short packet transmissions. Thus, it is foreseen that the degree of persistence of the ARQ scheme has to be adjustable, both to the different traffic classes and also to the reliability of the PHY layer at each link.

8.3.1.3 Envisioned Configuration in Different Deployment Scenarios

The basic characteristic of the WINNER RAN will be the multiple operating modes and deployment scenarios. Therefore, a common ARQ scheme through the WINNER RAN might not offer optimum results. Herein we investigate how to optimise the WINNER ARQ framework for different deployment scenarios.

As already mentioned in section 2.4.2.1, the main retransmission problem arises due to the handover of the UT not only between different APs but also between different RNs. In particular, the problem arises on the downlink as the receiver is moving and the path changes. For example in a 2-hop scenario with a RN between the UT and the AP if the data units are first transported to the RN and acknowledged by it, the corresponding AP will assume that all transferred data units have been successfully received and therefore it will release them from its transmitter buffer. However, if the UT moves and establishes a connection with another relay then some of the unacknowledged data units may be left in the old relay and hence, the UT might request retransmission of data units that are neither in the AP nor in the new RN. Therefore, end-to-end retransmission is required. This can be achieved either by buffering the data in all intermediate nodes and release them only when an acknowledgment is received from the destination or by applying two layers of retransmission.

Nevertheless, in the uplink direction the same issue does not arise since even if the sender (UT) is handed over, the path to the receiver is already established and the next hops should be able to forward the data units. Hence, a simpler retransmission scheme could be applied whereby the retransmission window advances and buffered data of a node are released upon the receipt of an acknowledgment from its next hop. That way the signalling and buffering requirements are minimised. Another advantage of this approach is the fact that no acknowledgement from the destination is required to be forwarded to the sender (UT) and therefore in case of a handover the problem described for the downlink does not arise. However, an important requirement for the employment of this scheme is the reliability and functionality of the RNs. In particular, if the RNs fail often then this scheme will result to loss of data which TCP may mistakenly interpret radio interface errors as congestion and therefore unnecessarily reduce the transmission rate. Therefore, a periodic assessment of the RNs might be needed for each deployment scenario by a management entity.

An interesting question regarding the layered approach is the termination points of the end-to-end retransmission scheme. As the UT may perform handovers between RNs that are associated with different APs appropriate termination points for the end-to-end retransmission scheme seem to be the UT and RANG. The main principle of the end-to-end scheme is that the RANG always sustain the data units in its buffer until the UT has acknowledged them. Observe that the proposed end-to-end retransmission protocol is envisioned to co-exist with the single hop scheme to enable fast and frequent retransmission. The lower layer protocols are thus terminated in AP, RN and UT whereas the upper layer protocol is terminated in the UT and in the RANG.

However, as the number of hops and the frequency of lower layer retransmissions are not necessarily known by the upper layer entity, the actual instances of data units arriving at the destination node are hard to predict. It is therefore difficult to configure the upper layer retransmission protocol. Additionally, as the number of involved lower layer entities in the data transfer path may also change based on the routing and handoff decision, it can be difficult for the upper layer entity to discover whether some missing data unit is actually lost or if something has changed in the data transfer path (resulting in varying “end-to-end” delays). It is therefore important to assess the problem of unnecessary retransmissions and other negative interactions between the protocol layers.

In order to minimise the end-to-end delays and uncertainties due to the unknown number of intermediate nodes two possible solutions could be studied. One solution could be to estimate the end-to-end delays and then adapt the upper layer retransmission timers accordingly. Alternately, we could allow the transfer of the upper layer termination point to different types of network nodes according to the specific topology and deployment scenario.

In particular, the requirement for a successful retransmission during handover is that the ARQ termination points should be one hop further than the handover point so as to avoid any loss of data. As the UT may handover between RNs or APs we propose that the ARQ termination points should be capable of being transferred from between different types of nodes such as an AP and the RANG according to the deployment scenario. In order to be able to act proactively to a handover we could associate the termination point transfer process with the triggers/measurements that identify handovers. This will require cross layer signaling in case the triggers are initiated at higher layers e.g. triggers based on QoS, UT location, velocity, network availability. As soon as a trigger for a handover is activated, the trigger aware node notifies the corresponding termination point by sending a message containing new ARQ information in order to terminate the current ARQ instance. Also, a decision process may be used to make the decision regarding the next termination point. Criteria to be used might include type of deployment scenario, number of relays, type of relays, relay ownership etc. Additionally, the current termination point sends a message to the new termination point to inform the node of the current state of the ARQ instance. This new message may include information including the UT id, current termination point node id, current sequence number etc. In order to make this process more efficient the ARQ scheme used during the handover duration could be a different one than the normal operation. For example, by employing an ARQ mechanism that requires less buffering, less bandwidth is wasted during the transfer of the ARQ termination point. Figure 8-13 shows the operation flow of the proposal.

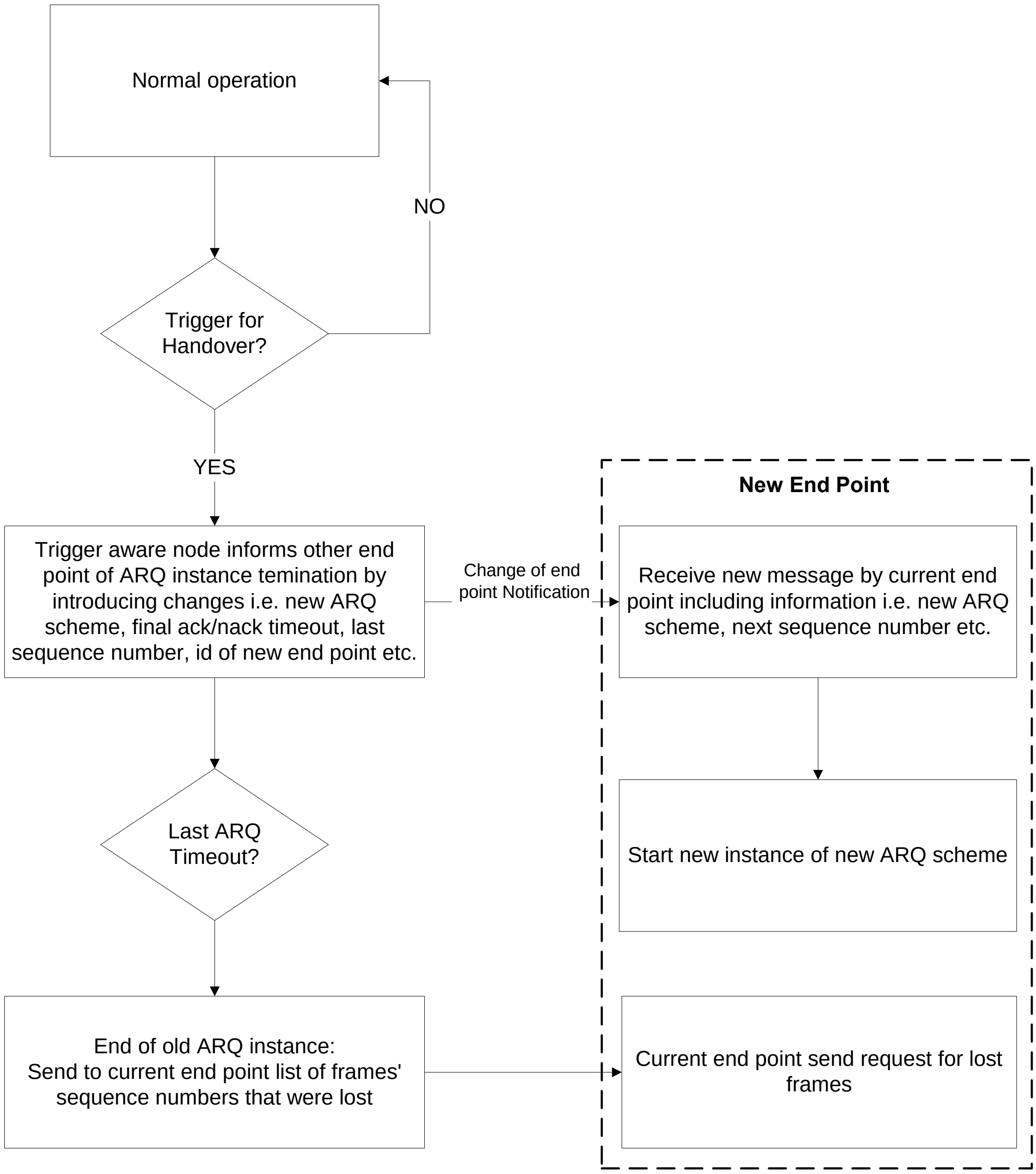


Figure 8-13 Flow diagram of ARQ transfer operation

Another particularly interesting and promising proposal within the WINNER framework is to exploit the concept of cooperative relaying. Here multiple receiving/transmitting RNs are involved in the reception/transmission of data in a similar manner as multiple APs are used during a soft handoff. This means that the data can be transferred through multiple and parallel paths simultaneously. In that case, several RNs close to the transmitting node (that can be either AP or UT) can be, simultaneously and independently of each other, involved in the reception/transmission of the same data unit. Moreover, the destination node (either AP or UT) could be also able to combine the signals from different sources (or relay nodes) exploiting useful information that is unnecessarily discarded in conventional relaying.

In the case of layered retransmission and conventional relaying if negative acknowledgements are received from several lower layer entities then retransmissions are performed until individual and independent receiving nodes collectively acknowledge all data units. However, for cooperative relaying, none of the individual nodes has necessarily received all data units. Instead there is a set of separate and independent receiver buffers (in different relaying nodes) where it is possible to find all transmitted data units. A crucial assumption is that the individual receiving nodes may relay the data further towards the destination node and therefore it is not necessary that all individual nodes receives all data units. If the relaying however fails the end-to-end retransmission scheme can recover the error.

Another retransmission proposal for cooperative relaying introduces distributed HARQ. Such a protocol requires two phases, at least initially, while further improvement can be achieved by avoiding the second phase if the destination has successfully decoded in the first one. Using feedback from the destination, additional information is provided by the relay only upon explicit request. An example can be defined as follows: the source transmits half of its message during the first time slot of duration 2 in a broadcast manner to the relay(s) and destination. Relay(s) and destination then both try to decode the message and send feedback on their decoding status in a broadcast manner to the other nodes. Consequently, all nodes have all necessary information to act appropriately during the (optional) second time slot. In case the destination was not able to correctly decode the source message while the relay was able to decode it, the latter will send additional redundancy in the second time slot. The destination then assembles the complete codeword and retries decoding. In case of decoding failure, a block error is declared. The same occurs when destination and relay simultaneously fail to decode.

9. Annex II: Examples of "Architectural Harmonization" of Deployment Concepts

9.1 HSDPA & WiMax

HSDPA-Enhanced Uplinkand WiMax are important and promising short-term evolutions in the field of wireless networks.

At a first stage, the two technologies can be seen as complementary: being HSDPA-Enhanced Uplink a packet oriented technology ensuring full mobility, and WiMax a broadband fixed wireless access (packet-oriented) ensuring at most portability (802.16-2004). In very simplified terms, HSDPA-Enhanced Uplink will be better than WiMax in terms of mobility, while WiMax can potentially ensure higher capacities.

It is therefore probable that an independent coexistence will happen initially; more likely, by different operators, HSDPA-Enhanced Uplink by Mobile Network Operators (MNO), WiMax by Fixed Network Operators (FNO) and/or Internet Service Providers (ISP), not excluding the case of MNO anyway interested in WiMax too.

However, at a second stage of development, WiMax will evolve to 802.16e (associated to 802.16g) and will become capable of supporting "nomadicity" (session continuity while the user terminal is moving), and potentially even mobility. This fact clearly leads to a more complex and interesting relationship between the evolution of mobile and fixed wireless networks, i.e., in the relationship between the evolution of 3G on one hand and evolution of WiMax on the other hand.

Some preliminary qualitative remarks about these relationships are here summarized:

- FNO/ISP may decide to deploy WiMax in order to offer WDSL-like applications, mainly for residential customers (without excluding "niche cases" for business customers too). This may be true for FNO/ISP in developed countries to complete their broadband data access infrastructure in both urban and rural areas; or to cover extended hot-spots in urban areas (not only indoor); moreover, for FNO/ISP in non completely developed countries WiMax might become the main broadband access option. Fixed applications will evolve to portability already in the context of 802.16-2004, when laptop-integrated user terminals will be available. A strong point for MNO/ISP will be the possibility of reusing to a large extent their IP-based core network, to which WiMax access points are connected as a new access technology. The critical points for FNO/ISP are the issues, such as the necessity of sites and of the related logistics and backhauling infrastructure; and the requirements for frequency band. Thus, the forecast for deployment plans by FNO/ISP should be cautious; nonetheless, the WiMax deployment by FNO/ISP is considered highly probable.
- Let's therefore assume that in a medium term perspective, some FNO/ISP will have their WiMax network deployed, for fixed broadband access mainly. In this case, the evolution towards mobility achieved through 802.16e will be extremely interesting, allowing potentially FNO/ISP to address at least partially the mobile market. Obviously, this possibility may have a heavy impact, and must therefore be taken into account; although, it should be on the other hand cautiously evaluated. There are several critical questions, such as: how easy will be the upgrade from 802.16-2004 to 802.16e? Which kinds of upgrade in the core network will be required? Will a "controller node" above the WiMax access points be required? These are still open issues.
- Also MNOs may be interested in WiMax 802.16-2004, to offer full mobile and nomadic services in a complementary way. The interest will be in principle stronger for MNO that do not (or not yet) plan to deploy HSDPA and Enhanced Uplink, which might try to bundle cellular mobile (mainly voice-oriented) services and wireless nomadic high-capacity data services. On the contrary, the evolution towards 802.16e may be seen as a potential danger. Focusing on 802.16-2004, a strong point of MNO is the availability of sites and logistics for cellular coverage; this advantage should be checked against the degree of similarity of WiMax and 2G/3G coverage characteristics, which could more or less enable co-siting of mobile and WiMax access points. Moreover, it should be considered the extent of the necessary upgrade of the backhauling infrastructure. On the other hand, an item to be studied in detail (possibly critical) is the degree of possible integration between the WiMax access network and the MNO core network; different options are currently under study, from feasibility and opportunity viewpoints.

From the above remarks, it is evident that both synergic coexistence and competition scenarios, related to the relationship of 3G and WiMax evolutions, are possible.

A final remark is on the "asymptotic" convergence of the B3G and WiMax evolutionary lines, from the viewpoints of technology principles: OFDM-based air interface, packet-oriented, IP-based network (IP backbone, IP protocols), advanced topologies (mesh, wireless multi-hop).The new solutions developed in WINNER should on one hand take the best of both evolutionary lines, from a specific technical viewpoint. At the same time, the final optimized WINNER solution should be properly positioned with respect to the scenarios outlined above, in terms of time perspective and of network deployment.

9.2 Feeders & Radio Remotization & Multi-Hop concepts

If an overall view on the considered deployment concepts is assumed, the access network infrastructure is in principle partitioned in the following segments:

- **Feeder segment**: it is a "transport network segment", i.e., related to the fixed connections between different network nodes, and does involve UTs. Typically, it supports the connection between a Base Station Controller and a Base Station. The feeder segment may be supported by different backhauling solutions. Several different transmission technologies may be used for the feeder segment. If wireless feeder technologies are adopted, this implies that the overall access network is composed of the cascade of two or more wireless stages. Two examples are: i) the use of microwave links to feed 2G/3G base stations and ii) a WiMax system used as feeder for several WiFi hot-spots (in a scenario sometimes called "hot-zone). This case is for some aspect similar to the "heterogeneous multi-hop" concept; however, it should be highlighted that in this case the two segments, although both wireless, are completely different and distinct from a functional viewpoint; no interaction is foreseen between the two wireless segment; there is no conceptual difference in the network architecture and functionality with respect to the case of wireline feeder.
- **Radio Remotization segment**: this is conceptually the remotization of the RF part of the access point; it can be seen as a link that remotize an "internal" interface of the access point itself. Due to the capacity required to remotize the radio part of the access point, an optical link is the only feasible option, leading to the "digital" Radio over Fibre (RoF), like the one currently standardized in CPRI (Common Public Radio Interface). The advantages of this solution may be the possibility of concentrating (hotelling) the central part of the access point; of improving possibilities for fast coordination between access points (sites) and simultaneously, to optimize the deployment/installation of the radio part, thus offering a further degree of freedom for the radio planning. A RoF like solution is particularly interesting in case of pico-base-stations.
- **Multi-Hop wireless segment:** this is the focus of WINNER interest, therefore it is important to highlight the conceptual difference with respect to the previous cases. In this case, the optimization of coverage is achieved "downside" with respect to the access point, not "upside", as in the previous case. The major advantage of this solution with respect to the previous one is the flexibility in the radio resource assignation, and the possibility of implementing mesh architectures. Moreover, from a practical viewpoint, it obviously does not require a fibre optic infrastructure.

The mentioned segments may in principle coexist in one access network deployment**.** Thus, the multi-hop deployment has the further advantage of being complementary with respect to other solutions emerging in the landscape of access network evolution.

On the other hand, several other combinations are possible: wireline feeder plus single-hop wireless; wireline feeder plus multi-hop wireless; wireless feeder plus radio remotization plus single-hop wireless; etc. The higher priority scenarios from the WINNER viewpoint are the wireline (or wireless) feeder plus multi-hop wireless. The combination "wireline feeder plus radio remotization plus multi-hop wireless" should be also studied in more detail, because it brings together different evolutionary lines.

9.3 IEEE 802.16 PMP and Mesh coexistence in multi-hop wireless concept

The Mesh air-interface can be applied to the backbone of a wireless network with a limited number of connections. Several User Terminals (UTs) shall be served by one node of the backbone network. For this purpose, the PMP air interface can be applied on the last hop towards the UT since it performs quite well for many single-hop connections. In addition, even the PMP mode with some proposed modifications can be used for a multi-hop connection, but the current version of the standard does not support this mode of operation. However, it is worth to investigate this opportunity by the combination of different protocol elements of the PMP and Mesh mode, and improve the performance of the Mesh mode in the multi-hop topology via respective modifications. This scenario is significant and complex enough to deserve an in-depth analysis based on simulations.

About the comparison of PMP and Mesh modes (for multi-hop support the PMP mode has to be modified), some preliminary performance results suggests that the Mesh mode can be advantageously applied for a low number of connections. Different to that PMP is better for large number of active connections.

This leads to the conclusion that a combination of the protocol elements of PMP and Mesh modes in multi-hop topologies tend to be a promising approach, to be investigated in the framework of the MAC design in the future taking into account the air-interface of WINNER. The coexistence of PMP and Mesh in multi-hop wireless network deployment is therefore currently under study in a more detailed way.

10. Annex III: Additional Information on Multi-hop Deployment Concepts

10.1 Comparison of a multi-hop with a single-hop network approach

10.1.1 On the Efficiency of Using Multiple Hops in Relay Based Networks

10.1.1.1 System Model for Computer Simulation

We consider a system consisting of one AP and a collection of FRN which are connected wirelessly to the AP. Network elements – one AP and 121 FRN's – are placed on a hexagonal grid. The relays have a double role – to relay radio packets between the access point and other relays (thus forming a feeder system) and to provide access services to user terminals. The simulation is focused on the feeder system functionality. The AP is located in the center of the grid in location 60, see Figure 10-1. The cell size of the hexagonal grid and implicitly the relative positions of the FRNs are adjusted function of the coverage radius of the FRN for UT access, such a way there is 100% coverage of the service area with minimum overlap. The FRN radius can be set as a simulation parameter. We define the attribute "in service" for each FRN, with value "true" if the FRN is located within a predetermined distance from the AP, called AP radius, and which can be set as a simulation parameter, and "false" otherwise. The number of "in service" FRN varies depending on the values of the AP and FRN radiuses, thus allowing the simulation program to be applied to a wide range of deployment scenarios. In Figure 10-1, the "in service" relays are depicted using a red triangle, while all relay locations are indicated by their number.

The relays are logically organized in concentric tiers around the AP using the hexagonal grid, see Figure 10-1. Each tier is indexed by its rank, starting with Rank = 0 for the AP and ending with Rank = 7 for the 7th tier.

The path loss model for each link includes log-distance model and normal shadowing. The normal distributed values for shadowing are independently generated for each radio path. For each link, the path loss is stored in a path-loss vector and reused as needed until the end of the simulation. This results in uncorrelated shadowing from relay to relay (even for relays along same geometrical direction).

Using the look-up table shown in Table 1, each link is allocated a modulation scheme with highest spectral efficiency allowed by its SNR. The upper limit of the spectral efficiency range is capped at 8 b/s/Hz.

A separate individual message of 1Mbit (arbitrary value) is sent to all relays located within a specified radius (AP service range) around the AP, simulating traffic in a real network with user terminals serviced by each relay.

The individual time required to transmit the message along a given hop from the transmitting to the receiving relay, is calculated as

$$MTT_{hop} = \frac{10^6}{B\eta} \qquad (37)$$

where MTT_{hop} is the message transfer time, [s], B is the channel frequency bandwidth, [Hz], and is the spectral efficiency for the respective hop, [bit/s/Hz].

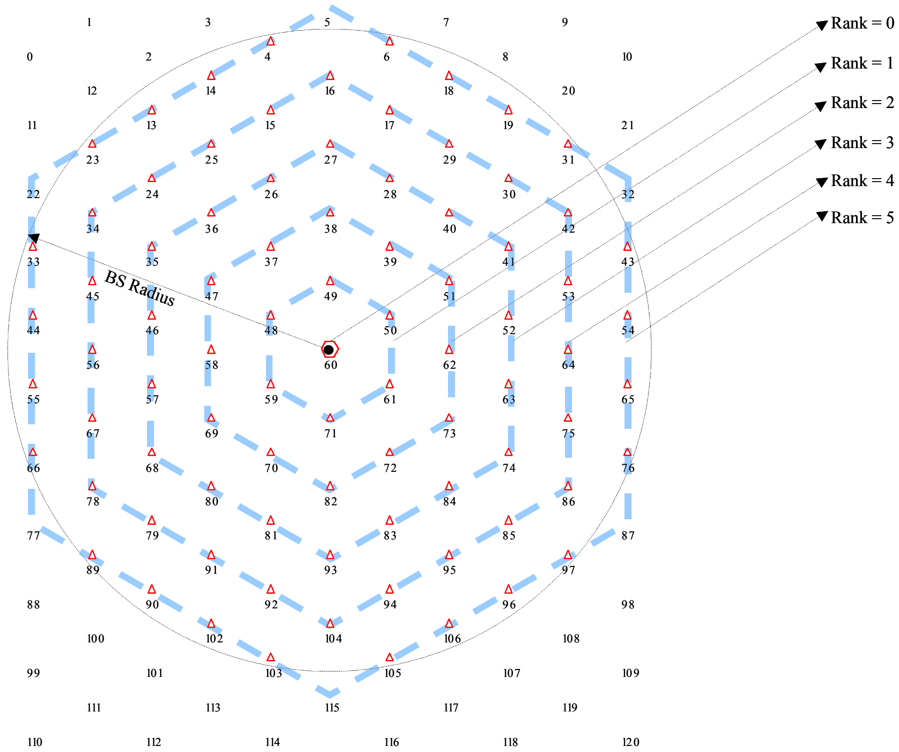


Figure 10-1 System layout on a hexagonal grid

The total time required for all relays to receive their messages is calculated as

$$MTT_{Total} = \sum_{i=1}^{121} I_i \sum_{j}^{n_i} MTT_{hop_{ij}} \tag{38}$$

where: I_i is an "in service" indicator, $I_i = 1$ if the relay i is within the AP radius, and $I_i = 0$ otherwise, and n_i is the number of hops required to transmit the message to relay I.

The ultimate objective is to save resources by improving the aggregate spectral efficiency (per link) and average spectral efficiency (per entire network). The formula for calculating the average spectral efficiency is

$$\eta = \frac{10^6 n_s}{MTT_{total} B} \tag{39}$$

Where η is average network spectral efficiency, [bit/s/Hz], n_s is the number of relays in service (equivalent with number of relays per AP) and B is the channel bandwidth, [Hz].

We assume that any relay can communicate with any relay and with the AP. When the AP has a message to be transmitted to a specific relay, the message can be passed from relay to relay until its destination. We want to determine the most efficient method of transmission, such that the aggregate message transfer time is minimized.

Table 10-1 Spectral efficiency vs. SNR

SNR [dB]	Spectral efficiency [bps/Hz]
0	0.1
4	1
6	1.33
6.8	1.5
7.8	1.75
10	2
12	2.67
13	3
15	3.5
17.7	4
19	4.5
21	5.25
26	6
31	7
35.5	8

10.1.1.2 Logical flow diagrams for the routing algorithms

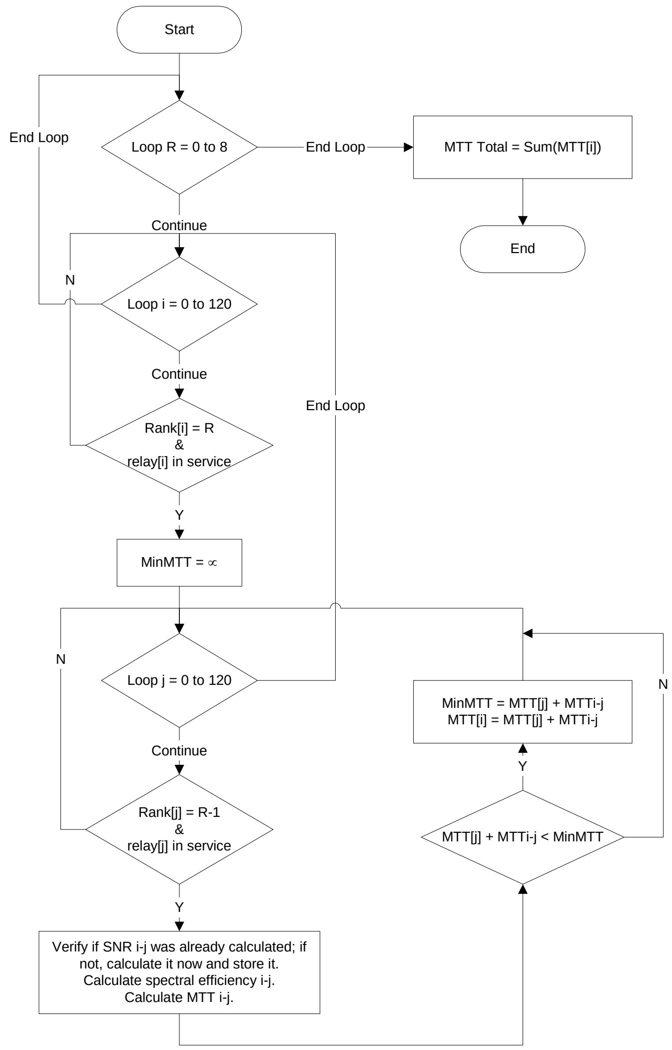


Figure 10-2 Greey multi-hop transmission – algorithm flowchart

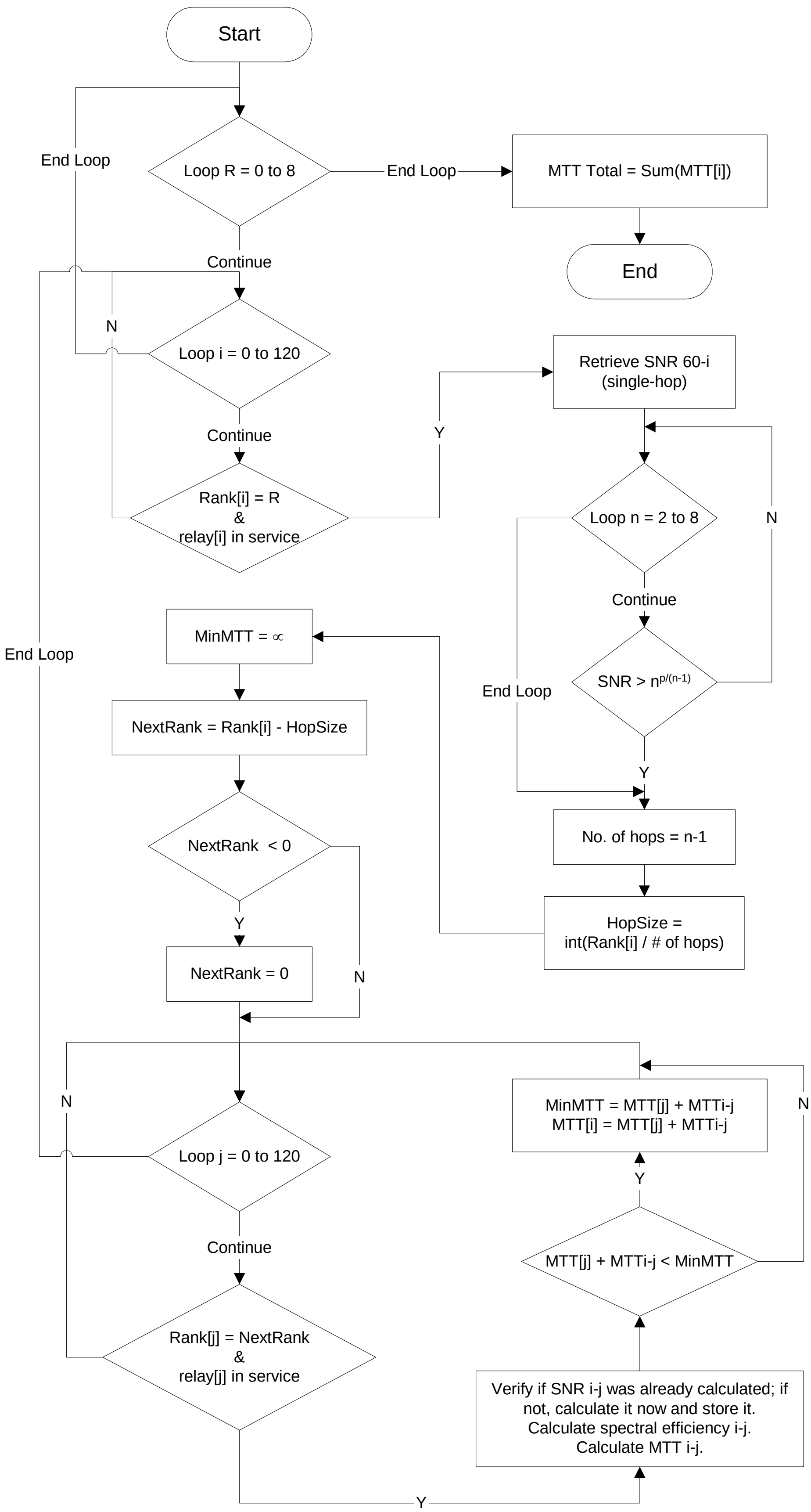


Figure 10-3 Transmission using MHC – algorithm logical flowchart

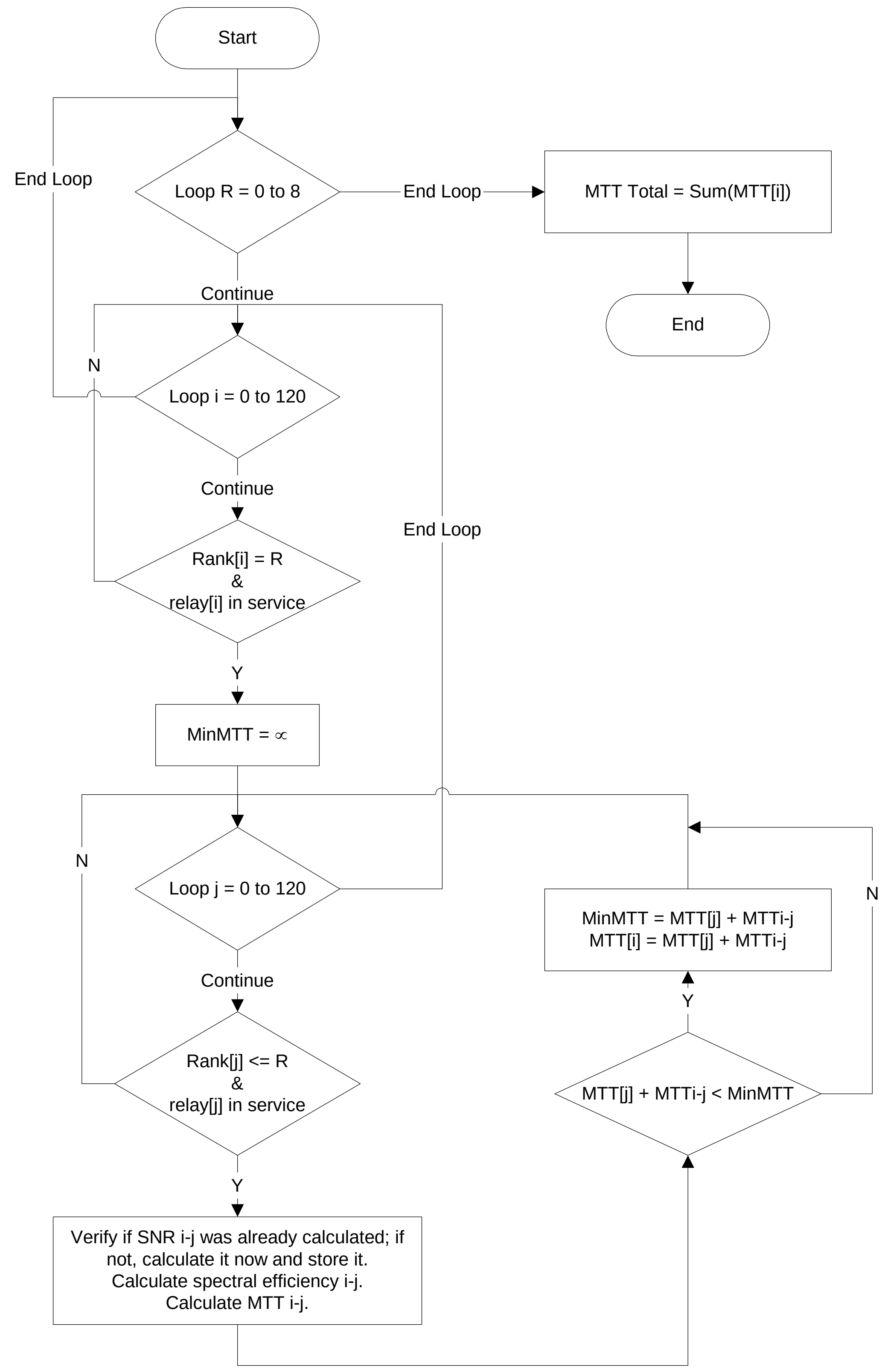


Figure 10-4 Exhaustive route search – algorithm logical flowchart

10.1.2 Performance comparison between Single and Multi-Hop Deployment

10.1.2.1 Cell size estimation

As first step towards the selection of Single- and Multi-Hop deployment, the maximum cell sizes were calculated based on the path loss model described briefly below. Because of practical limitations of the path loss model, the node placement was assumed to be in the street canyons which can be regarded as a valid assumption for small cell sizes, because of the expected costs for roof-top installation.

To approximately estimate the possible cell sizes the C/N was calculated with 24dBm transmission power. The results in Figure 10-5 show the C/N and throughput distribution obtained if the UT travels the way as shown in Figure 10-5. It can be seen that the cell size is large (more than 3 blocks) in any LOS connection while the maximum range for non-LOS (i.e. around the corner) is limited to approximately one block.

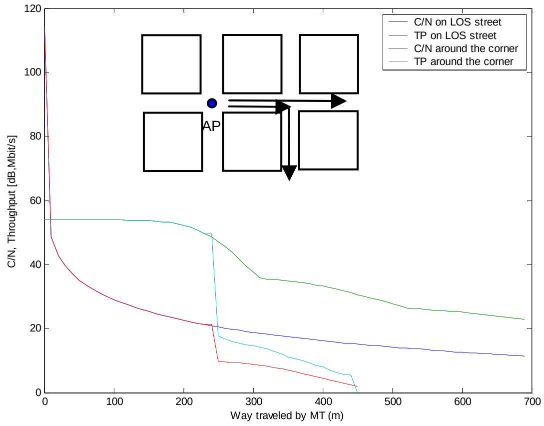


Figure 10-5 C/N and throughput for LOS and way around the corner

The path loss model used throughout the investigations is based on the model proposed in [11]. Although it is proposed for the indoor scenario, it can be quite easily extended to the outdoor case. The main idea of the model is to derive the local received power from site-specific parameters, which are:

1) mean free distance

2) transmission coefficients characterizing attenuation of obstacles positioned in the propagation path

3) reflection coefficient characterizing the amount of reflected signal energy due to obstacles in the propagation path

This model was selected because of its explicit consideration of the properties of the scenario such as shadowing effects due to obstacles in the transmission path. Compared to ray tracing based propagation models the model proposed in [11] is of considerably lower complexity.

10.1.2.2 Hiperlan/2 throughput curves

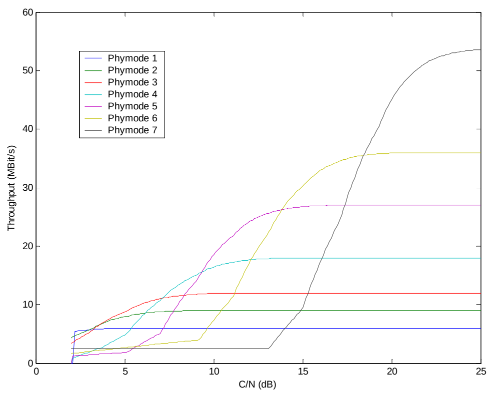


Figure 10-6 Throughput curves for HL-2 Physical Layer

10.1.2.3 Multi Hop throughput calculation

In order to calculate a mean throughput of a Multi-Hop cluster, which is comparable to that in the Single-Hop case, it is required to determine the time interval reserved for the transmission of AP and Relays belonging to a Multi-Hop cluster with its associated UTs in relation to the superframe size. You may find the definition of the superframe in [1]. The average throughput of the Multi-Hop cluster can than be obtained by taking the mean throughput for the direct AP-to-UT and Relay-to-UT communication into account. The total superframe size and the total active time for the Multi-Hop cluster can be constructed as follows. Please note that for the following calculations different throughput values have been taken into account for communication between AP-to-Relay and Relay-to-AP, respectively, while for AP-to-UT and Relay-to-UT links only a single throughput value based on the downlink has been used for simplicity reasons.

$$t_{Superframe} = R_{AP} \cdot t_{SingleAP} + R_{Rel} \cdot t_{SingleRel} \tag{40}$$

$$t_{MHCluster} = t_{SingleAP} + R_{RelCluster} \cdot t_{SingleRel} \tag{41}$$

$$t_{SingleAP} = t_{AP<->UT} + t_{AP->\mathrm{Re}l} \tag{42}$$

$$t_{SingleRel} = t_{Rel<->UT} + t_{Rel->AP} \tag{43}$$

$$t_{AP->\mathrm{Re}l} = \frac{T_{Rel<->UT}}{T_{AP->Rel}} \cdot f \cdot R_{RelCluster} \cdot t_{Rel<->UT} \tag{44}$$

$$t_{Rel->AP} = \frac{T_{Rel<->MT}}{T_{Rel->AP}} \left(1-f\right) \cdot t_{Rel<->UT} \tag{45}$$

with:

$t_{AP<->UT}$, $t_{Rel<->UT}$	direct communication time between AP/Relay and UT
$t_{SingleAP}$, $t_{SingleRel}$	total transmission time for a single AP/Relay
$t_{Superframe}$	total superframe time
$t_{MHcluster}$	total time occupied by Multi-Hop cluster
R_{AP}	number of independent resource partitions assigned exclusively to the AP (cf. Figure 4-8)
R_{Rel}	number of independent resource partitions assigned exclusively to the relay nodes (cf. Figure 4-8)
$R_{RelCluster}$	number of independent relay resource partitions per MH-cluster (cf. Figure 4-8)
$T_{AP<->UT}$	Average throughput of AP on the LOS streets
$T_{Rel<->UT}$	Average throughput of Relay to UT on the LOS streets (Side Street if viewed from the AP), not considering the AP-relay link
$T_{AP->Rel}$	DL average throughput of AP to relay link
$T_{Rel->AP}$	UL average throughput of relay to AP link
f	end-to-end DL/UL traffic asymmetry factor

Figure 10-7 shows the respective total time portions within a superframe. It should be noted that the superframe will look different in reality e.g. because of delay and switching point constraints.

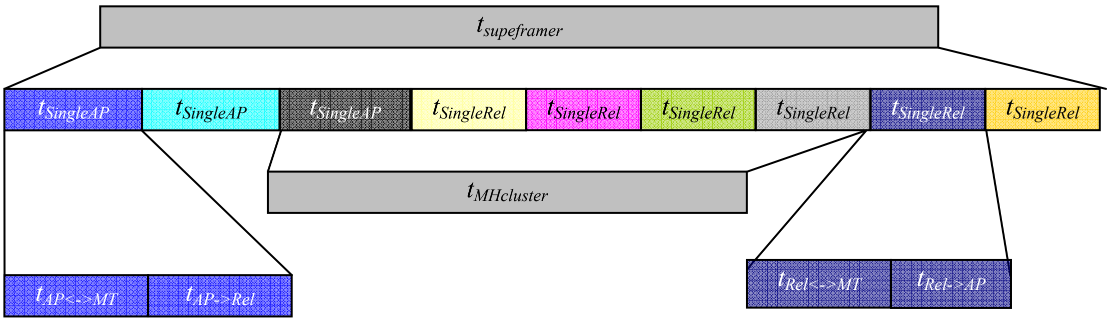


Figure 10-7 Total time partitions within the superframe (colours correspond to those used in Figure 4-8)

The required throughput values were obtained by calculations similar to those performed for the Single-Hop case. The throughput was calculated considering the strongest interferers in the first interfering ring and by assuming a utilization of the whole resource for the single link. The average throughput for the AP-to-UT and Relay-to-UT link was determined as mean of the maximum throughputs of each PHY-mode over all UT positions. It was found that on any streets providing a LOS towards the AP, the direct AP-to-UT connection offered the largest throughput except when crossing the 2nd street due to the interference of another AP using the same resource (black in this case). The relays showed throughput improvements on the side streets (NLOS-streets as seen by the AP). The calculated throughput values and the respective UT paths are shown in Figure 10-8, whereas in Table 10-2 the average throughput values are given.

An average throughput of a Multi-Hop cluster was determined as well by calculating the mean of the maximum throughputs obtained for either direct AP-to-UT communication or communication across a RN. The average throughput for a Multi-Hop cluster was found to be 29.7Mbit/s if the whole resource is completely utilized for the single link (either AP<->UT or AP<->Relay<->UT).

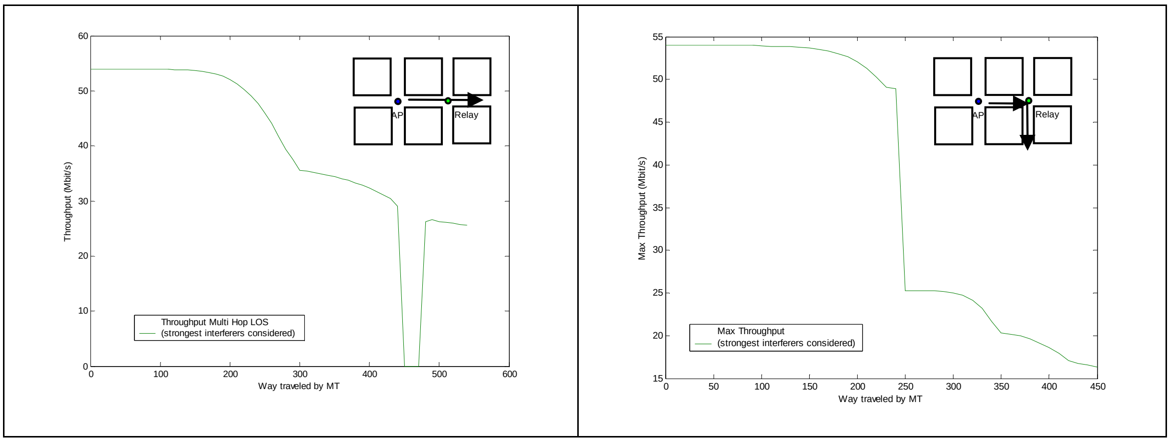


Figure 10-8 Throughput for LOS and NLOS connection

Table 10-2 Calculated Throughput values

$T_{AP<->UT}$	Average throughput of AP on the LOS streets	45.18MBit/s
$T_{Rel<->UT}$	Average throughput of Relay to UT on the LOS streets (Side Street if viewed from the AP), not considering the AP-relay link	40,56MBit/s
$T_{AP->Rel}$	DL average throughput of AP to relay link	47,5MBit/s
$T_{Rel->AP}$	UL average throughput of relay to AP link	24.5MBit/s

Equation (46) shows the total superframe size depending on the transmission times for the AP-to-UT and Relay-to-UT communication based on (40) to (45) and the parameters given in Table 10-2 and Table

10-3. In order to determine the MH-cluster throughput, the split between $t_{AP<->MT}$ and $t_{Rel<->MT}$ needs to be defined for which two solutions have been found.

$$t_{Superframe} \approx 3 \cdot t_{AP<->MT} + 16 \cdot t_{Rel<->MT} \tag{46}$$

R_{AP}	number of independent resource partitions assigned exclusively to the AP (cf. Figure 4-8)	3
R_{Rel}	number of independent resource partitions assigned exclusively to the relay nodes (cf. Figure 4-8)	6
$R_{RelCluster}$	number of independent relay resource partitions per MH-cluster (cf. Figure 4-8)	4
f	end-to-end DL/UL traffic asymmetry factor	0.5

Table 10-3 Multi-Hop Deployment parameters

The first solution is based on using an amount of resource per area similar to that used in the Single-Hop case. When taking the observation into account that the AP offers better throughput on the LOS streets, it can be seen in e.g. Figure 4-9 that each AP covers about the same area on the LOS streets as the four relays together. If all Multi-Hop clusters are considered, it can be seen that the APs cover about the same area with 3 resources as the relays with 6.(cf. Figure 4-8). If the users are distributed uniformly across the deployment area it is fair to assume that the APs should use twice the amount of resource (time) than the relays ($t_{AP<->UT} = 2 \cdot t_{Rel<->UT}$).

The second solution is based on alignment of the average throughputs archived in the Multi-Hop cluster. Since $\frac{T_{AP<->UT}}{T_{Rel<->UT}} \approx 1.11$ we can align the throughputs by selecting $t_{AP<->UT} \approx 0.9 \cdot t_{Rel<->UT}$

With the approaches mentioned and

$$T_{MHCluster} = \frac{t_{AP<->UT}}{t_{Superframe}} \cdot T_{AP<->UT} + \frac{R_{RelCluster} \cdot t_{Rel<->UT}}{t_{Superframe}} \cdot T_{Rel<->UT} \tag{47}$$

the MH-cluster throughput can be calculated to be 11.58MBit/s for $t_{AP<->MT} = 2 \cdot t_{Rel<->UT}$ and 10.85MBit/s for $t_{AP<->UT} = 0.9 \cdot t_{Rel<->UT}$.

10.2 Mobile relays

Here we present results with reference to the RAN functionalities found in Section 4.4.

Table 10-4 Parameters for large cell deployment

Large cell deployment	Case 1	Case 5	Case 2	Case 3
Pt(AP)(W)	20	20	20	20
Pt(Relay) (W)	10 (50%)	5 (25%)	2 (10%)	10 (50%)
d(AP_Relay) (m)	900	900	900	600
d(AP_UT) (m)	700-1100	700-1100	700-1100	400-800

Table 10-5 Parameters for small cell deployment – related to "pure" mobile relays

Small cell deployment	Case 4
Pt(AP) (W)	8
Pt(Relay) (W)	2 (25%)
d(AP_Relay) (m)	400
d(AP_UT) (m)	200-600

Table 10-6 Parameters for small cell deployment – related to UTs acting as mobile relays

Small cell deployment	Case 6	Case 7
Pt(AP) (W)	20	20
Pt(Relay) (W)	2 (10%)	2 (10%)
d(AP_Relay) (m)	500	500
d(AP_UT) (m)	400-600	400-600

ForTable 10-4 and Table 10-5 the UT and Relay step is 50 meters and the Relay antenna gain is set to 15dB. For Table 10-6 the Relay Antenna gain is 10 dB, the UT step is 25 meters. For all cases, we use a modified version of the Hata model ([4]), apart from Case 7 where we use the free space model. Thus, here we present the result of the results for mobile relays.

Figure 10-9 shows results for Case3, similar to Figure 4-25 (a), but for a "smaller" cell. Relay is at distance of 600m from the AP and the UT is at a distance of [400m, 800m] from the AP with a step of 50m. Thus, the figure shows that

- Compared to Case 1 at Figure 4-25 (a) we loose in gain between 4-8 dB. Still, even at the distance for Case3, we still get substantial gain for those UEs which are in the AP→Relay→UT configuration.
- Again we get the more gain for the relay positions of [-100, 100] meters. Difference between Min/Max point for the 650m case (for D(Relay_UT)=50m) is 27-3=24 dB, as in the previous cases.
- Results are very similar to those of Case 2, so we see that the same effect can be obtained either be reducing the Pt(Relay) from 10W to 2W (Case 2) or move the relay from 900 meters to 600 meters (Case 3).
- The difference as we move from 400→450→500→550 meters and from 800→750→700→650 meters is around 6/10/13 and 5/6/10 dB respectively. The reference point for the Relay is that of "0" meters.

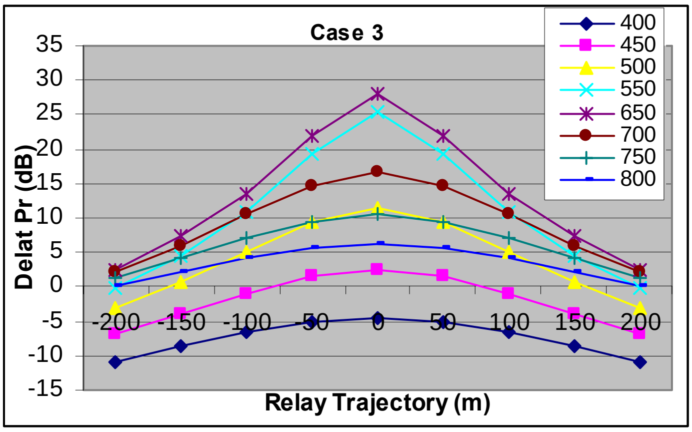


Figure 10-9 DeltaPr for Relay/UT closer to the AP (Case 3)

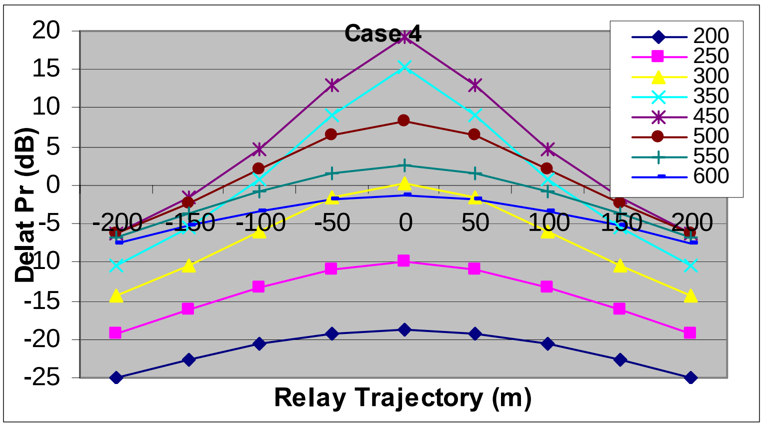


Figure 10-10 DeltaPr for a small cell

Figure 10-10 shows deltaPr for Case4 from Table 10-5. By reducing the Pt(AP), the Pt(Relay) and the distances of the Relay/UT from the AP, we try to model a smaller cell, with not so powerful mobile relays, in terms of the Pt(Relay). The main results are the following

- The gain (deltaPr) is very much reduced compared to the previous graphs. Whereas the max values for the previous cases were over 25/30 dB, in this case the max value is reduced to less than 20dB, for a d(Relay_UT) distance of 50m. This has mainly to do with the Relay/UEs being close to the AP and at the same time the power of the Relay being ¼ that of the AP power.
- The difference between the min (Relay at 200m) and max (Relay at 0m) values for the 450m case is 19-(-6)=25 dB, similar to the previous cases. Additionally, DeltaPr(0m)-DeltaPr(100m)=19-5=14 dB and DeltaPr(200m)-DeltaPr(100m)=5-(-5)=10dB
- The difference also for symmetrical places is larger. For instance, DeltaPr(450m) – DeltaPr(550m)>3 dB, whereas for Case 1 it was in the area of 1-2 dB. (Again the criterion for comparison is the d(Relay_UT)=50m distance).
- In general, as we see, for a smaller cell, the gain still is evident but only for certain cases e.g. for a radius of 50m around the Relay for the relay positions of -100, 100m.

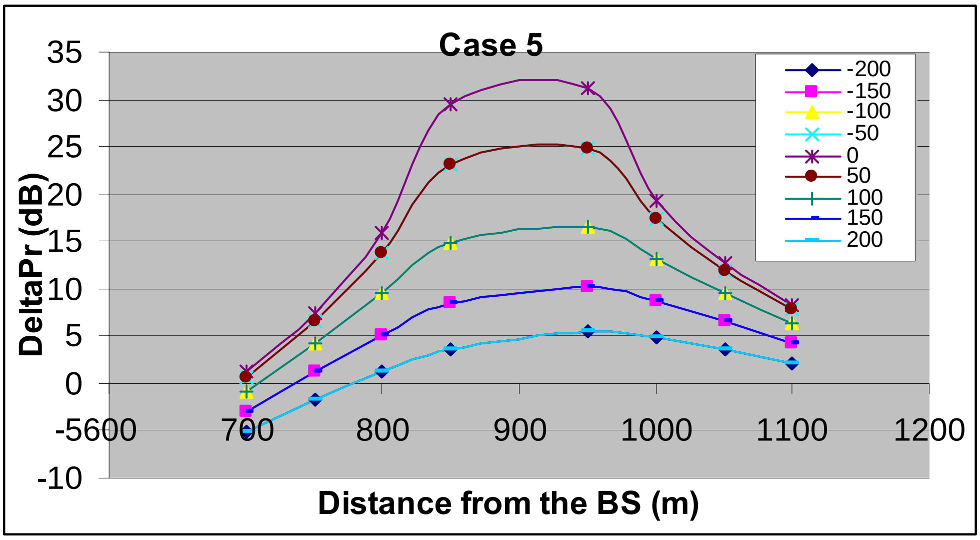


Figure 10-11 UT position on the left/right area of the relay

Figure 10-11 presents results for Case 5 from Table 10-4. It Figure 4-24 shows what we have briefly mentioned above, that for UT positions in the configuration AP→Relay→UT more gain is witnessed at the UT. Additionally, for UT positions symmetrical the Relay position (at this case that of 900m), the closer the UT is at the Relay, the higher the difference between the relevant DeltaPr is. For instance, for d(Relay_UT)=200m, we have a 1 dB<DeltaPr(800m)- DeltaPr(1000m)< 2 dB, whereas for D(Relay_UT)=50m we have 3 dB<DeltaPr(850m)- DeltaPr(950m)< 4 dB.

Figure 10-12 shows those results for Case6. The main difference is that now we use a 25-meter granularity for the positions of UT, so effectively we want to monitor what happens closer to the Relay. Similar results to the previous cases are obtained, although we see that for distances far closer to the Relay the gain we witness is not linearly increased. For instance, for the 525m case, the difference between min/max values is 24-(-14)=38 dB and for the same case we have abs(DeltaPr(200m) – DeltaPr(100m))=abs[(-14)-(-2)]=12 dB whereas abs(DeltaPr(0m) – DeltaPr(100m))=abs(24-(-2))=26 dB.

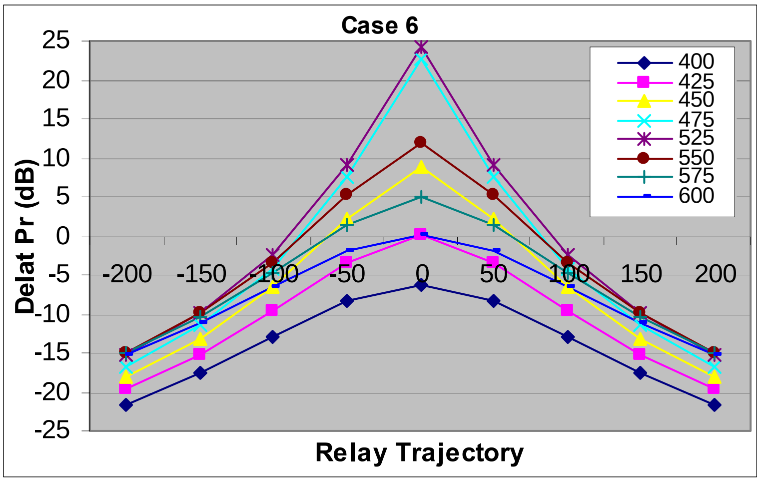


Figure 10-12 Results for Case 6 with UT granularity of 15 meters

Figure 10-13 shows similar same results as Figure 10-12, but for a different path loss model. In this case (Case 7), due to the small distances between the Relay (a high-end terminal) and the other terminals, we assume that there is LOS, and thus we use the free space model. This is a far optimistic model compared to the one we have used for the previous analysis (modified Hatta model) ([4]), but it is be used at this point to show the extent of difference. What is shown is that substantially good results are obtained and also that the difference between max and min values are smaller compared to previous cases and in some cases substantially smaller e.g. Case 6.

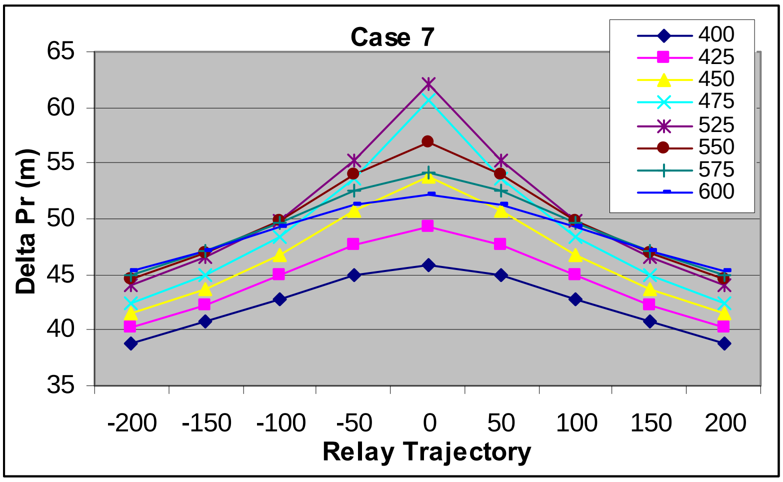


Figure 10-13 DeltaPr for Case 7 for free path model for UT/Relay channel

What has been shown so far is some initial results on the coverage that mobile relays can offer in comparison to the coverage that the AP can offer. In the above cases, (apart from Case7) we have just considered conventional cases of UTs being far away from the AP and not cases where e.g. reception in an area is bad due to severe channel conditions e.g. shadowing of a building, where results would be expected to be even more favourable for mobile relays.

What additionally would be interesting would be to present graphs for SIR measurements. At this point, we present just an initial SIR measurement assuming one receiver at the UT and no combining. Only one indirect path is used and all other paths from other relays are modelled as interference. The model to follow is that at presented in the cooperative relaying section. Additional SIR results for cooperative mobile relaying are shown in the relevant section. All SIR values will be calculated for all the positions of the Relay [-200, 200] with step 50m. The formula for the SIR values is the following

$$SIR = \frac{\Pr_i}{\sum_{j=1, j \neq i}^{K} \Pr j}, K = 8, i = 1 \ldots N \tag{48}$$

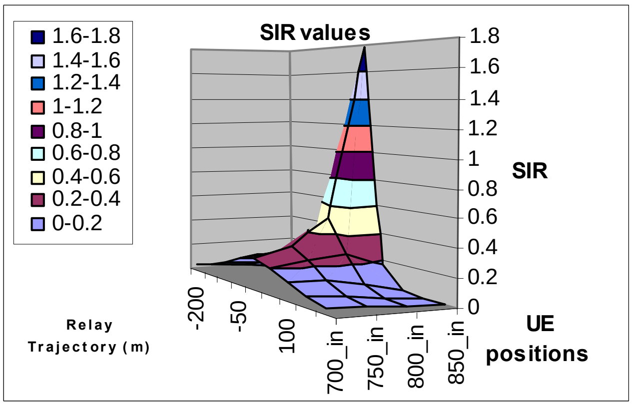


Figure 10-14 SIR measurements for Case 1

Figure 10-14 shows results for SIR for Case 1of Table 10-4 . As it is expected, the best SIR value is for the relay position aligned with the UT and AP. SIR values are mainly influenced by a number of factors such as number of mobile relay, Multiple Access technique etc.

What would be interested for further work would be to monitor what happens to SIR values for more complicated cases and also when we employ cooperative schemes and multiple transmissions are combined at the UT.

10.3 Routing and Forwarding in Multi-hop cellular networks

10.3.1 Routing and forwarding in ad-hoc networks and their appropriateness for multi-hop networks

In mobile ad-hoc networks the network topology is being changed, therefore the routing mechanism in such networks is a challenging task. There are two general approaches to routing in such networks: *topology-based* and *position-based* routing.

Topology-based routing protocols use the information about the links that exist in the network to perform packet forwarding. They can be further divided into *proactive*, *reactive*, and *hybrid* approaches. *Proactive* algorithms employ classical routing methods and maintain routing information about the available paths in the network even if these paths are not currently used. Obviously the main drawback of these approaches is the processing and signalling overhead for maintenance of unused paths. In contrast, *reactive* routing protocols only maintain the routes that are currently in use, however, they still have some inherent limitations; these protocols need to perform a route discovery before packets can be exchanged between communication peers which leads to a delay for the first packet to be transmitted. Moreover, even though route maintenance for reactive algorithms is restricted to the routes currently in use, it may still generate a significant amount of network traffic for the case that the topology of the network changes frequently. Finally, packet lost because of the route change is also an issue. *Hybrid* ad-hoc routing protocols usually combine local proactive routing and global reactive routing in order to achieve a higher level of efficiency and scalability. However, even a combination of both methods still needs to maintain at least those network paths that are currently in use, limiting the amount of topological changes that can be tolerated within a given amount of time.

Position-based routing algorithms eliminate some of the limitations of topology-based routing by using additional location information. They require the physical position information of the participating nodes. Commonly, each node determines its own position through the use of GPS or some other type of positioning services. A *location service* is used by the sender of a packet to determine the position of the destination and to include it in the packet's destination address. The routing decision at each node is then being made based on the destination's position contained in the packet and the position of the forwarding node's neighbours. The forwarding method here has a key role. Position-based routing thus does not require the establishment or maintenance of routes. The nodes have neither to store routing tables nor to transmit messages to keep routing tables up to date.

Obviously some of the advantages and disadvantages listed above are not matter of subject in the case of multi-hop cellular communications. Therefore, in the literature either completely new methods or a combination of the above mentioned methods have been presented for the case of multi-hop cellular network. Here are some general observations:

- The main challenge for ad-hoc routing is to establish and maintain the connectivity between the source and the destination. However this is not the case for multi-hop cellular networks, where there is the opportunity of utilizing "routes" within a network resource control mechanism to improve system performance and QoS provisioning. This can be realized by performing routing and other network functionalities in a cross-layer frame work. Therefore the new challenge for designing routing mechanisms in multi-hop cellular networks is to define noble objectives as well as to develop appropriate mechanism in this regards.
- Topology-based proactive methods can be used at least in the cell/sector coverage area or within an area including more than one cell/sector coverage area. In this case the routing table can be stored in and maintained by the access point. Specially in the case of utilizing fixed relays, in each time, a reasonable number of routes are feasible; this information also can be used for network performance optimization in the access point that can be jointly performed using other mechanisms such as power control etc.. In this case the communication overhead for updating the routing table, in an appropriate time periods should be a matter of concern.
- Reactive method can be considered as a part of a hybrid routing method specially for providing ubiquities network coverage for inter-system interconnection.
- However position-based methods inherently add extra complexity due to the fact of utilizing GPS positioning or any other positioning system, these method can be very helpful in multi-hop cellular networks. And this is because of the particular structure of the communication in both uplink and downlink. For example a position based method can be based on ids (or tags) that are assigned by the access point to the intermediate relays based on their position and their available resources to allocate into forwarding traffic. This is much simpler for the case of fixed-relays.
- Both topology-based and position-based methods can be considered for exploiting multi-user diversity and opportunistic resource control in different levels; multi-user diversity forwarding can be used as a part of a topology-based routing structure in the link level. Furthermore, it can be exploited through considering an enhanced topology-based routing that allows each node to choose the next hop from a set of given selections of nodes provided by the routing method.

Topology-based schemes may also be considered as a part of a network-wide routing method that manages inter-network interconnection.

10.3.2 Performance Evaluation of the Multi-Constrained QoS Routing Algorithm

An example visualization of the behaviour of the simulated network during the phase of neighbour recognition is depicted in Figure 10-15.

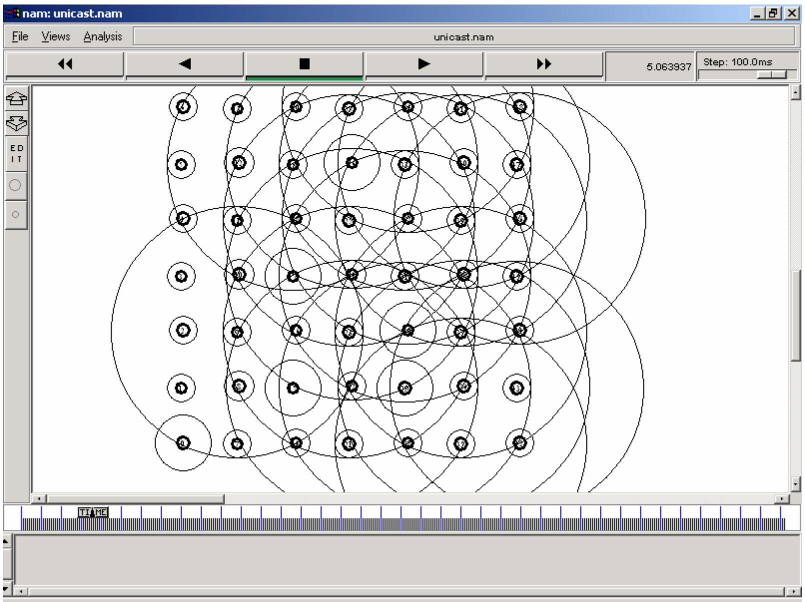


Figure 10-15 An example visualization of the behaviour of the simulated network during the phase of neighbour recognition

In the following example the node number 11 with the address 11.0.0.0 is considered, which has the neighbors numbered from 0 to 5 (in case of neighbor nodes there is no connection between their numbers and their addresses). At the beginning of the algorithm each node has the maximum *distance* and is placed in the *nodeQueue*. The source node is always assigned the zero *distance* value and the distances to its neighbors are calculated. The visited node is then moved to the *nodeSet* and another one with the minimum *distance* is extracted from the *nodeQueue*. After having visited all the nodes the routing table is generated, which has the following structure:

```
R_iface_addr->R_next_addr->R_dest_addr(R_dist).
```

```
nodeInstanceIdx 11 <-> 11.0.0.0
Node with address 11.0.0.0 is new to the list and is assigned the number 0
Node with address 26.0.0.0 is new to the list and is assigned the number 1
Node with address 2.0.0.0 is new to the list and is assigned the number 2
Node with address 34.0.0.0 is new to the list and is assigned the number 3
Node with address 0.0.0.0 is new to the list and is assigned the number 4
Node with address 1.0.0.0 is new to the list and is assigned the number 5
---------------------------------------------
nodeQueue:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1073741824)
2 : 2.0.0.0 (1073741824)
3 : 34.0.0.0 (1073741824)
4 : 0.0.0.0 (1073741824)
5 : 1.0.0.0 (1073741824)

minIndex = 0 <-> Node 0 has distance: 0
Current node has neighbors: (1,2,)
nodeQueue:
1 : 26.0.0.0 (1)
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (1073741824)
4 : 0.0.0.0 (1073741824)
5 : 1.0.0.0 (1073741824)

nodeSet: 0
0 : 11.0.0.0 (0)

---------------------------------------------
nodeQueue:
1 : 26.0.0.0 (1)
```

```
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (1073741824)
4 : 0.0.0.0 (1073741824)
5 : 1.0.0.0 (1073741824)

minIndex = 0 <-> Node 1 has distance: 1
Current node has neighbors: (3,)
nodeQueue:
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (1073741824)
5 : 1.0.0.0 (1073741824)

nodeSet:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1)

---------------------------------------------
nodeQueue:
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (1073741824)
5 : 1.0.0.0 (1073741824)

minIndex = 0 <-> Node 2 has distance: 2
Current node has neighbors: (4,5,)
nodeQueue:
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (4)
5 : 1.0.0.0 (8)

nodeSet:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1)
2 : 2.0.0.0 (2)

---------------------------------------------
nodeQueue:
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (4)
5 : 1.0.0.0 (8)

minIndex = 0 <-> Node 3 has distance: 4
Current node has neighbors: ()
nodeQueue:
4 : 0.0.0.0 (4)
5 : 1.0.0.0 (8)

nodeSet:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1)
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (4)

---------------------------------------------
nodeQueue:
4 : 0.0.0.0 (4)
5 : 1.0.0.0 (8)

minIndex = 0 <-> Node 4 has distance: 4
Current node has neighbors: ()
nodeQueue:
5 : 1.0.0.0 (8)

nodeSet:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1)
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (4)
```

```
---------------------------------------------
nodeQueue:
5 : 1.0.0.0 (8)

minIndex = 0 <-> Node 5 has distance: 8
Current node has neighbors: ()
nodeQueue:

nodeSet:
0 : 11.0.0.0 (0)
1 : 26.0.0.0 (1)
2 : 2.0.0.0 (2)
3 : 34.0.0.0 (4)
4 : 0.0.0.0 (4)
5 : 1.0.0.0 (8)

ROUTING TABLE
11.0.0.0->2.0.0.0->0.0.0.0(4)
11.0.0.0->2.0.0.0->1.0.0.0(8)
11.0.0.0->26.0.0.0->34.0.0.0(4)
11.0.0.0->11.0.0.0->26.0.0.0(1)
```

11.0.0.0->11.0.0.0->2.0.0.0(2)

10.3.3 Phased array antennas for routing in multi-hops context

An application of antenna arrays has been suggested in recent years for mobile communications systems to overcome the problem of limited channel bandwidth. It has been shown by many studies that when an array is appropriately used in a mobile communications system, it helps in improving the system performance by increasing channel capacity and spectrum efficiency, extending range coverage, tailoring beam shape, steering multiple beams to track many mobiles, and compensating aperture distortion electronically. It also reduces multipath fading, co-channel interferences, system complexity and cost, BER, and outage probability [22][23][24][25].

A phased array antenna uses an array of simple antennas, such as omni-directional ones (to keep the costs low), and combines the signal induced on these antennas to form the array output. Each antenna forming the array is known as an element of the array. The direction where the maximum gain would appear is controlled by adjusting the phase between different antennas. The phases of signals induced on various elements are adjusted such that the signals due to a source in the direction where maximum gain is required are added in phase. This results in the gain of the array (or equivalently, the gain of the combined antenna) which is equal to the sum of the gains of all individual antennas. The increase of the antenna gain is equal to 10LogN, where N is the number of elements, thus a reduction of transmission power can be reached too, even if our attention is focused mainly on network capacity increasing.

The pattern obtained from these antennas allows not to have a physical broadcast every time a transmission occurs, and for this purpose it has been studied the behavior of arrays in term of coverage with different number of elements (antennas) and different transmission powers using as transmission model the urban space attenuation model COST 231-Hata [26].

First of all each array can transmit within an angle of 120 degrees. For this reason each relay node should be equipped by three antenna arrays to cover 360 degrees. Moreover, despite of the low cost, the choice of two elements for each array is not suitable for our purpose because the resulting beam still covers a large area. Obviously higher is the number of elements, smaller is the portion of angle covered by transmission and fewer is the number of relays receiving the useless signal.

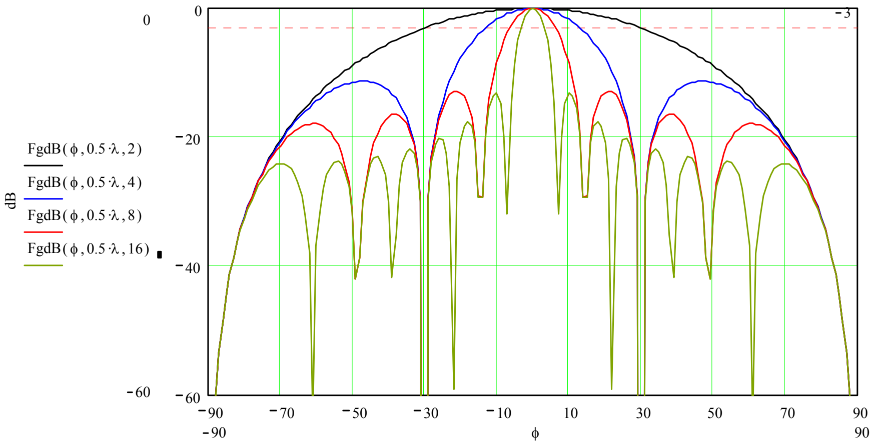


Figure 10-16 Group Factor in dB for N=2,4,8,16 elements set up at a distance of a 0,5*λ.

A right compromise between cost and coverage is reached with an array of four elements set up at a distance of 0.5 λ. Even the size of antenna is reasonable if we think the spectrum that will be released for 4G system will almost certainly located well above the 2 GHz band used by 3G systems.

Analysing the pattern obtained with four elements at the same phase (see blue curve on Figure 10-16), i.e. the direction of maximum gain is perpendicular to the array, the principal lobe covers an angle of 60 degree, but the signal is already 3dB under maximum at a spread of 25 degrees. In addition to the principal lobe, the first secondary lobe gives an important contribution while the attenuation of further lobes is very strong.

Hence in order to draw the relative coverage of these antennas, the principal lobe and the first secondary lobe have been considered. Taking in consideration the results of the application of urban space attenuation model COST 231-Hata [26] in worst conditions on the selected pattern with four elements and fixed radio range (i.e. fixed power transmission), the simplified model of antenna array coverage presented in above paragraph 4.5.5.2 has been selected.

10.4 Cooperative Relaying

10.4.1 Multi-antenna Aspects

SC-based Relay Detection:

$$P_{decode}^{(SC)} = \frac{L}{(m-1)!}\sum_{l=0}^{L-1}(-1)^l\binom{L-1}{l}\sum_{k=0}^{l(m-1)} b_k^l\left(\frac{m}{\bar{\gamma}}\right)^{m+k}\exp\left(\frac{\gamma_t(l+1)}{\rho\gamma_{link}\bar{\gamma}}\right)\sum_{t=0}^{m+k-1}\frac{(m+k-1)!}{t!}\frac{\left(\gamma_t/(\rho\gamma_{link})\right)^t}{\left(\frac{(l+1)m}{\bar{\gamma}}\right)^{m+k-t}} \tag{49}$$

where,

$$b_0^l = 1,\ b_1^l = 1,\ b_{l(m-1)}^l = \frac{1}{((m-1)!)^l},\ \text{and}\ b_k^l = \frac{1}{k}\sum_{j=1}^{\min[k,m-1]}\frac{j(l+1)-k}{j!}b_{k-j}^l \tag{50}$$

are recursively computed with $k = 2,3,\ldots,l(m-1)-1$.

In this mode of operation, the relay selects one from L branches available and on top of this it checks that the branch SNR is above the decoding threshold. We will first pursue the error performance at the relay, such error rate is required for the evaluation of the end to end system performance. Let us express this error rate as $P_e^{(\mathrm{r,sc})} = \frac{1}{v_{\mathrm{sc}}}\left(I_1^{(sc)} - I_2^{(sc)}\right)$, where $I_1^{(sc)}$ has been obtained as [61]

$$I_1^{(sc)} = \frac{L}{2\sqrt{\pi}(m-1)!}\sum_{l=0}^{L}(-1)^l\binom{L-1}{l}\sum_{k=0}^{l(m-1)} b_k^l\frac{\Gamma(k+m+1/2)}{(1+l)^{k+m}}B_y(k+m,1/2), \tag{51}$$

$y = \dfrac{2m(1+l)}{g\bar{\gamma} + 2m(1+l)}$ and $B_y(\cdot,\cdot)$ is the incomplete Beta function. Similarly, $I_2^{(sc)}$ can be expressed as [61]

$$I_2^{(sc)} = \frac{L}{2(m-1)!}\sum_{l=0}^{L}(-1)^l \binom{L-1}{l}$$
$$\times \sum_{k=0}^{l(m-1)} b_k^l \frac{1}{(1+l)^{k+m}} \left(\begin{array}{l} \Gamma(k+m) - \Gamma(k+m, \frac{m(1+l)\gamma_t}{\bar{\gamma}}) - \\ \frac{2}{\sqrt{\pi}}\sum_{p=0}^{\infty}\frac{(-1)^p}{(2p+1)p!}\left(\frac{2m}{g\bar{\gamma}}\right)^{-(p+1/2)} \frac{\Gamma(k+m+p+1/2 - \Gamma(k+m+p+1/2, m(1+l)\gamma_t/\bar{\gamma})}{(1+l)^{p+1/2}} \end{array} \right) \tag{52}$$

Finally, v_{sc} is given as

$$v_{sc} = \frac{L}{(m-1)!}\sum_{l=0}^{L}(-1)^l \binom{L-1}{l} \sum_{k=0}^{l(m-1)} b_k^l \left[\frac{1}{(1+l)}\right]^{k+m} \Gamma(k+m, (1+l)m\gamma_t/\bar{\gamma}) \tag{53}$$

MRC-based Relay Detection:

$$P_{decode}^{(mrc)} = \left(\frac{m}{\bar{\gamma}}\right)^{Lm} \frac{\exp(-m\gamma_t / \rho\bar{\gamma}\gamma_{link})}{\Gamma(Lm)} \sum_{k=0}^{Lm-1} \frac{(Lm-1)!}{k!} \frac{(\gamma_t / \rho\gamma_{link})^k}{(m/\bar{\gamma})^{Lm-k}} \tag{54}$$

$$P_e^{(\mathrm{r,mrc})} = \frac{1}{v_{\mathrm{mrc}}} \left(\begin{array}{l} \frac{\Gamma(Lm+0.5)}{2\sqrt{\pi}\Gamma(Lm)} B_x(Lm, 0.5) - \frac{\Gamma(Lm) - \Gamma(Lm, m\gamma_t/\bar{\gamma})}{2\Gamma(Lm)} \\ + \frac{1}{\sqrt{\pi}}\sum_{p=0}^{\infty}\frac{(-1)^p}{(p!(2p+1))}\left(\frac{g}{2}\right)^{(2p+1)/2} \frac{\Gamma(Lm+p+0.5) - \Gamma(Lm+p+0.5, m\gamma_t/\bar{\gamma})}{\Gamma(Lm)(m/\bar{\gamma})^{(p+1)/2}} \end{array} \right) \tag{55}$$

where $v_{mrc} = \dfrac{\Gamma(Lm, m\gamma_t/\bar{\gamma})}{\Gamma(Lm)}$. $\Gamma(\cdot)$ and $\Gamma(\cdot,\cdot)$ are the gamma and incomplete gamma functions respectively.

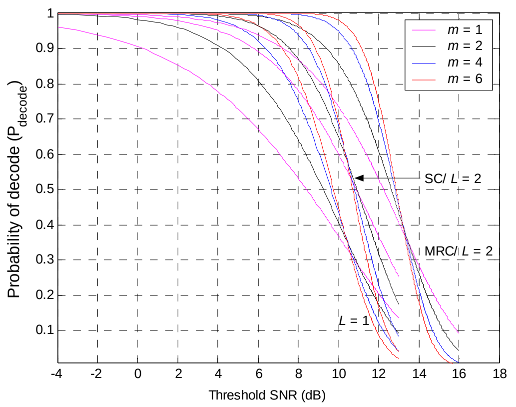
Improving the error performance at the relay will ensure reliable diversity combining of signals at the destination and hence enhancing the benefit obtained through cooperative relaying. Let us denote the actual error performance of the destination when it benefits from cooperation as $P_e^{(\mathrm{coop})}$, which implies that at least a relay did decode and forward its received signal. This error depends on the channels between the relay and destination, and between the source and destination. In this sense, the system can be described as being either balanced (symmetric network scenario) or unbalanced (asymmetric one). In the balanced channels situation the average SNR on all the links are assumed to be equal. For a system with equal-performance relays (i.e., same probability of decode and same error rate), the diversity obtained from cooperation can be obtained as [61]

$$P_e^{(coop)} = \frac{1}{2\sqrt{\pi}}\sum_{i=1}^{R}\binom{R}{i}(1-P_{decode})^{R-i}(P_{decode})^i \frac{\Gamma((i+1)m+1/2)}{\Gamma((i+1)m)} B_x((i+1)m, 1/2) \tag{56}$$

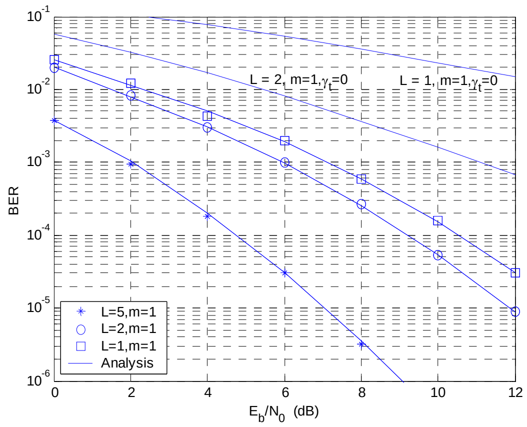
,

where $x = \dfrac{2m}{(2m+g\bar{\gamma})}$.

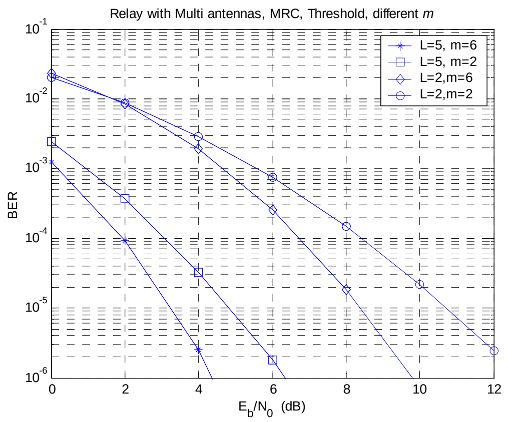
With these expressions and others in previous WINNER submissions, the end to end system performance can be evaluated.



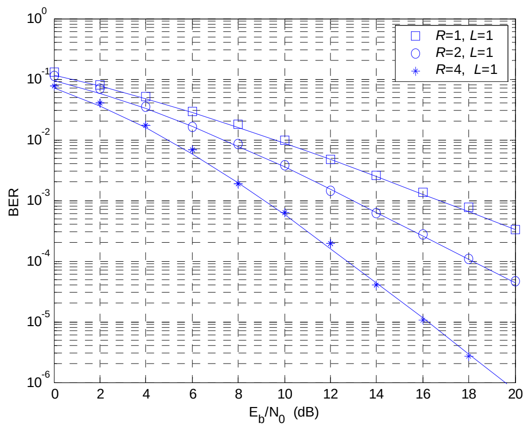
(a). Probability of decode of a relay with multiple antennas and also performing threshold decoding.



(b). Relay error performance: Threshold-based DF relaying with MRC detection at the relays (m=1).



c). Relay error performance: Threshold-based DF relaying with MRC detection at the relays (m=2, 6).



d). End-to-end error performance parallel relays with threshold DF (m=1). BPSK modulation is assumed in all links.

Figure 10-17 Sample results of multi-antenna parallel relays: Some simulations / analyses.

Table 10-7 SC-based relay detection: System comparison at BER = 10^{-4}

	R=*2*		R=*4*	
# of Antennas (L)	M=*1*	m =*2*	m=*1*	m=*2*
1	*18.0 dB*	*11.5 dB*	*13.0 dB*	*9.2 dB*
2	*13.2 dB*	*9.2 dB*	*9.2 dB*	*7.6 dB*
4	*10.8 dB*	*7.8 dB*	*7.6 dB*	*6.1 dB*

Furthermore, Table 10-7 has been given for the purpose of comparing the SC-based relay detection for different system configurations (R, L, m) at a given error performance (BER $=10^{-4}$). When R=2, L=4 system configuration is employed instead of R=4, L=2, we observe system degradation in the tune of 1.6 and 0.2 dB, for m=1 and m=2, respectively. However, we have to note that two detection chains are required for (R=2, L=4) as compared with four of R=4, L=2. The table also depicts that deploying two relays each with two antennas R=2, L=2 and the relay selects one at a time yields almost the same performance as deploying four relays each with one antenna (R=4, L=1, performance in Figure 4-37 (c)). It is worth mentioning that 2 detection resources are required in the R=2, L=2 case whereas the R=4, L=1 case necessitates 4 detection resources; besides deploying a relay is more expensive than installing one microdiversity antenna.

10.4.2 System Resource Constraint Combinations

The following table shows all of the system resource constraint combinations and resultant system connectivity models.

NCA	RC	DC	MCR	MCT	IC	Model
NCA	RC	DC	MCR	MCT	IC	FRFD
NCA	RC	DC	MCR	MCT	NIC	FRFD
NCA	RC	DC	MCR	SCT	IC	FRFD
NCA	RC	DC	MCR	SCT	NIC	FRFD
NCA	RC	DC	SCR	MCT	IC	FRFD
NCA	RC	DC	SCR	MCT	NIC	FRFD
NCA	RC	DC	SCR	SCT	IC	1R1D
NCA	RC	DC	SCR	SCT	NIC	1R1D
NCA	RC	NDC	MCR	MCT	IC	FR1D
NCA	RC	NDC	MCR	MCT	NIC	FR1D
NCA	RC	NDC	MCR	SCT	IC	FR1D
NCA	RC	NDC	MCR	SCT	NIC	FR1D
NCA	RC	NDC	SCR	MCT	IC	FR1D
NCA	RC	NDC	SCR	MCT	NIC	FR1D
NCA	RC	NDC	SCR	SCT	IC	1R1D
NCA	RC	NDC	SCR	SCT	NIC	1R1D
NCA	NRC	DC	MCR	MCT	IC	1RFD
NCA	NRC	DC	MCR	MCT	NIC	1RFD
NCA	NRC	DC	MCR	SCT	IC	1RFD
NCA	NRC	DC	MCR	SCT	NIC	1RFD
NCA	NRC	DC	SCR	MCT	IC	1RFD
NCA	NRC	DC	SCR	MCT	NIC	1RFD
NCA	NRC	DC	SCR	SCT	IC	1R1D
NCA	NRC	DC	SCR	SCT	NIC	1R1D
NCA	NRC	NDC	MCR	MCT	IC	1R1D
NCA	NRC	NDC	MCR	MCT	NIC	1R1D
NCA	NRC	NDC	MCR	SCT	IC	1R1D
NCA	NRC	NDC	MCR	SCT	NIC	1R1D
NCA	NRC	NDC	SCR	MCT	IC	1R1D
NCA	NRC	NDC	SCR	MCT	NIC	1R1D
NCA	NRC	NDC	SCR	SCT	IC	1R1D
NCA	NRC	NDC	SCR	SCT	NIC	1R1D
2CA	RC	DC	MCR	MCT	IC	2RFDFS
2CA	RC	DC	MCR	MCT	NIC	1RFD2H
2CA	RC	DC	MCR	SCT	IC	2RFD
2CA	RC	DC	MCR	SCT	NIC	1RFD2H
2CA	RC	DC	SCR	MCT	IC	2R2DFS
2CA	RC	DC	SCR	MCT	NIC	1R2D2H
2CA	RC	DC	SCR	SCT	IC	2R2D
2CA	RC	DC	SCR	SCT	NIC	1R2D2H
2CA	RC	NDC	MCR	MCT	IC	2R1D2S
2CA	RC	NDC	MCR	MCT	NIC	1R1D2H
2CA	RC	NDC	MCR	SCT	IC	2R1D
2CA	RC	NDC	MCR	SCT	NIC	1R1D2H
2CA	RC	NDC	SCR	MCT	IC	2R1D2S
2CA	RC	NDC	SCR	MCT	NIC	1R1D2H
2CA	RC	NDC	SCR	SCT	IC	2R1D
2CA	RC	NDC	SCR	SCT	NIC	1R1D2H
2CA	NRC	DC	MCR	MCT	IC	1RFD
2CA	NRC	DC	MCR	MCT	NIC	1RFD2H
2CA	NRC	DC	MCR	SCT	IC	1RFD
2CA	NRC	DC	MCR	SCT	NIC	1RFD2H
2CA	NRC	DC	SCR	MCT	IC	1R2D2S
2CA	NRC	DC	SCR	MCT	NIC	1R2D2H
2CA	NRC	DC	SCR	SCT	IC	1R2D
2CA	NRC	DC	SCR	SCT	NIC	1R2D2H
2CA	NRC	NDC	MCR	MCT	IC	1R1D
2CA	NRC	NDC	MCR	MCT	NIC	1R1D2H
2CA	NRC	NDC	MCR	SCT	IC	1R1D
2CA	NRC	NDC	MCR	SCT	NIC	1R1D2H
2CA	NRC	NDC	SCR	MCT	IC	1R1D
2CA	NRC	NDC	SCR	MCT	NIC	1R1D2H
2CA	NRC	NDC	SCR	SCT	IC	1R1D
2CA	NRC	NDC	SCR	SCT	NIC	1R1D2H

Figure 10-18 System Resource Constraint Combinations

Constraint combinations with destination and/or relay diversity combination but single channel reception and single channel transmission actually achieve less connectivity when each relay transmits on a separate orthogonal channel than when the source and all relays transmit on the same two channels. Intelligent use of the available channels results in identical connectivity to a system constrained to two available channels.

Although not practical in mobile relaying networks, constraint combinations with relay diversity combination but not destination diversity combination are of interest for fixed relaying networks where it is expected that fixed relays will have less resource constraints than the mobile destination.

It is assumed in the preceding derivation and the remainder of this paper that the half-duplex nature of wireless terminal hardware requires that each relay transmit and receive with different channels, implying a minimum of two orthogonal channels. However, it is theoretically interesting to briefly consider the system connectivity models that would result if in the future relays had full-duplex capabilities, for example by introducing advanced signal processing capabilities or some artificial isolation between the transmitter and receiver hardware. In this case there could be a single channel shared by all terminals, and the only other system connectivity constraints that would be relevant are relay and destination combination. There would therefore be four possible resultant system connectivity models depending on the relay and destination combination constraints: 1R1D, 1RFD, FR1D, and FRFD.

10.4.3 Example System Connectivity Models

The following figures present examples of the various system connectivity models along with their minimum cost constraint sets:

Single Relay Single Destination Two Hop (1R1D2H): This system connectivity model occurs when one relay is connected to the source and destination. Of the system resource constraint combinations that result in the 1R1D2H model the minimum cost constraint set is {2CA, NRC, NDC, SCR, SCT, NIC}. The following figure shows an example 1R1D2H model.

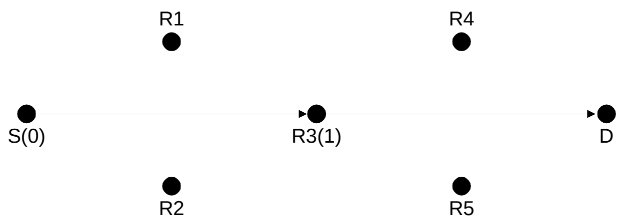


Figure 10-19 Example 1R1D2H System Connectivity Model

The 1R1D2H model is minimally two-hop connected.

Single Relay Single Destination (1R1D): This system connectivity model occurs when each relay is connected to one transmitter and the destination is connected to one transmitter. Of the system resource constraint combinations that result in the 1R1D model the minimum cost constraint set is {2CA, NRC, NDC, SCR, SCT, IC}. The following figure shows an example 1R1D model.

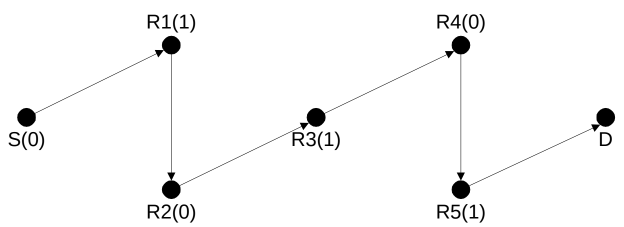


Figure 10-20 Example 1R1D System Connectivity Model

The 1R1D model is minimally multi-hop connected, and corresponds to the multi-hop models presented in [52][55][60].

Single Relay 2Chnl Destination Two Hop (1R2D2H): This system connectivity model occurs when each relay is connected to the source and destination. Of the system resource constraint combinations that result in the 1R2D2H model the minimum cost constraint set is {2CA, NRC, DC, SCR, SCT, NIC}. The following figure shows an example 1R2D2H model.

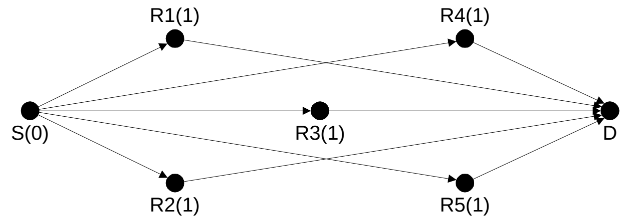


Figure 10-21 Example 1R2D2H System Connectivity Model

The 1R2D2H model corresponds to the multi-user diversity models presented in [50][54].

Single Relay 2Chnl Destination (1R2D): This system connectivity model occurs when each relay is connected to one transmitter and the destination is connected to a subset of transmitters on one channel. Of the system resource constraint combinations that result in the 1R2D model the minimum cost constraint set is {2CA, NRC, DC, SCR, SCT, IC}. The following figure shows an example 1R2D model.

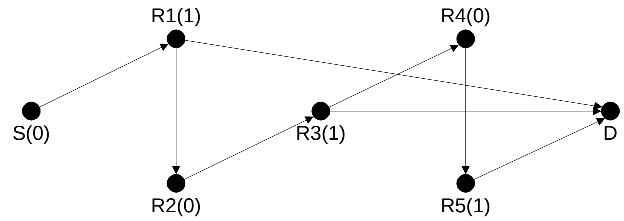


Figure 10-22 Example 1R2D System Connectivity Model

The 1R2D model corresponds to the multi-user diversity models presented in [50][54] under an alternate channel allocation where the source transmits on channel C0 and all relays transmit on channel C1. The source and destination are connected to all relays.

Single Relay 2Chnl Destination 2Chnl Source (1R2D2S): This system connectivity model occurs when each relay is connected to one transmitter, the destination is connected to a subset of transmitters on one channel, and the source is connected to a subset of receivers on both channels. Of the system resource constraint combinations that result in the 1R2D2S model the minimum cost constraint set is {2CA, NRC, DC, SCR, MCT, IC}. The following figure shows an example 1R2D2S model.

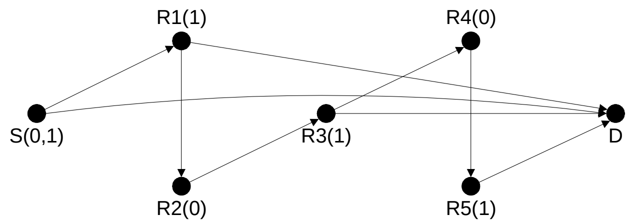


Figure 10-23 Example 1R2D2S System Connectivity Model

The 1R2D2S model has identical connectivity to the cooperative diversity models presented in [36][37][38][57][58] under an alternate channel allocation where the source transmits on C0 and C1 and all relays transmit on C1. The source and destination are connected to all relays and to each other.

Single Relay Full Destination Two Hop (1RFD2H): This system connectivity model occurs when each relay is connected to the source and destination and the source and destination are connected to each other. Of the system resource constraint combinations that result in the 1RFD2H model the minimum cost constraint set is {2CA, NRC, DC, MCR, SCT, NIC}. The following figure shows an example 1RFD2H model.

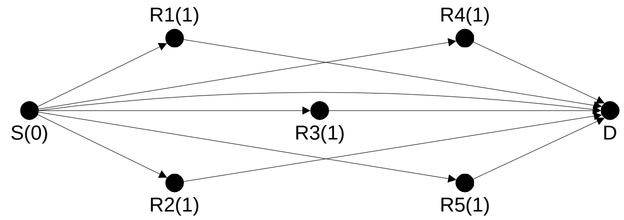


Figure 10-24 Example 1RFD2H System Connectivity Model

The 1RFD2H model corresponds to the cooperative diversity models presented in [36][37][38][57][58].

Single Relay Full Destination (1RFD): This system connectivity model occurs when each relay is connected to one transmitter and the destination is connected to all transmitters. Of the system resource constraint combinations that result in the 1RFD model the minimum cost constraint set is {2CA, NRC, DC, MCR, SCT, IC}. The following figure shows an example 1RFD model.

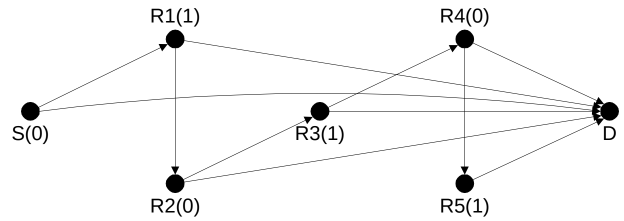


Figure 10-25 Example 1RFD System Connectivity Model

The 1RFD model corresponds to the cooperative diversity models presented in [36][37][38][57][58] under an alternate channel allocation where the source transmits on C0 and all relays transmit on C1. The source and destination are connected to all relays and to each other.

2Chnl Relay Single Destination (2R1D): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel and the destination is connected to one transmitter. Of the system resource constraint combinations that result in the 2R1D model the minimum cost constraint set is {2CA, RC, NDC, SCR, SCT, IC}. The following figure shows an example 2R1D model.

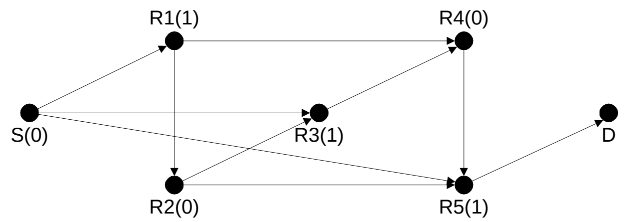


Figure 10-26 Example 2R1D System Connectivity Model

2Chnl Relay Single Destination 2Chnl Source (2R1D2S): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel, the destination is connected to one transmitter, and the source is connected to a subset of receivers on both channels. Of the system resource constraint combinations that result in the 2R1D2S model the minimum cost constraint set is {2CA, RC, NDC, SCR, MCT, IC}. The following figure shows an example 2R1D2S model.

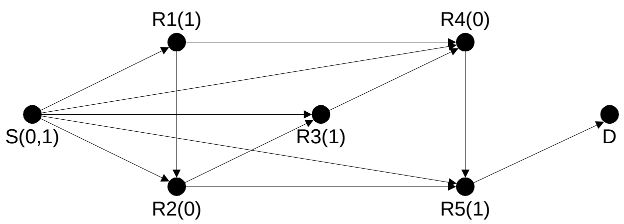


Figure 10-27 Example 2R1D2S System Connectivity Model

2Chnl Relay 2Chnl Destination (2R2D): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel and the destination is connected to a subset of transmitters on one channel. Of the system resource constraint combinations that result in the 2R2D model the minimum cost constraint set is {2CA, RC, DC, SCR, SCT, IC}. The following figure shows an example 2R2D model.

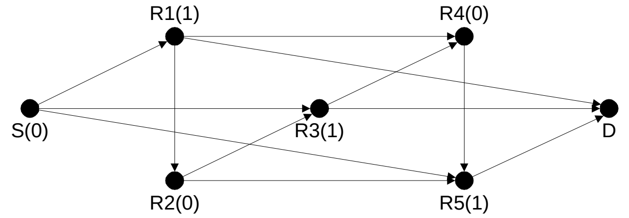


Figure 10-28 Example 2R2D System Connectivity Model

2Chnl Relay 2Chnl Destination Full Source (2R2DFS): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel, the destination is connected to a subset of transmitters on one channel, and the source is connected to all receivers. Of the system resource constraint combinations that result in the 2R2DFS model the minimum cost constraint set is {2CA, RC, DC, SCR, MCT, IC}. The following figure shows an example 2R2DFS model.

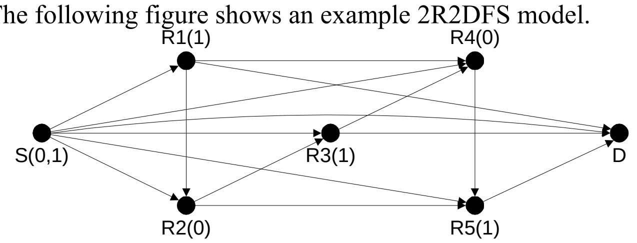


Figure 10-29 Example 2R2DFS System Connectivity Model

2Chnl Relay Full Destination (2RFD): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel and the destination is connected to all transmitters. Of the system resource constraint combinations that result in the 2RFD model the minimum cost constraint set is {2CA, RC, DC, MCR, SCT, IC}. The following figure shows an example 2RFD model.

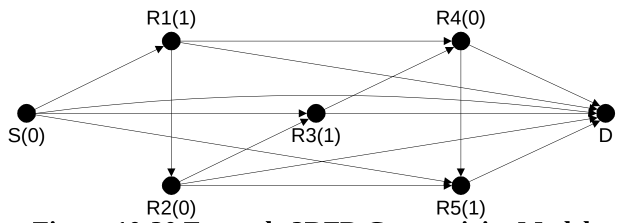


Figure 10-30 Example 2RFD Connectivity Model

The 2RFD model effectively extends the cooperative diversity models presented in [36][37][38][57][58] to the case where the relays belong to different cooperation groups in multiple tiers between the source and destination.

2Chnl Relay Full Destination Full Source (2RFDFS): This system connectivity model occurs when each relay is connected to a subset of transmitters on one channel, the destination is connected to all transmitters, and the source is connected to all receivers. Of the system resource constraint combinations that result in the 2RFDFS model the minimum cost constraint set is {2CA, RC, DC, MCR, MCT, IC}. The following figure shows an example 2RFDFS model.

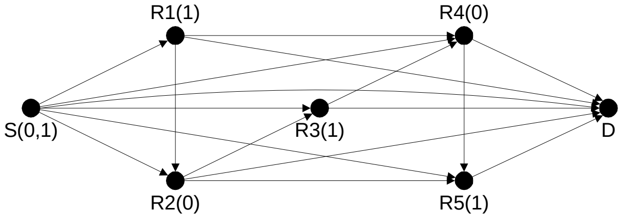


Figure 10-31 Example 2RFDFS System Connectivity Model

Full Relay Single Destination (FR1D): This system connectivity model occurs when each relay is connected to all transmitters and the destination is connected to one transmitter. Of the system resource constraint combinations that result in the FR1D model the minimum cost constraint set is {NCA, RC, NDC, MCR, SCT, NIC}. The following figure shows an example FR1D model.

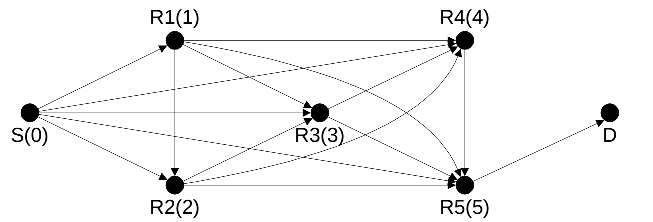


Figure 10-32 Example FR1D System Connectivity Model

Full Relay Full Destination (FRFD): Each relay is connected to all transmitters previous along the transmission path and the destination is connected to all transmitters. Of the system resource constraint combinations that result in the FRFD model the minimum cost constraint set is {NCA, RC, DC, MCR, SCT, NIC}. The following figure shows an example FRFD model.

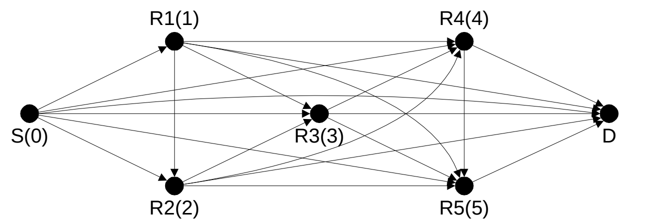


Figure 10-33 Example FRFD System Connectivity Model

The FRFD model is fully connected, and corresponds to the multi-hop diversity models presented in [51][52].

10.4.4 System Connectivity Model Simulation Results

All of the simulated network topologies have a single source terminal, five relay terminals, and a single destination terminal. All of the link distances are normalized with respect to the source to destination distance. The first network topology has the example terminal distribution shown in Annex V with the suboptimal (but illustrative) network connectivity also shown in Figs. 4-18 for the respective system connectivity models. The terminal distribution is symmetric with normalized link distances:

$$d_{S,D} = 3d_{S,R1} = 2d_{S,R3} = \tfrac{5}{4}d_{S,R4} = 3d_{R1,R2} = 2d_{R1,R4} = \tfrac{3}{2}d_{R1,R5}. \tag{57}$$

The second network topology has the example terminal distribution shown in Annex V with network connectivity that optimizes the error performance for the respective system connectivity model and relaying method combinations. The terminal distribution is identical to the first network topology such that:

$$d_{S,D} = 3d_{S,R1} = 2d_{S,R3} = \tfrac{5}{4}d_{S,R4} = 3d_{R1,R2} = 2d_{R1,R4} = \tfrac{3}{2}d_{R1,R5}. \tag{58}$$

The third network topology has linear terminal distribution (the relay terminals are fixed and collinear so that they divide the direct path between the source and destination terminals into six equal-length segments) with network connectivity that optimizes the error performance for the respective system connectivity model and relaying method combinations. The terminal distribution is symmetric with normalized link distances:

$$d_{S,D} = \tfrac{6}{1}d_{S,R1} = \tfrac{6}{2}d_{S,R2} = \tfrac{6}{3}d_{S,R3} = \tfrac{6}{4}d_{S,R4} = \tfrac{6}{5}d_{S,R5}. \tag{59}$$

The fourth network topology has central terminal distribution (the relay terminals are equidistant from the source and destination terminals and relatively close together) with network connectivity that optimizes the error performance for the respective system connectivity model and relaying method combinations. Let T_R be the set of all relay terminals. The terminal distribution is symmetric with normalized link distances:

$$d_{S,D} = 2d_{S,Ri} = 2d_{Ri,D}, \forall i \in T_R \text{ and } d_{S,D} = 8d_{Ri,Rj}, \forall i, j \in T_R. \tag{60}$$

The following figures respectively compare the BER of the system connectivity models for the suboptimal example network topology, example network topology, linear network topology, and central network topology using amplified relaying, decoded relaying with error propagation, and decoded relaying without error propagation. These highlight the impact of the system resource constraints by providing a comparison of the system connectivity models in terms of probability of error versus the signal to noise ratio of the single hop reference channel. The probability of error results are plotted such that the total allocated power of each system connectivity model is the same as that of the single hop reference channel that achieves the given signal to noise ratio. Although the simulations presented in this paper utilize the modulation scheme and network topologies noted above, it has been confirmed via further simulation that the qualitative results derived from these simulations generalize to other modulations schemes and network topologies.

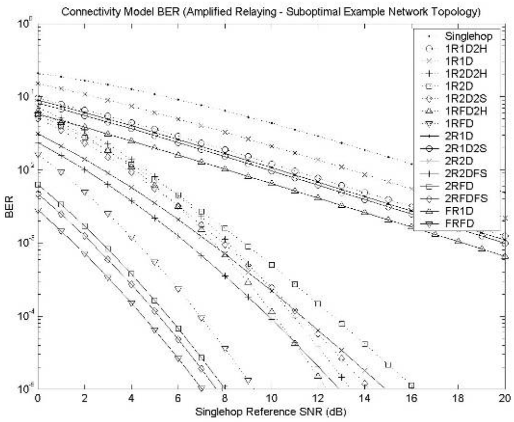


Figure 10-34
Amplified Relaying for Suboptimal Example Network Topology

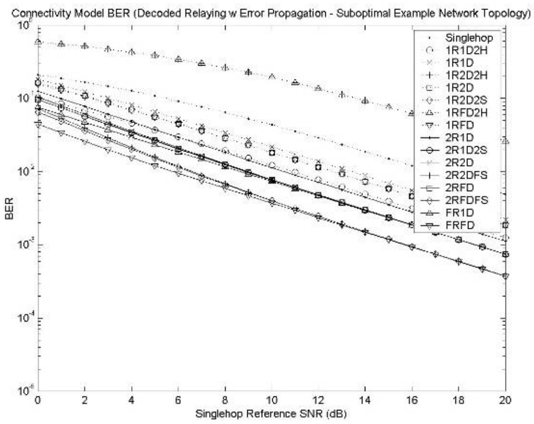


Figure 10-35
Decoded Relaying w Propagation for Suboptimal Example Network Topology

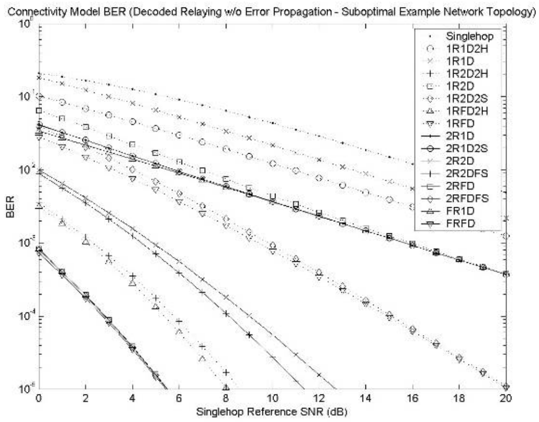


Figure 10-36
Decoded Relaying w/o Propagation for Suboptimal Example Network Topology

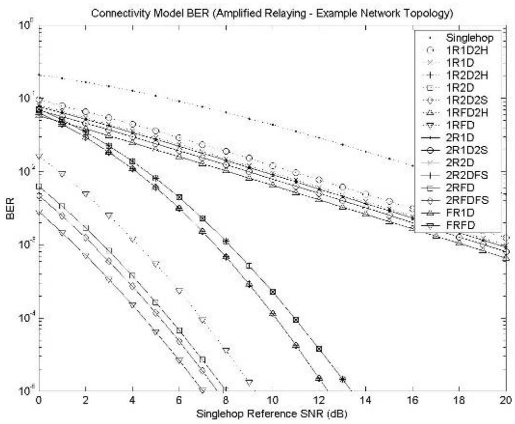


Figure 10-37
Amplified Relaying for Example Network Topology

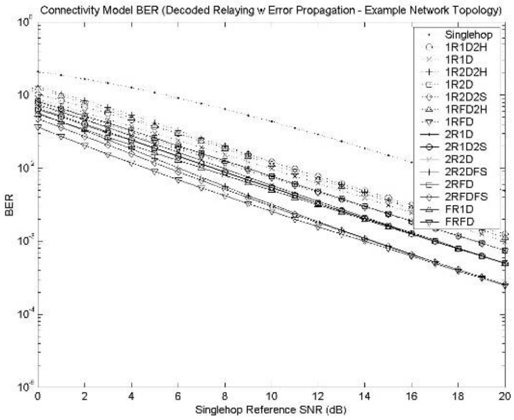


Figure 10-38
Decoded Relaying w Propagation for Example Network Topology

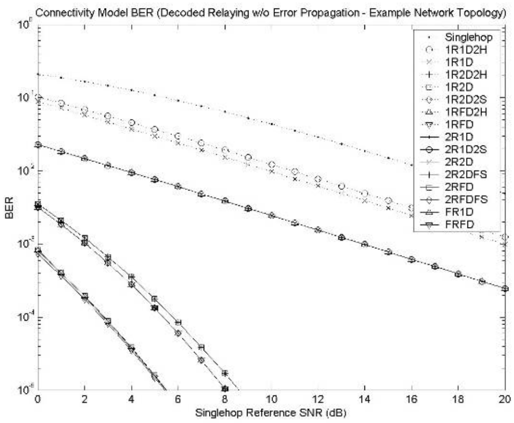


Figure 10-39
Decoded Relaying w/o Propagation for Example Network Topology

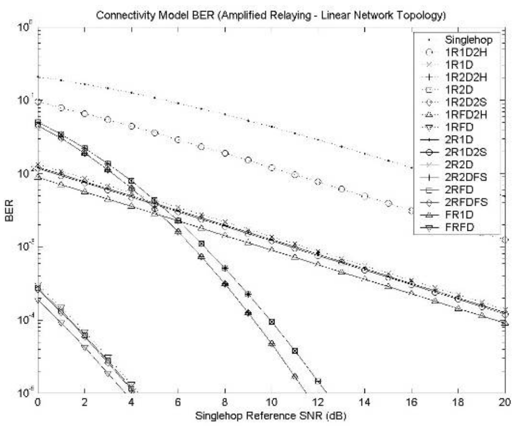


Figure 10-40
Amplified Relaying for Linear Network Topology

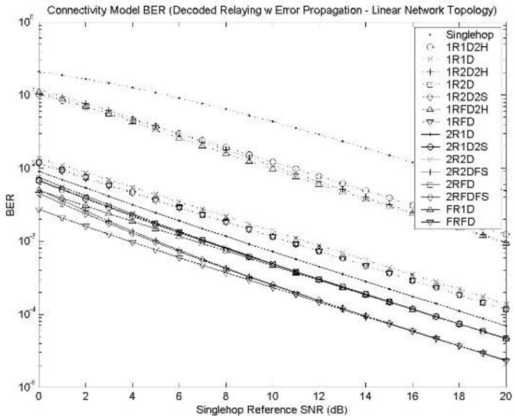


Figure 10-41
Decoded Relaying w Propagation for Linear Network bTopology

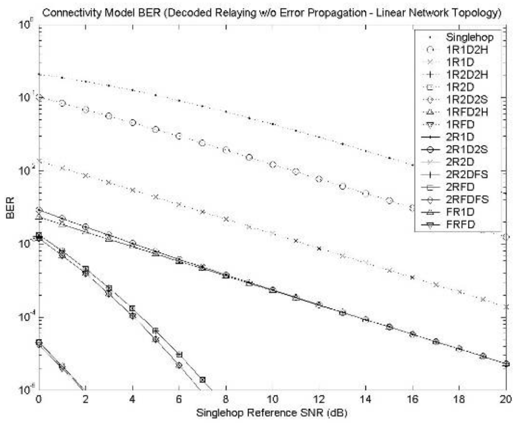


Figure 10-42
Decoded Relaying w/o Propagation for Linear Network Topology

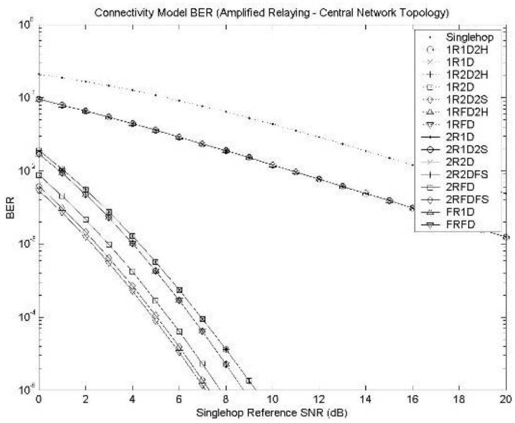


Figure 10-43
Amplified Relaying for Central Network Topology

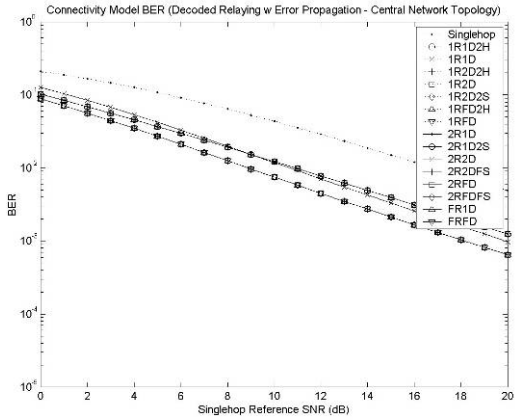


Figure 10-44
Decoded Relaying w Propagation for Central Network Topology

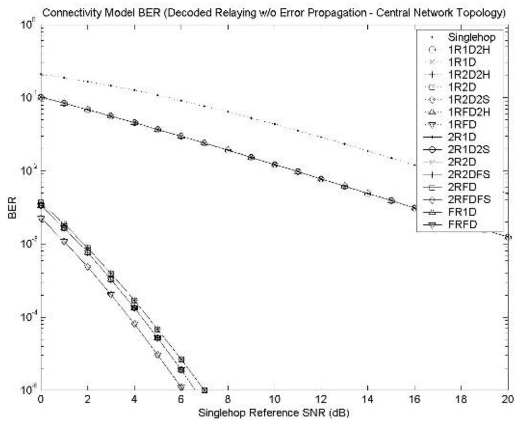


Figure 10-45
Decoded Relaying w/o Propagation for Central Network Topology

10.4.5 Cooperative mobile relaying results

10.4.5.1 Scenario 1

In this section, we present the rest of the results from Section 2.5.3.3 for cooperative relaying for Scenario 1. Figure 10-46 shows similar results as Figure 4-49 but we use 3 paths for the "good" signal and all other as interference. Thus, N=3 and K=6.

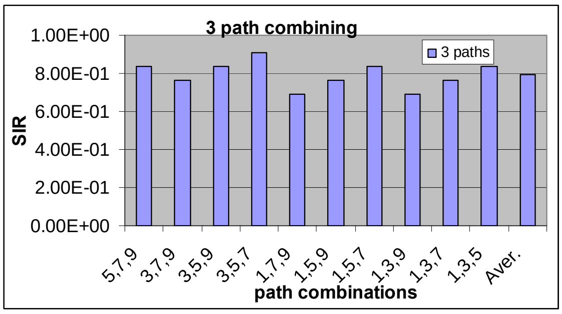


Figure 10-46 Results for M=9 and N=3

Figure 10-47 shows the same results as Figure 4-49 but as interference, we take into account only the odd numbered paths, specifically the 2-path combinations of paths 1/3/5/7/9. Effectively we have M=5.

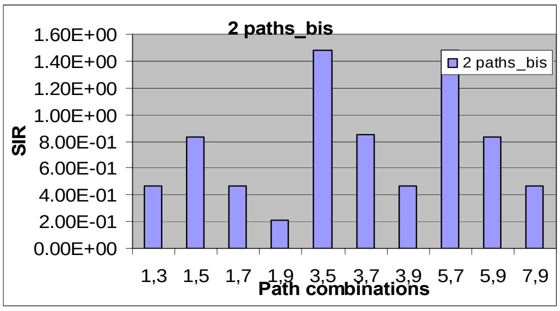


Figure 10-47 Results for M=5 and N=2

Figure 10-48 shows results for the case of N=3 and M=5. Again, we use as M and K only the odd numbered paths i.e. 1/3/5/7/9. For instance, for the case of N=1/9 we have K= 1/3/5

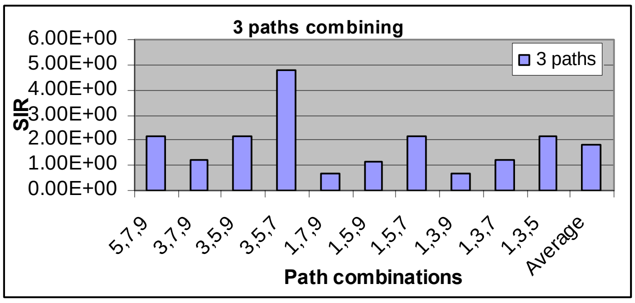


Figure 10-48 Results for M=3 and N=5

The main comments for the results presented above are the following:

- The more paths we use for combining, the better the results in terms of actual values.
- Not using many relays as interference produces significantly good results.

- The more paths we combine the more uniform the SIR values become, which is actually the target
- Additionally, the best results are obtained into positions that are closer to the UT. Thus, for selection of the best paths, positioning should be taken into account. Of course, in a realistic scenario being closer does not necessarily mean better received power levels. So, some kind of combining/selection should be in place for weighting which of the two is more important.
- A more complicated scenario with a higher number of relays should be in place in order to investigate what is the impact of more transmission from other relays at the UT.

A number of simplifications and assumptions have been taken into account for the previous analysis. These include uniform distribution of relays, trajectory on a straight line, fixed transmitted power, etc. Further work would include more complicated cases that will incorporate the use of the AP path, use of positioning/location to aid the calculations, election of the best 2 -3 paths / Find the best combinations, higher number of mobile relays, multiple power allocation / control strategies .e.g. mobile relay not transmitting at a fixed power, Relays with different trajectories etc. One of the main goals, under cooperative mobile relaying would be the projection/selection of those relays whose path combinations (always with reference to the time domain) will provide the best results in the UT in terms of actual values and uniform power levels.

10.4.5.2 Scenario 2

The rest of the results with reference to the cooperative mobile relaying for Scenario 2 are presented in the next section. First we present the table with the parameters we will assume for our simulations.

Table 10-8 ***Cases for Scenario 2 for connectivity results***

Case	Snapshot Number	Cell Max (m)	Number of MRN	Max Distance (m)	Ratio between UT/MR speed
1	100	20	50	10	0.7
2	100	20	150	10	0.7
3	100	20	300	10	0.7
4	100	20	10	10	0.7
5	500	20	50	10	0.7
6	1000	20	50	10	0.7
7	100	20	50	20	0.7
8	100	40	50	10	0.7
9	100	80	50	10	0.7
10	100	40	100	20	0.7
11	100	100	100	40	0.7

Another degree of freedom is the number of snapshots. Based on this, we run our simulation for a number of 100/500/1000 snapshots. This is represented by Case 4/5/6 in Table 10-8. Thus, in Figure 10-49 we present the same graph as in Figure 4-50 for all three cases for the snapshots.

What we see is that the number of average connections remains almost the same for the 500/1000 case, specifically 8/9 respectively. However, for the case of 100 snapshots this value is very low that of 1,5. Additionally, for Graph3 we see that more and larger variations are witnessed. However, the general profile of the min/max/average connections seems to stay “on average” the same for all three cases. Figure 10-50 shows also the coverage area, which is adequately covered by the mobile relays, similar to Case 3 of Figure 4-51.

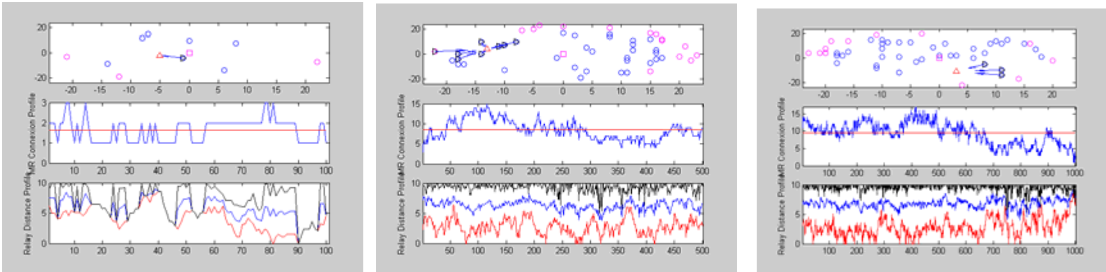


Figure 10-49 Case 4/5/6/- Graphs 1/2/3

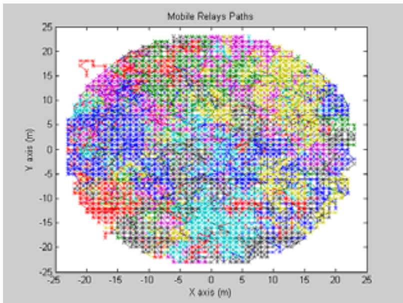


Figure 10-50 Coverage area for Case 6

Another degree of freedom is that of the cell range or as defined here the “Cell_Max”. To investigate the impact of the cell range, we use cases 7, 8 and 9 from Table 10-8. We use three values of 20, 40 and 80 meters around the AP. What the graphs in Figure 10-51 shows is the number of connections per snapshot per mobile relay. The X-axis is effectively the time domain (number of snapshots) and the Y-axis is the number of Mobile relays present in the cell area. Effectively, the summation of each “line” shows the total connections for each mobile relay. We modify the cell range and keep MRN number constant. For Case 7 we use a “Max_Distance”=20m, whereas for Case 8 and Case 9 we use a “Max_Distance”=10m

What we see is that for a relatively small cell (Case 9) we have a very small number of connections. Even for Case 8 (40 meters), we still have a small number of connections for the same number of mobile relays. However, this is not the case for Case 7, where not only we have a large number of connections (many UTs that can be used as potential mobile relays) but also high continuity and reliability for those connections. We have to bear in mind that both parameters are important and a combination of both of those should be used to select the best mobile relay(s) For instance, two mobile relays could be connected for 80% of the time, while one of those might be continuously connected and the other into blocks of time, which accumulated make up the 80% of time.

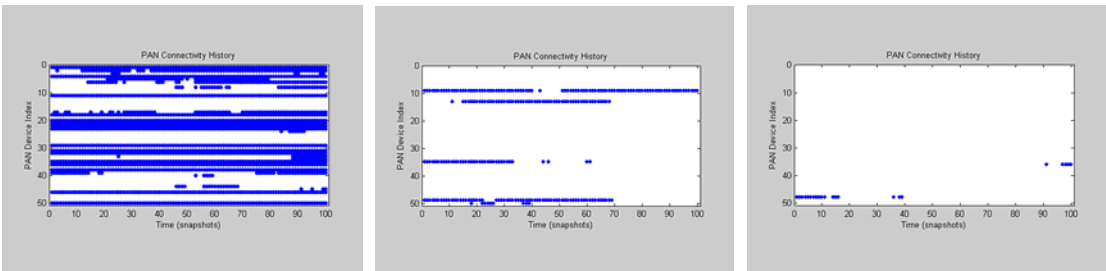


Figure 10-51 Case 7/8/9: Number of connections per relay for different cell ranges

Finally, we present two more cases. We modify the cell range “Cell_Max” from 20 to 40 and 100 meters and at the same time we modify the “Max_Distance” from 10 to 20 and 40 meters. This is a more combined approach and it represents Cases 10/11 from Table 10-8.

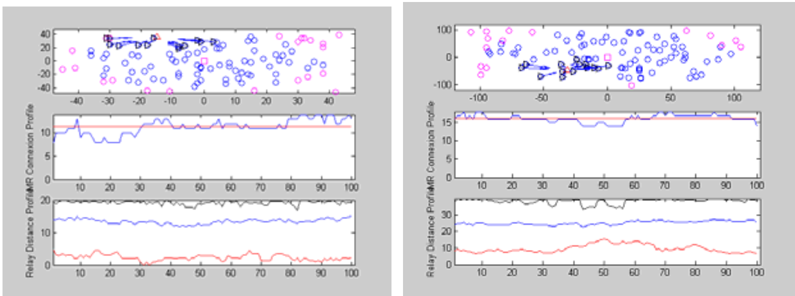


Figure 10-52 Case 10/11 - Connections

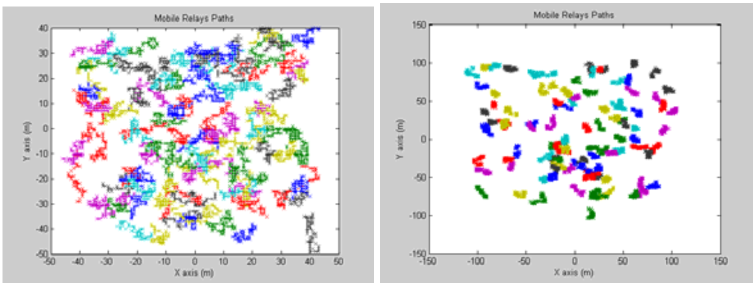


Figure 10-53 Case 10/11 – Coverage area

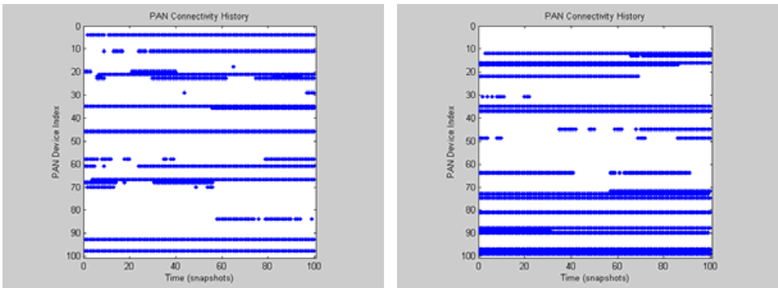


Figure 10-54 Case 10/11 – Number of connections for each MR

Figure 10-52 shows the three graphs. Comparing Graphs 2 in Figure 10-52 we see that for Case 11 the average number is 16, whereas for Case 10 is smaller and around 12. It also shows that for larger cells (Case 11), defined by "Cell_Max" we need to have an equally large area, defined by "Max_Distance" to look for potential mobile relays. Otherwise, we restrict a lot our possibilities. Graph 3 in Figure 10-52 shows that the average for min/max distance is almost doubled. The average for Case 10 is 14-15 meters and for Case 11 on average 22-25 meters. Figure 10-53 shows that the same number of mobiles are scattered in a larger area and a lower number of those are available per square meter compared to Case 10. This might influence also the continuity /reliability of connections with reference to the velocity. Figure 10-54 finally shows that for Case 11, even if we have a larger cell_area (Cell_Max), better results are witnessed, due to the larger area around the "target_UT" we aim to look for potential mobile relays. Thus, more connections are available and those seem very reliable, which as we stated previously, is what we aim for from our analysis.

10.5 Ranges Estimation for Wireless Feeder System Based on LMDS

From a WINNER point of view, the feeder system is related to the way to connect the WINNER APs with the transport network, that is, the system used to feed those APs. Of course, the transmission technology used by the feeder system can be in general terms, wireless or wire line. It is important to note that regardless the solution adopted in each particular case, the transmission technology to use would be exclusively for connecting the APs with its transport network, and therefore the WINNER users could not connect to this network, which is transparent for final users. In particular WINNER is interested to investigate and to develop the wireless solution for this feeder system, or in other words, with the system used to feed wirelessly the WINNER APs.

Hereinafter we introduce a general vision of wireless feeder systems, describing the technical parameters that were defined in the initial assumptions for WINNER scenarios, as well as the analysis and preliminary results of range estimations based on simple propagation models proposed for one of the most promising transmission technology for this purpose and which is currently used for wireless metropolitan environments, that is, the IEEE 802.16 or WMAN.

10.5.1 Wireless Feeder as one WINNER Scenario

In the internal report IR3.1 [113], produced by WP3 (new radio deployment concepts), it has been collected the initial assumptions for WINNER scenarios. The main contemplated scenarios were in-building (scenario 1), hot spot (scenario 2), urban/suburban (scenario 3) and rural (scenario 4). Further as complementary to these scenarios, it was decided to include the feeder scenario which is focused on the system for feeding wirelessly any of the APs of scenarios 2, 3 or 4. In this way, the wireless feeder system constitutes an important alternative to the wire line solution for feeding APs (or access points). In fact, the wireless feeder system has the potential to be developed far faster, less expensively and more flexibly than similar wire line systems. These potential benefits have motivated the inclusion in WP3 of the wireless feeder concept as complementary scenario to the other ones.

Although the feeder concept is well known, nevertheless it is included here because some innovative aspects can be identified, such as the feeding with NLOS conditions, as well as the cooperation of feeding and umbrella concepts. Interesting opportunities in terms of cost reduction of the wireless feeder infrastructure (versus wire feeding) have been potentially identified and therefore should be evaluated. For the previous reasons, the wireless feeder concept has been considered separately from the other WINNER scenarios.

Regardless there are a lot of proprietary solutions for implementing radio links, the radio transmission technology initially selected for the wireless feeder scenario has been IEEE 802.16 (WMAN) in the point to multipoint (PtM) mode, since the idea for the first phase of WINNER project was to use technologies based on existing systems supported for some important Standardization Organism. Besides it is convenient to remember that IEEE 802 LAN/MAN Standards Committee created in March 1999 the 802.16 working group on broadband wireless access and it focuses on standardizing US LMDS (Local Multipoint Distribution Service) systems.

In fact, at the moment there are several wireless transmission technologies like the traditional radio links and WLL/LMDS (Wireless Local Loop / Local Multipoint Distribution Service), which for example in UMTS were proposed for the access part in the transmission network, particularly for the link between radio network controller (RNC) and Node B (interface Iub) included in the radio access network of UMTS shown in Figure 10-55. For example, in environments where the population density is high and it is necessary a big bandwidth, a good solution when it is not justified the use of optical fiber due to the high costs and difficulties for installation, are the wireless access systems WLL/LMDS with licensed bands (e.g. 3.5 and 26 GHz respectively), and ranges from 2 to 20 Km depending on the operation frequency. However these systems have important problems for radio propagation at high frequencies and they are very sensitive to atmospheric effects like the rain. On the other hand for environments with lower population density so that lesser bandwidth requirements, it is possible to use conventional radio links with capacities between 2 Mbps and 32 Mbps, although nowadays there are equipments able to reach till 155 Mbps.

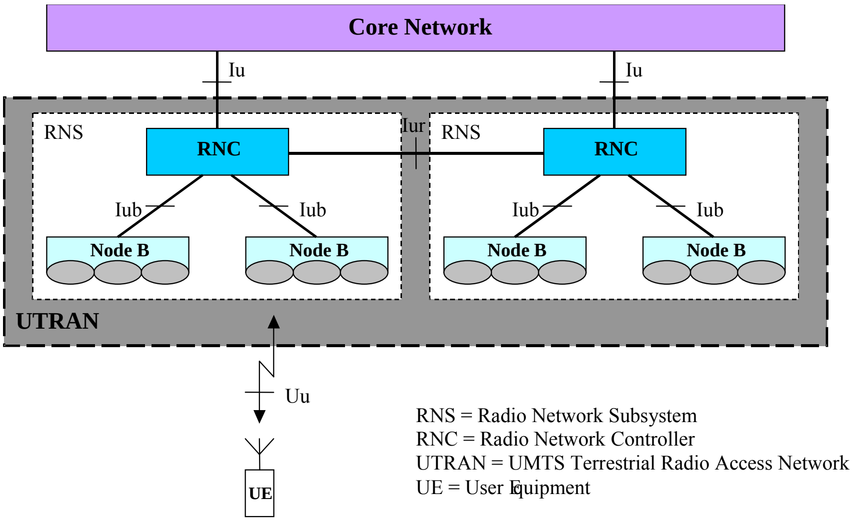


Figure 10-55: Radio access network for UMTS

Following with this trend, WINNER project will investigate the characteristics and requirements of wireless feeder system focused in some of the scenarios described in the internal report IR3.1. Although at this moment the future architecture for the WINNER access network is not clear yet, the wireless feeder concept to develop in WINNER project will tackle with the feeding of WINNER APs with wireless point to multipoint (PTM) links. Moreover one important characteristic of the wireless transmission technology used by the feeder system to develop in WINNER is that final users couldn't connect to this network. In other words, the transmission technology used for connecting WINNER APs with transport network would be especially thought for this purpose and will not be able to serve to final users.

10.5.2 General Vision of Wireless Feeder System

As we have mentioned before, there are different options for connecting the APs of a given network to its transport network, using a wire line or by means of a radio link depending on the available resources and the requirements of this connection. In other words, taking cost and reliability level into account, we should decide for each case what class of link is the most suitable for this connection. However the wireless feeder system to develop inside WINNER is devoted to radio link solution with special attention above all in point to multipoint topology.

Otherwise we have just pointed that the wireless feeder scenario can be seen as complementary to the other WINNER scenarios. For example the hot spot scenario (Manhattan zone) could be covered with an AP and possibly some homogeneous and heterogeneous relay nodes of a given technology. In addition, the AP need to be fed by means of some transmission technology, being the wireless feeding technology an important candidate to play this role, in alternative to traditional wire feeding solutions. The same concept is valid in principle for the other WINNER scenarios, in particular for feeding an urban/suburban or rural area.

Figure 10-56 shows the wireless feeder concept highlighting a functional partition from a system point of view. The wireless feeder system is constituted by a "Master Station" and one or more "Peripheral Stations". In the example outlined here, the transport node which acts as a traffic concentration point includes the wireless feeder master station, as well as the AP includes the wireless feeder peripheral. However, this is an implementation choice, and the concept remains valid even if this integration is not done. A distinguished point for the feeder concept included here is that the wireless link is not towards a final user but towards another node, such as in the next examples:

- To connect an AP (like WINNER scenario 1, more generally in-building) to the backbone network.
- To connect the AP of a hot-spot to a traffic concentration point.

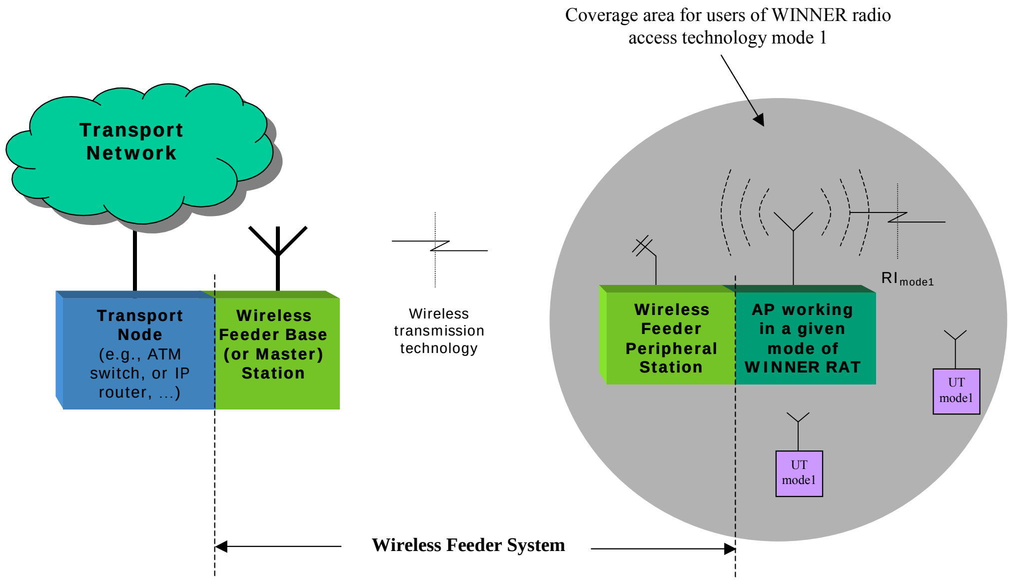


Figure 10-56 Wireless Feeder System

Some of the advantages or benefits for using a wireless feeder system, with respect to a wire feeder system, may be summarized in the following points:

- Overall feeding infrastructure lower cost (depending on the application context).
- Low installation and maintenance costs.
- Dynamic capacity allocation.

For existing wireless PtM systems with some tentative of standardization behind there are obviously important limitations concerning to maximum radiated power, sensitivity threshold, bandwidth modulation and coding schemes implemented, as well as others. Therefore in the best of the cases, the current maximum reachable throughput and range (e.g. 802.16) may be around few Mbps and Kms. In [124] is included a technical overview of the IEEE Standard 802.16, explaining the standardization process as well as the medium access control and physical layer features of this new standard. On the other hand, the initial objectives of Wireless Feeder System fixed in IR3.1 (described later on) were 10 Gbps (case NLOS) and 25 Gbps (case LOS) for cell throughput, and 5 Km (case NLOS) and 25 Km (case LOS) for cell size. So the final scope of WINNER about this topic, should be to study and investigate how to reach these objectives. In this sense and in a first approach there would be different possibilities, some of them addressed modifications of existing systems, and other focused on the development of new particular mode of WINNER radio interface specifically thought for wireless feeder systems.

In order to avoid confusion with the heterogeneous relay concept, at least in the context of WINNER project, we have to remember that the transmission technology involved in the wireless feeder system is only applied for connecting the APs of a given network with its correspondent transport network, and so users can not employ this technology for accessing to the network. On the other hand the connection between an AP and a heterogeneous relay is implemented using a radio technology that is also employed for connecting and serving final users. Further, assuming the common preliminary radio network architecture developed in WINNER as well as other European projects of the same frame work programme (FP6) like AN (Ambient Network) project, we could separate the heterogeneous relay concept and the wireless feeder concept, as it is shown in Figure 10-57. The logical nodes included in the transport network are the following:

- Access Router (AR): is a logical IP layer node that performs the tasks attributed to an Access Router as defined in relevant IETF specifications.
- Radio Access Network Gateway (RANG): is a logical network node terminating the data link layer. It terminates generic user plane protocols.
- ACS is a logical network node that controls the access to the radio interface resources. It terminates generic control plane protocols.

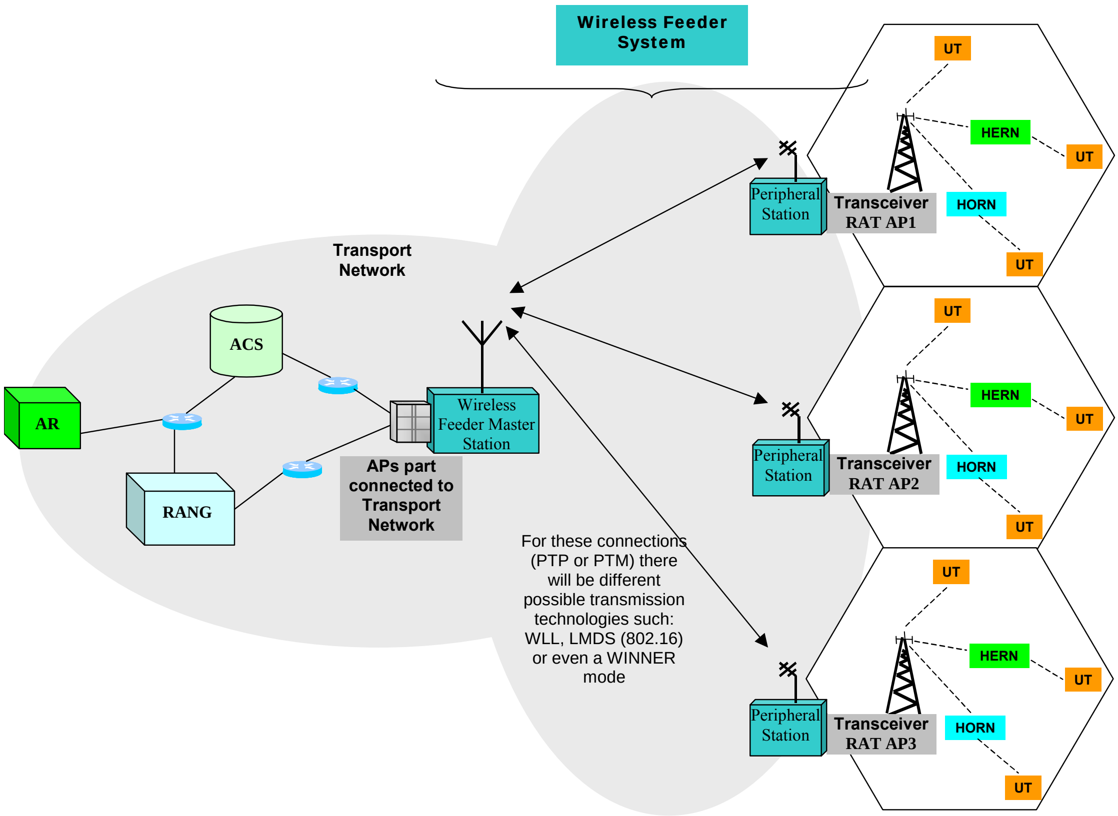


Figure 10-57: Wireless Feeder System for WINNER system

In this example it is illustrated a wireless feeder system composed by a master feeder station on the side of transport network and three wireless feeder peripheral stations on the access network side. Each of these peripheral stations is wirelessly fed by the same master station using point to point or point to multipoint connections and a given transmission technology. Obviously the selection of the topology as well as the characteristics of the transmission technology used in a particular case will depend on the requirements of total throughput and quality of service level that deployment plan have imposed for the area where the use of wireless feeder system is scheduled. For instance, some times in order to achieve the maximum engaged throughput could be better to use different frequencies for each wireless feeder peripheral station (some solution based on FDD duplexing methods), even different frequencies also for DL and UL links, instead of the same frequency for all the peripheral stations, multiplexing in time (some solution based on TDD duplexing methods) the data of each one of them and even the data for DL and UL. Of course there are many possibilities that should be analyzed in depth before deciding the best option for different scenarios and from different viewpoints (economical aspects, complexity, …).

10.5.3 Technical Parameters and Requirements for Wireless Feeder System

Other assumption done in IR3.1 about the wireless feeder scenario was to contemplate two different cases, for conditions of non line of sight (NLOS) and line of sight (LOS) respectively. This initiative has been contemplated also in [125] by the technology behind WiMAX (similar to the WiFi alliance but for Wireless Metropolitan Area Network). The NLOS case is devoted to the urban feeder (for instance feeder for hot-spot WLAN) and it has a special interest since its frequency requirements (< 6GHz) in terms of allocation should be considered together with the frequencies of the other WINNER scenarios.

The LOS feeder includes rural feeder scenarios, in addition to some particular urban cases with LOS conditions. For the sake of completeness, the ultra high capacity feeder application can be taken into account, up to the point to point high capacity boundary case; the concept is not new, but it may be worthwhile to consider in this case the new technologies in terms of potential upgrade of performances. It should be noted however that this last case addresses frequencies higher than 6GHz.

Table 10-9 shows the technical parameters that define each of these cases for the wireless feeder scenario. It is convenient to clarify that these parameters which defining the feeder scenario, are in accord with the connection or link between the feeder system and the fed nodes, that is, the master station and the peripheral stations. In this way the cell size is really related with the distance between the feeder system and the wirelessly fed node, and so on.

Table 10-9 Technical parameters for wireless feeder system

Parameter	Case NLOS	Case LOS
Mobility	Fixed	Fixed
Coverage/range	Urban/Suburban	Rural and some Urban/Suburban cases
Cell size (Km)	< 5	< 25
Cell throughput (Gbps)	5 – 10	10 – 25
Link data rate (Gbps)	< 1	1 (PtM) – 25 (PtP)
Spectrum & requirements (GHz)	< 6	> 6
Spectrum usage	Dedicated	Dedicated
Traffic parameter	Very high data rates High QoS data	Very high data rates High QoS data

10.5.4 Assessment of Current Technologies for Wireless Feeder System

Following we describe the simple propagation models used for the estimation of maximum reachable ranges in radio links point to multipoint based on existing systems supported for some important Standardization Organism, like is the case of LMDS systems recently standardized by 802.16 group. Of course, these ranges will depend on the used frequency band, the used modulation scheme, the used coding scheme, the required BER as well as other parameters. LMDS technology is applicable in many markets, from dense urban environments to rural areas, where there is no existing or poor wired infrastructure.

Regarding standardization, Table 10-10 summarizes some characteristics of the last standardization efforts performed in fixed broadband wireless access PtM communication systems.

IEEE designation	ETSI designation	Spectrum	Propagation conditions	Modulation	Bandwidth and bit rate	Duplexing alternatives
802.16 (12/2001) WMAN-SC	HiperACCESS (5/2002)	10 – 66 GHz (LMDS)	LOS	QPSK, 16QAM, 64QAM	20, 25 and 28 MHz. 32 – 134 Mbps for 28 MHz bandwidth	TDD, FDD
802.16a (1/2003) WMAN-SCa WMAN-OFDM WMAN-OFDMA	HiperMAN (4/2003)	2 – 11 GHz (MMDS)	NLOS	OFDM 256 sub-carriers QPSK, 16QAM, 64QAM	1.25 to 20 MHz flexible. Up to 75 Mbps for 20 MHz bandwidth	TDD, FDD

Table 10-10: Standardization efforts in fixed broadband wireless access (FBWA)

The main objective of any radio link is to deliver sufficient signal power to the receiver in order to achieve the compromised performance for a given communication, commonly specified as a minimum bit error rate (BER). From a receiver demodulator point of view this BER is a function of the signal to noise ratio (SNR), and high order modulation results in greater spectral efficiency but requires obviously higher signal to noise ratio (more transmitted power). However not only the path loss by free space propagation should be contemplated.

In general terms there are three different physical mechanisms which drive the radio propagation and determine the variation of the characteristics of radio signal. These are:

- Refraction in the earth's atmosphere alters the trajectory of radio waves and may change with time (term 4/3 earth radius). The index of refraction decreases monotonically with increasing height.
- Diffraction or scattering effects resulting from objects near the direct path (Fresnel zones).
- Reflections from objects, either near or far from the direct path.

To estimate the maximum reachable range in a wireless feeder system for the particular case where the used transmission technology is based on 802.16 standard, we have selected in a first approach two simple propagation models for LOS and NLOS conditions respectively. In further works, it will be convenient to analyze the propagation models proposed by WP5 of WINNER project, in particular for those models more applicable to the radio technology used by the wireless feeder system.

The frequencies used for each of the contemplated cases, have been 3.5 and 5 GHz for the NLOS condition, and 26 GHz for the LOS condition. It is important to remark that the propagation models proposed for estimating maximum ranges, contemplate neither interference effects not fading (multipath). In any case below are the typical sources of interference for LMDS systems:

Intra-System interference.

- Multipath.
- Cross polarization component.
- Adjacent channel interference.
- Co-channel interference.

Inter-System interference.

- Satellite systems.
- Other LMDS systems.
- Out-of-band interference.

Regarding the way to mitigate intra-system interference, follows general advices and recommendations to take into account in the design of LMDS systems.

- Multi-path.
    - Use highly directional antennas.
    - Give careful consideration to placement of antennas.
    - Use antennas with low side lobes.
    - Use robust modulation and error correction techniques.
- Cross polarization.
    - Major factor for systems which exploit polarization at same access point for frequency reuse need antennas with good cross-polarization.
    - Minor problem for systems which use polarization for separating access points.
    - Use robust modulation and error correction techniques.
- Adjacent channel.
    - Use constant envelope modulation.
    - Use linear power amplifiers.
    - Use robust modulation and error correction techniques.
- Co-channel interference.
    - Use highly directional antennas with low side lobes.
    - Deploy access points with maximum separation distance for same frequency, same polarization.
    - Use minimum transmit power and control TX power on return path.
    - Use robust modulation and error correction techniques.
    - Use adaptive interference suppression techniques.

Although for a particular path with LOS conditions has adequate Fresnel zone clearance, the path loss may differ significantly from free space under normal refraction conditions. The cause of this is probably multi-path propagation resulting from reflections. Besides, for digital transmission systems the multi-path propagation is an added problem and some extra considerations should be considered for digital systems.

- Depending on the relative phase shifts of the paths, the signals traversing them at a given frequency can add constructively to provide a gain with respect to a single path, or destructively to provide a loss. On longer paths in particular, the effect is usually a loss.
- Techniques such a spatial diversity may help.
- Some problems can be solved by using an adaptive equalizer in the receiver. Another way to attack some problems is to increase the symbol length while maintaining a high bit rate by using a multi-carrier modulation scheme such as OFDM. A common rule of thumb prescribes that the multi-path delay spread should be no more than about 10 % of the symbol length.
- The best way to mitigate the multi-path effects in some situations is to use highly directional antennas, preferably at both ends of the link. The higher the data rate, the more critical it becomes to use high-gain antennas.

10.5.4.1 Propagation model for NLOS conditions

The propagation model employed for estimating the maximum range in a wireless feeder system using LMDS as radio transmission technology for NLOS conditions is in according to IEEE 802.16.3c-01/29r4 recommendation [126]. It should be noted that although this model in fact was developed for line of sight conditions but without a total clearance between transmitter and receiver, in a first approach we have applied this model for NLOS conditions in WINNER scenarios, using antenna height and directivity [127], with frequencies lower than 6 GHz. The proposed set of propagation models are applicable to multi-cell architecture in scenarios with the following characteristics:

- Cells are lesser than 10 Km in radius, variety of terrain and tree density types.
- Under window or rooftop installed directional antennas (2 - 10 m) at the receiver.
- 15 – 40 m BTS antennas.
- High cell coverage requirement (80 – 90 %).

Moreover the wireless channel is characterized by:

- Path loss (including shadowing).
- Multipath delay spread.
- Fading characteristics.
- Doppler spread.
- Co-channel and adjacent channel interference.

It is to be noted that these parameters are random and only a statistical characterization is possible. On the other hand, usually beam-forming and MIMO antennas are used in order to improve the NLOS performance. The above propagation model parameters depend upon terrain, tree density, antenna height and beam width, wind speed, and season (time of the year).

The model covers three most common terrain categories found across the United States. Of course, other sub-categories and different terrain types can be found around the world. The maximum path loss category is hilly terrain with moderate to heavy tree densities (Category A). The minimum path loss category is mostly flat terrain with light tree densities (Category C). Intermediate path loss condition is captured in Category B. The model is based on extensive experimental data collected by AT&T Wireless Services across the United States in 95 existing macrocells at 1.9 GHz.

The proposed propagation model is summarized in the following equation and it is valid for distances between transmitter and receiver higher than d_0 (= 100 m):

Decoded Relaying w/o Propagation for Suboptimal Example Network Topology

$$L_{NLOS}(dB) = 12.44 + 20\log f + 10 \cdot \gamma \cdot \log\left(\frac{d}{d_0}\right) + s + \Delta PL_f + \Delta PL_h \tag{61}$$

where:

ΔPL_f is the frequency correction term given by (f in MHz):

$$\Delta PL_f = 6 \cdot \log\left(\frac{f}{2000}\right) \tag{62}$$

ΔPL_h is the receive antenna height correction term given by (h is the receive height between 2 m and 10 m):

$$\Delta PL_h = -10.8 \cdot \log\left(\frac{h}{2}\right) \quad \text{for categories A and B} \tag{63}$$

$$\Delta PL_h = -20 \cdot \log\left(\frac{h}{2}\right) \quad \text{for category C} \tag{64}$$

γ is the path-loss exponent with the following expression for h_b (height of the master access point in meters) between 10 m and 80 m.

$$\gamma = a - b \cdot h_b + \frac{c}{h_b} \tag{65}$$

a, *b* and *c* are constants dependent on the terrain category which values are reproduced below:

Constant	Terrain Category A	Terrain Category B	Terrain Category C
a	4.6	4	3.6
b	0.0075	0.0065	0.005
c	12.6	17.1	20

finally *s* represents the shadowing effects and follows lognormal distribution (typical value of the standard deviation for *s* is between 8.2 and 10.6 dB depending on terrain type).

10.5.4.2 Propagation model for LOS conditions

The LMDS systems at high frequencies (e.g. 26 GHz licensed band for this kind of systems in some European countries, or in general up 6 GHz for the wireless feeder system defined in WINNER in LOS conditions) are most susceptible to rain effects causing a reduction in the signal level. Rainfall causes depolarisation of the signals, leading to decreased signal level and decreased interference isolation between adjacent sectors and adjacent cell sites. Also at these frequencies, multipath fading should not be an important effect, and for LOS conditions shadowing and diffraction do not occur as often at lower frequencies. Moreover the height of the antennas as well as using directional antennas in a wireless feeder system plays a large role in reducing multipath effects.

For the previous raisons, we have selected a simple propagation model which only takes into account the path loss for free space propagation and the attenuation due to rain. In fact at millimeter wave, rain fading is the most dominant factor. Attenuation occurs due to absorption and scattering in rain. Fading due to rain attenuation is described empirically form link tests and point rainfall data. Due to raindrops are oblate rather than spherical, attenuation tends to be greater for horizontally polarized signals than for vertical polarized signals (IEEE 802.16cc-99/13).

Therefore the propagation model used for estimating the maximum range in LOS conditions has included the free space path loss and the signal attenuation caused by a given rain rate, like the following expression shows, where the frequency (*f*) is in MHz and the distance (*d*) in Km:

$$L_{LOS}(dB) = 32.44 + 20\log(f \cdot d) + A_{rain} \tag{66}$$

For estimating the long-term statistics of rain attenuation (Arain) we have used the simple procedure proposed by the recommendation ITU-R P.530-9 (2001):

- Obtain the rain rate R0.01 exceeded for 0.01% of the time (with an integration time of 1 minute). Although the recommendation includes different regions for Europe, we have only used the two regions contemplated for the Spain case which are H and K, with the values of 32 mm/h and 42 mm/h respectively.
- Compute the specific attenuation for the frequency, polarization and rain rate of interest using Recommendation ITU-R P.838.

According to this procedure the rain attenuation is described for the following expression:

$$A_{rain}(dB) = r \cdot d(Km) \cdot a \cdot R^{b}(dB/Km) \tag{67}$$

where:

r is the reduction factor (lesser than 1) since the rain affects only one part of the radio link and is calculated from the equation:

$$r = \frac{90}{90 + 4 \cdot d} \tag{68}$$

d is the distance between transmitter and receiver in Km, R is the rain rate exceeded for 0.01% of the time, as we have seen before, and a and b are constants depend on the frequency and polarization used. Below are the values of these constants for the case of 26 GHz, and for horizontal and vertical polarizations.

Frequency (GHz)	a_H	a_V	b_H	b_V
26	0.135	0.122	1.053	1.024

10.5.4.3 Range Estimation Results

From a general point of view for evaluating any wireless link would be necessary to know:

- The available radio frequency power.
- The available bandwidth.
- The required reliability (BER).

The goal of this section is to present a first estimation of maximum reachable ranges for LMDS systems, based on the simple propagation models, previously described, and for different radio propagation conditions (in terms of frequency bands and line or not line of sight). In these theoretical results, it has taken into account some radio transmission parameters recommended for 802.16, such the maximum transmitted power and the required SNR threshold for different kind of modulation schemes as well as for the bit error rates (BER) allowed in order to get a given QoS level.

The procedure used for the estimation of maximum reachable ranges has been the following:

- To calculate the total path loss for each of the propagation models used (LOS and NLOS).
- To calculate the SNR in the input of receiver taking the EIRP radiated, the total path loss, the noise power and the receiver antenna gain into account.
- To fix the required SNR for different modulation schemes and for a BER of 10-6.

So the signal to noise rate is calculated in according with the following expression:

$$SNR_{calculated}(dB) = EIRP_{tx}(dBm) + G_{rx}(dBi) - L_{(N)LOS}(dB) - N_{rx}(dBm) \tag{69}$$

where:

$EIRP_{tx}$ is the total radiated power in dBm (including transmitter antenna gain),

G_{rx} is the antenna gain in the receiver in dBi,

$L_{(N)LOS}$ the total path loss for LOS or NLOS conditions in dB,

and N_{rx} is the receiver noise floor which include the theoretical thermal noise (Boltzman's equation) plus the noise figure (NF) and implementation loss (IL) in the receiver itself, according with the following equation:

$$N_{rx}(dBm) = N(dBm) + NF(dB) + IL(dB) \tag{70}$$

where:

N is the thermal noise power in dBm ($N = k\,T\,B$),

k is the Boltzman's constant (1.38 x 10-23 J/K),

T is the system temperature, usually assumed to be 290ºK,

B is the channel bandwidth in Hz,

The thermal noise or channel noise is intimately tied to bandwidth used for the communication system, and it is the amount of radiation emitted, in the form of random or Gaussian noise, by the temperature effect of the transmitter.

According to 3GPP TS25104 the reference sensitivity level is the minimum mean power received at the antenna connector at which the Bit Error Rate (BER) shall not exceed the specific value indicated for a given modulation and coding scheme. In this way for the particular case that we have analyzed, the receiver SNR required for a BER level of 10-6 and for different modulation schemes and lower coding rates is shown below:

Table 10-11 SNR required for a BER level of 10-6

Modulation	Receiver SNR required for 10^{-6} BER (dB)
QPSK	9.8
16QAM	16.8
64QAM	23.0

Finally the fade margin is the amount of extra RF power radiated to overcome the multi-path phenomenon. Although this depends on the desired reliability of the link, and a good rule-of- thumb is to use between 20 and 30 dB. However for the proposed propagation models we have not considered any interference and multi-path effects. In any case the fade margin would be calculated by means of the following expression:

$$F_{m\arg in}(dB) = SNR_{calculated}(dB) - SNR_{required}(dB) \tag{71}$$

Hereinafter the graphic results obtained for NLOS and LOS conditions according to the proposed models for each of them. The figures were derived assuming the following values:

- Bandwidth: 20 MHz.
- Radiated power (EIRP): 47 dBm.
- Rx antenna gain: 18 dBi.
- Implementation loss: 2 dB.
- Noise figure: 5 dB.

10.5.4.4 Results for NLOS Condition

For estimating the maximum range in the NLOS case, two frequencies were used according to spectrum requirements (below 6 GH) stated in the preliminary assumptions of WINNER scenarios (IR3.1). These frequencies were 3.5 GHz and 5 GHz.

In accord with the proposed propagation model for NLOS conditions Figure 10-58 and Figure 10-59 show the total path loss for 3.5 and 5 GHz respectively, in terms of distance between transmitter and receiver for a wireless feeder system using a transmission technology based on 802.16 standard. Also it is illustrated the behaviour for two different antenna heights as well as for three different terrain categories.

- Category A: hilly terrain with moderate to heavy tree densities.
- Category B: intermediate path loss condition.
- Category C: mostly flat terrain with light tree densities.

As we can see for both frequencies the influence of terrain category is less significant in the case of using short antennas (10 m for master station and 2 m for peripheral station or AP) than in the case of using long antennas (80 m for master station and 10 m for peripheral station), although of course the path loss is bigger for short antennas.

Regarding the calculated SNR Figure 10-60 and Figure 10-61 illustrate the values of this parameter in terms of distance for 3.5 GHz and 5 GHz respectively, as well as for different terrain categories and antenna heights. Taking into account the required SNR for different modulation schemes, the maximum reachable ranges for 3.5 GHz and short antennas are between 800 m and 1.4 Km in terms of the used modulation scheme (64QAM, 16QAM or QPSK) and terrain category. These distances shorten a little bit for the case of 5 GHz, between 700 m and 1.2 Km. However for long antennas the ranges are between 2.6 Km and more than 5 Km in the case of 3.5 GHz, and between 2.2 Km and some more than 5 Km in the case of 5 GHz. As in the path loss charts, we can see that the influence of terrain categories is more important in the case of using long antennas than in the case of short antennas.

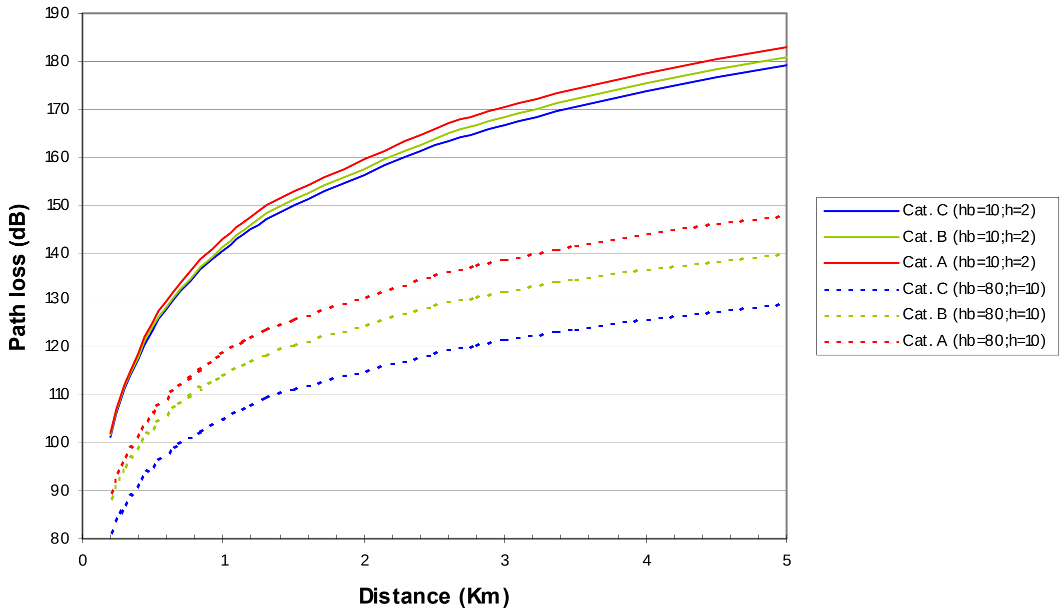


Figure 10-58 Path loss for 3.5 GHz and NLOS conditions (different terrain categories and antenna heights)

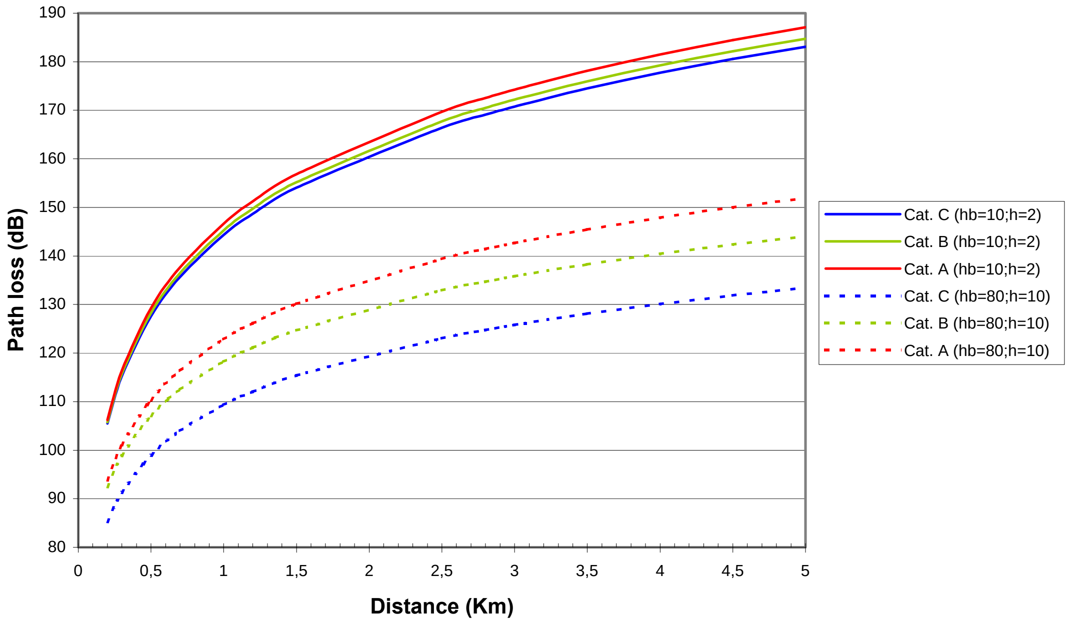


Figure 10-59 Path loss for 5 GHz and NLOS conditions (different terrain categories and antenna heights)

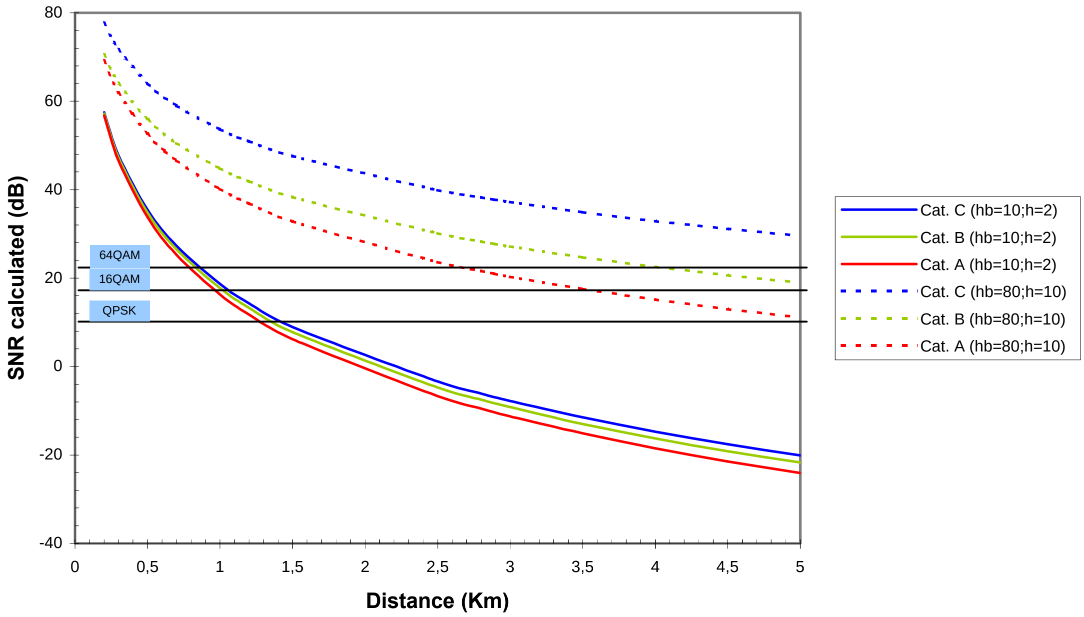


Figure 10-60 SNR calculated for 3.5 GHz and NLOS conditions (different terrain categories and antenna heights)

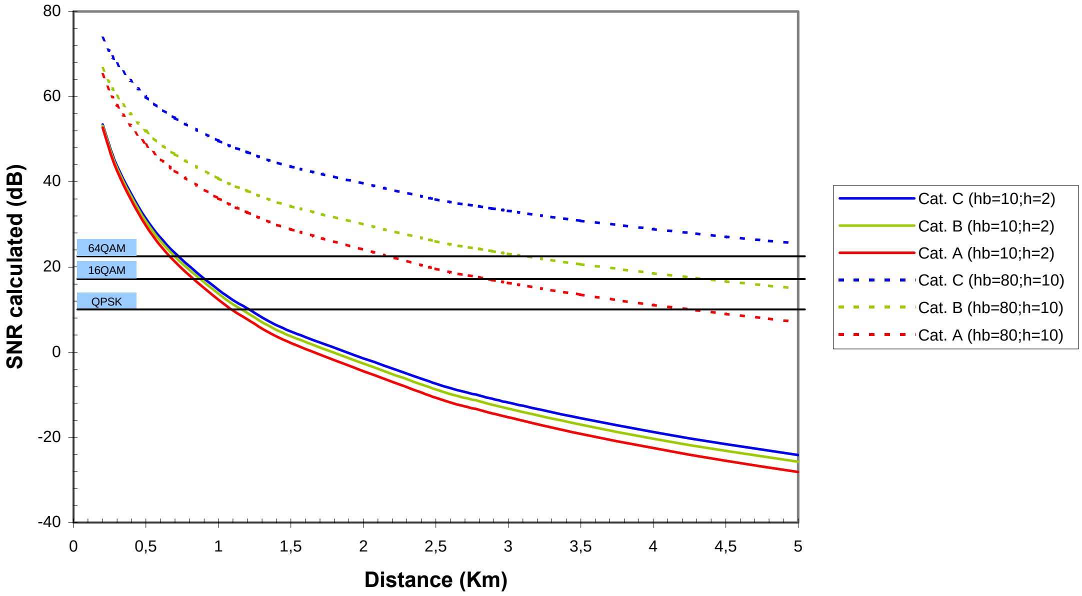


Figure 10-61 SNR calculated for 5 GHz and NLOS conditions (different terrain categories and antenna heights)

10.5.4.5 Results for LOS condition

For estimating the maximum range in the LOS case, only one frequency was used according to spectrum requirements (up 6 GH) stated in the preliminary assumptions of WINNER scenarios (IR3.1). This frequency was 26 GHz, which is included in the licensed band for LMDS systems.

Figure 10-62 shows the total path loss for 26 GHz in LOS conditions, in terms of distance between transmitter and receiver for a wireless feeder system using a transmission technology based on 802.16 standard. Also it is illustrated the behaviour for two different rainfall regions as well as for two different polarizations. For comparison besides it is included the path loss at this frequency but only taking free space propagation into account. According as distance is larger the influence of the polarization type is bigger, being vertical polarization the most favourable case.

About calculated SNR in LOS condition, Figure 10-63 illustrates the theoretical values of this parameter in terms of distance for 26 GHz, as well as for different rainfall regions and polarization types. Taking into account the required SNR for different modulation schemes, the maximum reachable ranges are between 1.8 Km and 4.5 Km in terms of the used modulation scheme, region and polarization. Only considering free space propagation it would be possible to reach ranges around the 25 Km using moreover the simplest modulation scheme contemplated in our analysis, that is, QPSK, since even for free space propagation the maximum distance is approximately 7 Km and 12 Km for 64QAM and 16QAM respectively.

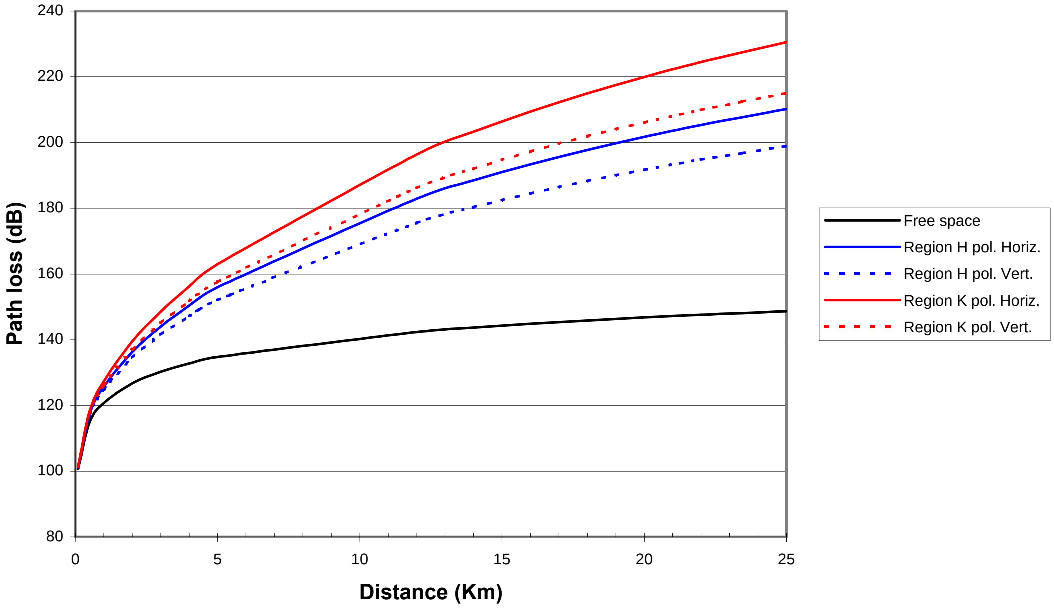


Figure 10-62: Path loss for 26 GHz and LOS conditions (different rain regions and polarizations)

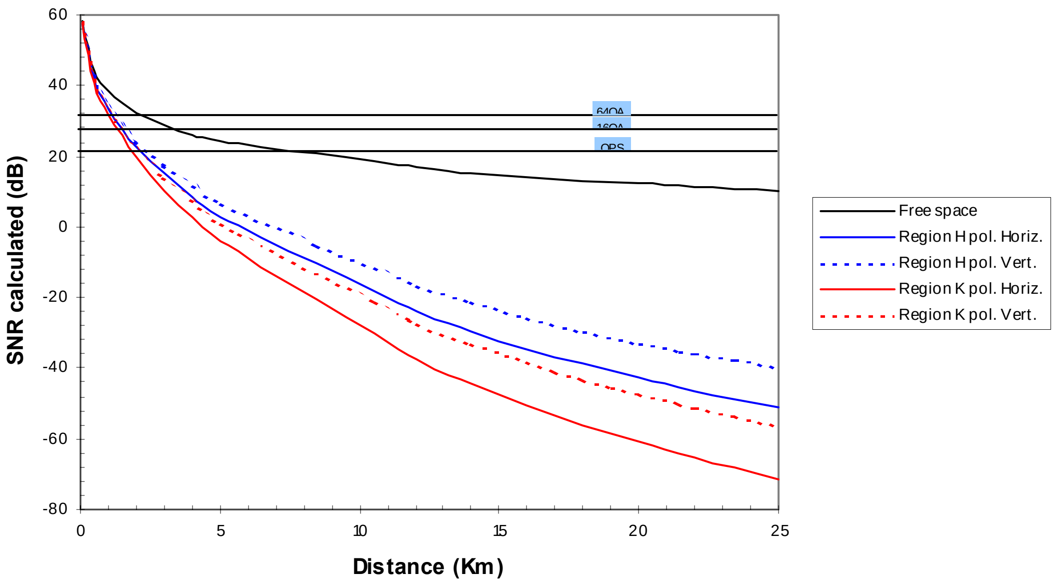


Figure 10-63: SNR calculated for 26 GHz and LOS conditions (different rain regions and polarizations)

10.5.5 Conclusions

The range theoretical estimation performed in wireless feeder system and based on optimistic (interference and multipath effects not considered) and simple propagation models for the particular case of using 802.16 as wireless transmission technology, has shown that the maximum reachable distances are far away of the proposed for this kind of systems in the IR3.1 (preliminary assumptions for WINNER scenarios). In particular for LOS conditions, the objective stated was of 25 Km while the results obtained in the theoretical analysis were between 1.8 Km and 4.5 Km. Also it is important to note that the current maximum bit rate for 802.16, around 134 Mbps is also far away from the fixed objective for wireless feeder system in the preliminary assumptions, between 1 and 10 Gbps.

So far, it is not decided the radio transmission technology more appropriate for using in WINNER wireless feeder system, and so at this moment it is not clear if WINNER radio interface will provide some particular mode working in a given frequency band (hence with the corresponding request of spectrum) specifically thought for this purpose, or if future versions or evolutions of 802.16 standard or others will contemplate the necessary requirements from different point of views (e.g. QoS level and total throughput) for this kind of systems.

In this last respect, recently we contacted with people involved in WP2 in order to know if they are contemplated inside the WINNER air interface modes, the development of a particular mode for feeding the WINNER APs in a wireless way. Although the answer to this question from task 2.7 report about the air interface concept proposal [117], was initially positive, the specific mode to which this report alludes is the "Wide area – Feeder links" mode. In fact this mode refers to the mode for connecting access point to access point, access point to relay node, or relay node to relay node, with cell ranges of 1 Km and peak data rates around 5 Mbps per MHz, but not explicitly for connecting the access points or APs to its transport network, that is, the wireless feeder system described previously in this section. Besides the achievable coverage and throughput figures of "Wide area – Feeder links" mode, so far based on only preliminary calculations, are far away of the established figures in the preliminary assumptions of WP3 [113].

So, at the moment the only mode contemplated in WP2 with feeder purpose, does not exactly adjust to the preliminary assumptions performed by WP3 for the wireless feeder system. However, further research focused mainly on the output power available in master and peripheral stations (obviously taking EMC aspects and power amplifier limitations into account), as well as antenna solutions, in particular the possibility to use highly directive antennas at both stations in the LOS case (offering large gains), could bring close the achievable ranges and throughputs of this mode to the initial presumed figures for wireless feeder system. Regarding the antenna solutions, it should be noted that the MIMO techniques may not applicable since multiple uncorrelated paths may not be attainable, and vice versa for the NLOS case.

Finally, related with the idea to use two different frequency bands in terms of line or not line of sight condition (< 6GHz for the LOS case and > 6 GHz for the NLOS case), we have to consider an important comment from WP2, which states that is not obvious that we can use two different spectrum portions for each of the cases, simply due to the availability and cost of spectrum. Anyway this is of course a political and also a business related question in the end very much out of our control of WINNER project, and therefore we should avoid a feeder concept that requires different spectrum allocations for the different cases. However if we could get different spectrum for those cases, then we should have a concept that can take advantage of this fact.

11. Annex IV: System Simulator Calibration Case

This chapter proposes a calibration case to be used in WINNER WP3. The proposal is compiled considering that the calibration case should be as simple as possible but at the same time including the most essential components and functionality expected from a system level simulator.

The calibration case comprises assumptions and models of different kinds. Below, the assumptions and models that belong to the same category are grouped together in separate sub-chapter. Every such sub-chapter first describes the assumptions made and then, in a second step, it is defined how these assumptions should be implemented and modeled in the simulator.

Table 11-1 General simulation parameters

Parameter	**Description/Value**
Static vs. dynamic simulations	Quasi-static simulations. Physical user mobility is not modelled but channel variations are modelled.
Simulator time step	22.5 μs one OFDM time slot duration
Simulation time duration (real time) [s]	TBD

11.1 User Behaviour

11.1.1 Assumptions

Table 11-2 describes the assumed user behaviour.

Table 11-2 User density, location and mobility assumptions.

Parameter	**Description/Values**
Number of users	Fixed number of users according to expected user density. Exact figure TBD
User location	Outdoors. Random location.
User mobility	No. Users are stationary.
Used service	File up- or downloading 50/50 ratio.

11.1.2 Models

11.1.2.1 User Distribution and Mobility Models

Users are distributed uniformly over the network Figure 11-1, the network borders are specified by hexagon borders. The users are not moving in the network. When a connection is ended (one file is uploaded or downloaded) a new user with a new position is created in the network. The new position is drawn from uniform distribution covering the whole network. This will create load variation between AP.

11.1.2.2 Traffic Models

Constant bit rate file up- and downloading. In CBR model constant packet sizes arrive with constant time interval. The file size is 3.0 Mbytes and the packet size is 1500 bytes, the packet interval is 0.0012 s resulting 10.0 Mbit/s offered traffic per connection. The service is characterized as a best-effort service.

11.2 Network Deployment

11.2.1 Assumptions

The network deployment assumptions are summarized in Table 11-3. The network is assumed to be a single-hop cellular network with antenna installation above rooftops (deployed in an urban or sub-urban area). These assumptions refer to the test environment C2.

Table 11-3 Network deployment assumptions

Parameter	Description/ Value
Deployment	Single hop communications above rooftop antenna installation, omni case, hexagonal cells.
Cell size (m)	
Hexagon brink[19]	100 [m][20]
Number of sites	19
Number of sectors per site	1

11.2.2 Models

11.2.2.1 Omni case

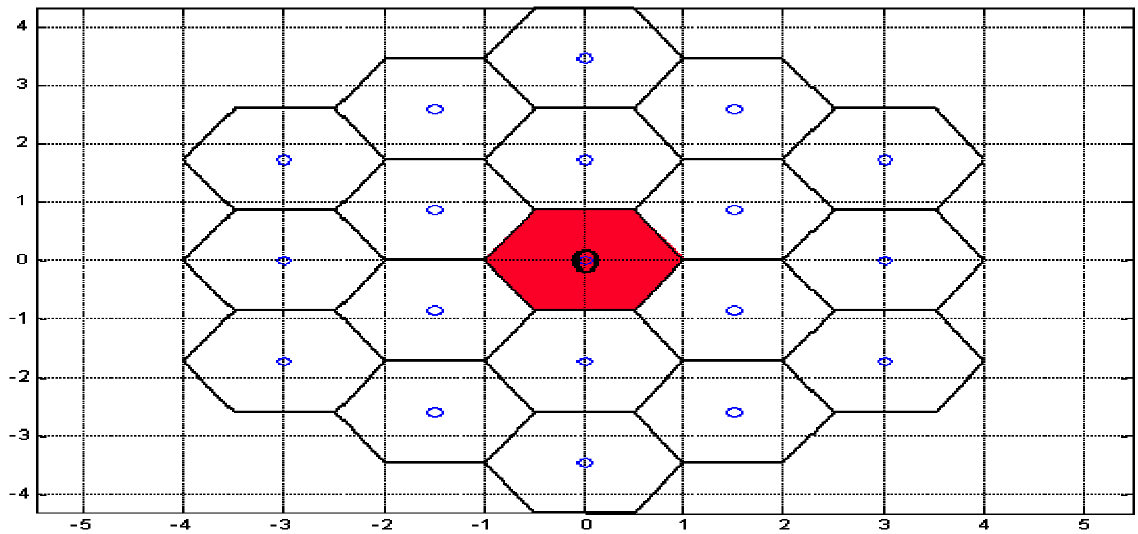


Figure 11-1 Omni case network includes 19 AP. Statistics are collected from all connections. The x- and y-axis are relative numbers so that with hexagon brink 100 [m] the coordinates are 100* x (or y). Area covered by network is 19*(sqrt(27)/2)*(hexagon brink)^2 [m^2]. AP separation sqrt(3)* (hexagon brink). With 100 m hexagon brink the network area is about 0.5 [km^2] and the AP separation is about 170 [m].

[19] This will be decided on the based of simple link budget calculation when the pathloss for test environment C2 are fixed.

[20] This comes from simple link budget calculation for used pathloss model see appendix.

11.3 Radio Link and Radio Propagation

11.3.1 Assumptions

Table 11-4 summarizes the radio link and radio propagation assumptions.

Table 11-4 Radio link and radio propagation assumptions.

Parameter	Description/Value
Carrier frequency	5 GHz –"worst case" for the path loss.
Channel BW	102.4MHz = 2048 x 50kHz
Active channel BW	83.2 MHz=1664 x 50kHz
DL max Tx power [dBm]	43
Number of AP antennas	1
AP noise factor [dB] (UL Receiver implementation loss)	6
UL max Tx power [dBm]	23
Number of UT antennas	1
UT noise factor [dB] (DL Receiver implementation loss)	9
AP Antenna pattern	Omni
Max AP antenna gain [dBi]	10
UT Antenna pattern	Omni
Max UTantenna gain [dBi]	0
DL total noise [dBm]	-86 (calculated from efficient BW and 9 dB noise factor)
UL total noise [dBm]	-89 (calculated from efficient BW and 6 dB noise factor)

11.3.2 Models

At the first step simple propagation model is used [97] and SISO channel is modeled. More detailed description of the model can be found from document [97] and more advanced models are constructed during the WINNER project.

11.3.2.1 Path loss Model

The general formulations of empirical path loss models for outdoor environment can be given in the [dB]

$$PL(d,f) = PL_{FS,1}(f) + 10 * n * \log_{10}(d) + C \qquad (72)$$

where $PL_{FS,1}$ (f) = 20 $\log_{10}$ (4πf/c) is the free-space path-loss at 1m distance,

d is the distance between the transmitter and the receiver,

f is the center frequency of the signal,

c is the velocity of light in vacuum,

n is the path-loss exponent that depends on the environment and

C is a environment dependent constant.

This model can be used in the simplified form

$$PL(d) = A * \log_{10}(d) + B \text{, [dB]} \qquad (73)$$

where A = 10n

B = $PL_{FS,1}$ (f) + C

and the values used for calibration are

$$PL(d) = 28.3 * \log_{10}(d) + 53.5 \ \ [\text{dB}] \tag{74}$$

11.3.2.2 Shadow Fading Model

Log-normal distribution with standard deviation (STD) 4.0 [dB].

11.3.2.3 Fast Fading Model

Fast fading simplification is the assumption that channel is constant during the super-frame DL or UL transmission period. The urban Micro model is utilized and each of the taps are Rayleigh faded. The assumed mobility for fast fading model is 3km/h, even though the UT do not move.

<table>
<tr><th colspan="2">Model</th><th colspan="6">Urban Macro</th><th colspan="6">Urban Micro</th></tr>
<tr><td colspan="2">PDP</td><td colspan="6">Based on SCM SL simulation</td><td colspan="6">Based on SCM SL simulation</td></tr>
<tr><td colspan="2">Total Delay-Spread (µs)</td><td colspan="6">0.86</td><td colspan="6">0.30</td></tr>
<tr><td colspan="2"># of clusters a.k.a. paths</td><td colspan="6">6</td><td colspan="6">6</td></tr>
<tr><td colspan="2"># of mid-paths per cluster</td><td colspan="6">3</td><td colspan="6">4</td></tr>
<tr><td>Clusters:</td><td>Relative Path Power (dB) /
Delay (µs)</td><td colspan="3">0
-1.425
-4.217
-7.852
-12.037
-14.919</td><td colspan="3">0
0.467
1.127
1.981
3.031
4.908</td><td colspan="3">0
-0.783
-2.775
-4.605
-5.513
-7.658</td><td colspan="3">0
0.261
0.429
0.608
0.811
1.019</td></tr>
<tr><td rowspan="2">Mid-paths:</td><td>Relative Path Power /
Relative Path Power (dB) /
Delay (ns)</td><td colspan="2">10 / 20
6 / 20
4 / 20</td><td colspan="2">-3.010
-5.229
-6.990</td><td colspan="2">0
7
27</td><td colspan="2">7 / 20
5 / 20
4 / 20
4 / 20</td><td colspan="2">-4.559
-6.021
-6.990
-6.990</td><td colspan="2">0
5
11
28</td></tr>
<tr><td>Delay-Spread (ns)</td><td colspan="6">10.2</td><td colspan="6">10.0</td></tr>
</table>

11.3.2.4 Minimum Coupling Loss

Minimum coupling loss models the minimum attenuation between transmitter and receiver and assumed to be 70 dB.

11.4 Transmission Scheme and Multiple Access Method

11.4.1 Assumptions

Table 11-5 describes the assumption regarding the basic transmission scheme and multiple access method. Since the definition of these parameters is one of the major objectives of WP2, the parameters below are only coarse assumptions targeting nothing but the simulator calibration. OFDM modulation is used in both uplink and downlink, and the time structure comprises symbols, slots, frames and super-frames.

Table 11-5 Assumed transmission scheme, multiple access method and and duplex arrangement.

Parameter	Description/Value
Duplexing	TDD
Basic access DL	OFDM – this was the most common assumption in WP2

Basic access UL	OFDM – use the same as for DL for simplicity
Number of sub-carriers	2048 – based on selected freq BW, easy for FFT
Sub-carrier spacing [kHz]	50
Sub-carrier noise [dBm]	-127
Number of used sub-carriers	1664 –outermost sub-carriers are used as guard carriers
Multiple access DL intra-cell (multiplexing)	TDMA – easy to implement
Multiple access UL intra-cell	TDMA
Multiple access DL inter-cell	(Frequency reuse 1 is assumed) The LA should handle badly interfered UT (different coding rate, modulation etc.)
Multiple access UL inter-cell	-“-
Slot (DL and UL). i.e smallest time unit that can be allocated for traffick.	1 symbols (22.5 µs)
Superframe	
Total length (N_{tot})	61 slots (A ms)
DL length (N_{DL})	25 slots (B ms)
UL length (N_{UL})	36 slots (C ms)
Cell info/system info overhead	Modelled in broadcast
Other overheads (TX/RX switching etc.)	-
Number of sub-bands in frequency domain (L)	L=1

11.4.2 Models

11.4.2.1 OFDM Link Quality Models

The link to system interface to be used in WINNER is effective SINR mapping. Two types of effective SINR mapping has been proposed in WP2, namely EESM and MIESM, another one based on exponential mapping and another one based on the pre calculated mutual information function. MIESM was find to be more accuracy, however the parameters and link results are not available at the moment ithus t was chosen to use the link quality model based on the received SINR and an exponential mapping to an effective SINR ($SINR_{eff}$). [More parameters available]

The calibration case is restricted to SISO and an OFDM transmission scheme is used in both uplink and downlink. We may hence assume that the bits in a data block are transmitted over different OFDM symbols and different OFDM sub-carriers but that spatial multiplexing is not employed. Due to time variations and frequency selectivity, we have a multi-state channel. In the calibration case it is assumed that the link quality model is based on the received SINR (for a particular receiver design) and that the set of SINR:s (that make up the symbols in the data block) are mapped to an effective SINR by exponential averaging. The effective SINR is then used to determine the block error probability from a one-dimensional lock-up table, which is created from an AWGN simulation (for the given MCS).

For an OFDM link in between a transmitting node *i* and a receiving node *j*, the SINR on sub-carrier *k* (in a certain time instant) can be calculated as:

$$SINR(i,j,k) = \frac{p(i,j,k) * PL(i,j) * X(i,j) * \left|H(i,j,k)\right|^2}{N_0 + I(j,k)} \quad (75)$$

Where

- p(i,j,k) is the transmitter power (from transmitting node i intended for receiving node j on sub-carrier k)

- PL(i,j) is the (distance dependent) pathloss in between node i and j calculated according to equation ((75)

X(i,j) is the slow fading in between node i and j.

- H(i,j,k) is the frequency response of the channel (in between node i and node j on sub-carrier k), calculated as the Fourier transform of the channel tap model described in Section 11.3.2.3.

- N_0 is the noise, calculated as the thermal noise over a 50 kHz bandwidth (sub-carrier spacing) plus the receiver noise factor. That is, given a DL noise factor of 9 dB and an UL noise factor of 6 dB, N_0 equals –118 dBm and –121 dBm in downlink and uplink, respectively

- I(j,k), finally, is the interference experienced at the receiving node j at sub-carrier k (modelled according to Section 11.4.2.2)

The SINR calculation in equation (76) assumes that there is no inter-carrier or inter-symbol interference, i.e. it is assumed that the length of the cyclic prefix always exceeds the maximum time dispersion of the channel. It is further assumed that the channel coherence time is much larger than the sub-carrier separation (i.e. the channel is flat within the sub-carrier) and that the channel coherence time is much larger than the symbol duration (the channel is constant over an OFDM symbol).

Having calculated the SINR of all data symbols in the block, the set of SINR should be mapped to an effective SINR value.

$$SINR_{eff} = -\beta * \log_e \left(\frac{1}{N} \cdot \sum_{n=1}^{N} e^{-SINR_n / \beta} \right) \quad (76)$$

The sum is taken over all data symbols n (n=1,2,…N) in the data block, transmitted over different OFDM sub-carriers and OFDM symbols (i.e. the block has a certain extension both in frequency and time). The block is the re-transmission entity. In the calibration model it is assumed that a single modulation and coding scheme is used inside one re-transmission entity. The value of the adjustment factor β is summarized in Table 11-6 for each employed MCS.

Table 11-6 Adjustment factor (β) for the different employed MCS:s.

	Modulation type	Coding rate	β
MCS1	BPSK	½	0.9
MCS2	QPSK	½	1.8
MCS3	16QAM	½	5.9
MCS4	16QAM	5/6	8.4

Finally, having determined the effective SINR of the data block, the block error probability is estimated from the one-dimensional lock-up table created by AWGN simulations (for the used MCS). In the simulations, a random experiment based on the estimated block error probability may be used to determine whether the block could be decoded correctly or not.

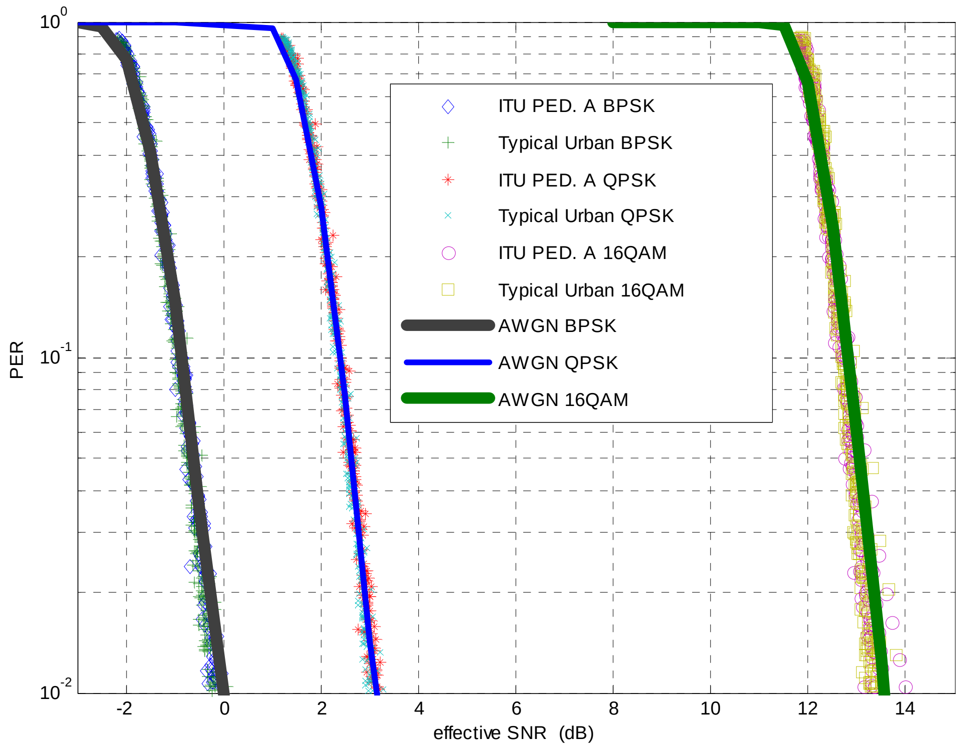


Figure 11-2 Effective SNR vs. PER using the exponential effective SINR mapping with β values from Table 11-6 and 100 channel realizations for both channel models More details see [102].

11.4.2.2 Interference Model

In order to calculate the SINR of each sub-carrier, the sub-carrier strength as well as the interference experienced on the sub-carrier must be known. In a multi-cell network, the interference originates from many different sources. This sub-chapter addresses the interference modelling.

For the calibration case, we propose to consider all interference sources in the interference calculation. Moreover, all interfering links are in general frequency selective, which will influence the interference that is experienced on the different sub-carriers.

The interference experienced by a receiving node j (that received information from a transmitting node i) on sub-carrier k is calculated according to equation (77) below.

$$I(j,k) = \sum_{l \neq i} p(l,k) \cdot PL(l,j) \cdot X(l,j) \cdot \left| H(l,j,k) \right|^2 \tag{77}$$

where

- p(l,k) is the power transmitted by node l on sb-carrier k (the sum is taken over all transmitting nodes except the node transmitting a signal intended for the receiving node j)

- PL(l,j) is the path loss between a transmitting node l and the receiving node j, calculated according to equation (77)

- X(l,j) is the slow fading in between node l and j.

- H(l,j,k) is the frequency response of the channel in between node l and j at sub-carrier k (calculated as the Fourier transform of the channel-tap model described in Section 11.3.2.3)

11.5 Radio Network Functionality

11.5.1 Assumptions

Table 11-7summarizes the assumed radio network functionalities.

Table 11-7 Assumed radio network functionality

Parameter	Description/ Value
Channel allocation/scheduling algorithm	Round robin between users, one user fulfil the super frame as much as possible
Adaptive coding and modulation	Yes, use limited set of predefined modulation and coding schemes. BPSK ½, QPSK ½, 16-QAM ½ and 16QAM 5/6
ARQ	Yes. HARQ type I
Maximum number of retransmission attempts (Nmax)	5
L2 ACK/NACK message	DL 1 symbols or 20 bytes if send with data UL 2 symbols or 20 bytes if send with data
ACK transmission period	4 super-frame ???
Radio over head for packets [21]	20 bytes ?????
Inter-cell synchronization[22]*	Yes
Inter-cell coordination (same super-frame structure)[23])	Yes
Power control DL	No –fixed Tx power is used
Power control UL	No –fixed Tx power is used
Handovers	Yes
Handover Margin	6 dB
Handover time interval	2 s
Cell selection	Based on path loss

Figure 11-3 shows the procedure of sending first packet when the decision of AP has already been made based on measurements made during earlier super-frames.

The procedure when call is arriving to Terminal might be different and will be studied in WINNER, but not in the scope of calibration case.

[21] This includes packet seq. num MAC address (if needed) etc. (Input needed from WP 3 and 2)

[22] Inter-cell synchronization means that adjacent cells use same time reference and that super-frames start at same time instant. Inter-cell coordination implies that the downlink and uplink transmission periods are the same in neighbouring cells, i.e., in such a network there is no interference in between terminals neither in between base stations (access points)

[23] see above

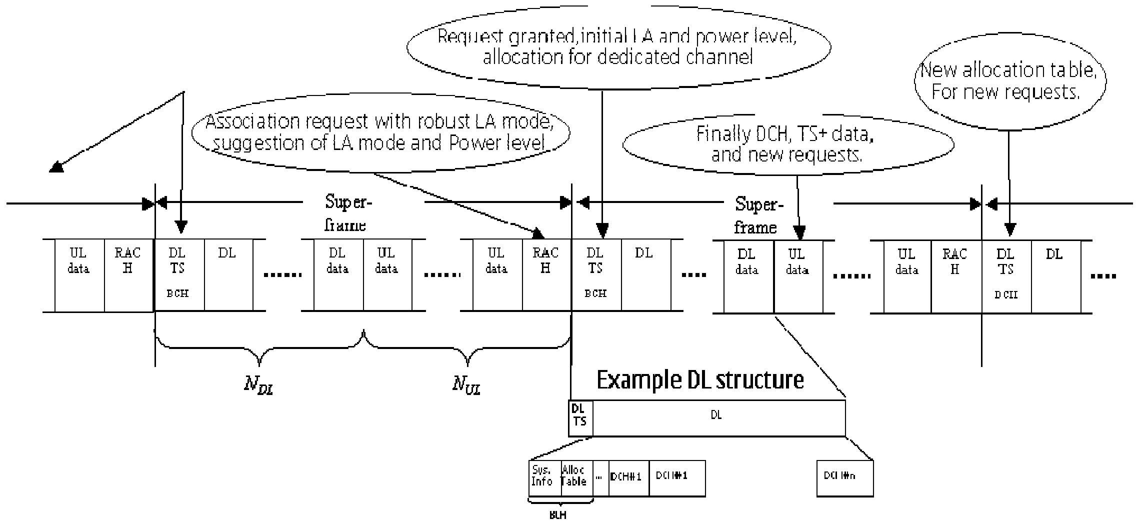


Figure 11-3 UL association measurements and procedure. This assumes that system info contains the Tx-power level information. One example DL structure, measurement can be done from BCH and info send to the Terminal in allocation table. Model simplification based for this example process is discussed in following sub-chapters.

11.5.2 Models

The basic radio network functionalities to be modelled in the system simulator include association to the network, scheduling, link adaptation and ARQ. Measurements of the radio channel are needed to utilise some of these functions.

11.5.2.1 Modelled Super frame structure

For simulation purposes some simplification is always necessary, one of them is to simplify the super frame structure, see Figure 11-3. For simulations the number of RACH can be fixed to two UL slot in each super frame, to model the overhead that associations and HO will create.

For the broadcast messages two DL slots are allocated in every super frame these includes TS, allocation table etc. (see Figure 11-3). The second one of these has to be received correctly to be able to receive/send anything during the super frame. Most robust modulation coding scheme is used in this slot. Initial LA is made based on the SINR measured during this slot. This is simplified model, and the assumption that slot two have to be received correctly in each super frame (to be able to receive/send) models the situation where new allocation table is send begin of each super frame.

Tx/Rx switching time is not yet taken into account, but when most dominant time overheads are known (UL timing advance etc) these are modelled by stretching the time slot duration, so that the whole super frame time duration correspond the “real” super frame time duration.

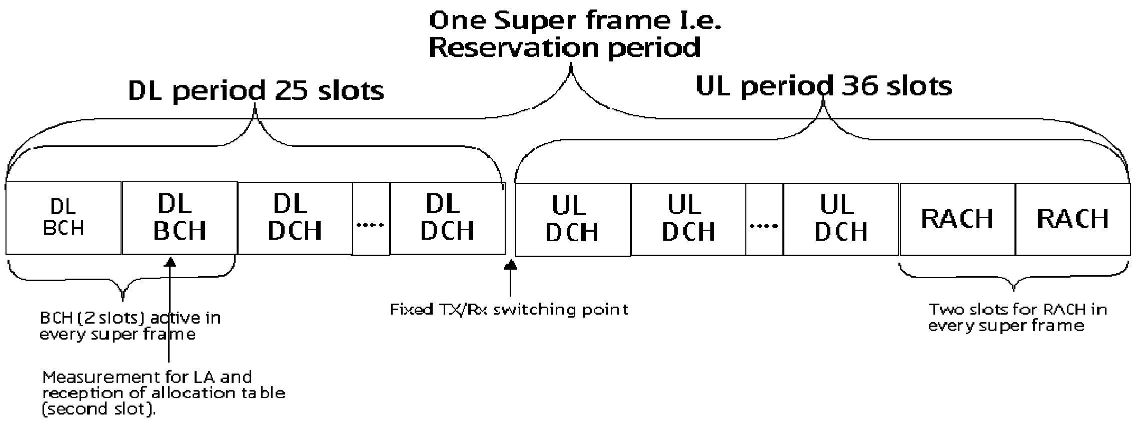


Figure 11-4 Simulator super frame structure. DCH can be re-placed with SHCH (shared channel)

11.5.2.2 Association and disassociation

Before Terminal can send first packet and/or start to download packets it has to measure the BCH channel received powers. Based on these measurements Terminal decides which AP it will associate. This association is based on received power levels (in calibration case), and Terminal will associate itself to the AP having strongest received power level. At the calibration case Terminal will always initiate the association process. Association request PDU is sent to the path loss-wise best source AP during

RACH slot reported in the broadcast information. In case of failed association random back-off (uniform 1-6 super-frame) is utilised to prevent simultaneous associations to block each other. Association is confirmed when following BCH message from the desired AP indicates of the UT admission, and informed to owner UT using association notification for appropriate UT state transition.

Disassociation routine is triggered only when new service arrives at idle UT, at the same time when

UT is given a new random position. Time limit is set for connection establishment from UT's new position (X super-frames or Y ms). Reaching the limit with unsuccessful association effort, UT is considered being out of coverage area and therefore it is re-positioned.

11.5.2.3 Modelled Packets

In the calibration case no packet segmentation is assumed. So that if the radio packet has bits to fully load one OFDM symbol and e.g.10 bit are left over then two slots are reserved to send this packet. Packet concatenation is applied, whenever within data flow the queue contains packets enough to fill the allocated DCH time, for the maximum utilization of channel capacity. In case of concatenation radio overheads are included only ones for the concatenated packet.

In DL only one training sequence (TS) is assumed, which is modelled in begin of the super frame. Thus the DL packet only contains the "data" (including radio overheads) from upper layer and is send during dedicated channel (DCH). The channel estimation based on TS is not modelled.

In UL in begin of each transmission one slot is reserved for TS. The reception of the TS is not modelled i.e. only the air interface overhead of TS is modelled.

11.5.2.4 Packet reception/transmission

For UL/DL packet the effective SINR is calculated from data symbols, i.e. TS is not included in the calculation. For DL and UL packet the receiving/transmitting terminal calculates the effective SINR to the second slot of the super-frame and the decision is this slot received correctly is made by comparing the effective SINR to the BLER look up table. In case this slot is received correctly reception/ transmission during super-frame is possible. Similar L2S interface is utilized for reception of all packet types (including TS).

For each received packet the effective SINR is calculated over the packet (whole IP packet) and the corresponding BER value is read from look-up table. PER value P is derived from BER value p according the following equation

$$P(p,l) = 1-(1-p)^{l} \tag{78}$$

The BER values include the FEC coding and interleaving effect since they are produced from one OFDM sized BLER values, see Figure 11-2. [If PER values for 1500 bytes packet are available this can be simplified, note that in Figure 11-2 the packet sizes differs due to different modulation and coding scheme, the time period is fixed]

11.5.2.5 Cell re-selection (Handover)

Handover results from UT originated cell re-selection. Only forward handovers are supported, meaning that no advance resource availability from the new cell is checked. The relevant data buffer content transfer between APs is triggered by notification from the selected AP to the old AP, guaranteeing lossless handover. Thus in simulator the data is immediately available in new AP and data transfer from AP to AP is not modelled. UT makes the HO measurements at fixed time intervals (e.g. 2 s). Received powers from different neighbour APs are stored. If the signal from stranger AP becomes adequately stronger according to a fixed HO margin, and from own AP weaker than a predefined minimum threshold, new association procedure is launched. During the Super frames when HO measurements are made no data can be received or send by this UT.

11.5.2.6 Resource request

For DL AP knows the wanted resources, for UL UT have to inform about the queued data. UT does that via resource request packets. This can be included in the UL burst (10 bytes overhead) or send as such (to get the first allocation). Regular resource requesting possibility is guaranteed for UTs by the scheduling algorithm.

11.5.2.7 Scheduling and allocation

This chapter describes two things the scheduling of resources between users, allocation of resources inside Super-frame. The allocation period is equal to one Super-frame. Round Robin method is the basic idea for scheduling among UTs data flows.

For signalling type of allocations AP maintains scheduling groups. Associated UTs are uniformly distributed and kept in these groups according the arriving, leaving of UTs. Scheduling groups correspond to a super frame interval whose length, in super frames, is equal to the number of groups. Each group corresponds to a certain super frame within the interval. Thus in each super frame a subset of associated UTs shares signalling turns. The maximum size of group is parameterized to be 4. The number of groups is determined by the rule to satisfy the maximum group size.

At the beginning of each allocation period and before going to actual user data queues, AP first selects the UTs in turn according to scheduling groups. These UTs are given DCH allocation of the length required for the transmission of a resource request packet, i.e. a symbol for training sequence and a symbol for request message. Hence regular resource requesting possibility is guaranteed for UTs.

As regards the scheduling of actual user data, AP chooses the next active UT following the previously served UT in its UT table, and reserves the required DL and UL DCH slots according to the DL and UL data buffer information of the chosen UT. For potentially remaining DCH slots of the prevailing super frame, AP allocates correspondingly for the following UTs as not to leave channel capacity unutilized. Thus the scheduling method is to make the most of the capacity, as fitting more users to DCH frame causes more overhead in terms of more time slots short of data and more UL training symbols as non-data carrying time slots.

Information about queued UL data is obtained through buffer status reports, or resource requests, transmitted by active UTs. At the moment the scheduling algorithm lacks QoS support. The allocation information is sent on BCH in the beginning of each super frame.

11.5.2.8 Power Control Model

Fixed transmit power is used in both uplink and downlink.

11.5.2.9 Adaptive Modulation and Coding Model

As a basic MCS for BCH and RACH transmissions, BPSK is BPSK ½. Four modulation and coding schemes are used in the DCH, as defined in Table 11-8

Table 11-8 Employed set of modulation and coding schemes

MCS	Modulation type	Coding rate
MCS1	BPSK	½
MCS2	QPSK	½
MCS3	16QAM	½
MCS4	16QAM	5/6

The QPSK ½ is selected as a initial MCS for all connections. Moreover, the link quality estimation is performed regularly. Link adaptation is similar on downlink and uplink. Faster MCS is requested if certain number of DCH packets are in sequence received correctly. Slower MCS is requested if also certain number of DCH packets is in sequence received erroneously. This provides though that transmissions are captured at receiver. Slower MCS is taken into use also if within fixed super frame interval any packet under transmission meets half of the maximum allowed retransmissions or the number of unacknowledged packets reaches a fixed upper bound.

11.5.2.10 ARQ Model

The network uses a HARQ type I ‘selective repeat’ retransmission protocol. The L2 signalling (ACK:s) is modelled throughout the simulations.

The ARQ modelling is performed as follows:

1) A packet is transmitted using one of the specified modulation and coding schemes.
2) At the receiver, the packet error rate (PER) is estimated according to the signal quality and the used modulation and coding scheme.
3) A random experiment, based on the PER, determines whether a packet error occurred or not.
4) When the UT/AP acknowledging time interval is reached the receiver generates a 10 byte cumulative ACK and needed selective ACK.
5) When receiving an ACK message, the sender removes the packet from the data buffer. If error has occurred the transmitter retransmits the data packet. If the packet was not delivered correctly after Nmax = 3 retransmission attempts, the lower MCS is selected (if the lowest MCS is used, the call is dropped).

11.5.2.11 Dropping Model

Also disassociation routine is triggered only when new service arrives at idle UT, at the same time when

UT is given a new random position, and this takes UT to PENDING state. Time limit is set for connection establishment from UT’s new position. Reaching the limit with unsuccessful association effort, UT is considered being out of coverage area and therefore it is re-positioned. If a packet is received erroneously Nmax times with the lowest MCS the call is dropped.

11.6 Calibration assessment criteria

For the calibration purposes following measures are decided to be under calibration:

- Packet RX power distributions (send/received by any AP). This is assumed to calibrate the propagation and user distributions mechanism.
- Packet C/I distribution (send/received by any AP), This is assumed to calibrate the interference calculation in the network and link quality model implementation.
- Packet delay distribution (send/received by any AP). This is assumed to calibrate the scheduling and allocation mechanism as well as overheads modelled in the simulators.
- Packet error rate PER (send/received by any AP). Calibrate LA.
- Fraction of blocked user (from any AP). Calibrate the ARQ and LA (the out-of-coverage figure).
- UT throughput (received by UT associated to the network). This is assumed to include most of the simulation features and take into account more visibly the traffic model and LA.
- AP DL/UL throughput (send/received by any AP), one of the assessment criteria which should take into account all simulation features.
- Dropping rate

11.6.1 Collected statistics

All of these statistics calibrate different models, but some of these are more essential. E.g. the throughput is one of the most essential measures, but if some differences exists in the throughput curves the reason is easier to identify through the calibration of middle step measures like packet SINR. It will be seen which ones are easiest to calibrate in practice and these are the starting points.

11.6.1.1 Packet/Block Rx-power level UL/DL

The Rx-power of packet received is defined to be the average (over all sub-carriers and block data OFDM symbols) received power. Since the channel is not assumed to change during the super-frame (calibration case assumption) the received power level do not change during the transmission of one packet.

11.6.1.2 Packet SINR value

The effective SINR value is calculated as described in chapter 11.4.2 and stored from all packets.

11.6.1.3 Packet Delay

The packet delay is measured for data packet, this is the delay from sender IP layer to receiver IP layer excluding all TCP/IP buffering.

11.6.1.4 Packet error rate

PER is collected from data packets.

Alt 1:

PER is increasing whenever packet is not received correctly. PER is collected from all packets.

Alt 2:

PER is collected for each user and cumulative distributions of connection PER are plotted.

11.6.1.5 Dropping rate

The number of dropped connections are compared to the total number of connections during the simulation.

11.6.1.6 UL/DL throughput

The throughput of each connection is measured at the application layer. Connection will start when the UT is created and end when the last packet is received. The throughput statistics is collected only from correctly ended services. The throughput is the correctly received connection data bit/connection duration. The cumulative distribution of the connection throughputs is plotted.

11.7 Definition of Time and Frequency Structure

A proposal for a time and frequency structure of the WINNER radio interface is one expected output from WP2.Various analysis, e.g. by means of computer simulations, is one contribution to reach such a results. Accordingly, simulations and simulator calibration must be based on some coarse initial assumption regarding the upcoming WINNER radio interface.

The proposed calibration case uses OFDM modulation in both uplink and downlink and TDD is used to separate the uplink and downlink traffic. In general, the smallest transmission entity in an OFDM system is a single symbol on a single sub-carrier. To form resource units of practicable size, however, several sub-carriers in frequency and symbols in time may be combined into blocks, see Figure 6-2

A block is assumed to contain M symbols in time and K sub-carriers in frequency (the bandwidth is divided into L sub-bands, each comprising K sub-carriers).

For calibration purposes, pure TDMA operation is proposed which implies that one user will receive all the sub-carriers, no multiple access in the frequency domain, Furthermore, in the calibration simulations the proposal is to have only one symbol per block in time domain (M = 1). However how many fast fading samples system simulator needs in frequency selective wide band 102.4 MHz is a question of initial LL/SL interface L.

We denote the time extension of one (M=1 block) as slot i.e smallest time unit that can be allocated to one user. Furthermore, a super-frame is defined as N_{tot} consecutive slots, out of which N_{DL} are assigned for downlink transmissions N_{UL} are assigned for uplink transmissions ($N_{tot} = N_{DL} + N_{UL}$).

Such a super-frame represents the reservation period in the TDMA/TDD system that is proposed for calibration purposes. A schematic picture of the super-frame structure is depicted in Figure 11-4. The picture assumes, for simplicity, pure TDMA operation. Signalling information indicating the destination of the downlink data allocations must be included in the downlink transmission. Likewise, some downlink pilot symbols for channel estimation are required. In Figure 11-4, one possible solution is depicted, in which the pilots and the signalling information is contained in the first downlink slot of downlink reservation period, and the whole DL frame (i.e. the time from transmission power ramp up to transmission power ramp down) might contain several allocations to different users. For uplink, pilot symbols must be included in all frames (i.e. time from transmission power ramp up to transmission power ramp down).

In uplink, moreover, some slots may be assigned for conflict-free transmission (centralized transmission scheduling), while other slots may be used for contention based data transmission (non-centralized transmission scheduling) or random access (RACH). This division and the information regarding transmission scheduling of the conflict-free uplink slots must also be broadcasted. It may further be needed to include a guard period in between the downlink and uplink transmission periods. Also the need for small guard band between UL packets might be needed.

12. Annex V: Acronyms

ACK	ACKnowledgment
ACS	Access Control Server
AE_{WX}	WINNER Access Equipment for Mode X
AMC	Adaptive Modulation and Coding
AN	Ambient Networks
AP	Access Point
AQM	Active Queue Management
AR	Access Router
ARQ	Automatic Repeat reQuest
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BLER	Block Error Rate
BS	Base Station
BTS	Base Transceiver Station
CCK	Complementary Code Keying
CDMA	Code-Division Multiple Access
CFP	Contention Free Period
CSMA/CA	Carrier-Sense Multiple Access/Collision Avoidance
CS_W	WINNER Connection Service
CQI	Channel Quality Indicator
DCF	Distributed Coordination Function
DIFS	Distributed Inter-Frame Space
DL	Downlink
DSSS	Direct Sequence Spread Spectrum
EDCA	Enhanced Distributed Channel Access
FBRN	Fixed Bridging Relay Node
FCS	Fast Cell Selection
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
FHSS	Frequency Hopping Spread Spectrum
FoM	Figures of Merit
FPLRN	Fixed Physical Layer Relay Node
FRRN	Fixed Routing Relay Node
FRN	Fixed Relay Node
GLL	Generic Link Layer
GPSR	Greedy Perimeter Stateless Routing
HARQ	Hybrid Automatic Repeat reQuest
HCF	Hybrid Coordination Function
HERN	HEterogeneous Relay Node
HORN	HOmogeneous Relay Node
HSDPA	High Speed Downlink Packet Access

HS-DSCH	High Speed Downlink Shared Channel
HS-PDSCH	High Speed Physical Downlink Shared Channel
IBSS	Independent Basic Service Set
IFFT	Inverse FFT
IP	Internet Protocol
L2T	Layer 2 Tunnel
LOS	Line of Sight
LUT	Look-Up Table
MAC	Medium Access Control
MAC-g	Generic Medium Access Control
MAC-r	Mode-Specific Medium Access Control
MBRN	Mobile Bridging Relay Node
MCM	Modes Convergence Manager
MIMO	Multiple-Input Multiple-Output
MPLRN	Mobile Physical Layer Relay Node
MRA	Multi-Radio Access
MRRN	Mobile Routing Relay Node
NACK	Negative ACKnowledgement
NLOS	None LOS
ODMA	Opportunity Driven Multiple Access
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PCF	Point Coordination Function
PDPC	Packet Discard Prevention Counter
PDU	Protocol Data Unit
PER	Packet Error Rate
PHY	Physical Layer
PS	Packet Switch
QAM	Quadrature Amplitude Modulation
QoS	Quality-of-Service
QPSK	Quadrature Phase Shif Keying
RAB	Radio Access Bearer
RAN	Radio Access Network
RANG	Radio Access Network Gateway
RAT	Radio Access Technology
RED	Random Early Detection
RLC	Radio Link Control
RN	Relay Node
RNC	Radio Network Controller
RRC	Radio Resource Control
RRC-g	Generic Radio Resource Control
RRC-r	Mode-Specific Radio Resource Control
RRM	Radio Resource Management
RS	Resource Controller

RTT	Round Trip Time
SAP	Service Access Point
SDM	Space Division Multiplexing
SDMA	Space Division Multiple Access
SDU	Service Data Unit
SIFS	Short Inter-Frame Space
SIR	Signal-to-Interference Ratio
SLC	Service Level Controller
SINR	Signal-to-Interference Noise Ratio
SNR	Signal-to-Noise Ratio
SS	Secondary Station
STTD	Space-Time Transmit Diversity
TDMA	Time-Division Multiple Access
TNL	Transport Network Layer
TTI	Transmission Time Internal
UL	Uplink
UMTS	Universal Mobile Telecommunications System
UT	User Terminal
UTRA	Universal Terrestrial Radio Access
WCDMA	Wideband Code Division Multiple Access
WiFR	Wireless Fixed Relays Routing
WINNER	WWI New Radio IP
WLAN	Wireless Local Area Network
WSE	Weighted Spectral Efficiency
WWI	Wireless World Initiative

13. Annex VI: References

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