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Convergence Techniques in a Multi-Standard Mobile Communication Environment

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Convergence Techniques in a Multi-Standard Mobile Communication Environment

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Contents

Chapter 1

Introduction

Mobile communication has changed peoples's life in many ways during the last decades. It is foreseen that there will be over billions of individual wireless subscribers worldwide in the coming years. The currently existing second-generation (2G) cellular standards Global System for Mobile Communications (GSM)/General Packet Radio Service (GPRS) subscribers will reach the number over a billion while Universal Mobile Telecommunications System (UMTS), High Speed Downlink Packet Access (HSDPA) and Enhanced Data rates for GSM Evolution (EDGE) systems each will collect hundreds of million of subscribers. These third-generation (3G) cellular standards have been implemented, viable high-speed handsets are being shipped in quantities and users are signing up for high-speed data services. Furthermore, plenty amount of local high-speed Wireless Local Area Networks (WLAN) are offered within public places. In Europe the similar product and mostly discussed standard is High performance Local Area Network seconde generation (HiperLAN/2).

Different kind of services are offered to the subscribers, including not only typical voice services but also the other data services with high and time variant Quality of Service (QoS) requirements. Especially the 3G cellular standards are trying to obtain subscriber by offering new services such as the video streaming and webbrowsing.

GSM can not support Voice-over-IP (VoIP) service due to its limited transmission data rate, while HiperLAN/2 can support the video streaming service with a satisfying quality to the subscribers, such as the Moving Picture Experts Group (MPEG) stream. Table 1.1 makes a list for some of the typical services inside the current mobile communication market and shows which standard(s) can support these services with satisfying quality.

Service	Traffic Model	GSM/GPRS	UMTS	WLAN
Voice call	Classical ON-OFF	Yes	Yes	Yes
Video telephony	VoIP	No	Yes	Yes
Video telephony	H.263 encoding	No	Yes	Yes
Video on demand	MPEG4	No	No	Yes
Webbrowsing	WWW 64Kbps encoding	Yes	Yes	Yes
Webbrowsing	WWW 2Mbps encoding	No	No	Yes
Messaging	Email	Yes	Yes	Yes

Table 1.1: Different wireless communication standards support different services

Although different standards are developed separately, the overlapping between the coverage area from different standards provides the subscribers a flexible choice. Emerging handset initiatives will play a pivotal role in those areas where more than one standards are available.

Multi-standard transceivers have already received a considerable amount of attention in the mobile communications community, especially for the support of different cellular standards. However most of these activities have focused on support for mobile roaming and are not directly for simultaneous operation.

The research work discussed in this thesis has been carried out in the frame work of the research for future mobile communication systems. One consideration of the fourth-generation (4G) mobile communication system is trying to enable the convergence between different standards. Convergence means a flexible and simultaneous use of multiple standards, aiming to enable an enhancement of services for the benefit of related actors, such as end users, network operators and service providers. Flexibility for end users and network operators, as well as efficiency of radio resource allocation are considered to be the benefits of standards convergence.

Convergence between standards indicates a parallel or alternate use of equipment and radio networks adhering to two or more physical layer transmission standards for the transmission of information related to the same communication. Different aspects of convergence are enabled either at radio link level or at network level, and therefore achieve different levels of benefits. Convergence management is then of great importance.

In this thesis, a concept of *Convergence Manager* is developed aiming to manage the convergence between multiple standards for the benefits of related actors, not only end users but also network operators and service providers.

Chapter 3 starts with the function of convergence manager and is followed by the brief description of important considerations for a convergence manager: scenarios, convergence algorithms and locations. A simulation scenario is defined based on the co-location of UMTS and HiperLAN/2, which will be used throughout this thesis.

Chapter 4 discusses three scenarios for the convergence manger: services scenario, standard scenario and user scenario. These three scenarios are considered as the baseline to define the simulation scenarios for the evaluation of convergence benefits. The service scenario focuses on the QoS requirements from different service classes. Example traffic models from different services are described. The standard scenario focuses on the time-variant radio interfaces. The link adaptation techniques are to be discussed for different standards.

The algorithm in convergence manager consists of two parts: network selection algorithm and adaptive scheduling algorithm. Different adaptive scheduling algorithms within each single network are discussed in Chapter 5 with various aspects including fairness oriented, channel oriented, queue oriented and cross layer adaptive scheduling algorithms. Performance improvements are observed from the dynamic mapping service quality requirements and instantaneous link channel quality. Chapter 6 describes the different selection algorithms for the multi-standard networks: pre-selection, single network selection and parallel network selection algorithms.A performance comparison between network selection algorithms is carried out within a single user scenario.

The location of convergence manager is a crucial issue for the architectural design of an environment, providing simultaneous usage of standards. It impacts the user's experience of service rendering and the service provider's ability to combine services in order to increase the value for the customer. Based on the convergence's granularity, the end user may also be involved in the network choice decision.

One possibility discussed in Chapter 7 of this thesis is to locate convergence manager only in the Multi-mode Terminal (@MT). There will be no change on the network side. Only network selection algorithms could be implemented by the convergence manger for this consideration.

Different applications to locate convergence manager @MT are described, as well as analytical performances of convergence manager with different network selection algorithms in the defined simulation scenario.

Another investigated convergence manager location is on the network side (@NET). Chapter 8 discusses mainly about the consideration to put convergence manager at the BaseStations (@BS). Combinations of adaptive scheduling and network selections algorithms can be used by convergence manager according to different scenario parameters. Significant system performance improvement of convergence manager is illustrated by the simulation results.

Chapter 9 compares the different location of convergence manager and evaluates each with advantages and drawbacks.

This thesis is concluded by a summary of the main results in Chapter 10.

Chapter 2

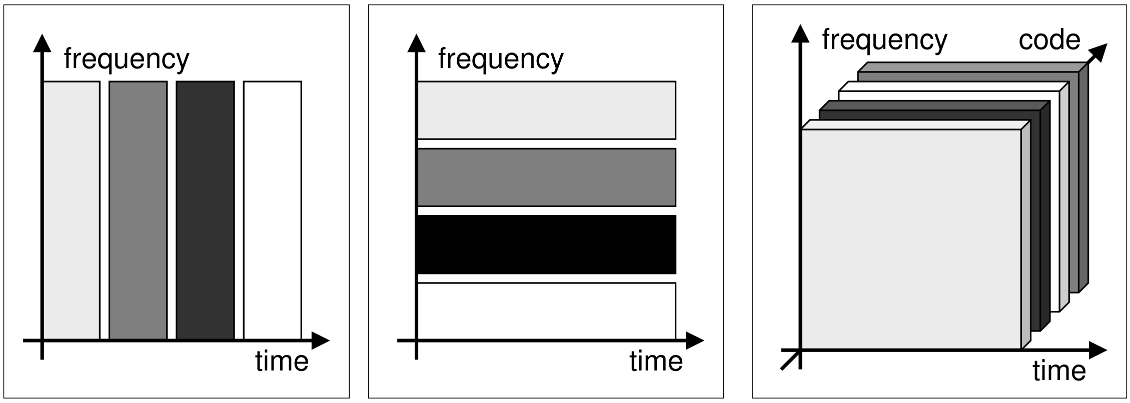
Multi-Standard Mobile Communication Environment

Europe has an acknowledged lead position in the development of different wireless communication and broadcast systems based on a range of well-established standards: 2G digital cellular networks based on GSM and 3G cellular networks based on new radio access technologies. There are multiple recognized 3G standards, mainly two 3G evolution paths: UMTS and Code Division Multiple Access system 2000 (CDMA2000). In addition to the communication systems based on cellular networks, WLAN or HiperLAN/2 products offer also high speed wide-band service for users.

2.1 Multiple Access Techniques

The characteristic functions of a mobile radio system are found especially in the data link layer and in the network layer, where it is defined how multiple users can communicate over the same channel simultaneously. This is controlled by multiple access methods. With the help of multiple access methods it is determined how a physical channel is divided into subchannels for individual users.

Resources of the medium can be split into several dimensions (e.g. time, frequency, code, space). Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA) and CDMA are three known techniques. They are shown in Figure 2.1.



(a) Time Division Multiple Access (**TDMA**)
(b) Frequency Division Multiple Access (**FDMA**)
(c) Code Division Multiple Access (**CDMA**)

Figure 2.1: Multiple access techniques: TDMA, FDMA and CDMA

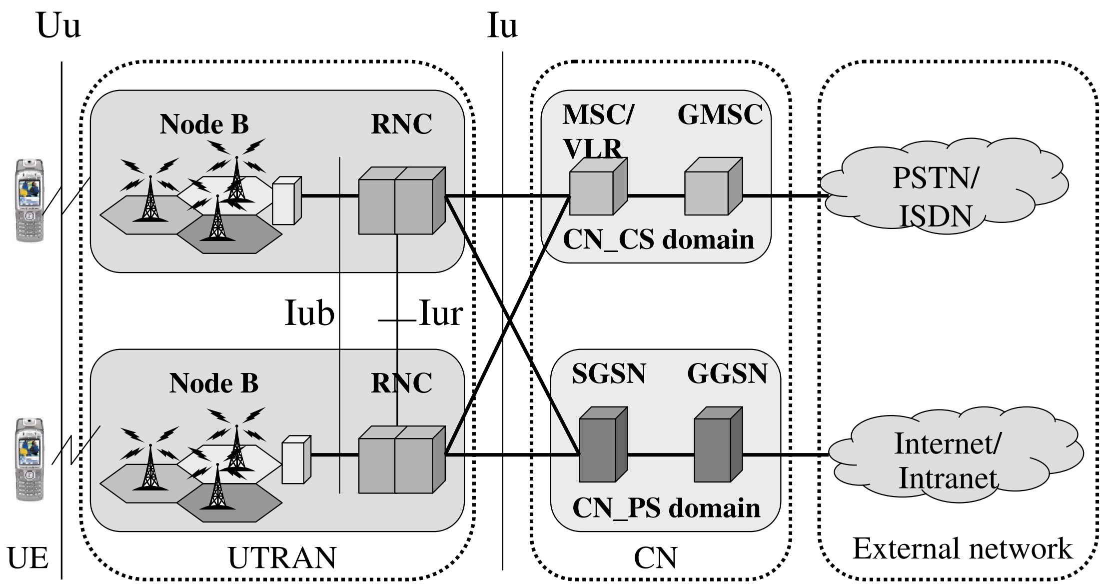
GSM is using TDMA, that means each user can allocate the whole bandwidth but only for a small time slot which limits the transmission data rate. GSM offers a data transmission service but the rate is limited to a maximum of 9.6 Kbit/s (Kbps), which matches perfectly to voice communication.

GPRS system brings the packet-switched bearer services to the existing GSM system. The maximum throughput is then raised up to 160 Kbps. GPRS brings a few new network elements to the GSM network, namely Serving GPRS Support Node (SGSN) and the Gateway GPRS Support Node (GGSN) [56].

EDGE [55] is a major enhancement to the GSM data rates. EDGE enhances the throughput per timeslot for GPRS by introducing the Phase Shift Keying (8-PSK) modulation in addition to the existing Gaussian minimum Shift Keying (GMSK) which is used by GSM and GPRS.

The data handling capability of 2G systems is limited to mainly voice services.

UMTS [60] can support high bit rate transmission which enables high quality multimedia communications with various QoS requirements by applying different radio access techniques. UMTS is using CDMA, which aims for a higher network utilization, supporting up to 2 Mbit/s (Mbps) data rate. In the current years, HSDPA has been designed as enhancement to UMTS and the packet data rate is increased up to 10 Mbps by means of link adaptation.



UE	User Equipment	RNC	Radio Network Controller
UTRAN	UMTS Terrestrial RAN	MSC	Mobile Service Switching Centre
SGSN	Serving GPRS Support Node	VLR	Visitor Location Register
GGSN	Gateway GPRS Support Node	GMSC	Gateway MSC
CN_CS	Core Network Circuit Switch	CN_PS	Core Network Packet Switch
Uu	Radio interface UE<> Node B	Iu	UTRAN<> CN interface
Iub	Node B<>RNC interface	Iur	RNC<>RNC

Figure 2.2: Network structure of UMTS

2.2 Cellular Network Standards: GSM and UMTS

A cellular network, e.g. UMTS network [32], consists of three interacting domains: Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). Figure 2.2 illustrates how an UMTS network is normally built.

The GSM network can be described with a similar structure, including CN, Radio Access Network (RAN) and UE.

The main function of the CN is to provide switching, routing and transit services for user traffic. The CN also contains the databases and network management functions. The basic CN architecture for UMTS is based on GSM CN plus packet switching domain while GSM networks are based on circuit switching.

The RAN provides the air interface access method for UE. GSM and UMTS are providing different service capacity based on different radio access techniques.

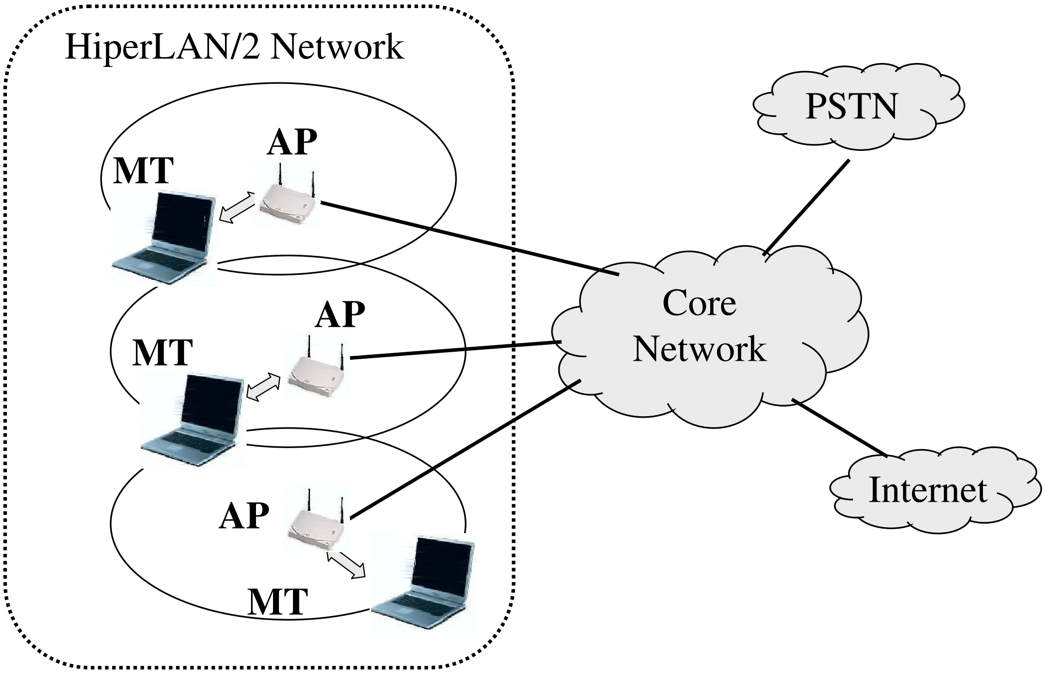


Figure 2.3: HiperLAN/2 network architecture

2.3 WLAN Standard: HiperLAN/2

WLAN [59] is a shared-medium communications network that broadcasts information over wireless links for all stations to receive. WLAN is widely considered to play a major role in wireless multimedia communications that require high speed transmission.

HiperLAN [28] is a standard that has been developed for what was conceived to be the next generation WLAN. The HiperLAN/2 operates in the 5 GHz frequency band with 100 MHz spectrum. A typical topology of a HiperLAN/2 network is depicted in Figure 2.3. The Mobile Terminals (MTs) communicate with one Access Point (AP) at a time over an air interface; while along with the movement, HiperLAN/2 automatically performs handover to the nearest AP. Ad hoc networks, where the MTs communicate directly, can also be created, but their development is still in early phase.

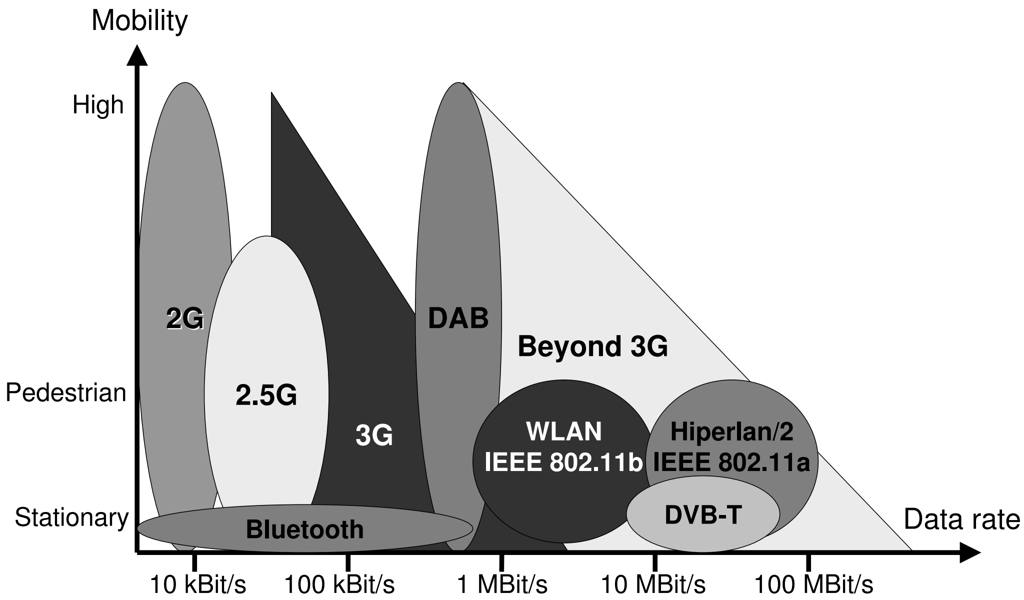


Figure 2.4: Mobility vs data rate for different cellular and WLAN systems

2.4 Convergence between Multiple Standards

Figure 2.4 shows the positioning of various mobile cellular and WLAN standards as a function of mobility and data rate. A similar figure can be drawn if mobility is replaced by cell size. Generally the performance of these standards shows that mobility or cell size is inversely proportional to data rate. The ongoing development in mobile systems are aiming to push the boundary toward upwards and right side of the figure.

It is unlikely that any one system will be able to provide support at both the limits of data rate and mobility together. Therefore if the user requires support for high mobility and high data rates at the same time, the user must have access to more than one wireless standard. In the current mobile communication market, there are various mobile cellular and WLAN standards available. An opportunity then appears that a user with a single terminal could be operating an application or several interlinked applications by making use of more than one standard [15].

2.4.1 Multi-Technology Access Network

In a possible vision of the 4G mobile communication system [39], all these wireless standards constitute possible access interfaces to a common core IP based network. Services are not cou-

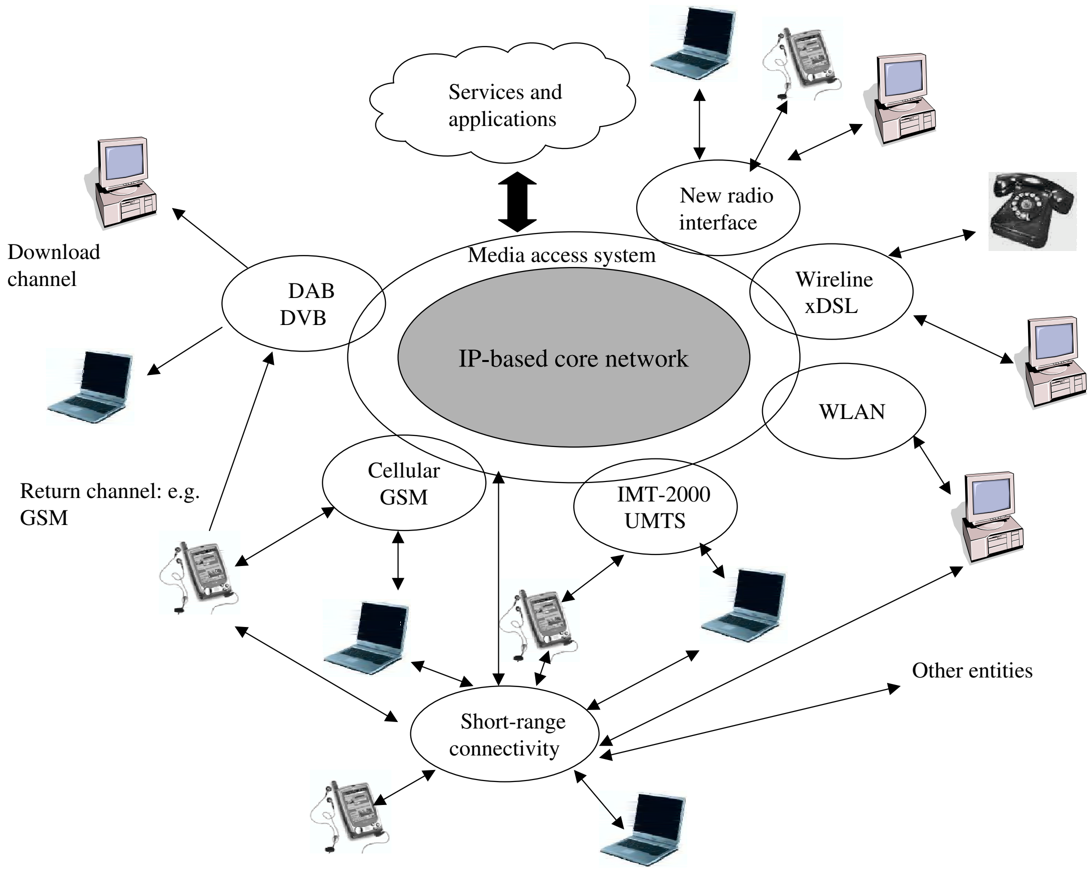


Figure 2.5: The multi-technology access network [39]

pled with any particular air interface. Services could be added onto the core IP network that would be available to all mobile users, regardless of the network they are using to access the core, as illustrated in Figure 2.5.

Through evolution of technology, new services have been or will be offered to subscribers, Data services have really taken off during the last couple of years with Short Message Service (SMS) and Multi-Media Message Service (MMS). GPRS packet data services are being launched, providing higher data rates and always-on capability. UMTS and its evolutions will provide even higher data rates, and a more comfortable offering of more demanding services. The evolution from 2G mobile communication system towards 3G mobile communication system will also lead to more convergence through a reduction of number of main 3G cellular technologies.

Of course there are technical challenges given the conflicting nature of these technical requirements if the user wants to use the standards simultaneously. But it still makes sense because that the combination of wireless resources and the sharing of services between various standards may offer great benefits to both the user and the operator.

2.4.2 Co-Location of Multiple Standards

Co-location of multiple standards happens when the coverage areas of different standards overlap. An example of co-location of multi-standards and the cell geography is illustrated in Figure 2.6.

In this kind of scenario, it can be observed that some end users have the possibility to use more than one standard if they are inside the multi-standards environment while other users can only use single standard due to their position. It is assumed in this thesis that all the end user equipments are able to access different mobile communication standards. When an end user could access to multiple standards, e.g. when coverage areas of the different networks overlap, there is no need to restrict oneself to a single interface [17]. Their usage can be combined in order to obtain the 'best' possible connection, according to a certain criteria. The convergence of multiple standard permits bandwidth aggregation, thereby allowing support for demanding applications that need high bandwidths. Furthermore, this can have additional advantages for some applications in terms of increasing reliability, where some or all packets can be duplicated and sent on multiple interfaces. Also, it can help in mobility management, where the delay associated with handovers can be significantly reduced when an alternate communication path exists.

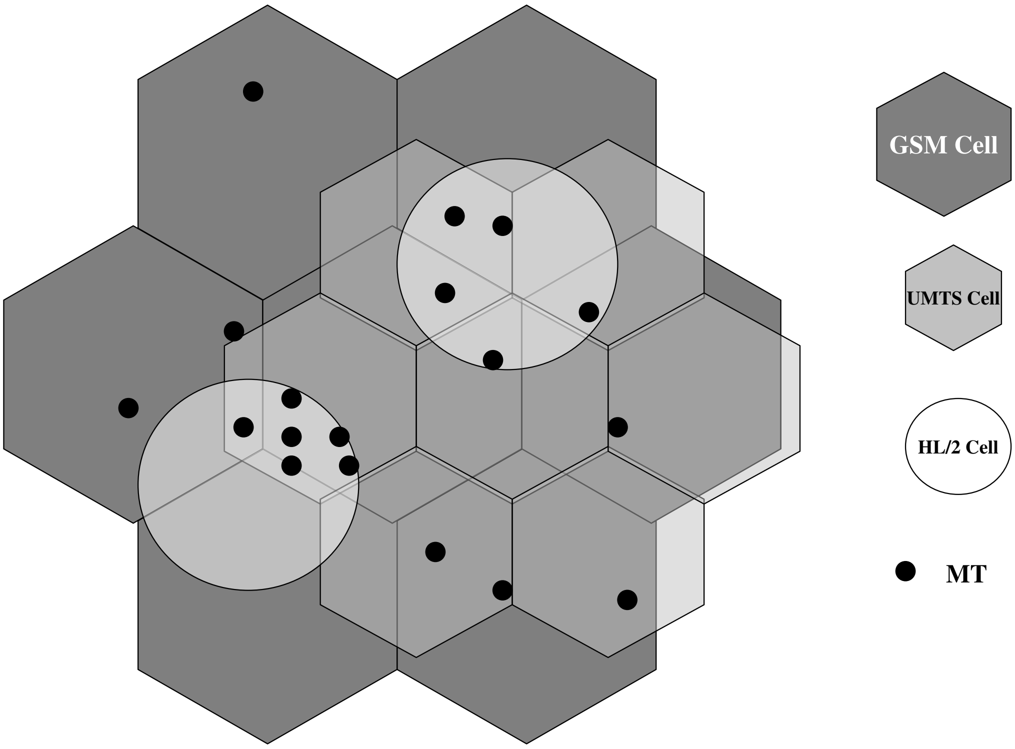


Figure 2.6: Co-location of GSM, UMTS and HiperLAN/2

2.4.3 Traffics of Different Services

Next Generation (NG) systems will provide a broad set of services and applications with different characteristics and target users. Data transfer, video-telephony, and multiple applications for E-commerce are foreseen, among many others, for deployment within NG systems [47].

In multimedia services, different types of applications have different characteristics and performance requirements. The large set of possible applications has been grouped into four main categories of service classes, according to 3rd Generation Partner Projects (3GPP) [3]. The four service classes are:

- Conversational class
- Streaming class
- Interactive class
- Background class

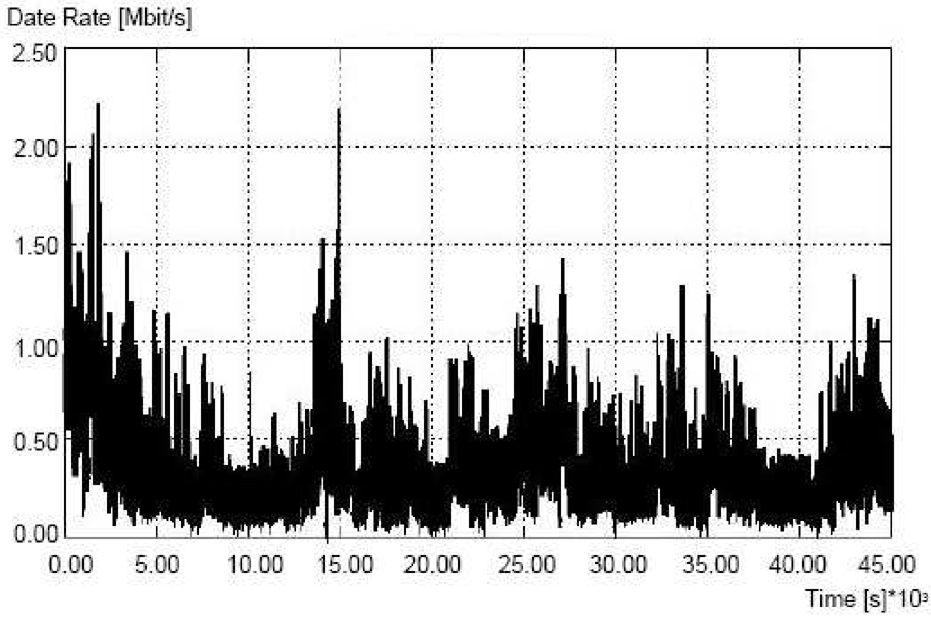


Figure 2.7: An example of a video data source with high and variable data rate

Video Streaming Model

An example of video signals with high and variable data rates is illustrated in Figure 2.7.

The traffics of a video streaming model have completely different characteristics comparing to that from a normal GSM speech call.The video streaming shown has an average data rate of 880 Kbps but different instantaneous data rates between 100 Kbps and 3 Mbps. Video streaming traffics cannot be supported by GSM due to its limited data rate while a voice call can be transmitted through both GSM and UMTS networks. It is even possible to split the traffics of a video streaming model and transmit them through both UMTS and WLAN network inside their coverage overlapping area. It is a big challenge to manage the different traffics with variable QoS requirements in an environment with multiple standards.

2.4.4 Simultaneous Use of Multiple Standards

The understanding of simultaneous use of standards is dependent upon the point of abstraction - what appears simultaneous to a user may be simple switching for the radio transceiver, whereas a transceiver simultaneously using two standards in soft handover will only appear to be using one standard to the user [53].

But not every case of simultaneous use of multiple standards benefits from the convergence between different standards. This thesis focuses only on those cases of simultaneous usage which enable and benefit from the convergence of multiple standards [11]. For example, talking on a telephone while transferring a file via WLAN is not considered in this thesis, since there is no convergence involved. This also rules out the case of using multiple instances of a single standard, e.g. two GSM channels, for increased data rate, since the convergence aspects are considered trivial.

Convergence between multiple standards means parallel or alternate use of equipment and radio networks adhering to two or more physical layer transmission standards for the transmission of information related to the same communication.

2.4.5 Convergence Technique

The 3GPP has performed an UMTS-WLAN convergence feasibility study in [5], [6] and [46]. The document identifies and describes six level of interworking, from the simplest common billing scenario with no requirement affecting 3GPP specification to service continuity or seamless service continuity requiring modification of 3GPP specifications. In most of the scenarios, the interworking requires in 3GPP specification . As the level of interworking increases, so does the amount of specifications, in the same time, more flexibility and benefits can be achieved from the convergence between UMTS and WLAN. According to this, the 3GPP provides a concept for connecting WLAN access networks to a GPRS core network as the next step in [5]. As proposed, several new network elements like the WLAN Access Gateway (WAG) and the Packet Data Gateway (PDG) are introduced into the UMTS environment. The WAG provides the possibility to charge for packet-based functionality as well to enforce the packet routing towards the PDG. In this way, packets of an authorized UE can only be exchanged between the WAG and a dedicated PDG.

Together with high data rate cellular access, HiperLAN/2 has the potential to fulfil end user demands in hotspot environments. HiperLAN/2 offers a possibility for cellular operators to offer additional capacity and higher bandwidths for end users without sacrificing the capacity of the cellular users, as HiperLANs operate on unlicensed frequency bands. The interworking solutions enable operators to utilize the existing cellular infrastructure investments and well established roaming agreements for HiperLAN/2 network subscriber management and billing. In [30] the requirements and architectures for interworking between HiperLAN/2 and 3G systems are described, as well as the operational requirements, user requirements, mobility, QoS mapping, security and standardization requirements to 3G system.

The co-location of multiple standards offers the potential benefits of high data rate and high capacities or mobility at the same time. Intelligent convergence techniques have to be developed in favor of both end users and network operators. It is necessary to control and monitor the transmission of different service traffics through different networks. Therefore a convergence manager with different locations in the networks is designed to implement different convergence techniques according to different scenarios.

Chapter 3

Convergence Manager

The convergence of different standards brings promising benefits into the system design for the future mobile communications. But at the same time, the system becomes much more complex comparing to separate networks. Therefore it is necessary to moderate and control the convergence among several standards. In order to benefit from the existing multiple standards, it is worthy to develop techniques to enable the convergence between multiple wireless standards, and to demonstrate how a user with a mobile terminal can be simultaneously connected to several networks operating according to different standards, allowing him to access a wider variety of services. For that purpose, convergence manager is designed to realize and control the convergence between multiple standards.

3.1 Function of Convergence Manager

The general function of convergence manager is to map services to multiple standards, enabling convergence between different standards. The mapping means that the traffics of the services, e.g. data packets from a streaming video service in packet-access network, are transmitted through different network(s) of different standards. In this case convergence manager will decide how to transmit user traffics through different networks.

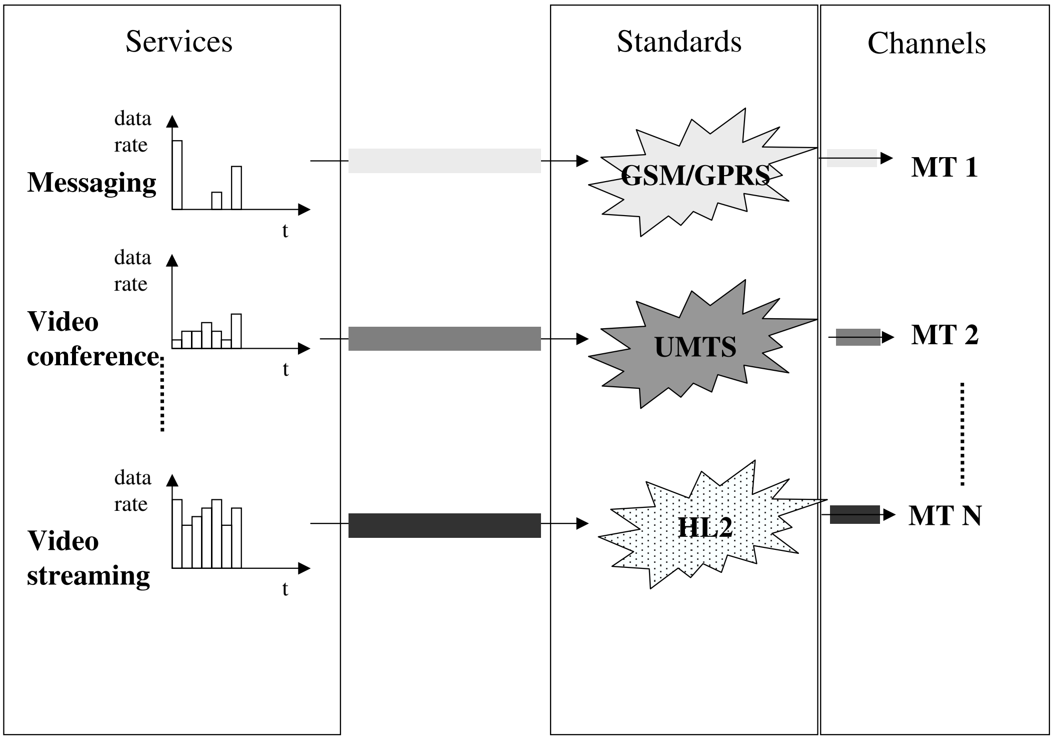


Figure 3.1: Transmission by different standards without convergence manager

3.1.1 Transmission without Convergence

In the current telecommunication market, different standards are designed separately for and can support different services. When there is no convergence between multiple wireless standards, service traffics will be mapped onto a limited number of standards according to their QoS requirement and the standard capacity. As shown in Table 1.1 in Introduction, video streaming service can be served only by HiperLAN/2 because of its high transmission data rate requirement, normal voice services will be carried by GSM and video telephony is properly mapped to UMTS, as shown in Figure 3.1.

3.1.2 Mapping with Convergence Manager

The situation is getting much more complicated, when a single MT can have access to multi-standard networks simultaneously. The concept of convergence between multiple standards would represent a large study to explore this field in detail for a given set of systems. One or more network entities shall be modified and a convergence manager shall be implemented

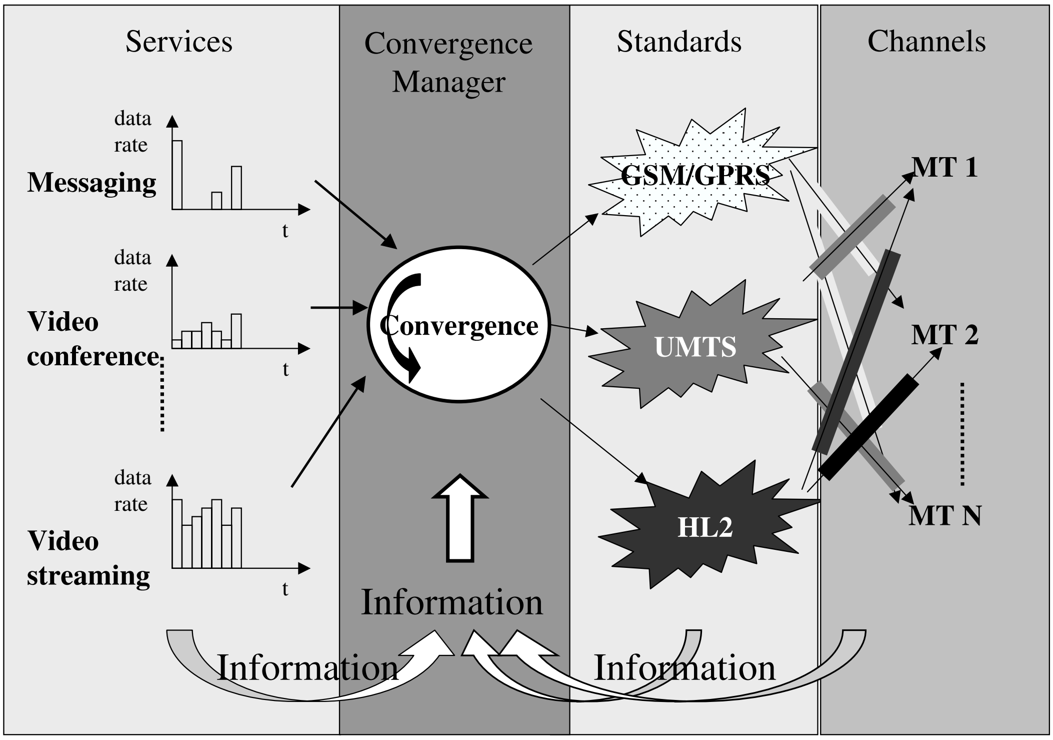


Figure 3.2: Mapping services to standards with convergence manager

somewhere inside the network. How to transmit the variant source traffics from different end users through networks based on different wireless communication standards is then the decision of a convergence manager [19]. The clever decisions contributing to an optimized system performance are relying on the algorithms inside convergence manager. For an optimal mapping of services to standards, convergence manager shall know the status of all networks involved, shown in Figure 3.2.

The decisions of convergence manager are based on different mapping parameters such as QoS requirements, network capacity, channel quality and convergence algorithm, etc. Where to insert convergence manager brings different level of flexibility and complexity at the same time related to the granularity of the convergence.

3.2 Scenarios for Convergence Manager

From both the operators' and end users' point of view, the convergence of standards can happen in different places and convergence manager can apply various scheduling algorithms. Not only

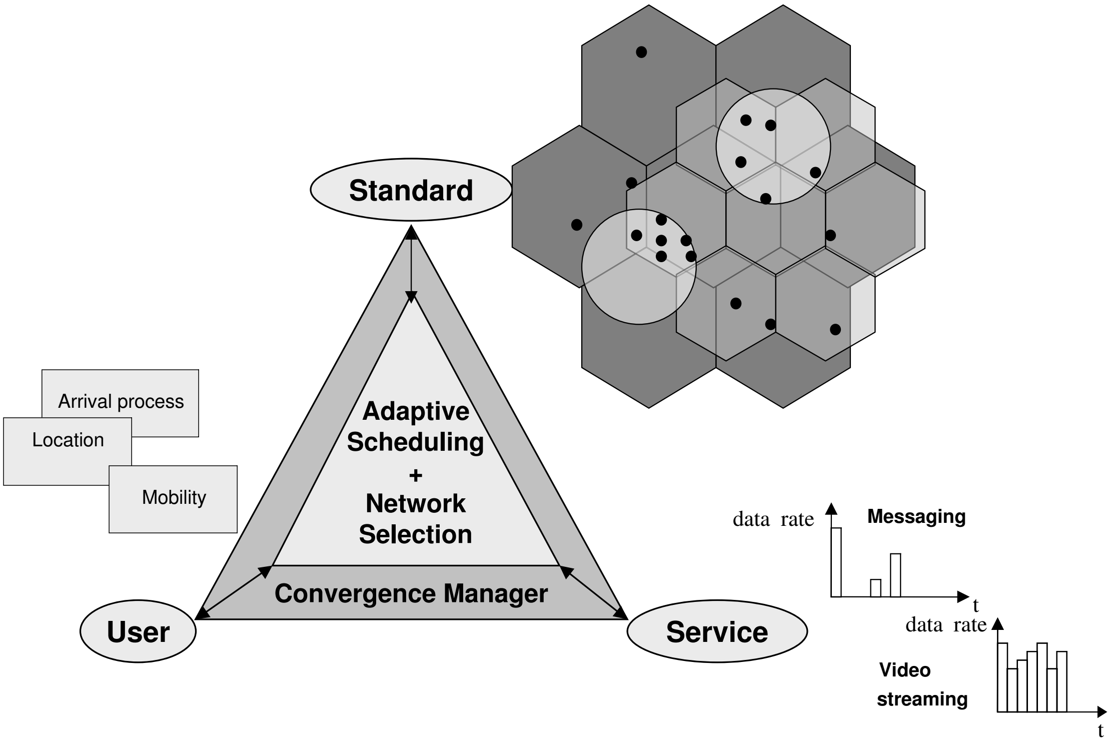


Figure 3.3: Scenario components for convergence manager

QoS classes but also wireless radio channel will influence the convergence technique in convergence manager. In order to assess the benefits and challenges of simultaneous use of standards, it is necessary to identify the relevant scenarios to study convergence manager with different scheduling algorithms.

A scenario for convergence manager consists of a *Service Scenario*, a *User Scenario* and a *Standard Scenario*. These scenarios are components and parameters that influence the performance of convergence manager. It is the task of convergence manager to take into consideration of the requirements from different scenarios.

Figure 3.3 illustrates the scenario components and parameters that influence the performance of convergence manager.

In multimedia services, various types of applications have different characteristics and performance requirements. Traffics from either a conversational class, a streaming class, an interactive class or a background class have various QoS requirements [16], which are to be considered by convergence manager.

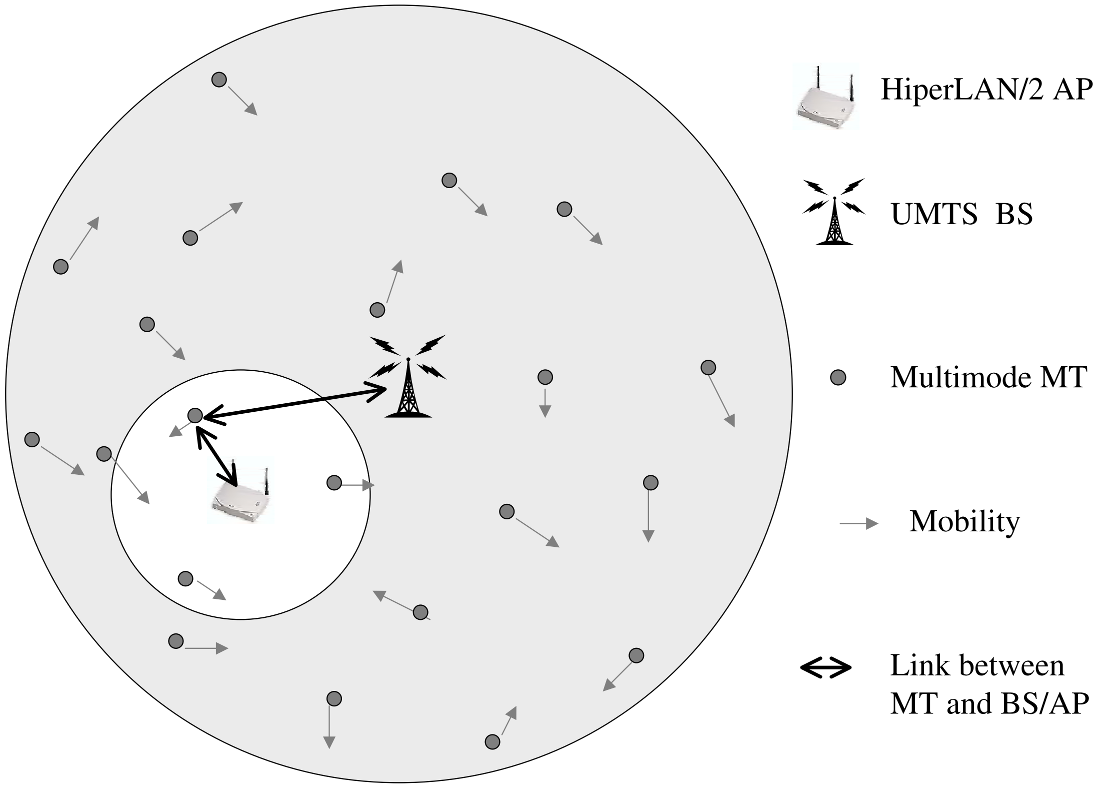


Figure 3.4: Convergence manager scenario with UMTS and HiperLAN

The standards scenario for a convergence manager is based on the co-location of multiple standards. The capacity limitation of each standard is mainly due to their multiple access techniques. Time variant radio channel quality is an important parameter for convergence manager too because link adaptation techniques are a part of the convergence technique.

A typical scenario for this thesis is illustrated as in Figure 3.4. Only a single cell is considered in this scenario comparing to the multi-standard co-location networks as described in Figure 2.6 in section 2.4.2.

UMTS has full coverage and one HiperLAN/2 cell is located inside the area with a smaller coverage than the UMTS. The HiperLAN/2 coverage area is assumed as another round area within the UMTS coverage area. There are numbers of MTs inside the area. Every MT is moving in a direction with fixed speed while the direction is randomly generated during the initialization phase.

The scenarios for convergence manager will be discussed in details in Chapter 4.

3.3 Algorithms in Convergence Manager

The algorithms in convergence manager are implemented by two different categories: network selection algorithm and adaptive scheduling algorithms within single network.

Scheduling within a single standard among end users is an important part of convergence manager function. Convergence manager is designed to find the best optimized usage of current network capacity: allocation of radio resource based on their availability, on the service requirement and the instantaneous situation of mobile radio channel. Link adaptation technique plays an important role during this process. Whether being fair to every end user or allocating more resources to more favorable users influences every single system performance significantly. By applying different scheduling algorithms convergence manager assigns different priorities among end users within the same scenario and offers different portions of resources for their service traffics.

The question of the network selection comes up when an end user could access to more than one networks based on different standards. Which network(s) is (are) the best choice for every single user is a complicated task, considering the requirements from different scenario components. Whether allowing change of network in the middle of an application and whether supporting parallel transmission through multiple networks simultaneously bring different levels of flexibility and complexity for the implementation. How to split the service traffic of a single user among resources from multiple networks influences greatly the overall system performance. Several network selection algorithms are defined and applied by convergence manager which try to improve the system performance.

The function of convergence manager will be fulfilled with a good combination of network selection and adaptive scheduling algorithms. The adaptive scheduling algorithms and the network selection algorithms for convergence manager will be discussed in Chapter 5 and Chapter 6.

3.4 Locations of Convergence Manager

3.4.1 Basic Assumption

The basic assumption is that the MT, enabled for a simultaneous usage of standards, has to contain a convergence manager instance in any case. Depending on the concept this convergence manager entity inside terminal can either work autonomously or cooperate with another convergence manager entity anywhere in the network. Solutions, which make use only of a convergence manager entity in the terminal, are, without specific enhancements, only able to support user defined network selection algorithms. For solutions, which make use of a pair of convergence manager entities, the interaction between these instances, which is implicitly required, enables that both network selection among multiple standards and adaptive scheduling among users in each single standard can be considered.

The location of convergence manager is a crucial issue for the architectural design of an environment, providing different level of convergence between multiple standards. It impacts the user's experience of service rendering and the service provider's ability to combine services increasing the value for the customer. The operator may also take advantage of the functionality of a convergence manager e.g. by utilizing more efficiently the implemented networks. Therefore, the implementation of convergence manager functionality affects many players in the value chain e.g. network operators, terminal manufacturers, terminal software vendors or network equipment manufacturers. Based on the introduced convergence the end user may also be involved in the decision process of choice of network. In a word that the benefits of convergence depends on the location of convergence manager.

Several concepts for convergence manager placement are going to be identified. As potential hosting entities, the MT, BS, CN, backbone network and server entities are all possible, each including significant pros and cons. To make the resulting architecture applicable to real world networks, it is furthermore of importance to consider also the migration path from existing standards towards a convergence manager enabled architecture. Chapter 7 and Chapter 8 describe two convergence manager placement concepts and point out specific advantages and constraints.

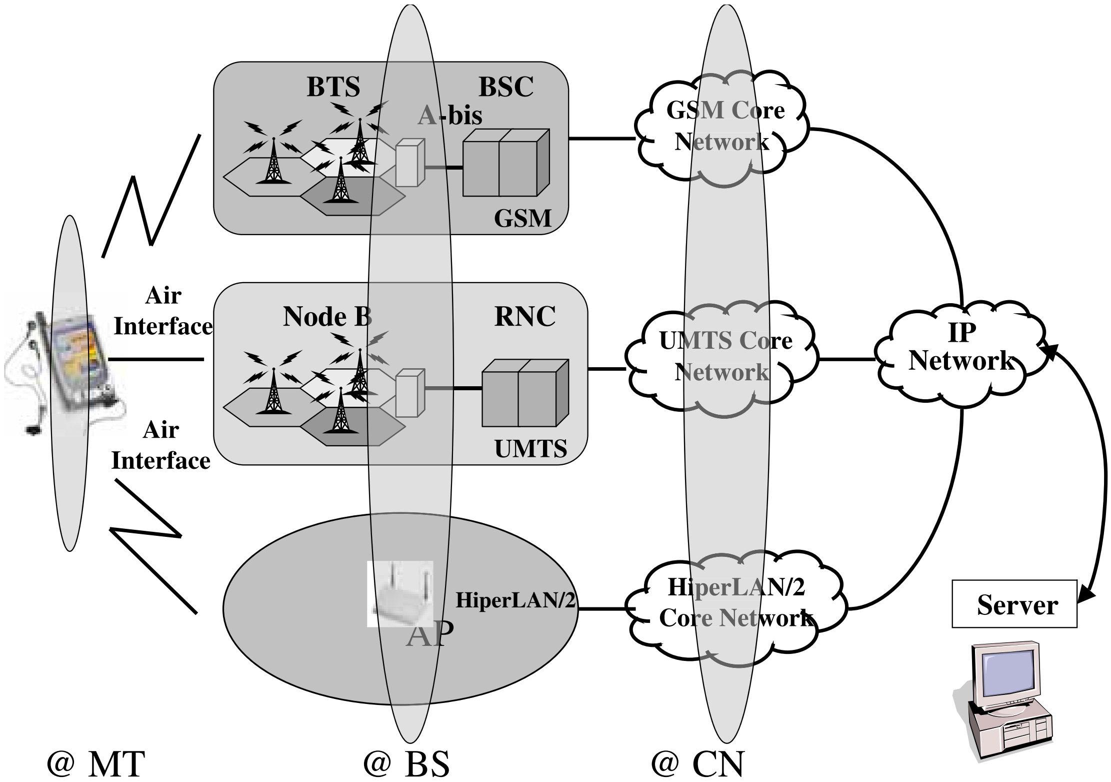


Figure 3.5: Possible locations of convergence manager

Chapter 4

Scenarios for Convergence Manager

Scenario for convergence manager are designed to test and demonstrate the function of convergence manager. Such a scenario for convergence manager consists of a *Service Scenario*, a *User Scenario* and a *Standard Scenario* [38]. This chapter is going to characterize these three components.

Simulations have been carried out in order to illustrate the analytical performance of the convergence manager. The simulation models are build up according to these three scenario components too.

4.1 Service Scenario

In multimedia services, different types of applications have different characteristics and performance requirements. The large set of possible applications has been grouped into four main categories of service classes, according to 3GPP [3]. The four service classes are:

- Conversational class
- Streaming class
- Interactive class
- Background class

4.1.1 Conversational Class

The most well known use of Conversational Class is telephony speech. But with internet and multimedia, a number of new applications will require this scheme, for example VoIP and video conferences. Real-time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception. The real-time conversation scheme is characterized by a low transfer time, because of the conversational nature of the scheme, and at the same time that the time relation (variation) among information entities of the stream must be preserved in the same way as for real-time streams. The maximum transfer delay is given by the human perception of video and audio conversation. Therefore, the limit for acceptable transfer delay is very strict (less than 200 ms), as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is both significantly lower and more stringent than the round trip delay of the interactive traffic case. Traffic is nearly symmetric. Real-time conversation fundamental characteristics for QoS are:

- Time variation among information entities of the stream
- Conversational pattern

4.1.2 IP Telephony (VoIP) Model

Many modules have been build in literatures [12], [48] and [13]. Based on the fact that conversational speech can be characterized by a sequence of talkspurts (service times, messages) which are separated by silent spurts (idle times), a VoIP model is described in Figure 4.1. In a conversational speech, each party is usually silent for more that 60% of the time. Therefore, a packet switched voice transmission can be modelled as a Markov model with two states of "silence" and "talk". When in "silence" no packets are generated, and when in "talk" packets at a constant rate are generated. During the activity phase, IP packets carrying the speech information are transmitted. This type of traffic can therefore be simulated as an ON-OFF model with activity and silent periods generated by an exponential distributed random variable with mean values t_{ON} and t_{OFF} respectively, as for the classical ON-OFF model.

The parameters to be used for this VoIP model is listed in Table 4.1:

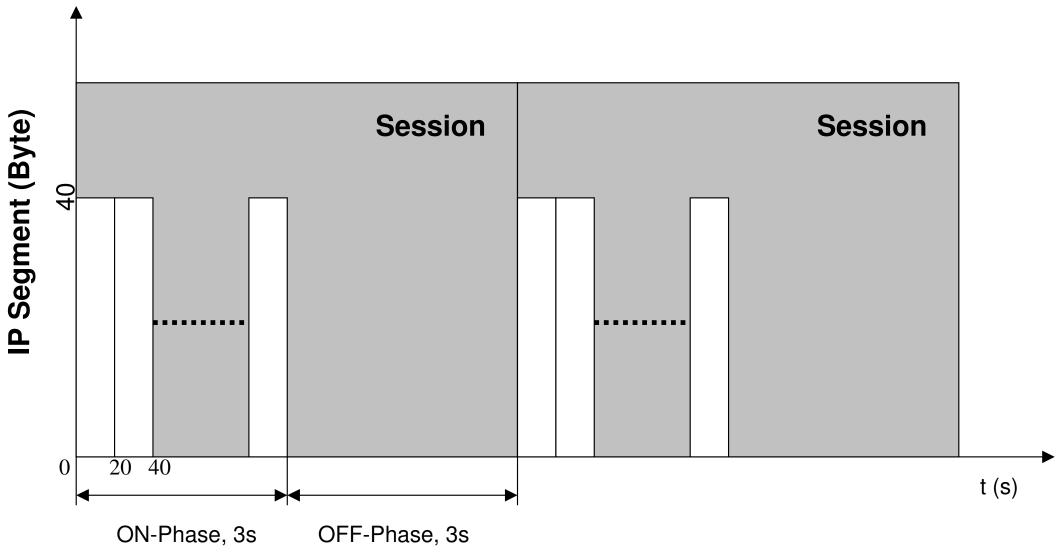


Figure 4.1: IP telephony transmission session

Activity Probability	50%
Mean ON-Phase, t_{ON}	3 s
Mean OFF-Phase, t_{OFF}	3 s
IP-Segment payload	32 Byte
IP-Segment overhead	8 Byte
IP-Segment time interval	20 ms
Mean data rate	8 Kbps

Table 4.1: Parameters of VoIP traffic model

4.1.3 Streaming Class

When the user is looking at (listening to) real-time video (audio) the scheme of real-time streams applies. Multimedia streaming is a technique for transferring data such that it can be processed as a steady continuous stream (e.g. a client browser can start displaying data before the entire file has been received). The real-time data flow is always aiming at a live (human) destination. It is a one way transport (applications are very asymmetric.). At the receiver, streaming data is played by a suitable independent media player application (downloadable from the Web or readily bundled to a browser) or a browser plug-in. This scheme is one of the newcomers in data communication, raising a number of new requirements in both telecommunication and data communication systems. Streaming services are mostly unidirectional. It is characterized by that time relations (variation) among information entities (i.e., samples, packets) within a flow must be preserved, although it does not have any requirements on low transfer delay. The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. But as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application [22]. Thus, acceptable delay variation is much greater than the delay variation given by the limits of human perception. Real-time streams fundamental characteristics for QoS are:

- Preserve time relation (variation) among information entities of the stream

4.1.4 Video Streaming Model

Moving Picture Experts Group (MPEG-4) [35], [22], [45] and H.263 encoded video is expected to account for large portions of traffic in future wireline and wireless networks [33] and [40]. An example of video signals with high average and variable data rates is illustrated in Figure 2.7 in section 2.4.3.

The simulations are using video streaming traffics from different sources with an average date rate of 880 Kbps, but different instantaneous data rates between 100 Kbps and 3 Mbps.

4.1.5 Interactive Class

When the end-user, that is either a machine or a human, is on line requesting data from remote equipment (e.g., a server), the interactive scheme applies. Examples of human interaction with the remote equipment are: web browsing, data base retrieval, and server access. Examples of machines interaction with remote equipment are playing a computer game interactively across the network, polling for measurement records, and automatic data base enquiries (tele-machines). Interactive traffic is the other classical data communication scheme that on an overall level is characterized by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate). Interactive traffic fundamental characteristics for QoS are:

- Requested response pattern within a certain time; round-trip delay time is therefore a key attribute
- Preserve payload content (packets must be transparently transferred, with low bit error rate(BER)

A typical example of interactive service is the access to a web server (web browsing service [23]). The model as shown in Figure 4.2 considers a sequence of packet calls during a www browsing session. The user initiates a packet call when requesting an information entity. During a packet call, several packets may be generated, constituting a bursty sequence of packets.

4.1.6 Web Browsing Model

WWW browsing sessions are basically unidirectional (downlink direction). In these types of sessions, a packet call corresponds to the downloading of a WWW document (e.g., web page with images, text, applets, etc.). After the document is downloaded, a reading time interval (by the user) is experienced.

The following parameters must be modelled in order to catch the typical behavior described in Figure 4.2:

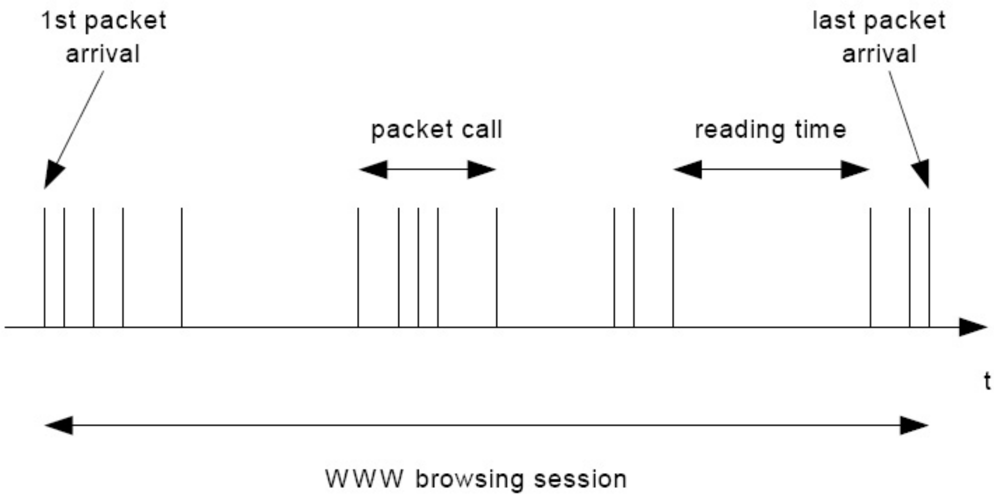


Figure 4.2: Typical WWW browsing session

- Session arrival process
- Number of packet calls per session
- Reading time between packet calls
- Number of packets within a packet call
- Interarrival time between packets (within a packet call)
- Size of a packet

4.1.7 Background Class

When the end user, which is typically a computer, sends and receives data files in the background, this scheme applies. Examples are background delivery of E-mails, SMS, download of databases and reception of measurement records. Background traffic is one of the classical data communication schemes that on an overall level is characterized by that the destination is not expecting the data within a certain time (they do not require immediate action). Thus, the scheme is more or less delivery time insensitive. Delay may vary from seconds to minutes or even hours. Another characteristic is that the content of the packets must be transparently transferred (with low bit error rate). Background traffic fundamental characteristics for QoS are:

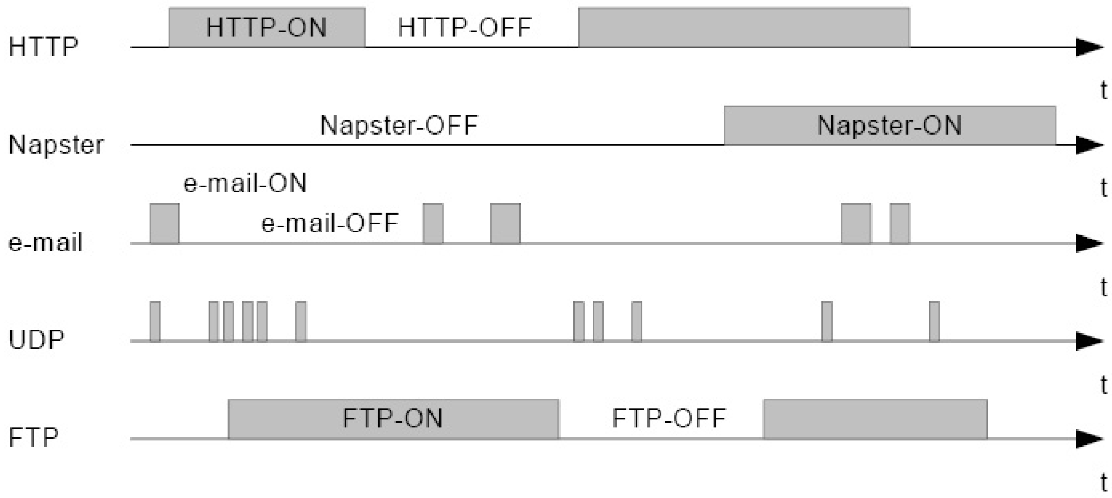


Figure 4.3: Classification of non-real time IP traffic streams

- The destination is not expecting the data within a certain time
- Preserve payload content (packets must be transparently transferred, with low Bite Error Rate (BER))

4.1.8 Email Model

Each application is completely described by its statistical properties, which comprise of an alternating process of ON and OFF periods with some application specific length or data volume distribution, respectively in Figure 4.3. Moreover, within each ON-period the packet arrival process is completely captured by the packet interarrival times and the corresponding packet sizes.

4.1.9 Service Classes Comparison

Table 4.2 resumes and compares the main characteristics of the different service classes.

The main distinguishing factor among these classes the delay sensitivities of the traffics. Conversational is meant for traffic that is very delay sensitive, while Background is the most delay insensitive traffic class. Conversational and Streaming are mainly intended to be used to carry real-time traffic flows. Conversational real-time services, like video telephony, are the most delay

	Conversational Class	Streaming Class	Interactive Class	Background Class
Important requirement	Stringent and low delay	Real time	Low bit error rate	Low bit error rate
Transfer delay	Minimum fixed	Minimum variable	Moderate variable	Big variable
Buffering	No	Allowed	Allowed	Allowed
Traffic structure	Symmetric	Asymmetric	Asymmetric	Asymmetric
Bandwidth	Guaranteed bit rate	Guaranteed bit rate	No guaranteed bit rate	No guaranteed bit rate

Table 4.2: Characteristics of different service classes

sensitive applications and those data streams should be carried in Conversational class. Interactive and Background are mainly meant to be used by traditional Internet applications, like WWW, E-mail, Telnet, FTP and News. Due to less stringent delay requirements, compared to Conversational and Streaming classes, both provide better error rate by means of channel coding and retransmission. Traffic in the Interactive class has higher priority in scheduling than Background class traffic, so background applications use transmission resources only when interactive applications do not need them. This is very important in wireless environment where the bandwidth is low compared to fixed networks.

4.1.10 Data Queues

It is to be expected that in future wireless communication systems, a mix of different services and applications will be required, e.g. voice, data and video will be taken into account, each of that is with different QoS requirement. Throughput, delay, reliability are typical QoS requirement parameters. In the simulation, a mixture of different services are also considered, that mobile terminals are associating with different traffic data models. The data traffics of every terminal are going to be first stored in a data queue for this mobile terminal. It is assumed that each mobile terminal always has sufficient queue length for their source traffic.

An important remark to mention is that the key purpose of the simulation is trying to evaluation the performance of convergence manager, whose function is to dynamically allocate the data packets from the data queues to different radio frames of different standards. The exactly parameter values for different service classes are not necessary to be absolute correct or realistic. It will be sufficient if they could be distinguished according to their typical QoS requirements by the parameters.

4.2 Standard Scenario

The standard scenario describes the technical means by which the user terminal can connect to and interact with services. Figure 2.6 in section 2.4.2 shows the standard scenario based on the co-location of multiple standard networks. There are some places where three different radio access technologies are available, but in other places only a subset of them are available to the user terminals. Or some BSs are capable of providing several different standards, but others provide only one single radio access technology. This example already includes a large number of different scenarios in a single picture. The input parameters should describe the different scenarios by identifying the relevant aspects here.

Figure 4.7 in section 3.2 focuses on a single cell area with full UMTS coverage. Inside the area there is small spot with HiperLAN/2 coverage. For the users inside the HiperLAN/2 spot, they are able to connect to both the UMTS BS and the HiperLAN/2 AP. The convergence manager will decide the transmission path for the source traffics with convergence algorithms, but based on the radio interface situation.

4.2.1 Time-Variant Radio Channels

The time-variant radio channel model is generated to describe the link quality between the MT the HiperLAN/2 AP or UMTS BS [9]. It is necessary to ignore some physical layer details for both standards but rather focus on the radio channel stochastic characteristics.

The time variant broadband radio channel [25] for both HiperLAN/2 and UMTS is given with the following components:

Pathloss

There are different models proposed in plenty of literatures to describe the path-loss within the coverage area. The path loss in the simplest free-space model is given as Equation 4.1:

$$L_{ps}[dB] = -147.6 + 20\log f + 20\log d \tag{4.1}$$

Where f is the carrier frequency of HiperLAN/2 and UMTS standards, given as 5 GHz and 2 GHz respectively. d is the distance between the MT and the AP or BS. It can be seen that in this model the path-loss will depend on the distance between the MT and the BS/AP.

There are other path loss models with more parameters and is suitable for different environment, such as the one-slope model in [31]:

$$L_{ps}[dB] = L_0[dB] + 10n\log(d) \tag{4.2}$$

Where L_0 denotes the path loss at a distance of $1m$, and n is the so-called slope factor. Both parameters depend on the propagation scenario.

Slow Fading

Distributions encountered in meteorology are notoriously non-Gaussian, usually because they have more extreme events than expected for a Gaussian distribution. However, a distribution that is normal in the logarithm of a parameter often provides a much improved fit to observations [50], [43] and [41]. Another advantage of the log-normal distribution is that it is positive-definite, so it is often useful for representing quantities that cannot have negative values. Log-normal distributions have proven useful as distributions for rainfall amounts, for the size distributions of aerosol particles or droplets, and for many other cases. The slow fading p_{a_s} is given by a log-normal distribution as in Equation 4.3:

$$p_{a_s} = \frac{1}{\sqrt{2\pi\sigma^2}}\exp(-\frac{(\ln a_s - \mu)^2}{2\sigma^2}) \tag{4.3}$$

here μ is the expectation value of the normal distribution, σ^2 is the variance. In the simulation, since it is calculated in dB domain, the standard deviation is given as $\sigma = 5.8dB$. The Slow fading interval can use the frame time in UMTS or HiperLAN/2.

Rayleigh Fading

The MTs are assumed to be with mobility, the fast fading is therefore necessary to be taken into consideration. The fast fading amplitude is given by a Rayleigh fading although it is normally assumed for narrowband systems. The Rayleigh fading amplitude is given in Equation 4.4:

$$p_{a_f} = \frac{a_f}{\sigma^2} \exp(\frac{-a_f^2}{2\sigma^2}) \tag{4.4}$$

where σ^2 is the variance. For the reason of normalization, the mean attenuation should be 1 as shown in Equation 4.5:

$$E[a_f^2] = 2\sigma^2 \tag{4.5}$$

The fast fading interval depends on the MT velocity. Within the interval, the fast fading is constant. From interval to interval, the value of fast fading is generated independently.

Normally the Rayleigh fading are considered for narrow-band channel fading while UMTS and HiperLAN/2 are both wide-band communication systems. One solution will be to integrate the fast fading e.g. in each subchannel in Orthogonal Frequency Division Multiplexing (OFDM) based HiperlAN/2 system. For reason of simplicity, the Rayleigh fading is used to describe the whole bandwidth as worst case.

Average Signal to Noise Ratio (SNR)

SNR is the ratio of the power of the wanted signal and the noise. Note that in CDMA the wanted signal power may not be the same as the total power from the wanted base station. Also the noise power may include interference. The time-variant average SNR is given as Equation 4.6:

$$SNR_{ins}[dB] = P_{trans}[dB] - P_{noise}[dB] - L_{ps}[dB] - p_{a_s}[dB] - p_{a_f}[dB] \tag{4.6}$$

The transmission power and the noise power are given in normalized way so that the on the edge of the coverage area the channel capacity can be set to 0. The MT outside the coverage area will not have any link.

Coherent Time

Digital transmission systems are characterized by multipath propagation and shadowing. In a mobile radio channel the situation becomes more difficult because the channel changes very rapidly due to the movement of the MTs. A time-variant behavior is produced. A frequency shift related to the transmitted frequency is produced according to the user mobility, known as Doppler frequency in Equation 4.7.

$$f_D = -f_0 \cdot \frac{v}{c} \cdot \cos(\phi) \tag{4.7}$$

In this equation the f_D is the Doppler frequency, f_0 is the carrier frequency, v is the speed of the MT, ϕ is the angle between the moving direction of MT and the directly link between the MT and BS, and c is the speed of light. Different Doppler frequencies will be found on different paths due to the movement of the MT. This leads to the time variance of both the channel impulse response and channel transfer function. The coherence time is the period T_c of the time interval $[t_0 - T_c/2; t_0 + T_c/2]$ over which the magnitude of the time frequency auto-correlation function is at least half of its maximum value. The transfer function of the channel changes only slightly during the coherence time. The following approximation as Equation 4.8 is used:

$$T_c = \frac{1}{f_{D,max}} \tag{4.8}$$

The coherence time indicates the maximum time interval when the channel quality can keep constant.

4.2.2 Link Adaptation Technique

Although it is known that indicator based link adaptation technique has much better performance [44], for reason of simplification, the mapping from SNR onto channel data rate is taken out based on the average channel behavior. In this case, the average SNR of a radio link is taken into account in accordance with some resulting average Packet Error Rate (PER) performance curves, in order to determine a suitable Physical (PHY) mode in HiperlAN/2 and Modulation Coding (MC) mode in HSDPA respectively.

PHY Mode Selection in HiperLAN/2

The scheduling algorithms in the convergence manager is based on the adaptive consideration. In order to improve the radio link capability due to different interference situations and distance of MTs to the AP, a multi-rate PHY layer is applied in HiperLAN/2 [29], where the "appropriate"

Modulation	Coding rate	Max. throughput (Mbps)
BPSK	1/2	6
BPSK	3/4	9
QPSK	1/2	12
QPSK	3/4	18
16QAM	9/16	27
16QAM	3/4	36
64QAM	3/4	54

Table 4.3: PHY modes in HiperLAN/2

PHY mode will be selected by a link adaptation scheme. The data rate ranging from 6 Mbps to 54 Mbps can be varied by using various modulation on each OFDM subcarrier and by applying different puncturing patterns to a mother convolutional code. Binary PSK (BPSK), Quadrate PSK (QPSK), 16 Quadrate Amplitude Modulation (16QAM) are used as mandatory modulation formats, whereas 64QAM is optional for both AP and MT. In HiperlAN/2 the link adaptation means chosen of PHY modes out of the 7 possible PHY modes as in Table 4.3.

Different criteria are conceivable for selection of an appropriate PHY mode for a given radio channel behavior: delay-oriented or throughput-oriented. The chosen of the PHY mode will be based on the expected Packet Error Rate (PER) for a given instantaneous radio channel. For PER estimation, conventionally the average received SNR together with the average PER performance curves are used [31]. An example of the PHY mode selection based SNR and PER is shown in Figure 4.4.

Adaptive Modulation Code (AMC) Selection in HSDPA

The HSDPA concept has been designed to increase packet data throughput by means of fast physical layer (L1) retransmission and transmission combining as well as fast link adaptation controlled by the Node B. The link adaptation is implemented as adaptive modulation and coding scheme selection according to the link channel quality.

By employing 15 multi-codes and 3/4 coding rate, the HSDPA can achieve up to 10 Mbps data rate. The combinations of different modulation and coding schemes as well as their achievable data rates are shown in Table 4.4. With time and code multiplexing of users, the theoretical data

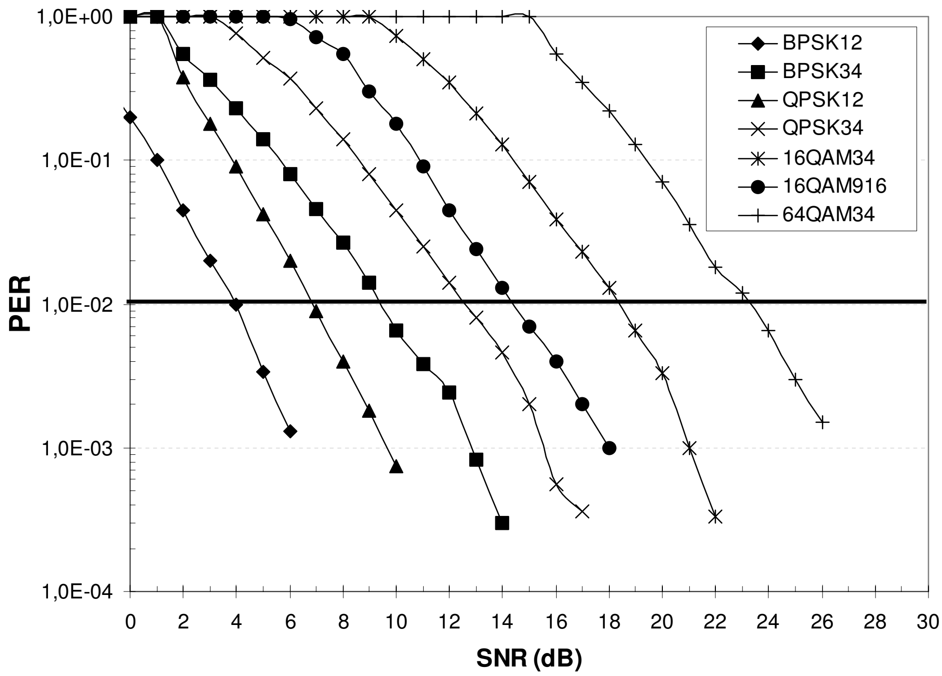


Figure 4.4: An example of HiperLAN/2 PHY mode selection based on SNR [49]

rates can be achieved by a single user or divided between several users. This provides the place for an optimized resource allocation among the users according to their capabilities. The basic principle of HSDPA is to adapt to the current channel conditions by selecting the most suitable modulation and coding scheme, known as AMC selection. Similar to the delay oriented PHY model selection in HiperLAN/2, the AMC selection can be applied based on the average PER and SNR performance curves.

	Modulation	Code rate	Max. throughput (Mbps)
1	QPSK	1/4	1.8
2	QPSK	1/2	3.6
3	QPSK	3/4	5.3
4	16QAM	1/2	7.2
5	16QAM	3/4	10.8

Table 4.4: Modulation and coding modes in HSDPA

4.2.3 Link Layer Frame

The radio frames are the interfaces between MT and BS. Radio frame structures are defined by every single standard.

HSDPA frame

[1], [2] and [4] define the physical layer of UMTS. The HSDPA concept has been designed to increase packet data throughput by means of fast physical layer (L1) retransmission and transmission combining as well as fast link adaptation controlled by the Node B. The transport channel carrying the user data with HSDPA operation is denoted as the High-Speed Downlink-Shared Channel (HS-DSCH) [36].

The Transmission Time Interval (TTI) in HS-DSCH is defined to be 2 ms (3 slots) to achieve short round-trip delay for the operation between the terminal and Node B. 16QAM is added to increase the peak data rate. SF is fixed to 16 but multi-code transmission as well as code multiplexing of different users can take place. The maximum number of codes can be allocated is 15. Each terminal may allocate different number of codes depending on the UE capacity and the channel condition. An example with two active users is illustrated in Figure 4.5, who are using the same HS-DSCH. Both users will have to check the information from the High-Speed Signal Control Channel (HS-SCCH) to determine which HS-DSCH codes to de-spread and other necessary parameters.

HiperLAN/2 Medium Access Control (MAC) frame

The air interface is based on Time Division Duplex (TDD) and TDMA, i.e., the time-slotted structure of the medium allows for simultaneous communication in both DL and UL within the same time frame, called MAC frame [8], [27] in HiperLAN/2. Time slots for downlink and uplink communication are allocated dynamically depending on the need for transmission resources. The basic MAC frame structure on the air interface has a fixed duration of 2 ms and comprises transport channels for broadcast control, frame control, access control, downlink and uplink data transmission and random access (Figure 4.6). All data from both AP and MTs is transmitted in dedicated time slots, except for the Random Access CHannel (RCH) where contention for the same time slot is allowed. The duration of broadcast control is fixed, whereas

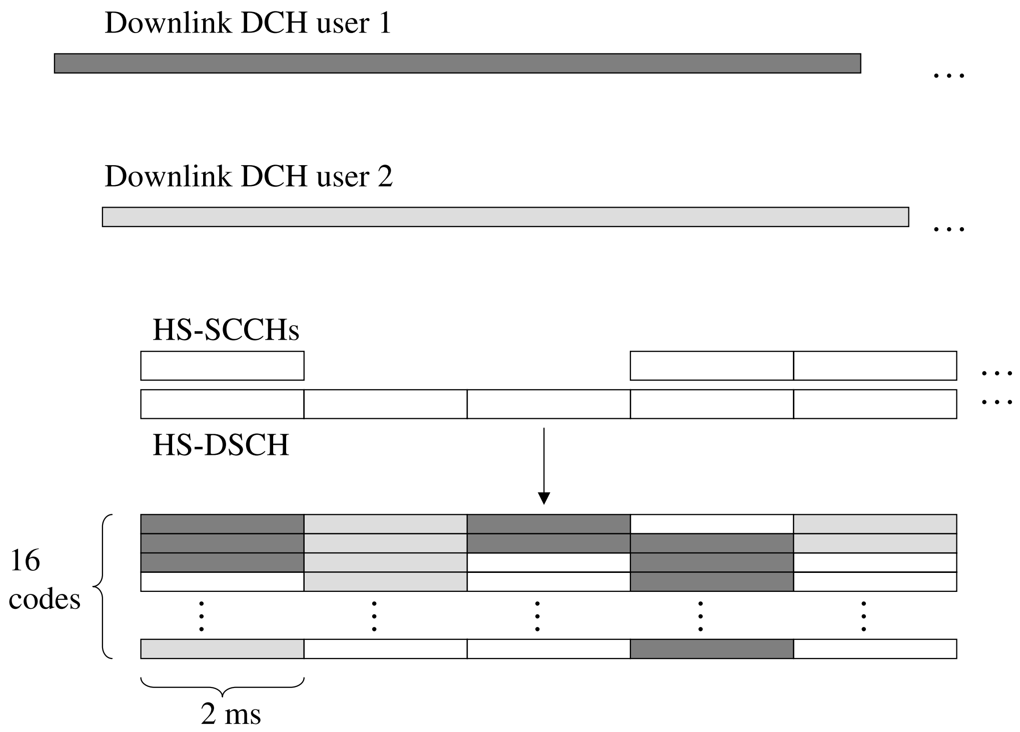


Figure 4.5: HSDPA link channel frame structure

the duration of other fields is dynamically adapted to the current traffic situation. The MAC frame and the transport channels form the interface between Data Link Control (DLC) and the PHY layer.

A MAC frame comprises a broadcast phase, downlink phase, uplink phase and a random access phase. The broadcast phase comprises fields for Broadcast Control (BCH), Frame Control (FCH) and Access Feedback Control (ACH). The uplink and downlink-phase carries traffic from/to an MT in bursts of packets called Long CHannels (LCH) and Short CHannels (SCH). A LCH contains control or user data and a SCH contains only control data (e.g., acknowledgements and Resource Requests in the uplink) and is always generated by the DLC layer.

4.3 User Scenario

The end users are the owner of the mobile terminals, they have also important affects to convergence manager, e.g. their preference to select the transmission standard. End users in the real world have different roles, a business man will use his mobile terminals different from a student.

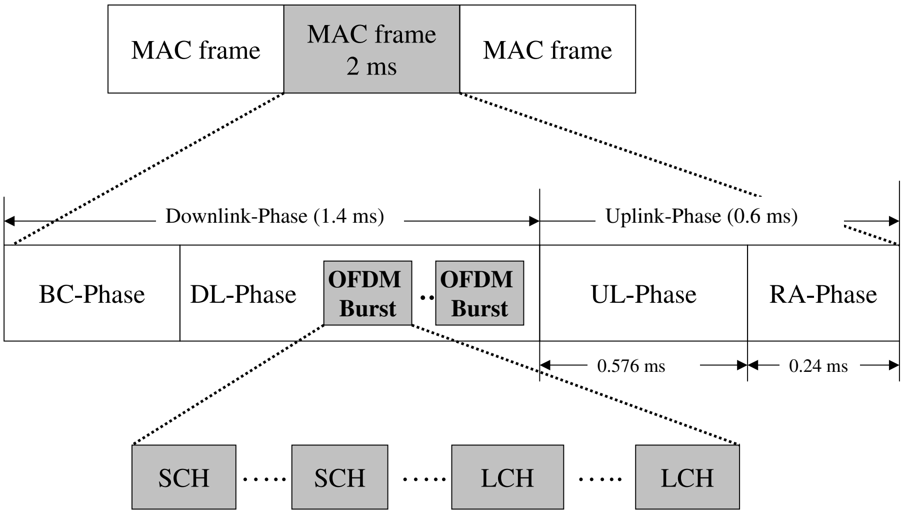


Figure 4.6: MAC frame structure in HiperLAN/2

End users may use their mobile terminals in different situation, calling mother may differ from watching football games per mobile telephone. End users will choose different contracts for their telephones, paying by time or by load. User satisfaction is one of the criteria to evaluate the performance of convergence manager.

4.4 Scenario Parameters for Simulation

In order to evaluate the convergence manager, it is necessary to set the three scenario components into parameters which are important for setting up simulations [18].

4.4.1 Service Scenario Parameters

The service scenario is defined by the type and characteristics of the service and the applications that the end user handles. For each convergence manager service scenario, the relevant characteristics of those services can be simplified to the following input parameters:

- Number of different services
- Data rate requirements (constant or variable)
- In case of variable, expectation value, variance and probability density function

- QoS requirements, including:
  - Maximum delay
  - BER or PER
- Connection oriented or connectionless

4.4.2 Standard Scenario Parameters

The input parameters for standard scenario may include:

- Number of different standards and BS capabilities, i.e. which different radio access technologies can be provided by a given basestation
- Geography of the cells and spatial distribution of the BS, i.e. which cells from different radio access technologies are available at any given point
- Antenna characteristics
- Radio channels and their propagation parameters

4.4.3 User Scenario Parameters

Input parameter in the user scenario related to the behaviors of the mobile terminals can be:

- Number of users (given by per cell or per km^2)
- Spatial distribution of users (uniformly or non-uniformly distributed)
- For each mobile terminal the following information may be given:
  - Position
  - Available standards
  - Service used by this user
- Traffic model
  - Arrival process
  - Probability density function (pdf) of service duration

4.5 Simulation Scenario: UMTS coverage with HiperLAN/2 Cell

Before we will discuss about the performance and benefits of convergence manager in the coming chapters with different considerations, it is necessary to talk about the scenario used for the simulation. A single-cell scenario is considered. Two wireless communication standards are considered here:

- HSDPA for UMTS
- HiperLAN/2

The geometric coverage is given in Figure 4.7. UMTS has full coverage inside the round area and one HiperLAN/2 cell is allocated inside the area which has smaller coverage than the UMTS. We give the HiperLAN/2 coverage area as another round area within the UMTS coverage area. There are numbers of mobile terminals inside the area, every MT is assumed to be able to work within both standards, named as in this thesis multi-mode MT. It is assumed that there is always a convergence manager functionality integrated inside the terminal so that it can access two standards simultaneously. whether a convergence manager is implemented inside the network or the BSs will bring different results and they will be discussed in the coming chapter.

The MTs can be divided into the following groups, taking each observing moment:

- MTs outside the UMTS coverage will not be served until they move into the UMTS coverage area. The MTs outside the UMTS coverage are not included in the result calculation.
- MTs inside the UMTS coverage but outside the HiperLAN/2 hotspot can only transmit their traffic over UMTS.
- MTs inside the HiperLAN/2 hotspot coverage area have the possibility to transmit through both UMTS and HiperLAN/2 standards. For those MTs, the convergence manager will decide the route of their source traffic through which standard(s).
- Two cases are considered according to the distribution of the MTs inside the UMTS coverage: uniform or hotspot distribution. In case of hotspot, there is ten times the MT density within the hotspot comparing to the rest of the coverage area.

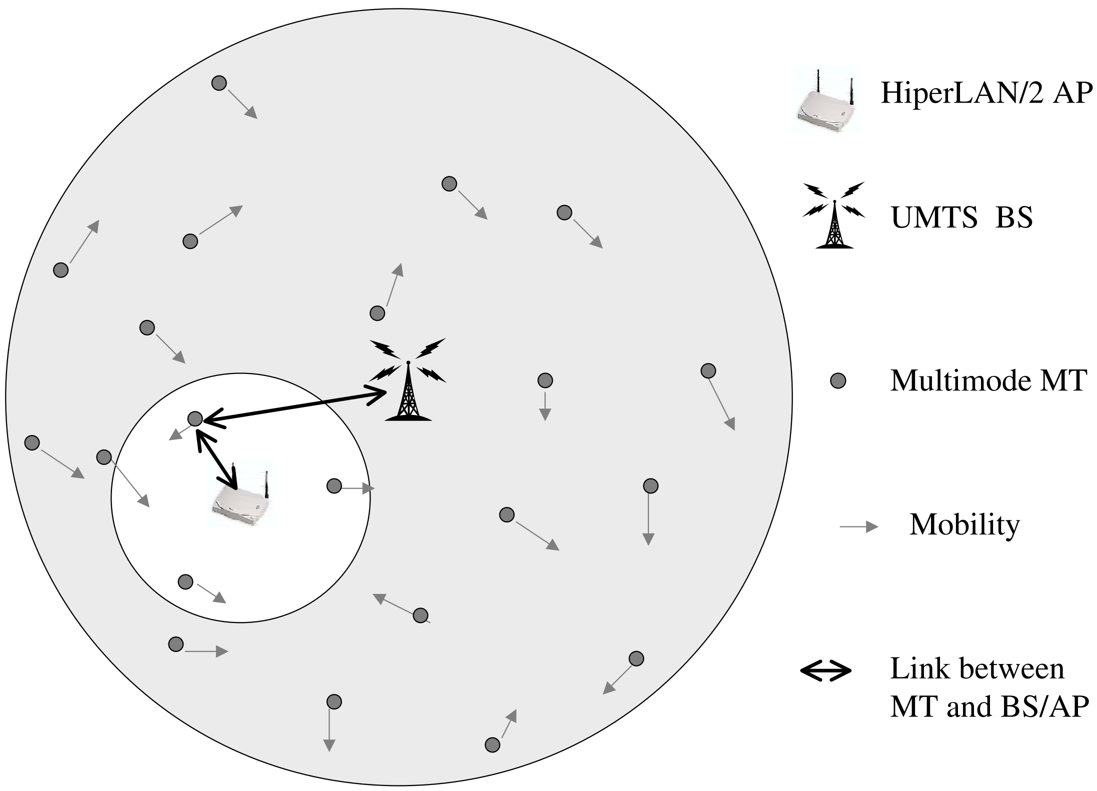


Figure 4.7: Simulation scenario with UMTS and HiperLAN

VoIP and video stream are used as source model. The source traffic can be considered as time-variant, with typical data rate requirement (VoIP with mean 8 Kbps while Video stream with 880 Kbps as required data rate). The traffics are segmented into packets with fixed length.

The basic assumptions concerning the parameters for the simulation scenario are summarized in Table 4.5.

Parameter	Set-up Value
Source Traffic	VoIP and Video streaming, MT with single service
Frequency	$f = 5GHz$ for HiperLAN/2 and $f = 2GHz$ for HSDPA
Slow fading	Lognormal distribution, $\sigma = 5.8dB$
fast fading	Rayleigh fading $\sigma^2 = 0.07$
Modulation	BPSK, QPSK, 16QAM, 64QAM
Coding	Convolutional codes, Code Rate $R = 1/4, 1/2, 3/4, 9/16$
Connection direction	Downlink
Multiple access	TDMA for HiperLAN/2, CDMA for HSDPA
Radio Frame length	2 ms for HiperLAN/2 MAC frame (1.38ms used for downlink), 2 ms for HSDPA radio frame
Packet size	54 Bytes
Cell size	Round, $Radius = 150m$
BS for HSDPA	Center of cell
AP for HiperLAN/2	(50m, -50m), $Radius = 50m$
MT	Uniform distribution or Hotspot cell
Mobility of MT	Radome starting direction, fixed direction after starting, $1m/s$ or $10m/s$
Adaptation	Delay oriented PHY mode and AMC mode selection

Table 4.5: Summary of the parameters used in the simulation scenario

Chapter 5

Adaptive Scheduling Algorithms in Convergence Manager

This thesis is dealing with a broad-band, down-link dominated, time and frequency multiplexed wireless mobile system, trying to study the dynamical mixing of different types of traffic over fading channels. The function of convergence between different standards can be described as allocation of wireless resources based on their availability and on the service requirement.

The technique of convergence manager then will include two parts: selecting the network(s) of multiple standards for each end user and scheduling of multi users inside each standard network. The adaptive scheduling algorithms are going to assign different priorities to end users within each single network, aiming for the optimized system performance. These priorities are at the same time important information for the network selection algorithms.

In this chapter the adaptive scheduling algorithms used in convergence manager and their performances are going to be discussed.

Adaptivity scheduling is crucial in order to obtain improved spectral efficiency in the 3G and future mobile communication systems. The often discussed adaptive scheduling schemes are normally based on the real-time prediction. In order to achieve good scheduling performance, it is always necessary to obtain an accurate long-term channel prediction. If such prediction cannot be made, non-predictive approaches will be applied, such as coding and interleaving, perhaps with link adaptation. To achieve useful result for a scheduler, it is also necessary to have access to the different requirements set for different traffic classes.

There have been many different scheduling algorithms discussed in the literatures [20]. There are scheduling algorithms aiming at providing fairness among the served users, whereas others aiming at minimizing the average waiting time, or other parameters to evaluate the performance such as average discarding rate, average served number of users, etc.

In general, most of the scheduling algorithms can fit into the following two categories:

- Fairness oriented
- QoS oriented

5.1 Fairness Oriented Scheduling

5.1.1 Generalized Processor Sharing (GPS)

GPS for wirelined channels means that the resource will be shared among the users in a way so that for any time interval, the resource will have been serving a user i at least as much as it guaranteed share, s_i, of the total service rate. The share for the user c_i is expressed by Equation 5.1:

$$s_i = \frac{r_i}{\sum_{u=1}^{U(t)} r_u} C(t) \tag{5.1}$$

where r_i is the rate weight for the user i, which dictates what portion of the service capacity $C(t)$, available at time t, should be given to user c_i. Furthermore, $C(t)$ is the total capacity at time t, with $U(t)$ being the number of users at time t.

The fairness criterion is also known as another way of description. Let $W_i(t_1, t_2)$ denote the assigned bandwidth, or transmission rate, to the user c_i during a time interval [t_1, t_2]. If $s_i(t)$ is allocated according to Equation 5.1, and

$$W_i(t_1, t_2) = \sum_{t=t_1}^{t_2} s_i(t) \tag{5.2}$$

then

$$\left| \frac{W_i(t_1,t_2)}{r_i} - \frac{W_j(t_1,t_2)}{r_j} \right| = 0 \tag{5.3}$$

for any two clients c_i and c_j. Equation 5.3 is the fairness criterion. It means that a user that needs service, during any short time interval $[t_1, t_2]$, will at least get a share of the total service according to it pre-defined service rate share r_i. This can be guaranteed as long as the sum of all the guaranteed service rate shares does not exceed that total service rate.

GPS [10] is not really a scheduling algorithm by itself. But it is used in the scheduling community as a benchmark for how an idealized fair scheduler should behave. The GPS discipline requires a fluid model for the traffic that it served and all users waiting to be served can be served simultaneously. In another word, the server has infinite capacity and the traffic are infinitely divisible in the same time. There is no such case in real scenarios, therefore the GPS cannot be realized exactly. But its behavior can be approximated in different ways.

There are variants out of GPS, for example Weighted Fair Queuing (WFQ) [57] tries to assign resources to the clients according to their pre-defined service rate requirements. It is indeed a packet-by-packet approximation of the GPS discipline.

5.1.2 Proportional Fair Scheduling (PF)

The proportional fair scheduling algorithm (PF) [37] for wireless channels allocates the resource to the user that can transmit at the highest rate, relative to its average achieved throughput during some window of past transmissions.

In its most basic form, the PF algorithm assigns the resource to the user indicated by

$$\arg\max_i \frac{(C/I)_i(t)}{T_i(t)} \tag{5.4}$$

where $(C/I)_i(t)$ is the current carrier-to-interference ratio for user i at time t, and $T_i(t)$ is the achieved data throughput for user i, in a limited time window, up to time t. PF scheduling is resource efficient, in the sense that users that have received worse service for some time, will be compensated in accordance with their weights in Equation 5.4.

The wireless channel variations offer the possibility to achieve multiuser diversity gain, if the variations be efficiently exploited. The multiuser diversity gain results from the possibility to utilize the wireless channels at higher transmission rates that their average possible rates, by utilizing temporarily good transmission conditions and avoiding temporarily bad channel conditions, while still utilizing all channel resources all the time.

There are variants from Equation 5.4, for example, normalization is performed with the average attainable throughput, which results in the scheduler assigning the resource to the user with the best resource quality as compared to its average resource quality.

It is observed that PF factorizes large channel variations. To overcome it, it is proposed to take stochastic distribution (e.g. pdf) into consideration, looking at the entire distribution of the transmission rate instead of the recent average transmission rate.

5.1.3 Modified Proportional Fair Scheduling (MPF)

The aim of the QoS scheduling algorithms is to take the channel properties into account, in order to efficiently utilize that resource when they offer good service, and at the same time pay attention to throughput and delay requirements from the users, in order to offer attractive and predictable services.

A Modified Proportional Fair Scheduling (MPF) algorithm is given in [14] with the modification aming to support delay sensitive services. MPF gives an increased absolute priority to users that have exceeded their delay constraints. At the same time, it gives up the possibility to achieve the scheduling gain from the channel variations.

Among the mentioned algorithms, MPF is dependant to the channel and queue states. It can take into account the externally assigned priority for different QoS requirements. Several scheduling algorithms are implemented inside convergence manager with the similar approach, taking both channel and queue states and even the external priority into account.

It is important to mention that here only the instantaneous channel quality are taken into account, which means a short-term scheduling. There are another direction in the scheduling algorithms research, where the methods utilize prediction of future channel qualities, in order to schedule a long-term transmission. They are believed to achieve a more holistic optimization. But those scheduling algorithms are not discussed in this thesis.

The convergence manager with suitable scheduling algorithms assigns priorities to the data packets and data queues for the order of transmission. The following described scheduling algorithms can be distinguished by the fact if they take into account the channel state information, and/or the data sources requirements when fixing the priorities for the data packet.

5.1.4 Round Robin (RR)

The Round Robin algorithm (RR) is very fair and simple. The resources are uniformly distributed between users. It could be implemented as that the users take turns in utilizing the service for a

Service	Traffic Model	GSM/GPRS	UMTS	HiperLAN/2
Conversational	voicecall	high priority	high priority	low priority
Streaming	Videosteam	low priority	low priority	high priority
Interactive	Webbrowsing with 144kbps	not supported	middle priority	middle priority
Background	FTP with 64kbps	low priority	low priority	low priority

Table 5.1: Pre-defined service priority within different standards

equal period of time, or for a pre-defined amount of work, in a cyclic way. There are also variants of the RR, for example, the different clients will be served according to a pre-defined rate weight.

5.2 Queue Oriented Scheduling

For the allocation of data packet from user source traffic queue, only information from higher layers such as QoS parameters is used. The instantaneous radio channel states are not taken into account for scheduling in this types of algorithms.

5.2.1 Shortest/Longest Queue (SQ/LQ)

The user with the shortest/longest queue length in a given time frame will be assigned with the highest priority, without taking into account the channel state information of each standard. The priority may change from time frame to the next one after the last transmission.

5.2.2 Pre-defined Service Priority (SP)

Different services/applications and traffic source models have different characteristics with different QoS requirements. In the same time, the capacity of different standards are limited according to the air interface technologies. It is then possible to give the priority to the user according to their service in each standard. For this purpose, Table 5.1 has been built with some example traffic source models and their given priorities in different standards according to their capacity limitation.

5.3 Channel Oriented Scheduling

In this case, exclusively information from the lower layers (PHY layer and instantaneous radio channel characteristics) is exploited. The scheduler tries to adapt the resource allocation to the dynamic and stochastic channel behavior, e.g. by assigning priority to user who experiences a favorable radio channel.

5.3.1 Best Channel (BC)

The algorithm in convergence manager based on the absolute best channel will optimize the data throughput, but on the other hand, always give priority to the users close to the BS, who can achieve high SNR. It is not fair to the other users who may not have so favorable position.

5.3.2 Relative Best Channel (RBC)

To apply fairness aspects, it might be useful not to choose the user with the *absolutely* best, but rather the *relatively* best radio channel. RBC is a scheduling algorithms that could be used by convergence manager. The relative transmission rate state $trsraterelative_i$ for user i is expressed in a quantized form by the number of the currently achieved transmission rate $trsRate_i$ and the average transmission rate $trsrateave_i$ (averaged over the last several time frame) as:

$$trsraterelative_i = trsrate_i - trsrateave_i \tag{5.5}$$

Comparing with the absolute best channel algorithm the user with a low average transmission rate (e.g. a user far away from the Base Station) is given a much better chance of allocation if the channel state instantly improves.

5.4 Cross Layer Adaptive Scheduling

The complex tasks in mobile communication systems is commonly grouped into different layers, ranging from the application (highest layer) to the air interface (lowest layer). The well-known ISO/OSI-model [34] consists of seven different layers. In such an hierarchical approach, a layer communicates exclusively with the layers directly above and below it, so that try to keep the effort which must be spent for the information exchange between layers to a minimum. It is the main advantage of the conventional approach that the layer-specific protocols can be developed and optimized independently of each other. However this approach is limited in flexibility and the protocol optimization can only be processed on the basis of available information from adjacent layers. The transmission of the high data rates from different applications with heterogeneous and variable QoS parameters demands a high flexibility of the involved layers and the communications structure. The OFDM transmission technique in HiperLAN/2 is part of the PHY layer, but continuously generates detailed information about the current radio channel conditions, which can be used for a channel prediction and for an adaption procedure in a cross layer approach. Similar concepts can also be applied to the UMTS systems, e.g. in HSDPA system.

In [52], [58] and [61] the requirements of a cross layer consideration or PHY and DLC layer are considered. It can be expected that in this case a cross design and optimization of both layers with a larger degree of interaction will lead to a considerable capacity enhancement of the overall system [58] and [62].

In this case, information from both higher and lower layers is taken into account. It is then to consider the channel statistics as well as the data source statistics for the construction of radio frame in different standards. The concept has been described in Figure 5.1:

5.4.1 Shortest Transmission Time (STT)

In the beginning of every radio frame, the time needed for the transmission of all data packets for each user is calculated and the times are sorted. This transmission time is influenced by the number of packets inside the queue and additionally the instantaneous channel state. Priority is given to the user whose packet require the shortest transmission time. The transmission time t_i for user i at time t is calculated as:

$$t_i(t) = \frac{N_i PL(t)}{C_i(t)} \tag{5.6}$$

Where, N_i is the number of packets inside queue i, PL is the packet size and $C_i(t)$ represents the currently channel capacity (maximum transmission data rate) at time t. From this equation, it is apparent that the resource allocation takes into consideration the requirements of higher layers and as well as the stochastic behavior of the radio channel.

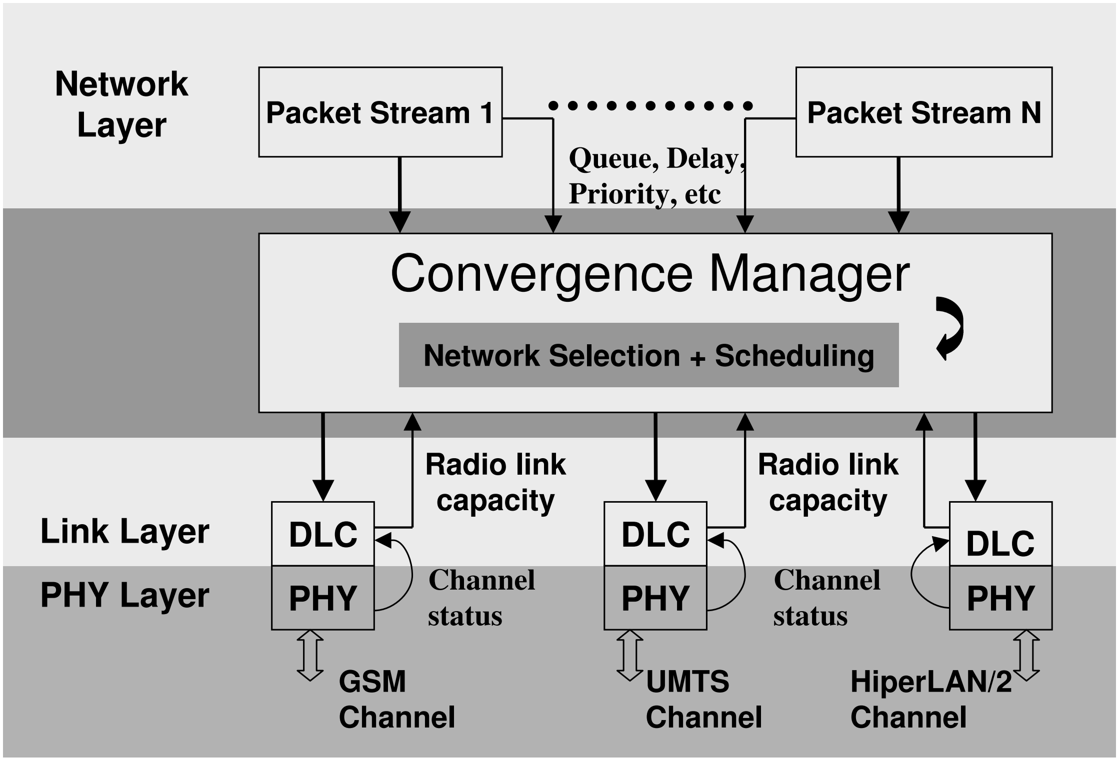


Figure 5.1: Cross Layer consideration in convergence manager

Algorithm	Channel Orient	Queue Orient	External priority	Fairness	QoS Orient
RR	No	No	No	Yes	No
SQ	No	Yes	No	No	Yes
SP	No	No	Yes	No	Yes
BC	Yes	No	No	No	No
RBC	Yes	No	No	Yes	No
STT	Yes	Yes	No	NO	Yes

Table 5.2: Summary of scheduling algorithms in convergence manager

5.5 Performance of Scheduling Algorithms in Convergence Manager

The discussed scheduling algorithms in convergence manager are listed in Table 5.2, according to whether they are dependent from the channel status, queue status, external priority, fairness and QoS requirements.

5.5.1 Comparison Scenario

The performance of a scheduling algorithm in convergence manager depends on a lot of parameters. Channel variance, user distribution, Service component are all important elements that could affect the performance [21]. It is really difficult to compare different scheduling algorithms and to say which one outperforms the others based on the absolute performance curves in different scenarios. A very simple simulation scenario is used to illustrate the performance of some of the scheduling algorithms which are used in convergence manager. We can have an idea of the performance comparison between different scheduling algorithms based on such a scenario.

A HiperLAN/2 cell with multiple users is considered as the simulation scenario (Figure 5.2). The access point is located in the center of the cell, and with full coverage inside the cell. The users are uniformly distributed inside the coverage area. Each user is moving in a fixed direction with the same speed. The starting directions of the users are randomly generated. Only downlink situation is considered and the MAC layer is structured as described in section 4.2.3. The PHY

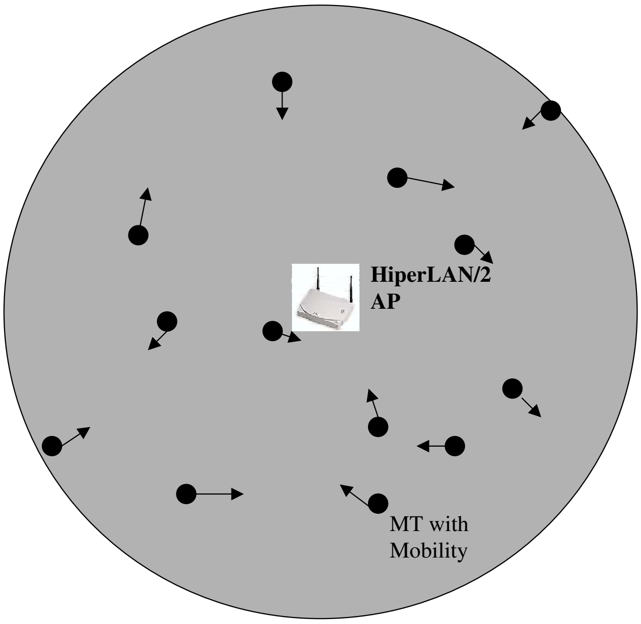


Figure 5.2: Simulation scenario for convergence manager adaptive scheduling algorithms

mode selection is applied as described in section 4.2.2. The queue length for every user is assumed to be infinite long so that all the source packets can be stored inside the queue. The packet based traffic model is designed such that every user is receiving every 40 ms a segment with variable length but the same mean data rate (1 Mbps). The segment will be discarded if not all the packets in the segment are transmitted during the 40 ms time frame. No additional signalling delay is inserted when applying adaptive techniques.

The parameters used to compare several scheduling algorithms are average system throughput and average discarding rate among all of the end users.

5.5.2 Performance Comparison

This performance comparison between several scheduling algorithms are shown in the following figures. Figure 5.3 and Figure 5.4 show the average system throughput or discarding rate versus number of users inside the coverage area. The system throughput increases when more users are inside the area. But sooner or later, the system throughput will reach the AP capacity limitation which means overloading.

The following system performances could be observed:

- When there are few users inside the scenario, the discarding rate is very small. This means that no overload situation happens and the system throughput performance is limited by the average data rate of the service. In such kind of situation, there is no benefits, if implementing complicated adaptive scheduling algorithms in convergence manager.

- When more users are inside the scenario and overloading happens for the AP, the system performance will reach the saturation. The maximal achieved system throughput is far away from the theoretical capacity of HiperLAN/2 as 54 Mbps. The worst case is shown by RR that almost every packet segment is discarded. It has the worst performance since there is no adaptive technique used. In such kind of situation, it is necessary to implement adaptive scheduling algorithms by convergence manager.

- The RBC has worse performance comparing to the absolute BC, which is not surprising since the consideration of fairness decreases the QoS related system performance.

- In Figure 5.3 the BC has better throughput performance than the SQ while Figure 5.4 shows the contrary performance with discarding rate. Whether the channel oriented scheduling algorithms will perform better than queue oriented scheduling algorithm is hard to say, since there are so many parameters who could change the performance, e.g. user distribution, user mobility, channel time-variance and even the traffic model.

- It is clear to see in both figures that the cross layer adaptive scheduling algorithm STT outperforms all other algorithms.

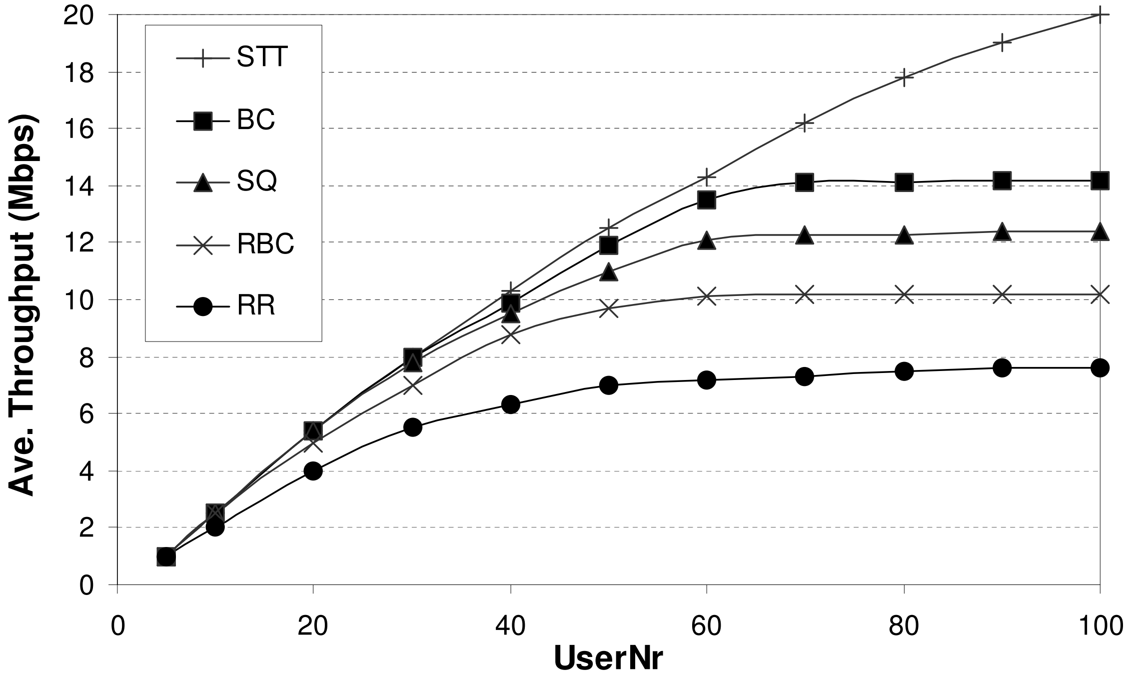


Figure 5.3: Average system throughput performance of different adaptive scheduling algorithms

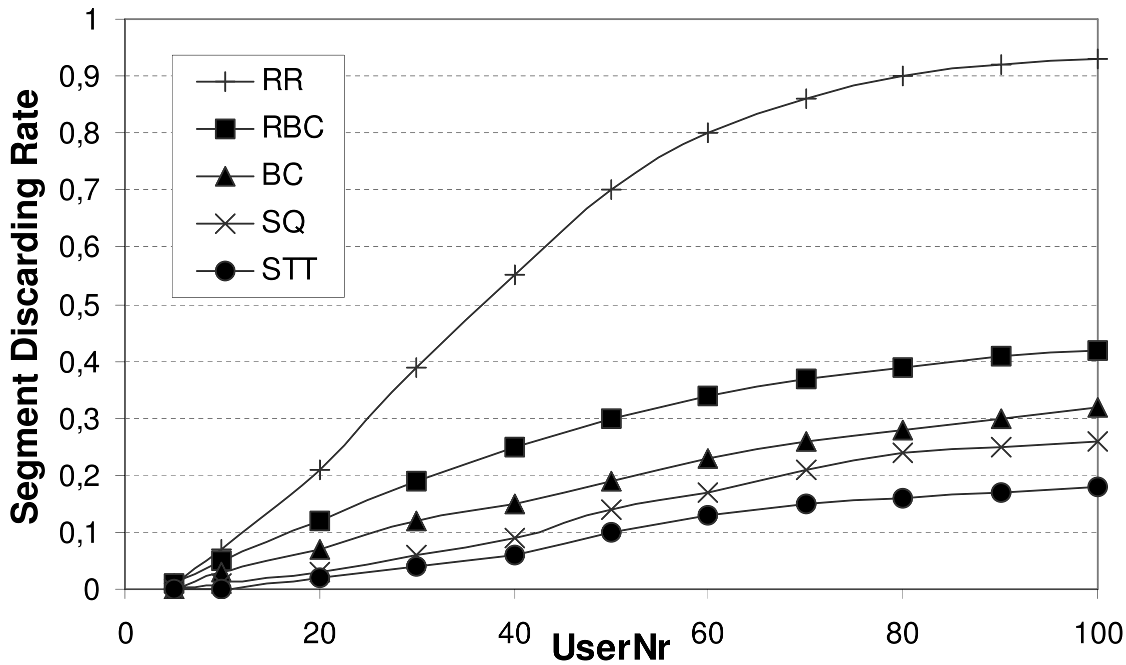


Figure 5.4: Average discarding rate performance of different adaptive scheduling algorithms

Chapter 6

Network Selection Algorithms in Convergence Manager

The technique inside the convergence manager consists both network selection and adaptive scheduling inside each standard. The convergence manager with network selection algorithms enables that an end user with a mobile terminal can be simultaneously connected to several networks operating according to different standards, allows him to access a wider variety of services.

In this chapter, several network selection algorithms for convergence manager will be discussed. They can be divided into different categories by whether allowing change of networks during one application or parallel transmission through multiple networks. According to them, different level of flexibility and complexity will have to be implemented by the convergence manager.

An Always Best Connected (ABC) concept [17] as network selection algorithm exists in an environment where several different types of access networks and different devices are available to a user. The user can choose the access network and device that best suits his or her needs, depending on the applications that he or she is currently running, and change whenever something better becomes available. Which is the best solution strictly depends on the application scenario, on the requirements and on the perspective (e.g., the 'best' from the user perspective does not necessarily coincide with the 'best' from the network operator perspective). The ABC concept was traditionally the vision behind vertical handover system from GSM or UMTS accesses and networks to unlicensed spectrum standards such as IEEE 802.11x. In that sense, ABC has always been about multi-access networks, although currently the concept has been extended to include some new dimensions, such as adaptive applications, reconfigurable terminals and networks, and operator competition. The new scenarios, network, service and terminal environments, and business models implicit in positing ABC as a key and integral feature of the 4G wireless world will require the setting down of careful definitions of many new terms and system components, first among which is the definition of ABC itself. From a user's perspective, whenever a phone or data call is made, the user would like to choose the access network that currently offers the 'best' price-performance ratio (e.g. minimizing cost, or maximizing the level of perceived quality of service, or minimizing cost provided that a certain quality of service constraints are satisfied). Users are not interested in the underlined standards in use, but only in the availability of services. In this way, an ABC system should not be confined to the implementation of multi-access standards for different radio networks, it must consider user's needs and requirements.

6.1 Pre-Selection Algorithm

Different services/applications and traffic source models have different characteristics with different QoS requirements as has been discussed in section 4.1. At the same time, the capacity of different standards are limited according to the air interface technologies. The pre-selection algorithm in convergence manager is considered as a pre-defined static mapping table between the services and standards. For that purpose, an extension of the Table 4.2 in section 4.1.9 has been built as Table 6.1, where some example traffic source models and supporting technologies are selected based on the information type of each service/application.

This network selection algorithm is very simple to be implemented, only a mapping table stored inside the mobile terminal will be sufficient to carry out the function. And there will be no additional delay introduced during neither the decision phase nor the handover phase.

This consideration is focusing on the standard selection procedure, where simultaneous use of multiple standards are implemented as vertical handover between standards. This means each MT are transmitting through via only one standard in any time but might via multiple standards if considering the whole process. From this point of view, the convergence managers are used to optimize the selection of standards. Several single network selection algorithms are described in Figure 6.1.

	Conversational Class	Streaming Class	Interactive Class	Background Class
Important requirement	Stringent and low delay	Real time	Low BER	Low BER
Transfer delay	Minimum fixed	Minimum variable	Moderate variable	Greatly variable
Buffering	No	Allowed	Allowed	Allowed
Traffic structure	Symmetric	Asymmetric	Asymmetric	Asymmetric
Bandwidth	Guaranteed bit rate	Guaranteed bit rate	No guaranteed bit rate	No guaranteed bit rate
Traffic Model	Speech telephony	Video stream	Webbrowsing	FTP
GSM/GPRS	High priority	No	No	Low Priority
UMTS	High priority	No	Low priority	Middle Priority
HiperLAN/2	Low priority	High priority	High priority	high Priority

Table 6.1: Pre-defined mapping between services and standards

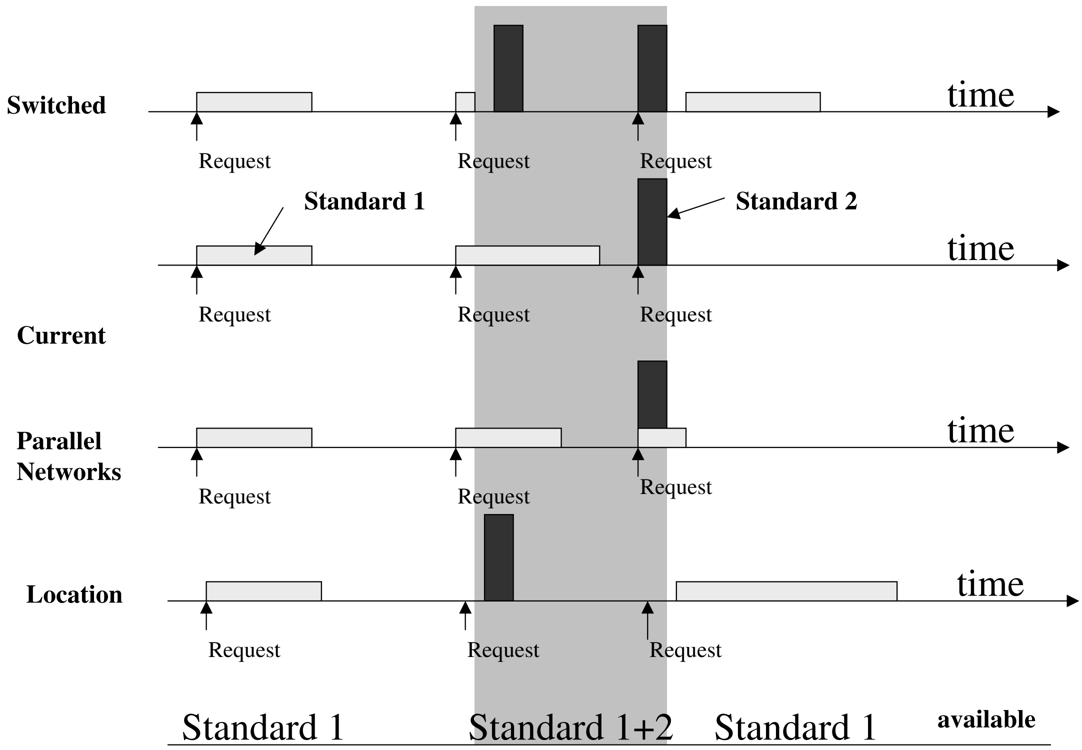


Figure 6.1: Network selection algorithms in convergence manager

6.2 Switched Algorithm

When the service requests arrives, the access network providing the highest data rate is selected from a list of currently available access networks. This algorithm supposes that convergence manager enables the switching between access networks while mid of application. At the end of each coverage period MT switches between standards to keep continuous connectivity and also to make sure that the transmission takes place always from the highest data rate bearer however parallel transmission on multiple standards is excluded. There are complexities associated with switching between different wireless access networks while in the middle of transmission, e.g. it requires close coordination of the convergence manager and involved BS/AP, significant interoperability and accurate synchronization for the transfer. An additional delay could be inserted into the transmission in order to modelling the time for the handover. And it will be seen that this delay affects the performance dramatically. But if the handover delay could be limited to a small amount of the whole radio frame time duration, this network selection algorithm shall be considered as with promising benefits.

6.3 Current Algorithm

The current algorithm is similar to that of switched algorithm in the sense that the standard providing the highest data rate is selected. However in contrast to switched algorithm, the transmission takes place via only one standard in order to avoid the complexity. The selection of standard is done when the request arrives and transmission starts, thus the vertical handover delay can be saved and in the real case, it is much easier to be implemented inside the convergence manager.

6.4 Location Algorithm

In location algorithm it is assumed that the convergence manager knows the mobility context of the user and the geographical coverage of wireless standards. For example, the familiar environment gives the chance to save the usual route of MT , e.g. where the WLAN AP is located and these info can be provided to the convergence manager by means of GPS. The Mobility context includes current location, moving direction and speed etc. Integration of sensors inside the MT may catch the mobility information and save it inside the MT.

The users mobility context and the knowledge of standards coverage area is used to make more intelligent and informative standard selection decision. The main consideration is to avoid the standard switching during one transmission process due to the complexity.

Assuming the simulation scenario for convergence manager as in Figure 3.4 in section 4.5, if a request arrives when the user is located in the area where only UMTS is available, the algorithm estimates that whether the transmission can be completed before the user enters the HiperLAN/2 cell. If yes, then select UMTS for this application otherwise wait until the user enters HiperLAN/2 cell and download via HiperlAN/2 then. Similarly if a request arrives when the user is inside the HiperLAN/2 cell, the convergence manager will estimate whether the download can be completed before the user leaves the HiperLAN/2 cell. If yes, the convergence manger will select HiperLAN/2 for the user otherwise UMTS.

6.5 Parallel Networks Selection Algorithm

Obviously the multi-standard network selection algorithms are more complicated than those who select only single standard. When the end user is located inside the coverage of multiple standards and has access to the different networks at the same time, the idea of the parallel network selection algorithm is to carry out the transmission parallel within multiple networks. This algorithm applies fully the concept of simultaneous transmission over different networks. But this algorithm brings the challenge of how to split the traffics of every user queue among different networks at the same time. The convergence manager has to somehow give priorities to different standards and decides the sequence of packets through different networks.

On the other side, this network selection algorithm fits well for those services who require high reliability. Repeating the same packets via different network routes will for sure brings benefits according to diversity gain. But for throughput oriented data services, it is almost impossible to implement this algorithm in the convergence manager.

There are ideas such as to transmit the packets from the beginning of data queue via one standard network while the packets from the end of data queue via another network in order to reduce the complexity for this algorithm. Due to to many open questions this algorithm can only be regarded as the theoretical best performance with convergence among multiple standards.

6.6 History Based Location Algorithm

The Location algorithm suffers from wrong decision due to the out of date channel data rate. Both cellular standards and WLAN provide variable date rate for every application, e.g. they do not guarantee a fixed data rate. So in this situation the mean data rate that would be achieved during the current download process can be inferred from the history, e.g. mean data rate achieved during the last file download process. The location algorithm is modified and named as "History Based Location Algorithm". History is used to predict the mean data rate for the current application. The algorithm performance depends on how long history could be used. The mean data rate during the last considered time is used as the predicted mean data rate for the current transmission.

6.7 Performance of Network Selection Algorithms in Convergence Manager

The simulation scenario for convergence manager is considered as in Figure 3.4 in section 4.5. A generic file download process is assumed to be the source traffic. Users generate requests to download files according to a Poisson process. The file size distribution may be constant or uniformly distributed. A single user is assumed so that we could have an idea of the performance of different network selection performance.

There are two kinds of delays associated with a file download process. One is the delay before the download started, named as *initial delay* and the other is the *actual download time*. The sum of the two gives the *total download time*. Figure 6.2 shows the mean total download time versus file size.

With reference to Figure 6.2, a couple of observations can be made:

- A single standard is not able to provide a realistic download time for full range of file sizes. It can be observed that small file download can be completed much faster through UMTS only method which is because of full coverage of UMTS resulting in zero initial delay whereas file download via HiperlAN/2 operation is often penalized with large initial delay. As the file size grows the limitations arising from the inherent low data rate of UMTS become visible and it takes longer time to complete large file download. On

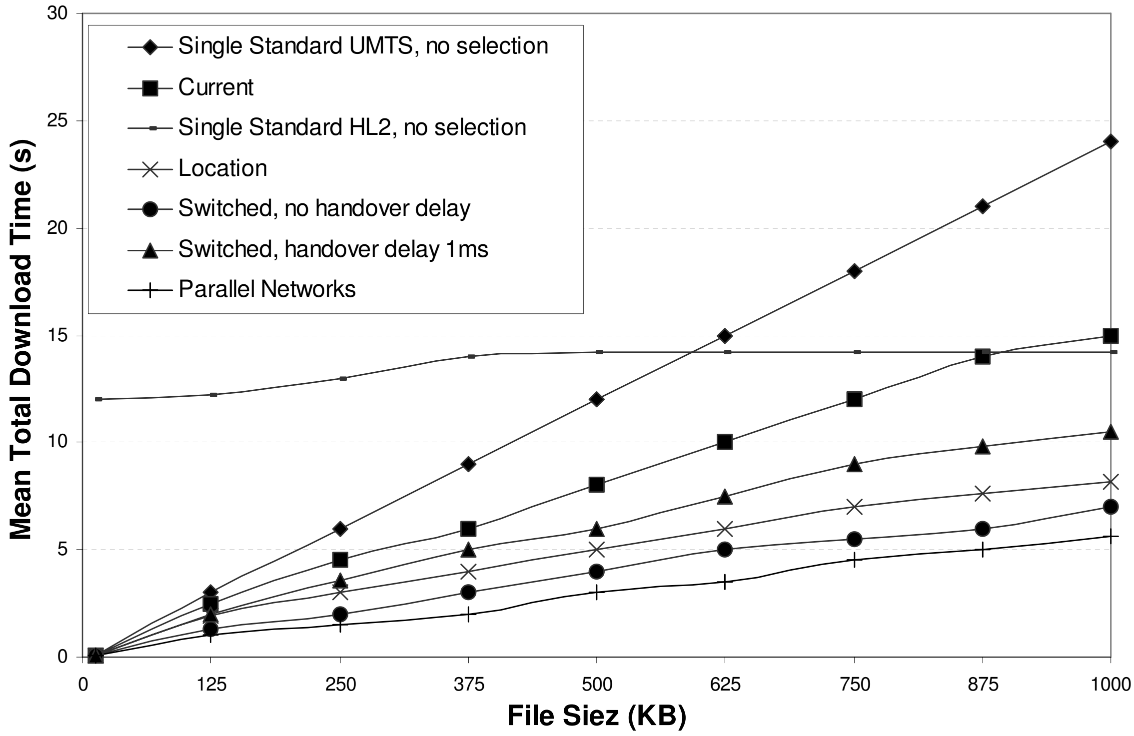


Figure 6.2: Delay performance of network selection algorithms with single user

the other hand the total download time via HiperLAN/2 is almost independent from file size. It is because of the high data rate offered by HiperLAN/2. The delay performance is more related to the location of the user.

- The current algorithm offers better delay performance than single network situation for the entire range of file size. It is due to the regional availability of HiperLAN/2 cell during which higher data rate reduces the file download time. However the wrong selection of the "current best" selection of standard limits the system performance too.

- The switched algorithm has a very good overall delay performance. The handover delay affects the performance largely, system with 1 ms extra delay has performed similar to the current algorithm.

- Location algorithm is of great interest because it helps to prevent the above said problem in current algorithm by using the knowledge of user-mobility and coverage-range-information of UMTS and HiperLAN/2. It exhibits the delay performance quite close to the switched algorithm over all range of file sizes and also it avoids the complexities associated with switching mid way of transmission as is the case with switched algorithm.

- The parallel network algorithm exhibits the lowest overall delay without considering signalling overhead. The lowest overall delay in this case is because it always downloads via both possible network, therefore ensures the highest data rate. It can be considered as the theoretical lower bound of performance for comparison purpose. But the performance is very near to that of the switched algorithm.

If considering multi-user inside the scenario, the situation is getting much more complicated because of multi-user interferences. The performance of the different network selection algorithms in the convergence manager in a scenario with multiple users will be discussed in details in Chapter 7 and Chapter 8.

Chapter 7

Convergence Manager @Mobile Terminal

The location of the convergence manager is a crucial issue for the architectural design of an environment, providing simultaneous usage of standards. The location has several definitions as following:

- The physical location where the hardware or software implementation for convergence take place, e.g. MT, BS/AP, Network Control Center, and Server, etc.
- The logical location in which layer of the seven layers defined in ISO/OSI model, e.g. PHY layer, DLC layer, or IP/Network layer, etc.

In the first approach, the location of convergence manager impacts the user's experience of service rendering and the service provider's ability to combine services to increase the value for the customer. The operator may also take advantage of the functionality of a convergence manager e.g. by utilizing more efficiently the implemented networks. Therefore, the implementation of the convergence manager functionality affects many players in the value chain e.g. network operators, terminal manufacturers, terminal software vendors or network equipment manufacturers. Based on the granularity of the introduced convergence the end user may also be involved in the decision process of choice of network.

For the second approach, the location of convergence manger impacts the level of flexibility and optimization. The convergence manager can then achieve different level of convergence between different standards. The different location of layers in the communication model will bring the limitation to the information exchange in convergence manager, therefore affects the achievable advantage for convergence manager.

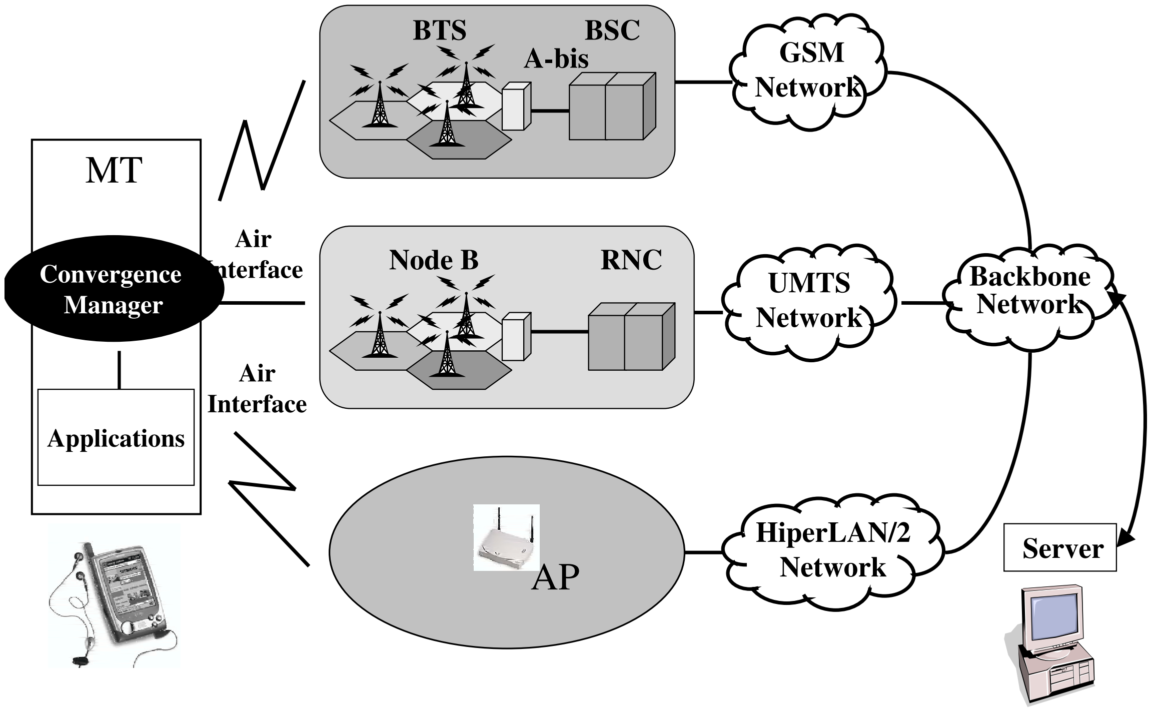


Figure 7.1: Convergence manager is located @MT

In most of the cases, it is difficult to distinguish between these two approaches, different physical location means mainly at the same time different level of information exchange in the abstract model.

Figure 3.5 in the previous section 3.4 has shown the possible locations of the convergence manager inside the networks which are based on different wireless standards. The terminal, access network components, core network components, backbone network components and server entities are all potential hosting entities, each including significant pros and cons. The scheduling algorithms and the network selection algorithms implemented by the convergence manager will affect the system performance, while at the same time, the location of the convergence manager limits somehow the application of the different algorithms. Some algorithms show surprising performance inside multiple user scenario comparing to that with single user. In some cases, some algorithms are even not applicable.

In this chapter, the concept of a single convergence manager instance in multi-mode MT is discussed (Figure 7.1). The convergence between multiple standards is to be realized through this entity of convergence manager but without any change in the existing networks.

An example application for such a convergence manager is to switch the video channel of an ongoing video conference from UTRAN to WLAN, whenever WLAN is available. Because of the higher available bandwidth in WLAN a better video codec can be used which increases the service quality for the user.

There are several possibilities, how the convergence entity can operate on the terminal. Two applications are investigated in the following sections. Section 7.1 shows a generic way, how the Session Initiation Protocol (SIP) [42] can be used to direct media streams or data channels via different access technologies. Because SIP runs at the session layer. This approach is named as the session layer concept. Section 7.2 provides an investigation about a specialized solution, enabling convergence functionality for a dedicated class of traffic, the file transfer traffic.

7.1 Session Layer Convergence

7.1.1 Inter Media Service (IMS) and SIP

The 3GPP has standardized the so-called IMS [7], which will be used for a fully IP based multimedia scenario, including also session establishment, starting with Release 5. SIP [42] will be used for this purpose. This convergence manager can reside between the session layer and the lower layers of a MT. The flexibility of SIP enables such architecture without any changes on the existing standards.

The IMS provides a proper solution to the users to reach more than one physical access points simultaneously. The MT first has to send registrations in his home network. During the registration process, different IP addresses could be assign to the MT.

After having obtained two different global IP addresses, the next step is to establish connections using available standards simultaneously. Including here the end-to-end scope of the session layer, it is no longer necessary to operate a convergence manager element in the network.

The convergence manager entity in the MT has also a reduced functionality. The most important task of the convergence manager is now to monitor the changing conditions on the lower layers and to crosscheck changes with the algorithms. If there are consequences for an ongoing session or for sessions, which shall be initiated new, the convergence manager has to indicate this to the session layer. The principle architecture of the MT regarding to the convergence manager is shown in Figure 7.2.

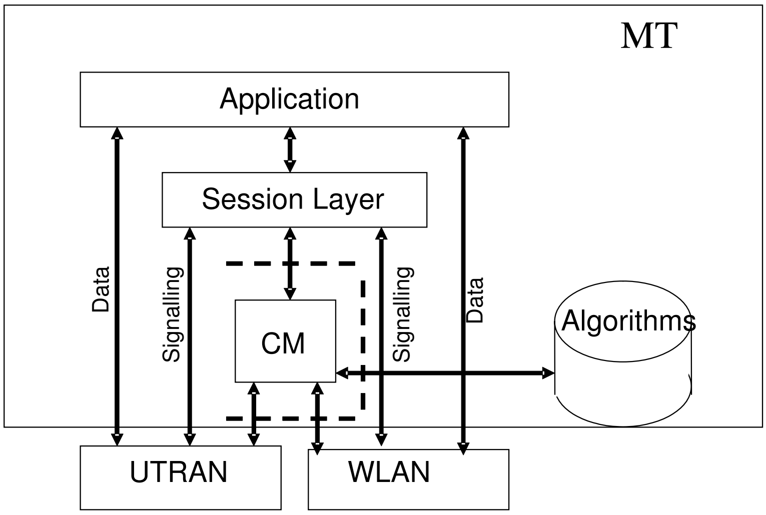


Figure 7.2: Terminal Architecture including convergence manager

The next step is that the session layer convergence manager should establish connections, making use of the lower layer possibilities according to the network selection algorithms defined in the policy base.

7.1.2 Video Conference Session in UTRAN and WLAN

A video conference session in UTRAN and WLAN can be used to explain the SIP approach. Because of the higher available bandwidth in WLAN a better video codec can be used, which increases the service quality for the user. It can be assumed that the signalling traffic shall go via the UTRAN. In this scenario, at first the UE has no WLAN coverage and therefore establishes both streams, audio and video, using the IP address obtained from the GPRS IP-Control Area Network (IP-CAN). Figure 7.3 shows the scenario in this stage.

After the recognition of the convergence manager that WLAN is now available too an end-to-end signalling interaction has to take place, including additionally several QoS related message exchanges. This delay may not be favorable for several applications. However, it is the procedure, which the 3GPP has foreseen for such conditions of changing session parameters. The flexibility of the IMS and especially of the SIP provides us with the ability to reuse the already defined session layer interactions for our purpose and therefore the proposed solution provides the requested convergence manager behavior. Figure 7.4 shows the resulting traffic flows.

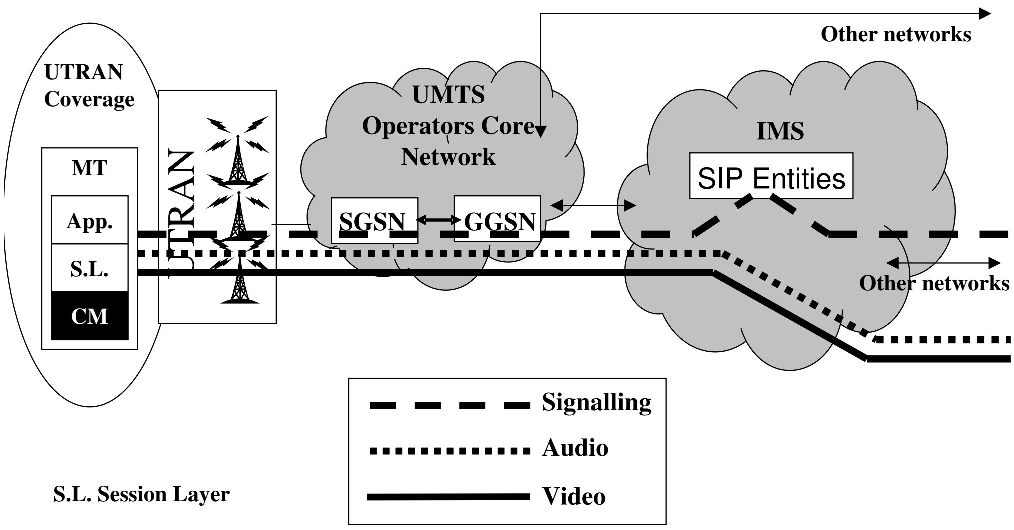


Figure 7.3: Video conference session without WLAN coverage

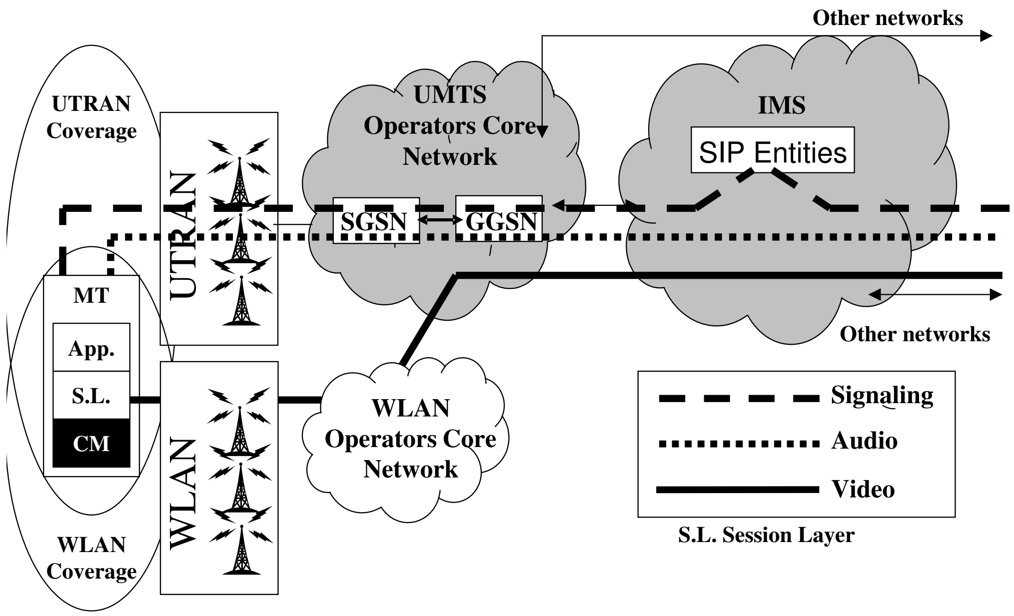


Figure 7.4: Video conference session using WLAN and UTRAN (video via 3G network)

There is no additional standardization effort required for an enhancement or an introduction of protocols for the proposed solution. However, the session layer requires the convergence manager entity and the algorithm in the terminal to enable behavior according to the wishes of the user. The limitation of using the discussed scheduling algorithms come from the session base, not packet base.

There are many open points in this proposal, such as the interacting signals between the session layer and lower layer and the algorithms in convergence manager need to be investigated more. Furthermore it is important to investigate this delay in more detail.

7.2 Standard Switching Implementation for FTP Download

One particular application in the background service class has interesting properties within the convergence manager concept: The FTP application uses the FTP protocol. This application, i.e. the file transfer and file download, has very loose delay requirements but can still utilize a high data rate. This makes it an interesting test case to demonstrate the possible benefits from convergence manager.

7.2.1 FTP Application and Protocol

The FTP protocol (RFC-959, STD-9) offers an important feature compared to any other of the protocols involved: By means of the "RESTART (REST)" command it is possible to start a file transfer at an arbitrary location in the file. This feature is not a required feature in the FTP standard (RFC-959), only optional, but many currently available FTP server programs have implemented this feature. Additionally, it is possible to stop a file transfer while in progress by means of the "ABORT (ABOR)" command. However, pretty much any network protocol inherently offers such a feature and grateful recovery from it, because events such as unplugging a network cable or losing radio contact to the access point will always stop a running connection. The combination of these two available commands means that the aborting of a current file transfer and its resuming is already possible with many off-the-shelf FTP server programs.

7.2.2 Switching Between Standards

An FTP client can implement convergence between multi-standard by switching its FTP connection from one standard to another, thus implementing the switching between standards. It is important to note that this does not require any cooperation in any other part of the network. It only requires the FTP protocol feature "RESTART", which may or may not be implemented by a given FTP server. Apart from this, no further modifications of the involved devices are necessary.

The characteristics of an FTP standards-switching algorithm can be described as:

- Without changes in the network
- Without changes at the FTP server
- Without changes in the radio standards
- Without any cooperation from the telecommunication service providers
- Even without any modifications of available physical layer implementation in a mobile equipment
- The only required modification is inside the client's application-level program

A client FTP application starts an FTP connection by opening a TCP/IP connection over a specific network interface of the client's computer. This network interface corresponds to a specific Communication System Standard, e.g. a 100BaseT-Ethernet, 802.11b WLAN, GSM-Circuit Switched Data (GSM-CSD) etc. The client application can then implement the switching of standards by closing the TCP/IP connection over the old network interface and opening a new TCP/IP connection over the new network interface.

Figure 7.5 shows the additional elements in a FTP session with Switching of Standards enabled. The convergence manager collects information from the network interfaces. It controls both the FTP session and the operating system's routing tables accordingly.

7.2.3 Other Applications

The Switching of Standards works well in the context of an FTP File Download. The prerequisites for this are:

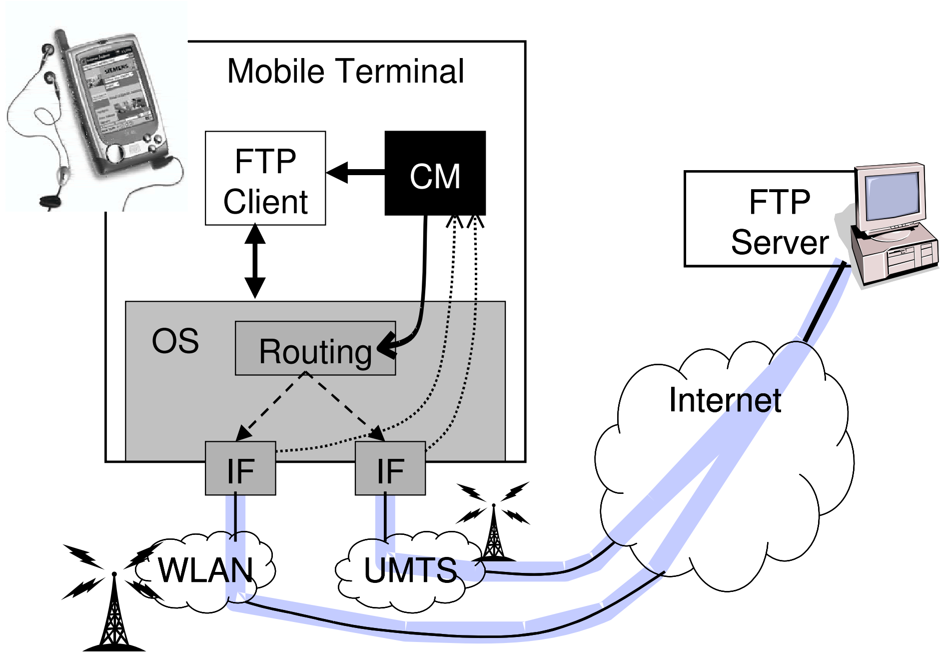


Figure 7.5: FTP session with switching of standards

- The FTP protocol supports required protocol functions, i.e. aborting and restarting a connection.

- The file download application is insensitive to delays and can easily tolerate several seconds of transfer delay.

One may ask whether other applications are suitable for such an implementation of Switching of Standards. In fact, there are many other applications which have only loose requirements for the transfer delay. Potential candidates for similar switching between standards implementations might be:

- E-Mail/ messaging

- Web Browsing

- Video on Demand

7.3 Convergence Manager Performance in Multi-User Scenario

The performance of a convergence manager with different network selection algorithms in a single user scenario has been investigated in the previous chapter section 6.7. The simulation results have proven that integrating a convergence manager in MT will improve the system performance. According to different network selection algorithms in the convergence manager, different granularity of convergence benefits are achieved. It has to be mentioned that those results are based on a single user scenario, where no interferences between users happens. If there are more than one user requiring for transmission at the same time, the network resources shall be divided. While the location of the convergence manager in MT prevents the end users getting information of other users, therefore no adaptive scheduling could be implemented.

Here the same scenario as Figure 3.4 in section 4.5 is continuously used but with multiple users inside the scenario. The performance of the convergence manager with different network selection algorithms will be investigated.

7.3.1 Connection Set Up Procedure

It is assumed that the MT will send connection requests to both UMTS BS and HiperLAN/2 AP if this MT is within the coverage area of both UMTS and HiperLAN/2. Otherwise the MT will send connection requests only to the UMTS BS. A typical connection set up procedure between the MTs and BSs can be described as in Figure 7.6:

The MTs are all located inside the UMTS cell. The MT2 is sending requests to both UMTS BS and HiperLAN/2 AP assuming it is inside the HiperLAN/2 cell while MT1 is sending request only to UMTS BS. At the beginning of each radio frame time, the BS/AP from the network side will read into their request memory and calculate the request number from the MTs and take them as the number of active user for the this radio frame. The radio resource will be divided among the active users in a way of fairly assigning same data rate to every requested user. In the UMTS radio frame the spreading factor is assumed to be 16. Every requested user is assigned one chip code which is equivalent to a data rate of 240 Kbps if there are less than 16 requested users inside the UMTS cell. If the number of requested users is between 16 and 32 then one chip

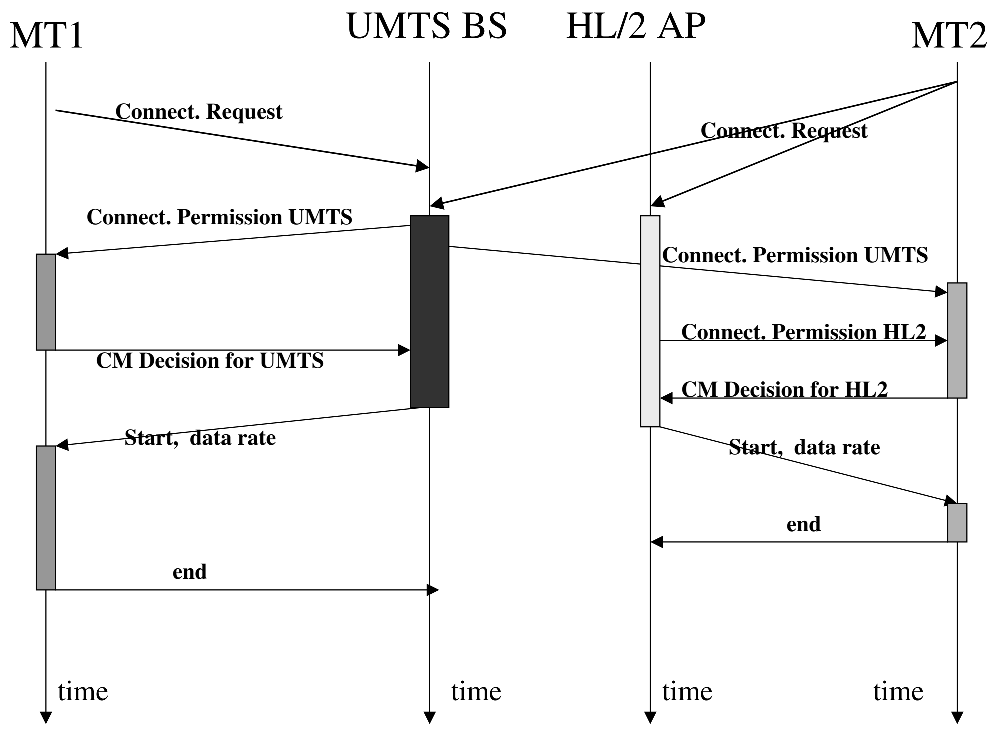


Figure 7.6: Connection set up between MTs and BSs/APs

code is used for two users in time multiplexed fashion so as the effective data rate becomes 120 Kbps.

Similarly if the number of users is between 32 and 64 the effective data rate is then 60 Kbps. If the MT is located inside HiperLAN/2 cell and has a available connection, the HiperLAN/2 AP will assign 6 Mbps to each MT and will transmit them using Round-Robin scheduling. The connection permissions with supported data rate from the available standards are sent back to the MTs by the BS/AP. The convergence manager will then make decision via which standard(s) will the file to be downloaded. After receiving the decision from the convergence manager, the BS/AP will inform the MT to start transmission. The decision of convergence manager will be updated for the next radio frame since the BS/AP are talking with convergence manager at the beginning of each radio frame. The BS/AP will be informed in the end of the transmission and to delete the request from their memory. The transmission is carried out after the decision of convergence manager based on the information exchange between the convergence manager entity inside the MT and the network side. Comparing to the normal connection set up procedure

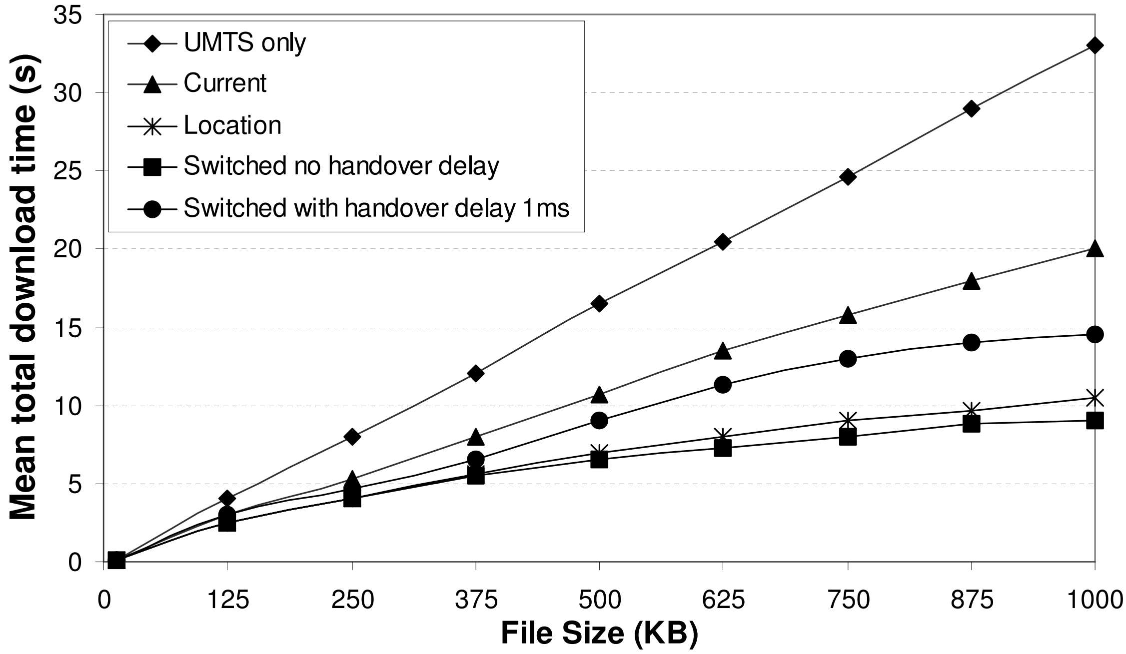


Figure 7.7: Delay performance of convergence manager @MT: Multi-User uniformly distributed

within current mobile communication network, the convergence manager brings changes only at the MT side. There is no change at the network side necessary.

Different network selection algorithms might lead to different decision by the convergence manager. Neither the network BSs nor the MTs can implement any adaptive scheduling due to lack of information. At the same time, the inference between MTs increases along the number of requested users inside the coverage area. The performance of convergence manager in such situation will be limited.

7.3.2 Performance of Uniform Distribution of MTs

The MTs are assumed to be uniformly distributed inside the UMTS cell. The network selection algorithms which have been discussed in Chapter 6 are implemented by the convergence manager.

Figure 7.7 shows the mean total download time versus file size performance, given that there are 16 MTs inside the simulation area.

Comparing with the system performance in single user case as shown in Figure 7.7, a couple of observations can be made:

- UMTS is not able to provide a realistic download time for full range of file sizes. A second standard is necessary to provide realistic service for large sized file download.

- The switched algorithm exhibits the lowest overall delay if no handover delay occurs. The handover delay affects the performance largely, system with $1ms$ additional delay, which is 50% time of the radio frame, has performed similar to the current algorithm. In reality, the handover delays are much smaller than half of the frame length, their performance are believed to be quite close to that without handover delay.

- The current algorithm offers better delay performance than single standard for the entire range of file size. But the benefits is limited due to the regional availability of HiperLAN/2 cell.

- Location algorithm exhibits the delay performance quite close to the switched algorithm over all range of file sizes and it avoids the complexities related to switching mid way of file download as switched algorithm. It is a good compromise between the performance and complexities.

- The overall system performance in multi-user case is worse than that of single user case. The reason for that lies not only on the data rate division among multi-user but also on the interferences between users. With knowing neither which networks to be selected by the other users nor how the network resources are to be used up by multi-users, the convergence manager is not that intelligent, wrong decisions always happen. Anyhow, the convergence manager brings great delay benefits to the multi-users, comparing to single standard. The overall downloading time with the convergence manager using location algorithm is only 30% of that without a convergence manager for large file size.

Figure 7.8 shows the mean discarding rate versus file size performance for different algorithms. It is important to understand what is meant by discarding rate here. The new request of a MT will be discarded before the previous request have been completed. The mean discarding rate is dependent of the mean total download time and the request arrival process. While only those served file downloads contribute in calculating the mean total download time and those discarded

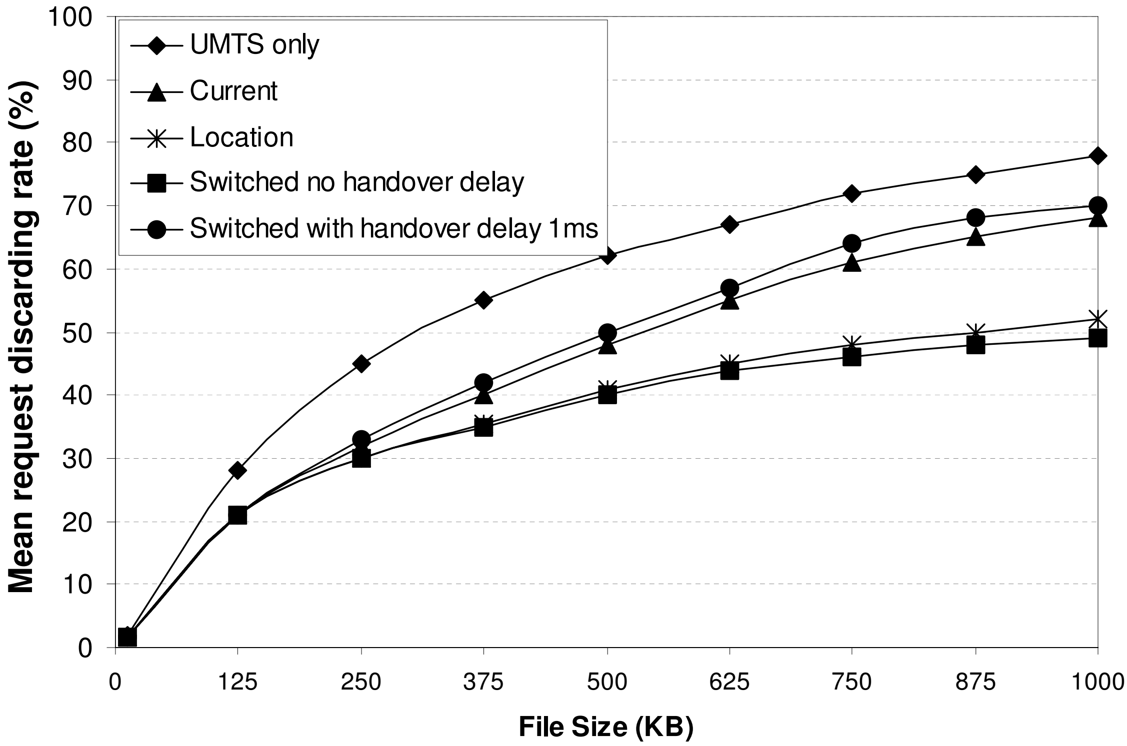


Figure 7.8: Discarding rate performance of convergence manager @MT: Multi-User uniformly distributed

requests do not effect the mean total download time. Therefore it is important to consider the mean discarding rate as well to evaluate the performance of convergence manager.

It is pretty clear that switched algorithm without additional handover delay has the lowest discarding rate because it is the theoretical lower bound as discussed before. The location algorithm exhibits mean discarding rate very close to switched algorithm over entire range of file sizes and it outperforms all other algorithms. The location algorithm is of great advantage because it helps to prevent the overhead problem during handover in current algorithm by using the knowledge of user-mobility and coverage-range-information. The discarding rate with convergence manager could be reduced up to 40% comparing to that of the system without convergence manager for large file size.

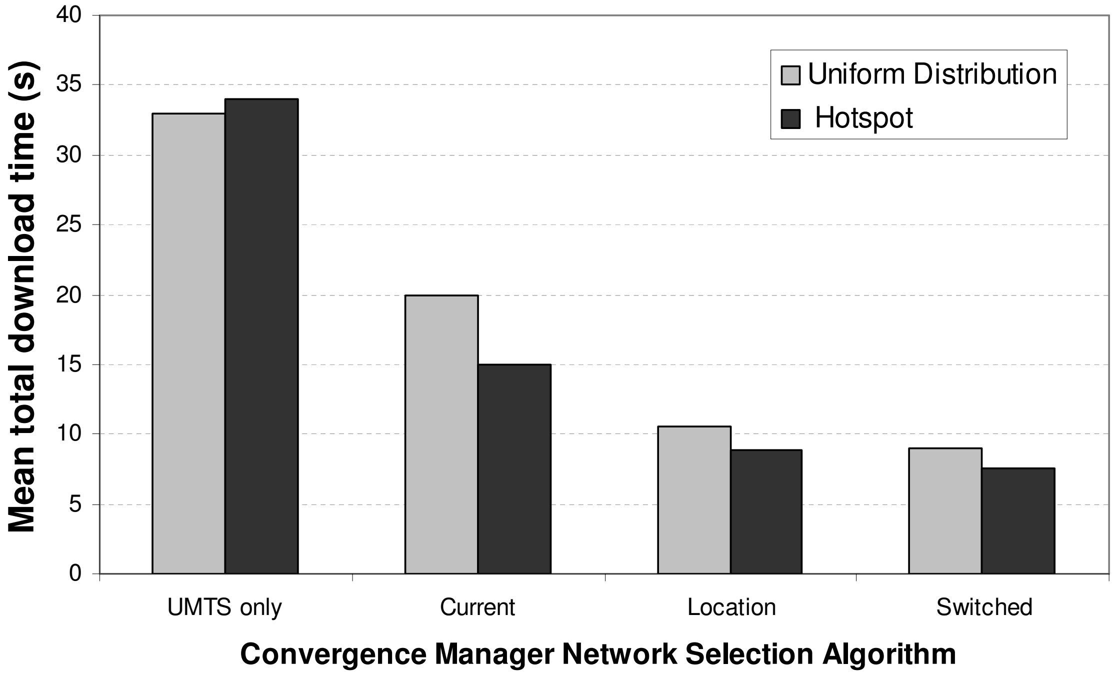


Figure 7.9: Hotspot effect to convergence manager @MT

7.3.3 Hotspot Effect

Another user distribution is considered inside the simulation scenario: a hotspot which means that the user density inside the hotspot area is ten times comparing to the rest area. A clever solution will be to place the HiperLAN/2 coverage over the hotspot area.

Figure 7.9 compares the mean total download time performances of different network selection algorithms between the uniform and hotspot distribution of MTs inside the simulation scenario. The file size is assumed to be 1000 KB and there are 16 MTs inside the scenario. There is almost no change to the system performance with single standard. But the system performance improves for all convergence manager network selection algorithm because that MTs transmit more often through HiperLAN/2 with a larger capacity.

7.3.4 Mobility Effect

User mobility is another parameter influencing the performance. Increasing the moving speed of the end user from 1 m/s to 10 m/s changes the performance of location algorithm greatly in

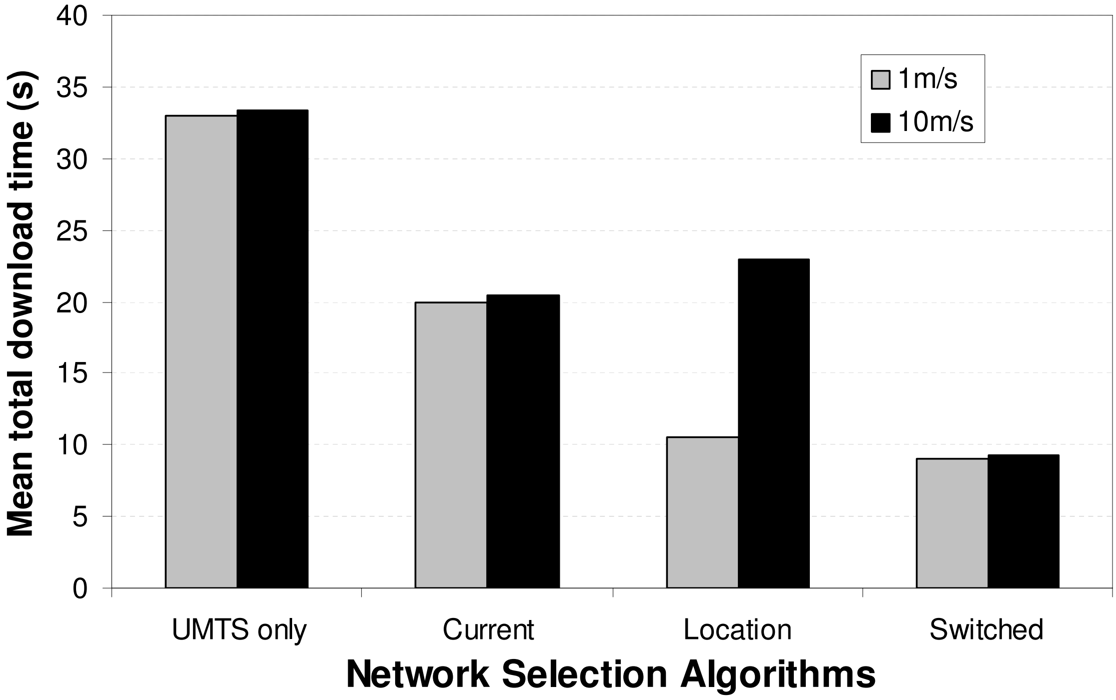


Figure 7.10: Mobility effect to convergence manager @MT

an uniformly scenario (Figure 7.10) given requiring file size 1000 KB. The location algorithm decides more often for UMTS because that the MTs can stay much less time in the HiperLAN/2 coverage. For situation with high mobility, the location algorithm is not any more a good network selection algorithm.

7.3.5 Multi-User Interference Effect

Figure 7.11 shows the mean values for the total file download time for different numbers of users and Figure 7.12 shows the mean discarding rate. The results have been shown for file size of 125 KB.

An important observation is that the performance of location algorithm gets further away from the switched algorithm by increasing the number of users. It indicates that the wrong decision by the convergence manager happened more often using this algorithm than the other convergence network selection algorithms. When user makes a request outside the HiperLAN/2 cell, the convergence manager estimates that whether the download can be completed before leaving the HiperLAN/2 coverage. Location algorithm makes use of the users mobility, UMTS/HiperlAN/2

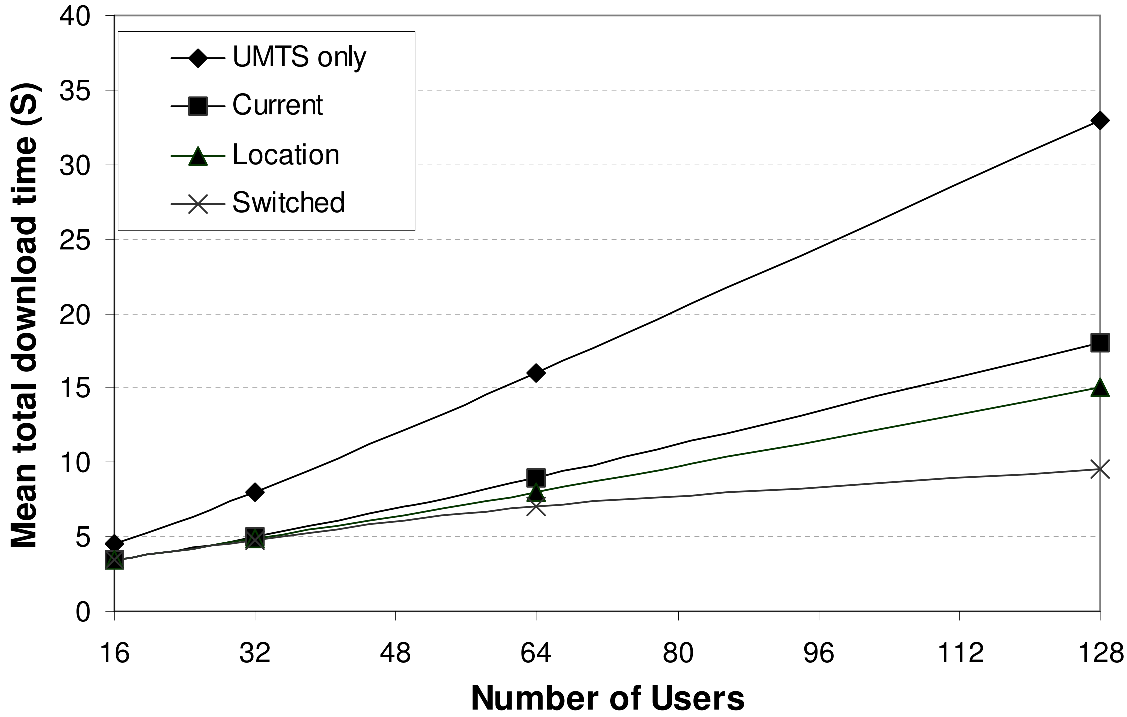


Figure 7.11: Delay performances of convergence manager @MT versus number of users

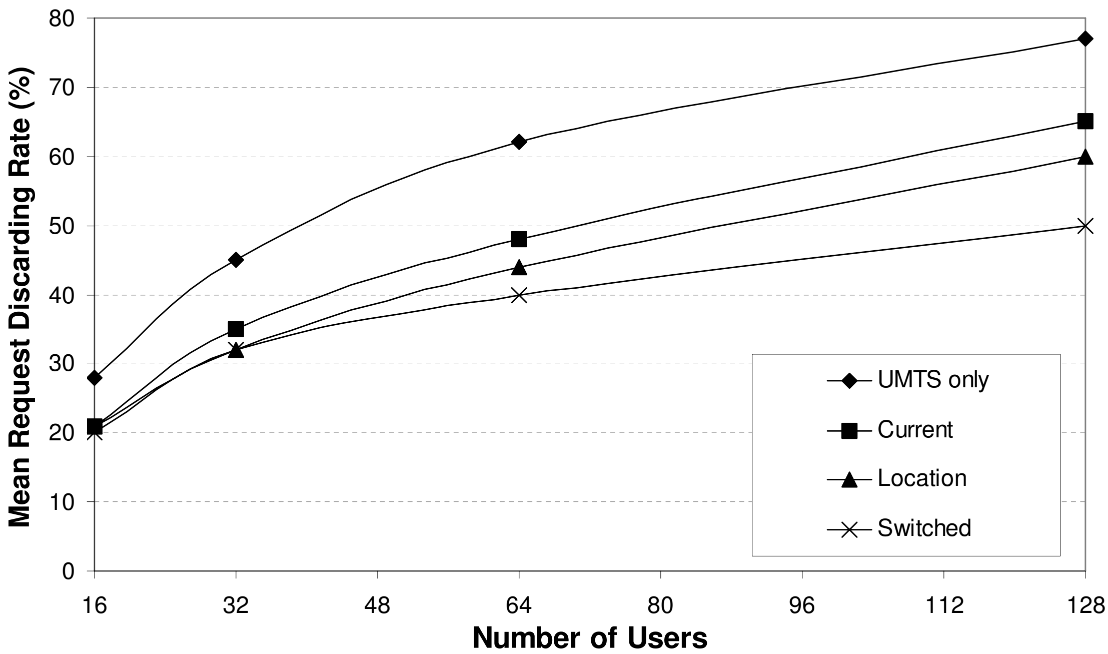


Figure 7.12: Discarding rate performance of convergence manager @MT versus number of users

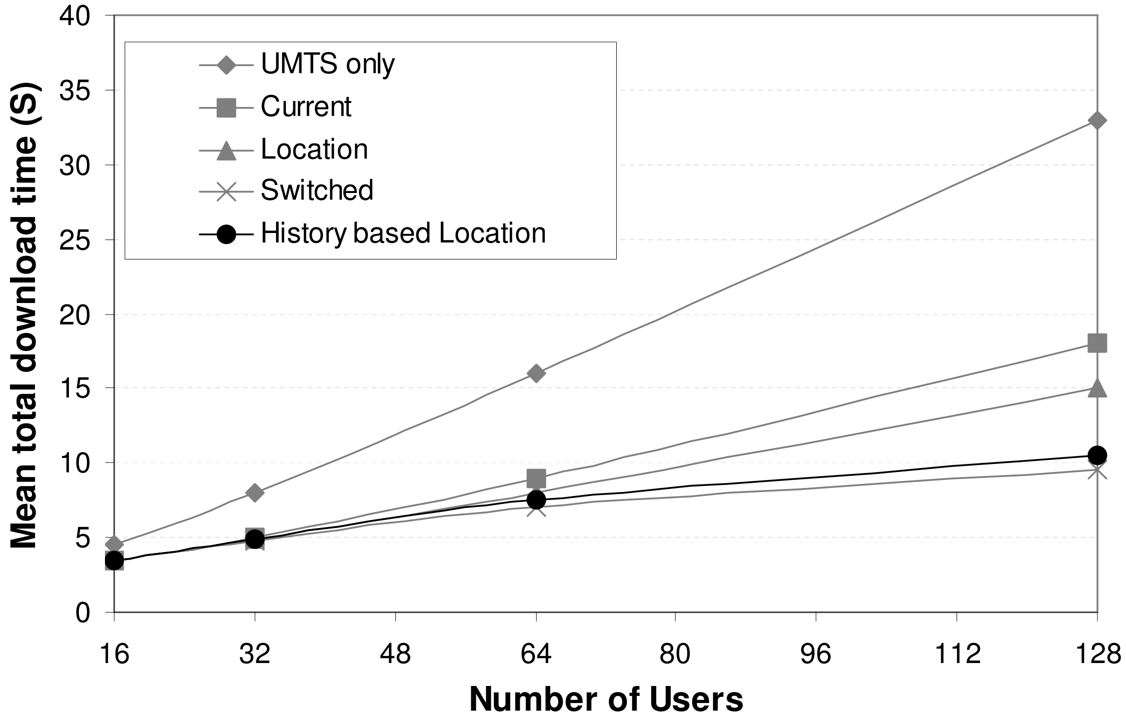


Figure 7.13: Delay performance of history based location algorithm

coverage information and the data rate of available standards in order to make estimation. It is clear that the convergence manager should know the correct data rate from respective standard for a correct estimation. In case of single user all available capacity is given to the single user when he makes request to download a file. It means that the user will be assigned one chip code (240 Kbp) from UMTS or full time slot from HiperLAN/2. It implies the correct estimation at the time when request arrives. But in case of multi-user, the date rate from HiperLAN/2 and UMTS is divided among the requested users. And the data rate changes according to number of requested users. The estimated download time at the moment when request arrives is not any more correct if new request arrives before the download finishes. Wrong decision is made by the convergence manager based on the wrong estimation.

History Based Location Algorithm outperforms the location algorithm in multi-user scenario. The mean data rate during the last radio frame time is used as the predicted mean data rate for the current radio frame. A performance improvement can be seen as high light in both Figure 7.13 and Figure 7.14 with the History Based Location Algorithm. In both figures the performance of the history based location algorithm is getting near to the switched algorithm.

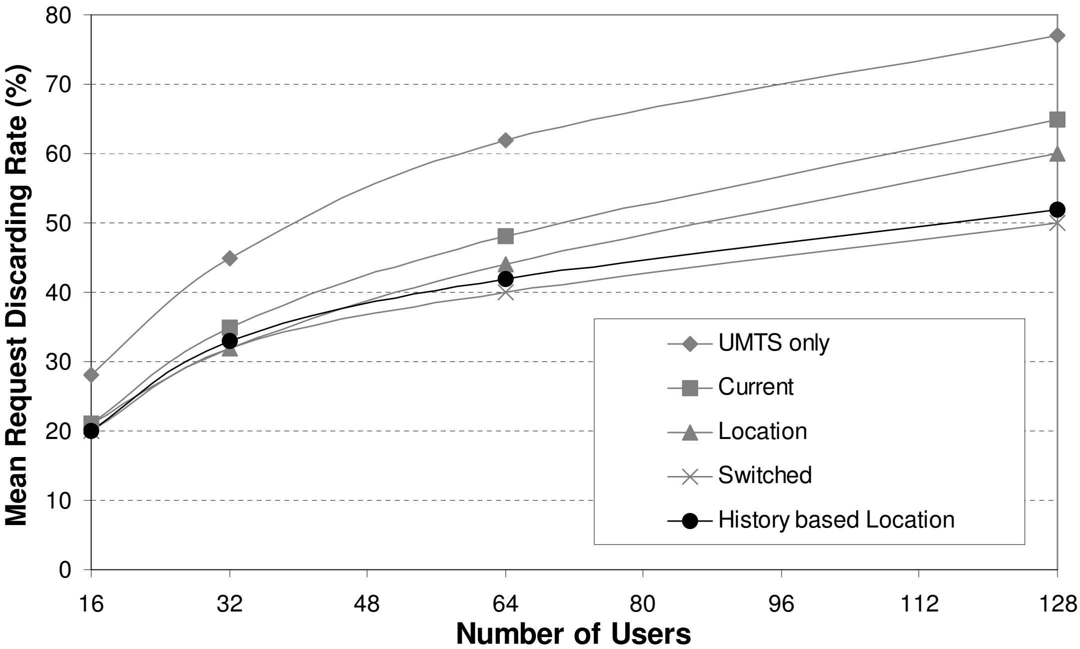


Figure 7.14: Discarding rate performance of history based location algorithms

7.4 Summary of Convergence Manager @MT Performance

The assumption that the convergence manager has the precise knowledge about user mobility and geographical coverage of wireless standards for the location and history based location algorithms may not be true any more in variant situations. The performance will be degraded if there is any error in the information for convergence manager. Even if keeping the assumption, the performance of history based location algorithm still depends on how accurate we can estimate the mean data rate for the current file download. Obviously if there is a rapid changing situations (because of fast moving speed or load fluctuations) the previous radio frames are too old to be used for mean data rate. The standard selection process is likely to be wrong and the performance will be degraded.

Indeed the performance limitation lies on the fact that the convergence manager entity is only located inside the multi-mode MT. Due to the location of the convergence manager, a limited level of information is exchanged between the convergence manager and the BS. Convergence manager can enable mostly the session based standard switching while a true simultaneous transmission on packet base is not suitable. Without the accurate instantaneous data rate, convergence

CM algorithm	Performance	Complexity	Limitation
Switched	Very good	High	Handover delay
Current	Good	Low	Wrong selection
Location	Good	High	Knowledge of mobility and geographic, accurate transmission data rate
History based Location	Very good	High	Memory of history, accurate transmission data rate
Parallel Networks	Best	Very high	Not suitable for convergence manager @MT

Table 7.1: Summary of the network selection algorithms of convergence manager @MT

manager performances is limited. The evaluations of different algorithms that could be used by the convergence manager are listed in Table 7.1 .

Chapter 8

Convergence Manager @Network

The location of convergence manger impacts the level of flexibility and optimization. In the previous chapter, the convergence manager is only located @MT. Neither the radio access networks nor the core networks has any change. The function of the convergence manager is limited with network selection algorithms and no adaptive scheduling could be carried. In spite of the limited level of flexibility and optimization, the convergence manager still brings obvious performance improvements into the multi-standard system.

The consideration to locate the convergence manager on the network side is much more complicated but with great interest. The convergence manager can be implemented at the network side (@NET), either in BS (@BS) or CN (@CN).

The basic assumption as mentioned in section 3.4 is followed here. It is assumed that there is a pair of convergence manager entities, one inside the multi-mode terminal and one on the networks for multiple standards. The convergence manager @NET is the master entity who monitors and controls the transmission inside networks while the convergence manager @MT is a passive entity taking charge of the cooperation with the master convergence manager @NET.

Several concepts for the convergence manager @NET are going to be identified. As potential hosting entities, the BSs, CNs, backbone networks and server entities are all possible, each including significant advantages and drawbacks. To make the resulting architecture applicable to real world networks, it is furthermore of importance to consider also the migration path from existing standards towards a convergence manager enabled architecture. There are following considerations related to this location:

- The convergence manager can be a functionality located in the BS (@BS), e.g. BS of UMTS network or AP of HiperLAN/2, with access to information not only from different user service requirements but also from the network and radio air interface side. Currently the BSs can support only single standard. But the co-location of the standards enables the fast communication between BSs or APs from different networks. It makes sense to implement a copy of the convergence manager inside the co-located BSs and APs. If in the future the antenna technique will support the different frequency bands within single BS, the convergence manager is to be located inside the multi-standard BS.

- The convergence manager may be located deeper inside the radio access network [51], e.g. the Base Station Controller (BSC) of GSM network or the Radio Network Controller (RNC) of UMTS network, so that the mapping of services to standards can only be changed slowly relative to changes in the radio channel. In this case, as a simplification, the mapping of services to standards is considered to be static for a given user (or at least static for a substantial period). The common situation will be to split the media streams onto different RAdio Bearers (RABs), which have different characteristics e.g. in terms of QoS, bandwidth, etc. UMTS already does this RAB selection related to different traffic classes. The convergence manager would be an integral part of the BSC/RNC. The convergence manager on a media stream level would function similar to the RAB selection functionality in UMTS RNC.

- The convergence manager may be located inside the core network (@CN) components, e.g. SGSN,GGSN. Since theses core network components may have inferences to different networks, such as GSM and UMTS core network, the data flows may be routed to these networks. Interfaces can be set up to interface other standards from the core network internally. In case of GSM and UMTS, the access networks may be connected to the same SGSN and splitting of media streams onto the access networks may be performed there.

8.1 Convergence Manager @BS

If the convergence manager is located @BS (Figure 8.1), the MTs can access different standards simultaneously with the following possible optimizations:

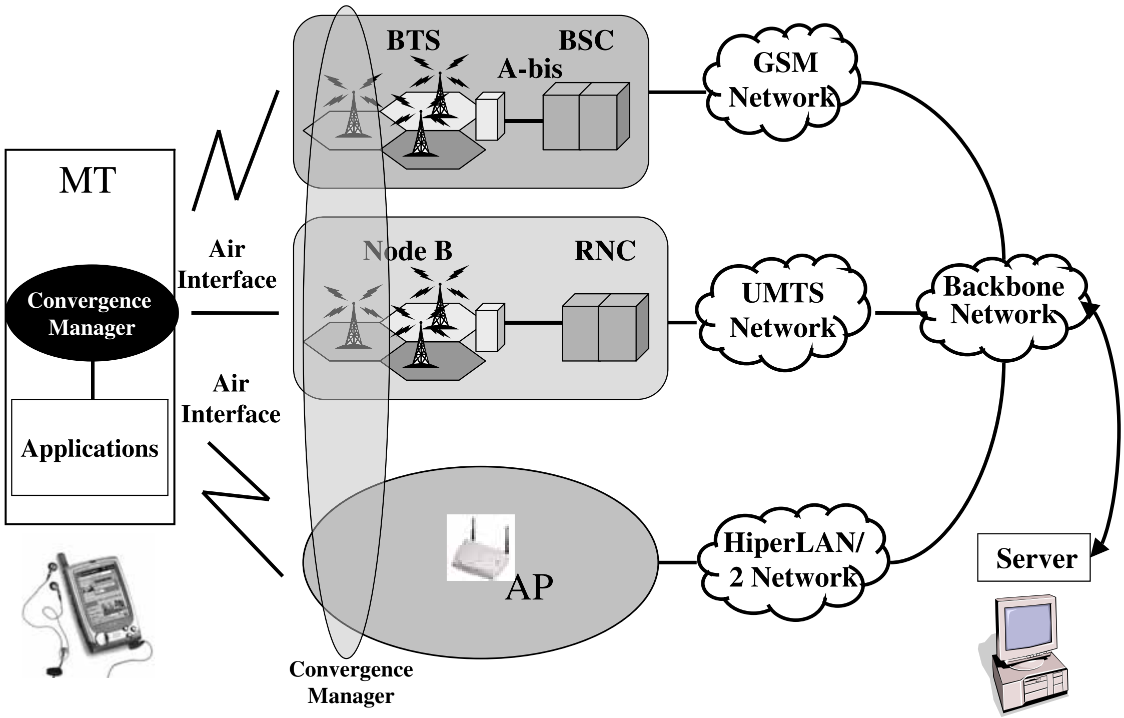


Figure 8.1: The convergence manager is located @BS

- Can optimize radio interface technology choice
- Can optimize network route
- Can optimize PHY/AMC modes

The location of the convergence manager offers the opportunity to collecting much more information from the network side. Based on those information the convergence manager can achieve a "global view" within the networks. More flexibility and optimization are brought into the multi-standard system, since the convergence manager @BS can apply both network selection algorithms and adaptive scheduling algorithms.

Cross layer optimization is enabled because the convergence manager @BS has the direct interface to the radio channels (Figure 5.1 in section 5.4). It receives the information from both upper layer, given as traffic signalling and possibly lower layers in the form of radio access technology signalling. It can be expected that in this case a cross design and optimization of both layers with a larger degree of interaction will lead to a considerable capacity enhancement of the overall system. The detailed information of the source and radio channel is extremely important for the convergence manager.

8.2 Performance of Convergence Manager @BS

Different combination of network selection algorithms and different adaptive scheduling algorithms as discuss in chapter 6 and Chapter 5 could be used by the convergence manager, thanking for the high level of flexibility accessing information from the location @BS. It is therefore the strategy of the convergence manager to keep the connection as long as possible for every single user.

The performance of the convergence manager is investigated by simulation. Quantitative results are shown in the following sections. Several parameters are used for the performance curves. A served user means that if all packets inside the user data queue are transmitted during a given time slot. For example if every 40 ms the users will receive new traffic packets into their queues, the user is marked as served given that all the packets from previous 40 ms are sent out. Otherwise the rest packets will be discarded and cleaned out of the queue of that user. Served number of user is a good parameter to evaluate the benefits to both network capacity for network operator and user satisfaction for end user. Mean discarding rate is almost an invert parameter to the served user number, it is more to the interest of end users. Beside that mean transmission delay is used as parameters which will depend on the number of transmitted packets and the actual transmission data rate. Obviously if following the 40 ms update, the mean transmission delay will be always smaller than 40 ms. This parameter is important for those services with high delay-oriented quality requirements. The allocation of radio resource among different networks by convergence manager is shown by parameter system throughput, which is of great importance for network operators to evaluate the benefits of convergence manager.

8.2.1 Single Standard (UMTS) with Uniform User Distribution

The MTs are uniformly distributed within the scenario, there is only UMTS available. Figure 8.2 shows how the system capacity comes along with the increasing number of users. It is assumed that the bandwidth is equally divided among the MTs.

If all users are using VoIP as traffic model, a single standard UMTS has sufficient capacity to transmit the data packets from all the users, which is shown in the figure that the served number of users are the same as the number of users inside the cell. The system is not requiring for an additional standard for these "light" services. But if everybody is trying to watch the video stream, the UMTS system capacity reaches its limitation very fast, maximum 11 MTs can be

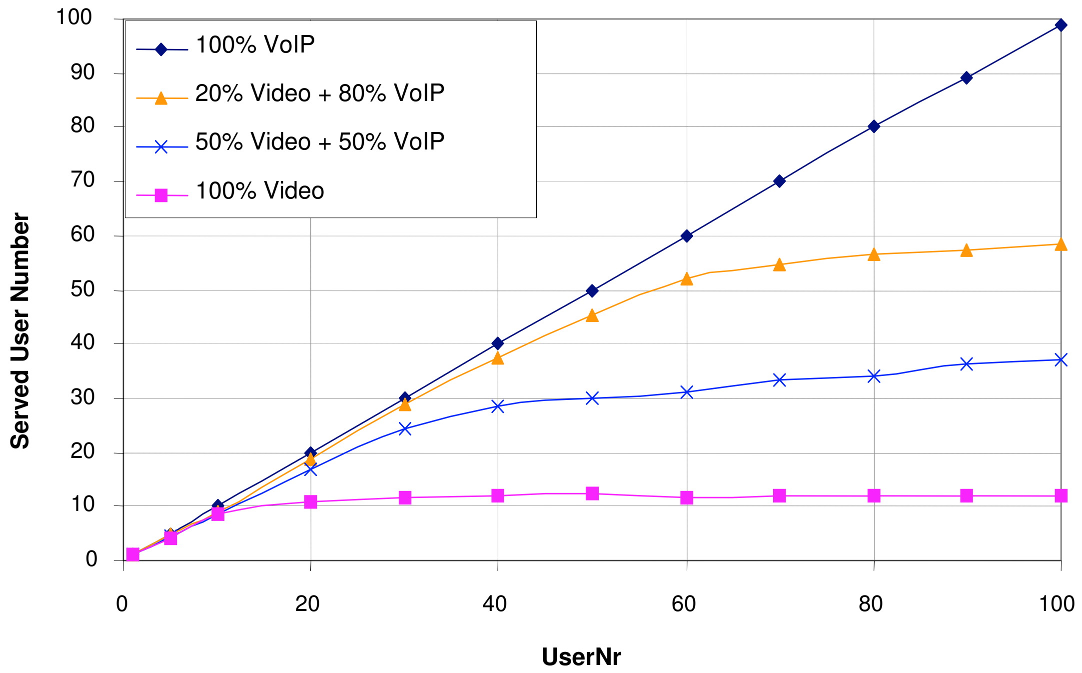


Figure 8.2: Performance of UMTS uniform cell with different service components

transmitted in the given interval time. Additional bandwidth is necessary to support the services in a realistic way.

Figure 8.3 takes a look at the situation when 50% of the MTs are using VoIP and 50% using video stream. The served user number with video stream loses its linearity very early and comes to a saturation of max. 11 users rapidly. Since the service which requires high data rate will need very long time to finish their transmission, and at the same time, the service requires small date rate will have less chance to transmit. The served number of users comes to a saturation much earlier if more services require high data rate. It is necessary to somehow optimize the allocation of the network resource between different users if the network operator wants to serve more users. The service quality for the end users will be improved at the same time if they need less time to finish their application and the connection will be less likely dropped out. A convergence manager is necessary even within a single standard, although the function will be adaptive scheduling instead of converge different networks based on different standards.

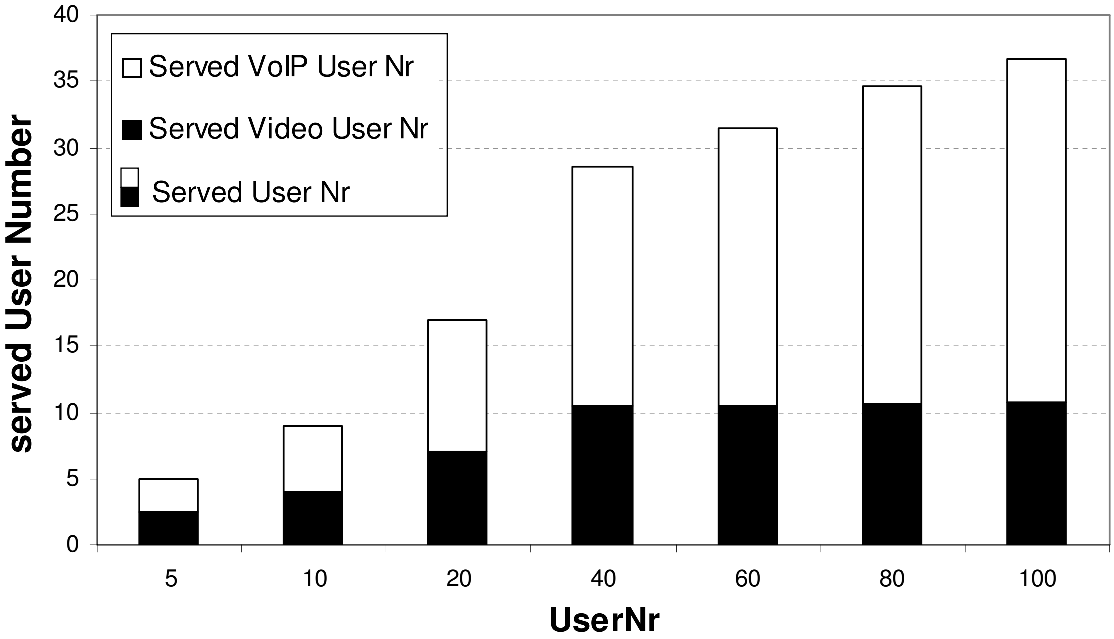


Figure 8.3: Single standard performance for different services

8.2.2 HiperLAN/2 and UMTS with Uniform User Distribution

Single standard as UMTS does not have sufficient capacity for users with services requiring high transmission data rate. The simulation scenario used is based on the "UMTS Coverage with HiperlAN/2 hotspot " as Figure 3.4 in section 4.5.

The users inside the HiperLAN/2 cell will have the possibility to use both standards. For them it is absolutely necessary to have convergence manager to manage the simultaneous use of UMTS and HiperLAN/2. It is expected that the convergence manager(s) are going to bring benefits for not only the users inside HiperLAN/2 cell but also those outside the HiperLAN/2 cell but inside the UMTS coverage. The switched network selection algorithm will be used by the convergence manager which means that the MTs could be transmitted through HiperLAN/2 once they enter the HiperLAN/2 cell. No adaptive scheduling algorithms are used by the convergence yet, the capacity of HiperLAN/2 and UMTS will be divided equally among the users who have access to them, respectively.

The position of the HiperLAN/2 cell in this situation will not affect the system performance very much, while the system performance depends more significantly on the size of the HiperLAN/2 cell.

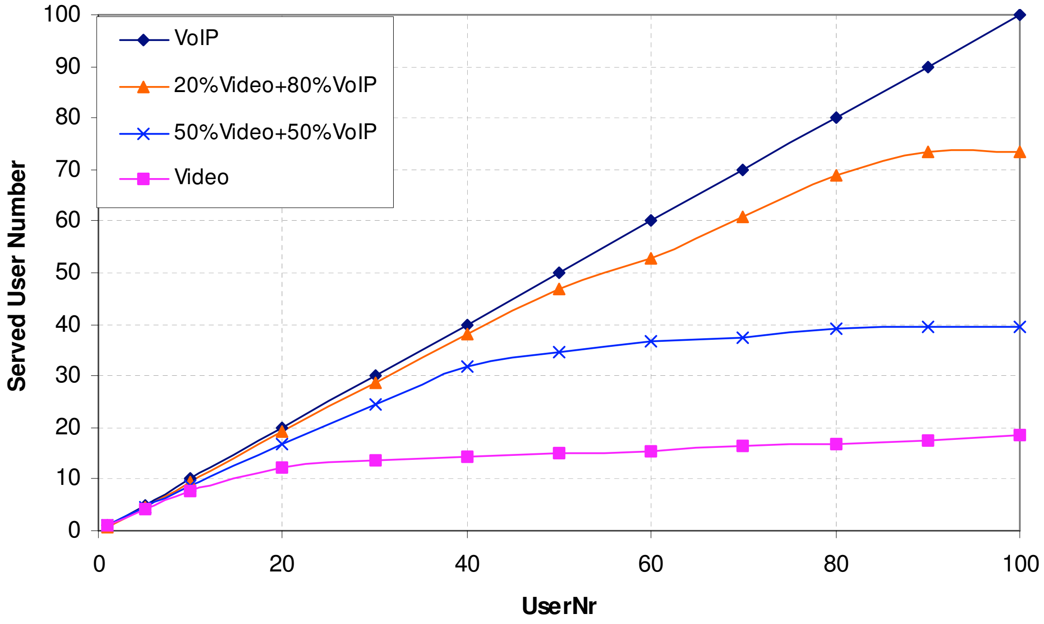


Figure 8.4: Performance of different service components with two standards

Figure 8.4 illustrates the system performance when HiperLAN/2 cell is 10% of the UMTS cell size. The number of the served users are slightly increased.

When comparing Figure 8.4 with Figure 8.2, it can be found that the served number of user in the scenario with an additional standard HiperLAN/2 is not improved that much compared to when there is only one standard UMTS. It is interesting to take a further look at the system performance with service component of 50% VoIP and 50% video stream. Figure 8.5 makes a capacity comparison between the single standard and convergence between two standards. The convergence manager enables to add the capacity of HiperLAN/2 into the system, which brings great benefits to the users with high data rate demanding services. The served number of users requiring video stream is increased by 50% comparing with single standard case. The overall served user number is therefore improved although for users requiring VoIP service there is almost no improvement.

Throughput is another important parameter to illustrate the system capacity. As shown in Figure 8.6, the system throughput with two standards equals to the sum from both UMTS and

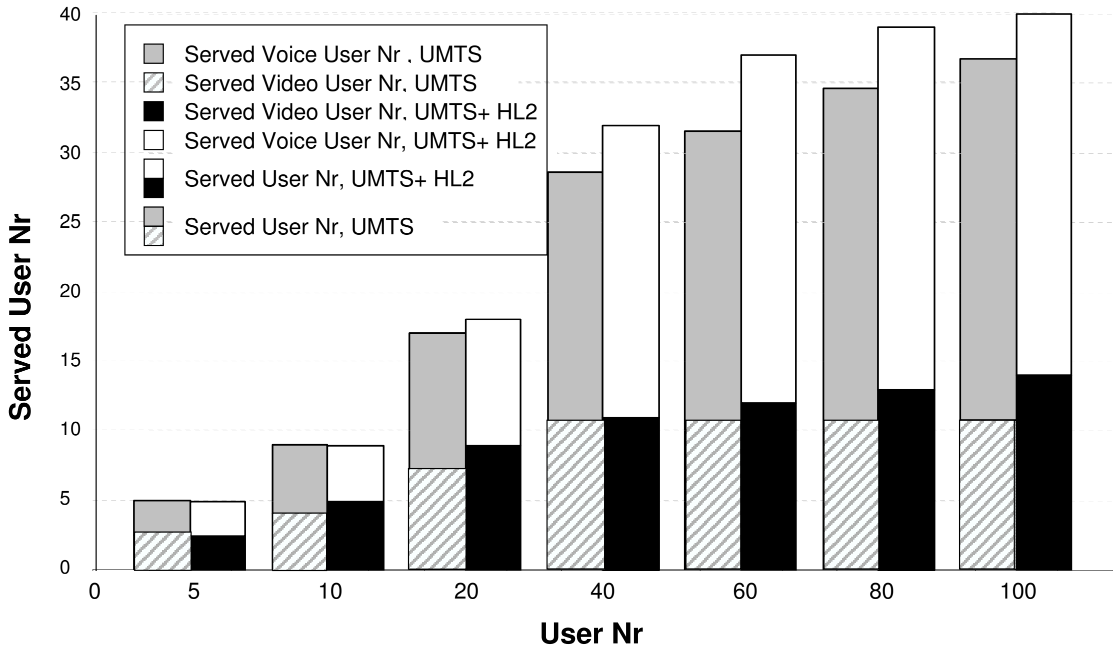


Figure 8.5: Convergence benefits of two standards to different services

HiperLAN/2. The implementation of convergence manager @BS enables the improved source allocation among multi-standard networks. The system capacity improvement shall thank to the convergence manager, who allocates the services requiring high transmission data rate to the network with high capacity and at the same time, saves the capacity in the network with less capacity for the users without any other access possibility. The convergence manager uses the switched algorithm trying to allocate as many traffics as possible to HiperLAN/2. If increasing the size of the HiperLAN/2 cell, the system performance will be increased too. The benefit of the convergence manager increases if there is more users inside the scenario. For situation of overload inside networks, it is obviously necessary to integrate a convergence manager.

In Figure 8.6 the benefits of the different scheduling algorithms in the convergence manager is illustrated as well. The location of convergence manager @BS enables cross layer adaptive scheduling inside every single network. The STT scheduling algorithm brings almost 100% throughput improvement compare to that with a RR algorithm.

It can be observed that the average throughput of the HiperLAN/2 is far from its offerable data rate 54 Mbps. The limitation of the performance inside HiperLAN/2 is due to the the uniform distribution of the users: only part of users are located inside the HiperLAN/2 coverage. The HiperLAN/2 network is not fully used in such kind of scenario.

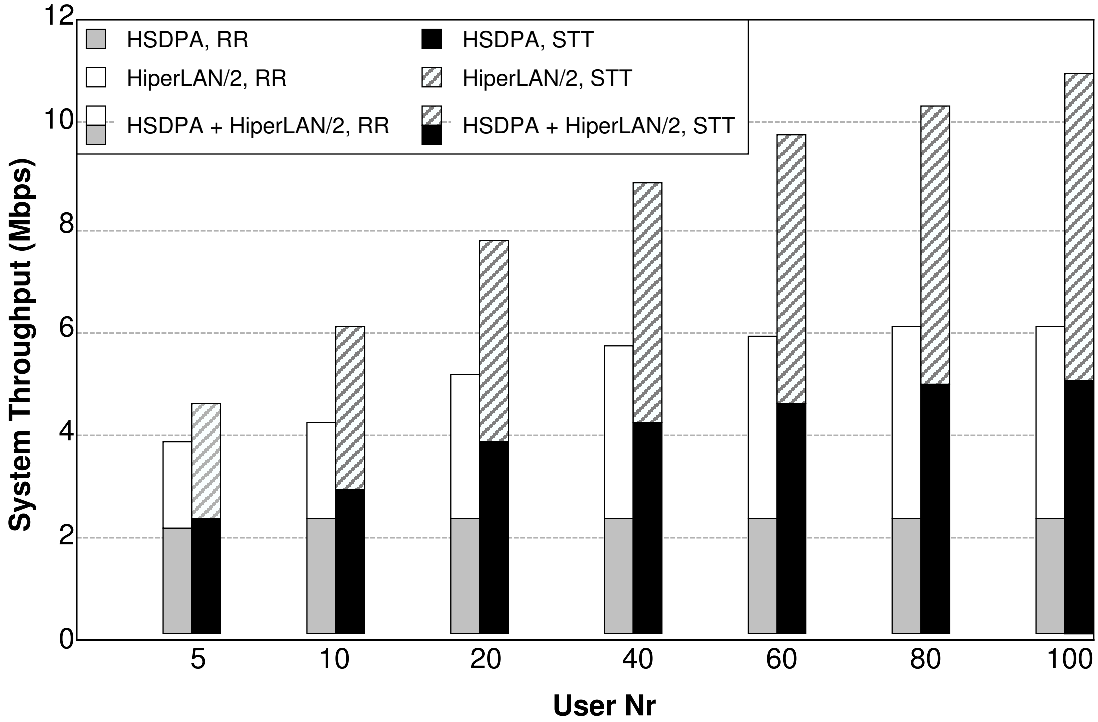


Figure 8.6: System throughput performance with convergence manager @BS

It has to be mentioned that the performance could be very different with different network selection algorithm in the convergence manager. If using service based network selection algorithm in the convergence manager, which means that the users with video stream service via HiperLAN/2 while users with VoIP via UMTS regardless of their location, the system performance is even worse than that of single standard. Those network selection algorithms as a static mapping between services and standards are not proper for time variant situation.

8.2.3 Single Standard (UMTS) with Hotspot

In the real life, it is not that often to find the uniformly distributed users inside the coverage area. On the contrary, it happens quite often that there is an area with more users than the other area, namely hotspot. It is not necessary to limit the definition of hotspot to only geographic distribution. A group users with intensive service requirements can be considered as a hotspot as well. Examples in our daily life for a hotspot can be that, in a football station many football fans are trying to share their joy with more friends on the other side of the telephone connection, or in the airport more than 200 passengers are trying to spend their time waiting for their airplane.

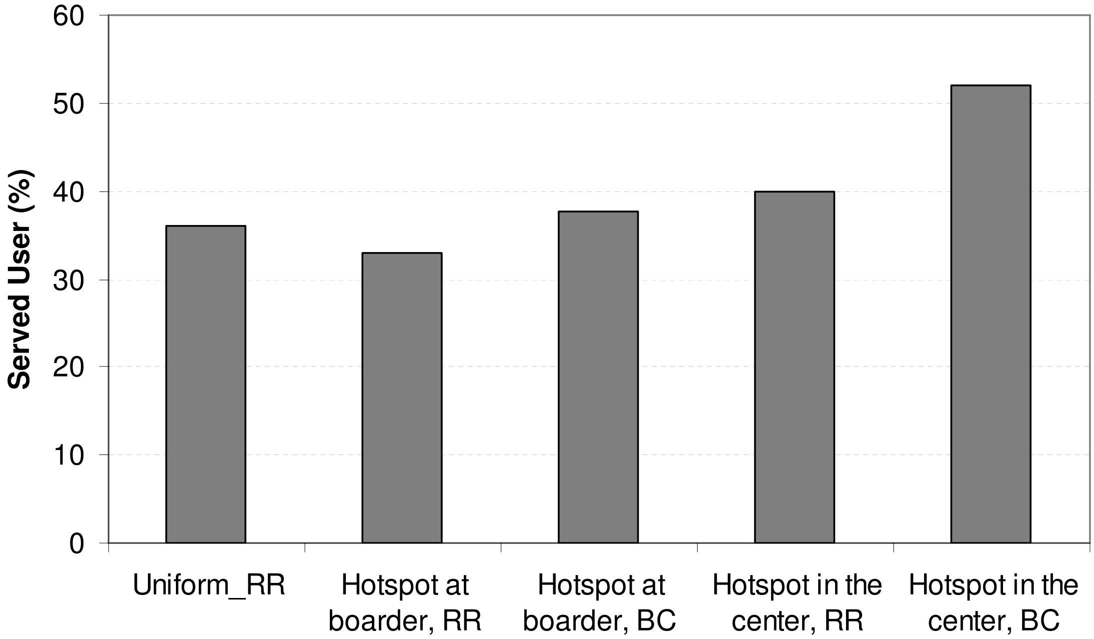


Figure 8.7: Hotspot position affects the performance with single standard

It is quite often nowadays in the GSM network that the connection is rejected or broken after suffering from bad connection quality inside the hotspot. Both the user and network operator have to suffer from the limited network capacity.

It can be observed in Figure 8.7 that system performance is related to the position of the hotspot. A system that more users are far away from the basestation has in average worse performance than if more users are near to the base station. 50% of the 100 users inside the scenario is requiring for voice service while the other 50% for video stream service. The system performance with the hotspot located at the boarder of the UMTS cell is even worse than that of the uniform user distribution. The channel oriented scheduling algorithm of convergence manager is suitable for such kind of scenario, the system performance is compensated because of a more efficient allocation of the source traffic to the channel capacity. The MT in favorable position is assigned higher priority for transmission and the system throughput performance is then maximized at the same time. The area near to the BS is in average with good channel quality, and the high density of users in this area increases the system performance. However still a high amount of users are not served due to the single system capacity limitation. Single standard is insufficient for hotspot scenario. Making use of convergence between co-located multi-standard is good solution in order to improve the system performance.

8.2.4 HiperLAN/2 Hotspot in UMTS Cell

As seen in section 8.2.2, using multiple standards simultaneously will generally improve the system performance. A single standard as UMTS is not sufficient for scenario when most of the users are located in unfavorable position. One solution of a network planner will be to allocate a HiperLAN/2 AP inside the hotspot, which enables that the MTs inside the hotspot may have access to both UMTS and HiperLAN/2. The system performance is improved with a convergence manager using different network selection and adaptive scheduling algorithms to fulfill variant QoS requirement from different user services.

Since HiperLAN/2 (up to 54 Mbps) has much higher system capacity compare to UMTS (up to 10 Mbps using HSDPA) the convergence manager prefers HiperLAN/2 when selecting networks.

Figure 8.8 illustrates the system performance when the hotspot is located at the border of the cell. It can be considered as the worst case of the system performance. The system capacity performance is greatly improved by the convergence between UMTS and HiperLAN/2 inside the scenario. The system shows sufficient capacity even if there is large number of users inside the hotspot. The convergence manager using adaptive scheduling algorithms STT and switched network selection algorithms increases 60% served user number comparing to single standard given 100 users inside the scenario. The gain is even going to be larger if increase the user number.

The gain of convergence manager is even larger when the users are requiring for higher data rate. It is assumed in Figure 8.9 that 50% of the users inside the scenario is requiring video stream instead of only 20% as in Figure 8.8. In Figure 8.9 the served user number is doubled comparing to single standard case, given 100 users inside the scenario.

More service components with higher data rate requirement and increasing user number will bring more traffic load to the system, pushes the system capacity reaching the capacity limitation. For the situation of networks with heavy loads or when the network is overloaded, it is a good solution to implement the convergence manager @BS. Clever combination of the network selection and adaptive scheduling algorithms in the convergence manager is important in order to achieve the maximized convergence benefits between multiple standards.

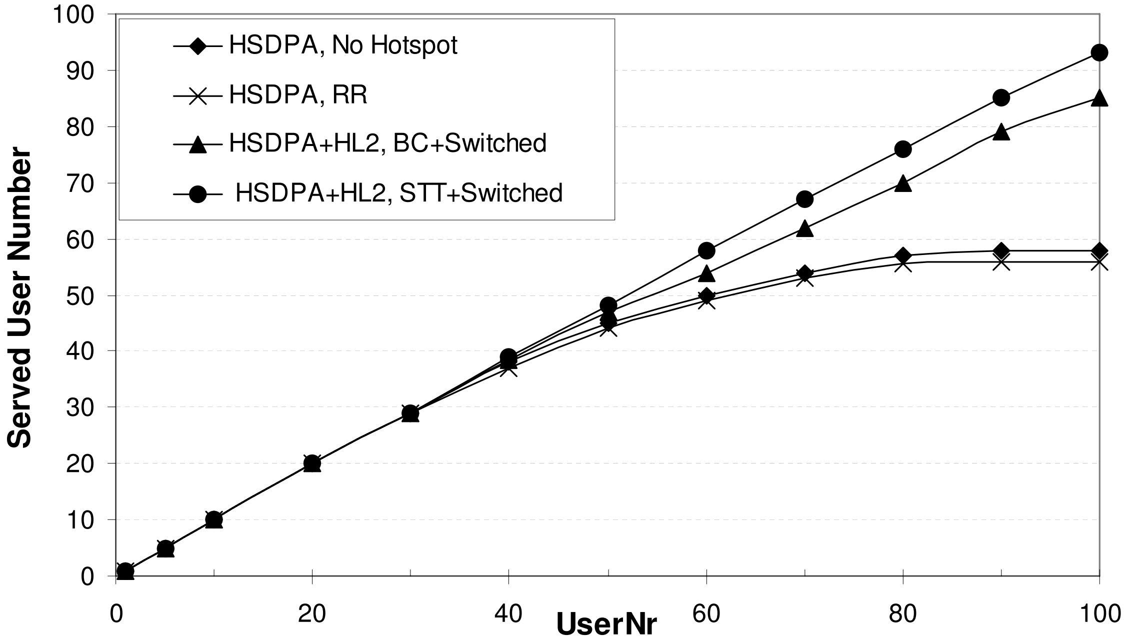


Figure 8.8: Convergence manager @BS performance, 20% video and 80% VoIP

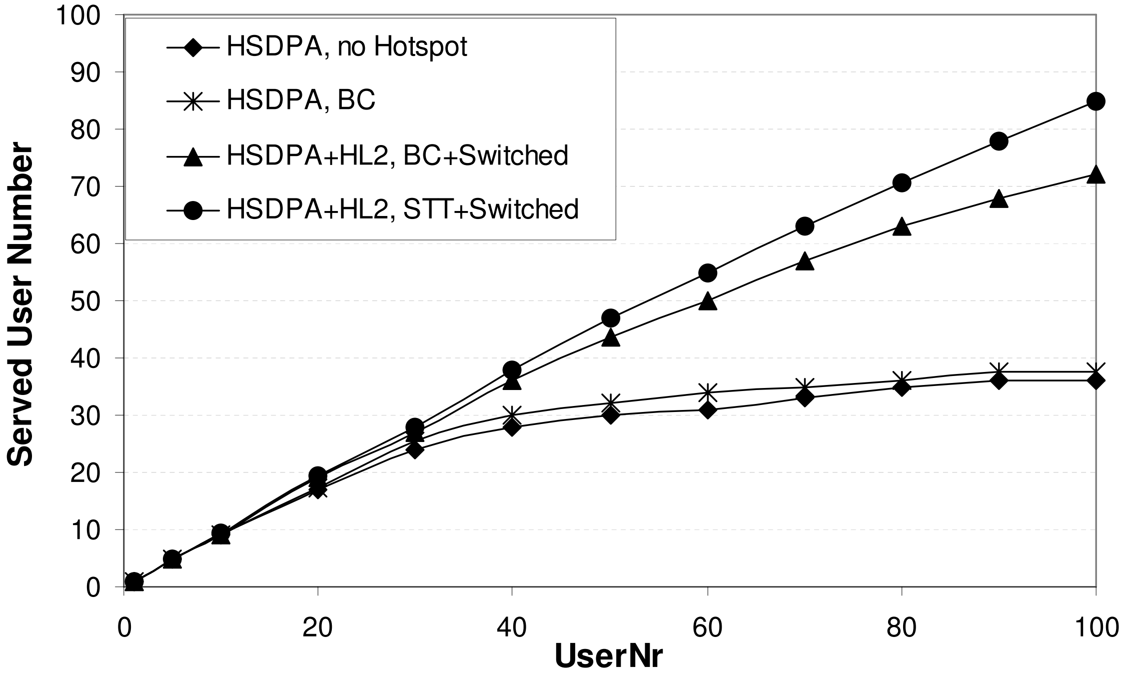


Figure 8.9: Convergence manager @BS performance, 50% video and 50% VoIP

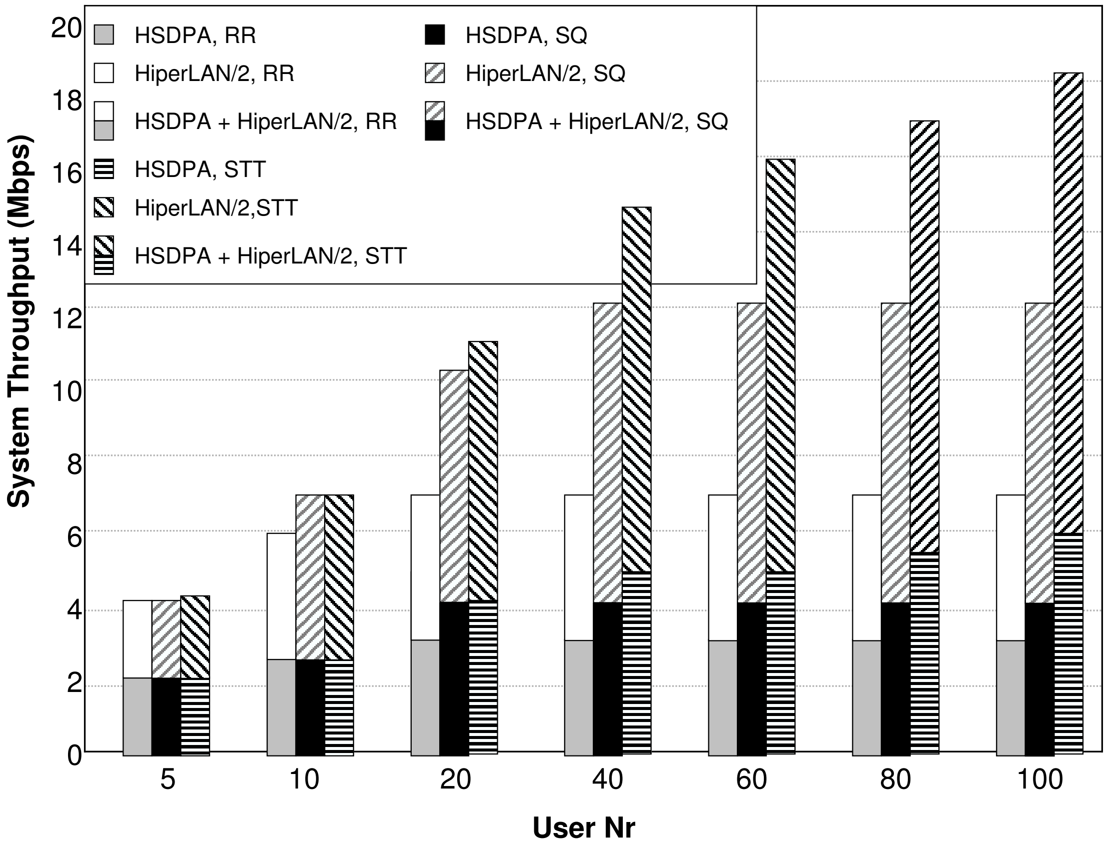


Figure 8.10: System throughput performance of convergence manager in UMTS cell with HiperLAN/2 hotspot

Figure 8.6 illustrated the throughput performance of convergence manager when the users are uniformly distributed. The results shown in Figure 8.10 are carried out within the HiperLAN/2 hotspot scenario. Similarly the switched algorithm is used by the convergence manager considering the higher preference of HiperLAN/2. The best performance is given by the convergence manager using cross layer link adaptation scheduling algorithm. Queue oriented scheduling algorithm SQ in convergence manager performances anyhow better than RR. The system average throughput is improved by 200% using STT comparing to RR. It has to be mentioned that the STT algorithm is requiring convergence manager to be updated along the time variant channel information while SQ is focusing only on the queue length of MTs. Better performance indicates at the same time higher complicity.

Delay performance could be used as a parameter for end user to evaluate the benefits from convergence manager. The average transmission delay in Figure 8.11 and segment discarding rate

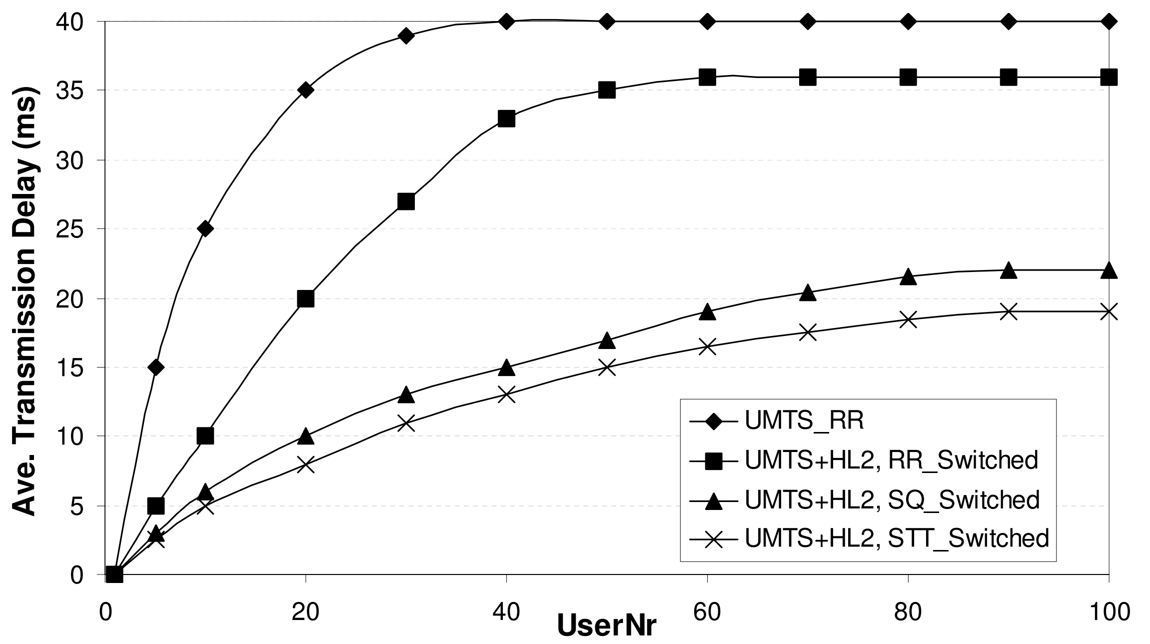


Figure 8.11: Average transmission delay performance of convergence manager @BS in UMTS cell with HiperLAN/2 hotspot

for the end users in Figure 8.12 prove that the convergence manager can improve the system performance by 50% in the same time as the network operators. According to the simulation setting that every 40 ms the end user will receive new packet segment, the average transmission delay is then limited up to 40 ms as shown in Figure 8.11. Therefore, the average delay performance and the discarding rate performance give similar curves with the same scheduling algorithm and network selection algorithm in the convergence manager.

8.3 Convergence Manager @CN

The convergence manager may be located @CN too. In this scenario the splitting of traffic may be carried out in the core network, e.g. in SGSN, GGSN (Figure 8.13). Since these entities may interface at least the GSM and UMTS network, the data flows may be routed to these standards. Interfaces can be set up to interface other standards from the core network internally. The possibility to implement this depends on how tight the access networks interoperate. In the case of GSM and UMTS, the access networks may be connected to the same SGSN, and spitting of media streams onto the access networks may be performed there.

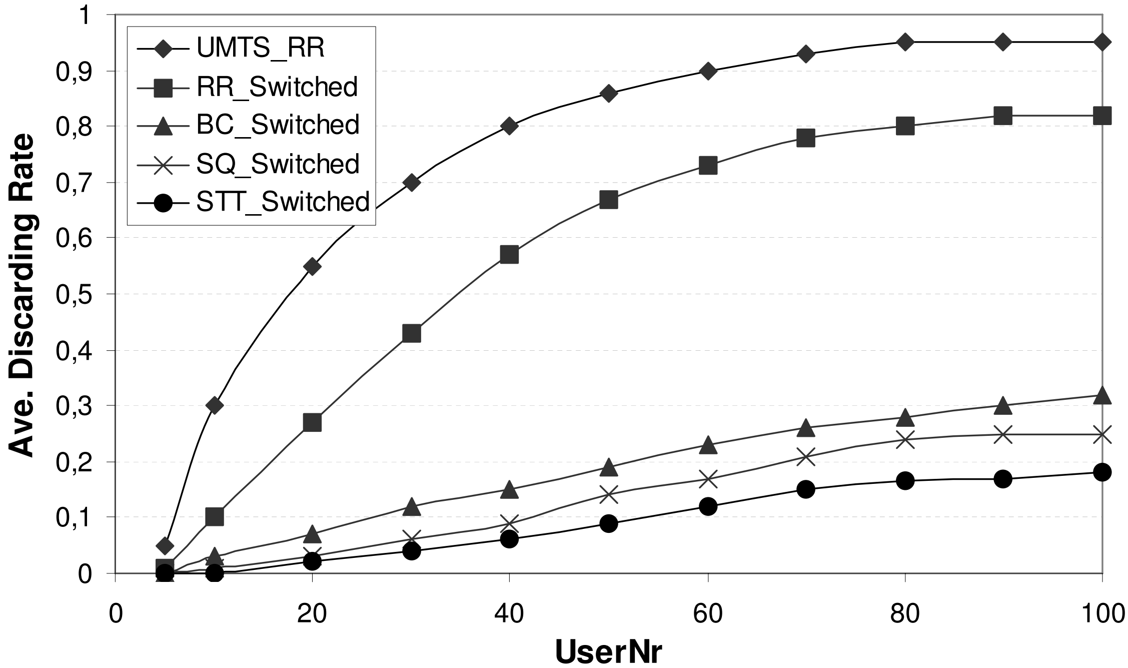


Figure 8.12: Discarding rate performance of convergence manager @BS in UMTS cell with HiperLAN/2 hotspot

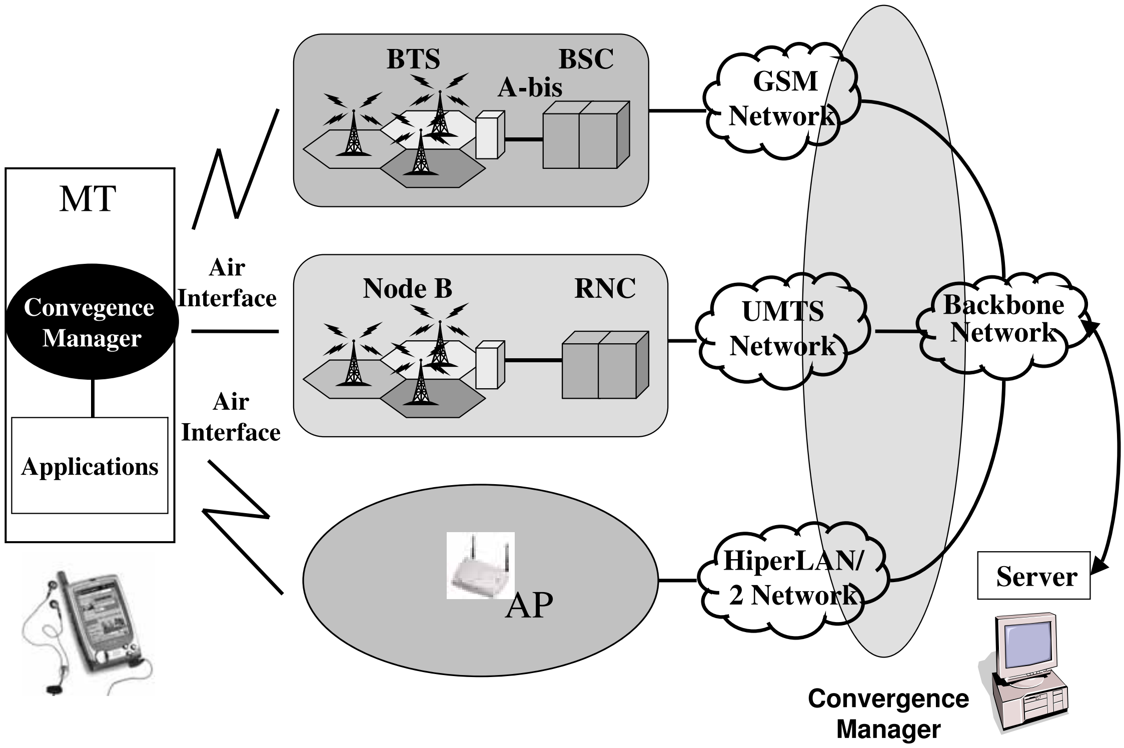


Figure 8.13: The convergence managers is located @CN

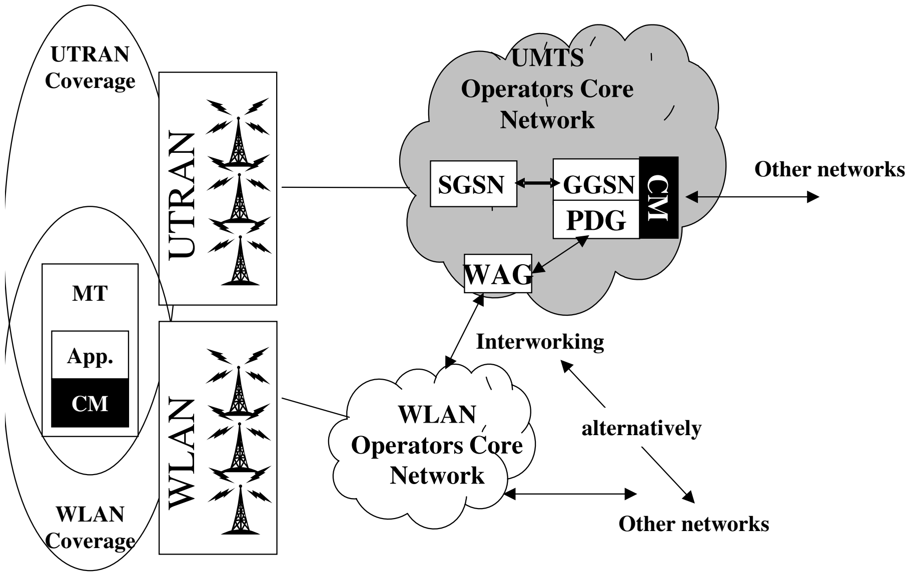


Figure 8.14: Concept of melting GGSN and PDG in convergence manager @CN, [19]

8.3.1 Melting GGSN and PDG

The similar applies to WLAN in case WLAN is also connected to the UMTS core network by tight interworking [54]. For loose coupling, the splitting/reassembling can be performed on a melted GGSN-PDG component. This concept is an outcome of the FLOWS project and therefore the following Figure 8.14 introduce this principle in details

In this concept all the convergence manager specific tasks can be handled on the network layer. There is no requirement to involve higher layers, e.g. the session layer. That means, the application interacts with the remote application providing a single point of contact. Except cases of changing session parameters, there is no necessity to interact end-to-end on the session layer because of the additional usage of WLAN, which means no end-to-end signalling interaction has to take place. Therefore, in such cases an immediate change of the connection conditions can be performed. It is important to note that in the described manner not only user defined or more exactly MT defined algorithms can be executed. Also operator specific rules can be considered, brought in by the network sided instance of convergence manager.

This approach of melting together GGSN and PDG provides the requested convergence manager behavior. The architecture is compatible to the 3GPP standards, what means that, except the matter of a combined GGSN-PDG component, no changes would have been required on existing standardization work. However, the interaction between the convergence manager in the network and the other instance on the MT would also have to be standardized. Therefore, an additional control plane protocol would have to be defined and to be included in the 3GPP procedures. If operator based policies shall also be realizable, standardized means for the collection of parameters would have to be found. Additionally, a defined interface between the terminal sided convergence manager and the higher layers would be helpful. Finally, it would have to be investigated, how scalability issues can be avoided on the GGSN-PDC component, caused by convergence manager operations.

Chapter 9

Convergence Manager Location Comparison

The location of convergence manger impacts the level of flexibility and optimization. The terminal, radio access networks, core networks, backbone networks and server entities are all potential hosting entities, each including significant advantages and drawbacks. In Chapter 7 and Chapter 8 the convergence manager @MT and @BS have been discussed in details and their performance have been illutrat by simulation results.

The basic assumption in this thesis is that the MT which is enabled for a simultaneous usage of multiple standards, has to contain a convergence manager instance in any case. This convergence manager can either work autonomously or cooperate with another convergence manager entity anywhere on the network side.

9.1 Summary of Convergence Manager @MT

The consideration to locate the convergence manager @MT brings significant benefits into the overall system. The convergence manager enables the MT to switch between different standard networks either on packet or application level. The convergence manager entity integrated inside the MT will communicate with the BSs for APs on the network side. Only the MTs will have to be modified by integrating an entity of convergence manager. There is no change required on the network side. No or very light standardization effort is required for an enhancement or an introduction of protocols for the proposed solution.

Without knowing the information from the other MTs to build up a global view over every single standard, the function of the convergence manager, as discussed in Chapter 7, is limited with only network selection. No enhancement on the network side is possible due to that no adaptive scheduling could be carried out by the convergence manager.

Current algorithm is very simple to implement by choosing the standard with higher transmission data rate at the beginning of every application. All the packets of an application will be transmitted through one single network according to the decision of the convergence manager. While switched algorithm allows the change of standard without interruption during a single application. The convergence manager will transmit the packets through another network which offers higher transmission data rate once it detects a new standard. The handover delay while switching between standards degrades the performance of the convergence manager. The location algorithm is trying to estimate the transmission time for an application based on the assumption that the MT knows about the user mobility and geographical coverage information of standard networks. The location algorithm is somehow a compromise between the current algorithm and switched algorithm. The idea of the location algorithm is trying to avoid the complexity due to change of standards during an application with switched algorithm but at the same time to exploit cleverer decisions than the current algorithm. Parallel network selection algorithm allows transmission of packets via different networks simultaneously and is supposed to give the theoretical performance bound. But this algorithm is extremely difficult to be implemented. The simulation results carried out in single user case have proven their performances: The parallel network selection algorithm gives the best performance and switched algorithm in very near to that without taking handover delay into consideration. The location algorithm outperforms the current algorithm. And overall making use of multiple standard networks enabled by the convergence manager gives better performance than that with a single standard.

The interferences between MTs decrease the performance in multi-user case. At most the location algorithm suffers from the wrong information of the transmission data rate. An improved algorithm namely history based location algorithm is developed making use of transmission data histogram. The performance of the convergence manager using history based location algorithm is improved and slightly worse than the switched algorithm inside multi-user scenario.

In the hotspot scenario the system performance decreases with only a single standard, but the convergence manager @MT improves the system performance taking advantage of convergence between multiple standards.

The assumption that the convergence manager knows the precise knowledge about user mobility and geographical coverage of wireless standards for the location and history based location algorithms may not be true any more. The performance will be degraded if there is any error in the information for convergence manager. Even if keeping the assumption, the performance of history based location algorithm still depends on how accurate the estimated data rate for the current transmission. Obviously if there is a rapid changing situations (because of fast moving speed or load fluctuations) the previous radio frames are too old to be used as mean data rate. The standard selection decision is likely to be wrong and the performance will be degraded.

Indeed the performance limitation lies on the fact that the convergence manager entity is only located @MT. Due to the location of the convergence manager, a limited level of information is exchanged between the convergence manager and the network side.

9.2 Summary of Convergence Manager @BS

More flexibility and benefits can be achieved by another consideration to locate the convergence manager @NET. Chapter 8 has focused on the concept of convergence manger @BS and discussed about the performance in details.

It is assumed that there is a pair of convergence manager entities, one @MT and another @BSs for multiple standards. The convergence manager @BS is the master entity who monitors and controls the transmission inside networks while the convergence manager @MT is a passive entity taking charge of the cooperation with the master convergence manager @BS.

The location of the convergence manager offers the opportunity to collecting much more information from the network side. The convergence manager @BS can be informed about not only the multi-user inference but also the instantaneous channel quality of different wireless standard. Based on those information the convergence manager can achieve a "global view" within the networks. More optimizing possibilities are brought into the multi-standard system, since the convergence manager @BS can apply both network selection algorithms and adaptive scheduling algorithms.

The convergence manager enables to apply link adaptation technique based on channel quality inside every single standard and which brings great benefits into the system. As known from the performance comparison between adaptive scheduling algorithms in Chapter 5, the adaptive scheduling algorithms with cross layer optimizing outperforms the other algorithms, such as pure queue oriented or pure channel oriented algorithms. As known from the comparison between network selection algorithm in Chapter 6 and Chapter 7 the switched algorithm is the best choice considering the performance and implementation complexity. In Chapter 8 the performance of the convergence manager with combination of different scheduling algorithm with the switched algorithm are discussed in different scenarios.

A single standard will reach its performance limitation very fast when either more MTs are requiring for services or there are more service components requiring high data rate. Introducing an additional standard and enabling the transmission via multiple standards will increase the system performance by the convergence manager. In the hotspot scenario the convergence manager increases the system capacity significantly and offers much better overall delay and throughput performances. The benefits of convergence manager gets even more significant when the average data rate requirement increases. A convergence manager is necessary to be implemented for network in heavy load or full load situation.

The location of the convergence manager @BS faces the challenges such as the standardization efforts and the signalling for the cooperation between different radio frames. Furthermore the pricing model and billing system for the network operator has to be updated as well. The situation gets more complicated in case the different radio access network belong to different network operators.

9.3 Comparison between @MT and @BS

Figure 9.1 shows the throughput performance comparison between the convergence manager @MT and @BS inside the same simulation scenario (UMTS coverage with HiperLAN/2 over hotspot) as defined in section 4.5. It is assumed that 50% the MTs are using VoIP and 50% the MTs are using video streaming traffic models. The MTs are following the hotspot distribution, a HiperLAN/2 cell is covering the hotspot area.

The poor level of optimization and cost of multi user inferences have limited the throughput performance of convergence manager @MT very far away from that when the convergence manager is @BS. The average throughput of the overall system with convergence manager @BS using STT and switched algorithm is 17 times higher than that if the convergence manager @MT. Even

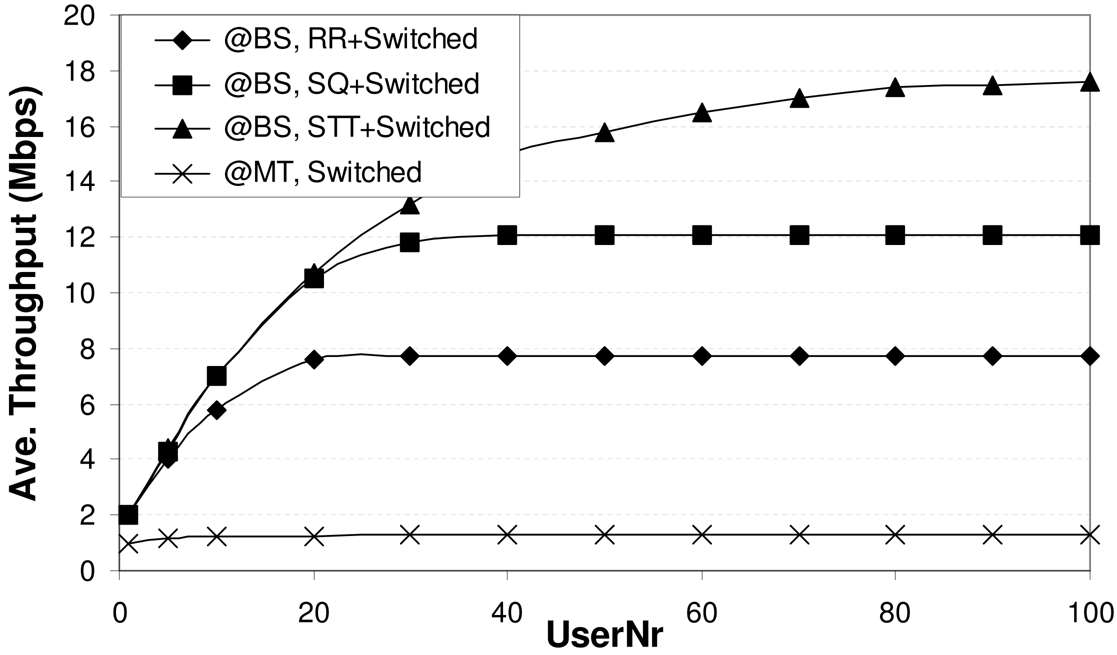


Figure 9.1: Served user number performance of convergence manager, comparison between @MT and @BS

if the convergence manager @BS is using the less favorable scheduling algorithm RR, the system performance achieves 8 times comparing to that if the convergence manager @MT.

The same result is shown in Figure 9.2 concerning served number of users inside the scenario. Under over load situation, the convergence manager @BS can satisfy 86% of the end users inside the scenario while the convergence manager @MT achieves only 10%.

The main advantages and drawbacks of each consideration are listed in Table 9.1, as a comparison between different convergence manager location.

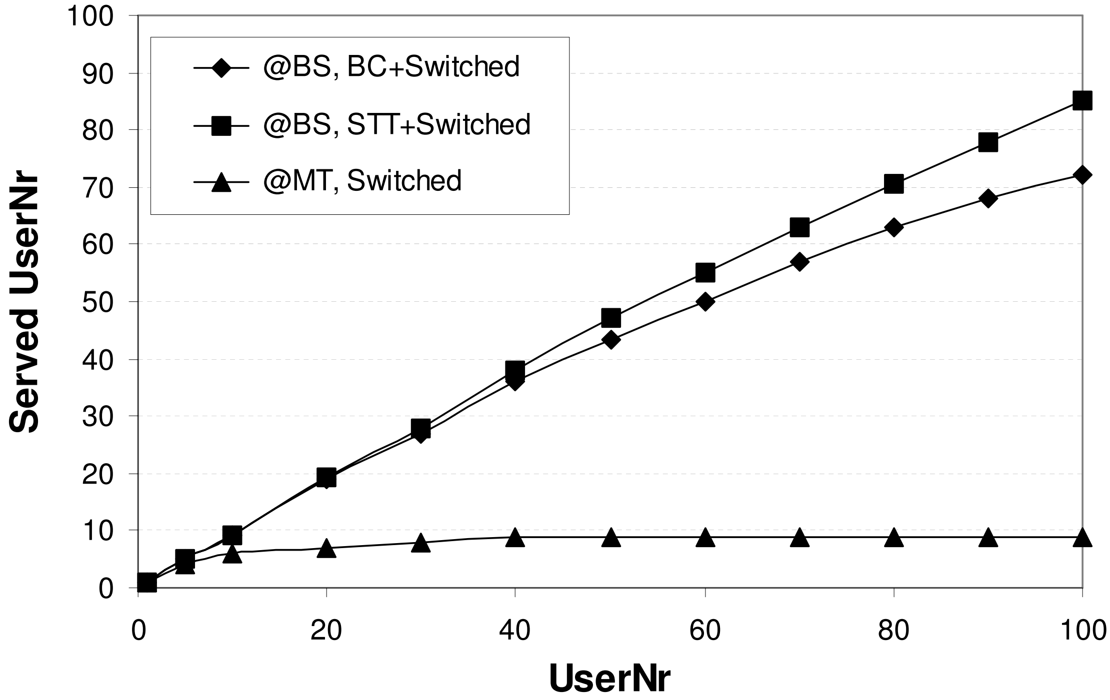


Figure 9.2: Served user number performance of convergence manager, comparison between @MT and @BS

Convergence Manager Location	@MT	@BS
Adaptive Scheduling	No	Yes
Network Selection	Yes	Yes
Benefits	No change in network	Significant performance improvment, optimization of radio interface, optimization of network route
Challenge	Limited benefits	Standardization efforts, hardware implementation, network management

Table 9.1: Comparison of the convergence manager locations

Chapter 10

Conclusions

The research work covered by this thesis is focusing on a flexible and simultaneous use of multiple standards. The convergence between mobile communication standards is aiming to enable an enhancement of services for the benefit of related actors, such as end users, network operators and service providers. Flexibility for end users and network operators, efficiency of radio resource allocation are considered to be the benefits of standards convergence.

In this thesis, a concept of Convergence Manager is designed to answer the open points related to the convergence between multiple standards.

The general function of the convergence manager is to monitor and control the mapping of service traffics into multiple networks based on different standards. The convergence manager is to enable parallel or alternate use of equipment and radio networks adhering to two or more physical layer transmission standards for the transmission of information related to the same communication.

The scenarios for the convergence manager are categorized as service scenario, standard scenario and user scenario. Service scenario includes the different QoS requirements from various service classes: conversational class, steaming class, interactive class and background class. Example traffic models from different services are described. User scenario defines the parameters for the convergence manager related to the user distribution, mobility and QoS requirements. The service scenario is based on the co-location of multi-standard, provides the possibility of accessing to networks of different standards. These three scenarios are considered as the baseline to define the simulation scenarios for the evaluation of convergence benefits.

The algorithm in convergence manager consists of two different categories: network selection algorithm and adaptive scheduling algorithm. A fixed mapping table between standards and service applications are offered as a pre selection algorithm for convergence manager. But the dynamic mapping between standards and services according to the instantaneous link channel quality and QoS requirements is of most interest and is investigated in details.

The cross layer adaptive scheduling algorithm STT outperforms the other scheduling algorithms inside single standard scenario. The link adaptation technique enables the PHY-mode selection and MC selection in the physical layer depending on the instantaneous channel quality. The cross layer optimization takes into consideration of both traffic QoS requirements and link adaptation technique on the radio interfaces. Pure channel oriented algorithm BC and pure source traffic oriented algorithm SQ/LQ performance bring very good performance too. It is also necessary to mention that fairness oriented algorithms such RBC, RR perform worse in general.

The Switched network selection algorithm which allows the user to switch to the network with higher data rate even during mid of transmission gives the best performance inside single user scenario. The current algorithm offers better delay performance than single standard. However the wrong selection of the current best selection of standard limits its performance. Location algorithm is trying prevent wrong selection by using the knowledge of user-mobility and coverage-range-information of available standards. It exhibits the delay performance quite close to the switched algorithm over all range of file sizes and at the same time it avoids the complexities associated with switching mid way of transmission. Parallel network selection algorithm allows transmission of packets via different networks simultaneously and is supposed to give the theoretical performance bound. But this algorithm is extremely difficult to be implemented in real situations.

The location of the convergence manager is a crucial issue for the architectural design of an environment, providing simultaneous usage of standards. The convergence technique is of great importance to network operators in order to optimize their network resource allocations. Based on the granularity of the introduced convergence the end user may also be involved in the decision process of choice of network.

One possibility discussed in this thesis is that convergence manager located only @MT. Convergence manager is a functionality which communicates with different communication network and tries to select the "best" standards for the applications of MTs. The advantage of this possibility of convergence manager location is that no change is required on the network side except an entity to communicates with the convergence manager. No or very light standardization effort is required for an enhancement or an introduction of protocols for the proposed solution. While at the same time, the performance of the convergence manager suffers from the lack of information on the network side. No enhancement from network side could be implemented since no adaptive scheduling algorithm could be implemented by the convergence manager. Only network selection algorithms could be implemented by the convergence manger.

More flexibility and benefits can be achieved from another location of the convergence manager @BS. A multi-mode BS is proposed for the future wireless communication network but it is not a must. The standardization efforts is necessary here to implement the convergence manager entity into the network protocol and to implement the optimization technique cross different layers inside communication system model. Another challenge is to define the handover signals between different standards. In spite of the challenges, this proposal is still with great meaning for a convergence manager concept. The location of convergence manager @BS brings the possibility to apply network selection algorithms and enables cross layer optimization technique based on the instantaneous channel condition of each individual MT.

A scenario based on co-coverage of HSDPA and HiperLAN/2 is used for the simulation. A combination of network selection and adaptive scheduling algorithms can be implemented by the convergence manager. The STT together with switched algorithm is proven to be the most optimized among all others. The higher the traffic load comparing to the system capacity is, the more benefits that convergence manager can bring into the system.

The consideration to place the convergence manger @CN is another promising solution for convergence between standards. This proposal will require only modifications inside the network nodes, which is of advantage than requiring modification inside the BS/AP. But the flexibilities in the system are limited because the channel oriented adaptive scheduling algorithms could not be applied by the convergence manager without access to the radio channel.

The convergence manager consideration indicates a good solution for network planning. In case of hotspot distribution among end users, the most effective solution is to introduce a WLAN cell covering the hotspot area. The convergence between multiple standards achieves an optimized allocation of user traffics among network resources, benefits not only the end user inside the hotspot but also those who are outside the hotspot area.

The convergence manager brings the possibility to design new services for the future mobile communication at the same time. Based on the performance of convergence manager within different scenario, it can be conclude that the convergence between different standards will be a strong candidate for the future mobile communication system design and the concept of convergence manager is for sure with a convincing future.

Appendix A

Business Model of Convergence Manager in Telecommunication Market

A business model describes how the companies do their business and with whom in the telecommunication market. If takes a look into the current telecommunication market, who are the actors? those words are familiar, such as end user, T-Mobile, Deutsche Telekom, EPlus, O2, AOL, etc. But what kind of roles are they playing if using a business model to describe the telecommunication market and what will be their goal along with activities? when trying to realize convergence between different standards, what will be the benefits for different actors and what can be the potential challenge for them?

A.1 Related Actors

A.1.1 End User

An end user is an actor that pays for using services transported over a wireless network. Price and service are always important for them. Generally they are always seeking for the best achievable quality of services with the lowest price.

A.1.2 Network Operator

A network operator is an actor that makes revenue from providing network access and transport services over his wireless network.

A.1.3 Service Provider

A service provider, or somewhere else known as, a content provider, is an actor that makes revenue from providing end user services over a wireless network.

A.1.4 Single Service Provider

This is the traditional "monopoly" situation in the wireless telecommunication business. The network operator also provides the end user services, such as voice telephony and internet access.

A.1.5 Multiple Service Provider

In most cases, the end user subscribes to a service provider not to a network operator directly. The user wants to access a certain service he finds useful. In this case, the network is merely a bearer of the service, a way for the content to reach the user. Subscribing to a network operator will be an issue for the service or content provider, not for the end user. The service provider will choose to buy the wireless access service from the network operator.

This model offers the best conditions for competition at the service level. Many service providers can get involved in the operation phase, generating a high expected revenue for the network operator, who in turn gets an incentive to maintain, enhance and develop the network.

Of course there should be and are more than one network operator to choose from for the service provider. For example in the 3G, different network operators will own different standardized access technologies.

A.2 Service Level and Price

The simplest model for pricing is to pay for the service that you get, completely proportional to the usage. The end users will pay their service provider for the services they enjoyed. The service provider will then pay the network operator for the services offered to his customers. In this case, the revenue for both service provider and network operator will increase if more users are served in a given time with a guaranteed service quality. In this case, it is extremely important for the network operator to optimize the user of the network resources, aiming to achieve the most revenue by offering the most service out of its capacity limitation. If the network operator is operating different wireless communication networks (e.g. GSM, UMTS, WLAN, etc) at the same time, it is necessary to control the simultaneous use of several networks. Similarly if the service provider have contracts with different network operators, a is as well important for the service provider in order to serve most possible users while requiring lest capacity from the operator.

In the current business models, there are normally a Service Level Agreement (SLA) [26] between the network operator and service provider. It defines the QoS requirements from the service provider to network operator and puts a limit on the amount of resources from network operator to the service provider. The SLA may include the limitation on :

- Number of simultaneous served users
- Throughput and delay
- Connection latency
- Price
- Time limitation that the SLA should be fulfilled
- Penalties for not fulfilling the SLA

The SLA can guarantee a certain revenue for the network operator if all requirements are fulfilled. Breaking the SLA will lead to the penalty for the breaking party. Therefore it is necessary for both network operator and service provider to control the network usage and agree on an optimized SLA.

In [26] six different price models are introduced to wireless services. The price models describe the different relationships between the service level and the revenue for network operator. Among these models, the service level can be understood as number of the served user under a certain SLA or the average data rate delivered to per user.

The technology development and standardization for 3G and 4G should be aimed at creating a benefit for both end users, service providers and network operators. They should come to agreement that their expect will be profitable for them. The physical layer of such a system should then allow a high flexibility in terms of resource usage policies that enables fast allocation of channel resources where they are best utilized. It is important for the network operators to be more flexible and efficient and to offer cheaper access to the end users.

A.3 Operator Capacity

Fulfilling all requirements will guarantee a certain revenue for the network operator. Breaking the SLA will lead to penalty. A good chance for the network operator to earn more money is overbooking the resources. The operator signs SLAs that he will most probably not be able to fulfill when the demand becomes high, e.g. at peak hours or so called heavy load. In these cases, scheduling will play an important role in minimizing the damage by efficiently allocating the available resources to the remaining clients. Especially if the network operator is operating different networks simultaneously, e.g. 2G network, UMTS network and WLAN, an good opportunity is provided by a Convergence Manager to maximize the operator revenue by optimizing the simultaneous multi-network usage to the requests coming from different wireless communication systems. Considering about the price models discussed in [26], the criterion to evaluate the convergence manager will be increasing served number of user under certain QoS requirements and increasing average data rate per user.

Figure A.1 shows a randomly generated example how the total system capacity and the demand of guaranteed services may vary over time. In some moment (peak time) the system will not have sufficient capacity to serve all the end users. The Convergence Manager is desirable to fill up the gap between the demand and the total service capacity aiming for the maximum revenue for the operator.

A.4 User Satisfaction

End users are always looking for the cheapest service to fulfill their requirement. There are also SLA between the end user and the service provider, defining the limitation on price, priority,

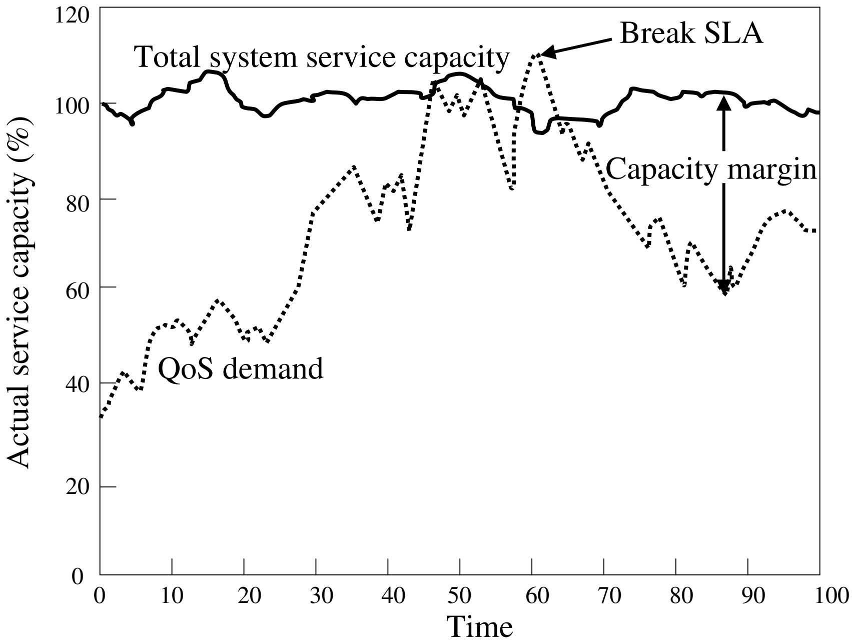


Figure A.1: Example of a system service capacity and how the service demand may vary over time.

latency. According to the different price model, the end users are classified into different target groups: cheapest price model will normally lead to the limitation of service quality and expensive services have always high priority. In the situation of shortage of network capacity, the end users with lest SLA might not be served. It gets even more complicated with the co-existing of GSM, UMTS and WLAN. Several wireless links can be provided to some end users, and it is then up to the end user which one(s) to use. If the end user will use different wireless link in the same time, issues such as billing, handover, hardware requirement will have to be taken into consideration. The user satisfaction will have to be improved by applying the convergence manager to control the simultaneous use of different wireless standards.

The user satisfaction function [24] is a function of all the variables that affect the user's perception of the access network. It can be factors such as:

- Average Data Rate (ADR)
- Handover Frequency and Latency (HF, HL)
- Power Consumption (PC)
- Cost (price of service) (C)
- Hardware Harmonization Requirement (HHR)
- User Preference (UP)

It can be written in function like:

$$UserSatisfaction = f(ADR, HF, HL, PC, C, HHR, UP, ...) \tag{A.1}$$

It is still a question how to describe the user satisfaction [24]. Normally the parameters are given with different weighting factor. The relative importance of different parameters will vary from user to user and depend on what type of service are being used. Furthermore, it can be assumed part of the function that depends on average bit rate and handover frequency and latency can be separated specifying the Quality of Service perceived by the user. The other part of the function can be separated into a function that corresponds to the actual cost of using the access networks seen from the user's point of view. The user satisfaction function will then take the form:

$$UserSatisfaction = g(ADR, HF, HL) - h(C, HHR, UP) \tag{A.2}$$

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Abbreviations

2G	Second Generation
3G	Third Generation
3GPP	3rd Generation Partnership Project
4G	Fourth Generation
ABC	Always Best Connected
ACH	Access Feedback Control
ADR	Average Data Rate
AMC	Adaptive Modulation Code
AP	Access Point
BC	Best Channel
BCH	Broadcast Control
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BS	BaseStation
BSC	Base Station Controller
C	Cost
CDMA	Code Division Multiple Access
CN	Core Network
DAB	Digital Audio Broadcast
DLC	Data Link Control
DSL	Dedicated Service Link
DVB	Digital Video Broadcast
EDGE	Enhanced Data rates for GSM Evolution
FCH	Frame Control Channel
FDMA	Frequency Division Multiple Access

FLOWS	Flexible Convergence of Wireless Standards and Services
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GMSK	Gaussian minimum Shift Keying
GPRS	General Packet Radio Service
GPS	Generalized Processor Sharing
GSM	Global System for Mobile Communications
GSM-CSD	GSM Circuit Switched Data
HF	Handover Frequency
HHR	Hardware Harmonization Requirement
HiperLAN	High performance Local Area Network
HiperLAN/2	HiperLAN second generation
HL	Handover Latency
HSDPA	High Speed Downlink Packet Access
HS-DSCH	High-Speed Downlink-Shared Channel
HS-SCCH	High-Speed Signal Control Channel
IEEE	Institute of Electrical and Electronics Engineers
IMS	Inter Media Service
IP	Internet Protocol
IP-CAN	IP-Control Area Network
ISO/OSI	International Organization for Standardization /Open Systems Interconnection
KB	KByte
Kbps	Kbit/s
LCH	Long CHannel
MAC	Medium Access Control
Mbps	Mbit/s
MC	Modulation Coding
MMS	Multi-Media Messaging Service
MPEG	Moving Picture Experts Group
MPF	Modified Proportional Fair
MT	Mobile Terminal
NG	Next Generation
OFDM	Orthogonal Frequency Division Multiplexing

PC	Power Consumption
pdf	probability density function
PDG	Packet Data Gateway
PDU	Protocol Data Unit
PER	Packet Error Rate
PF	Proportional Fair
PHY	Physical
PSK	Phase Shift Keying
QAM	Quadrate Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrate Phase Shift Keying
RAB	RAdio Bearer
RAN	Radio Access Network
RBC	Relative Best Channel
RCH	Random Access Channel
RNC	Radio Network Controller
RR	Round Robin
SCH	Short CHannel
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SMS	Short Message Service
SNR	Signal to Noise Ratio
SP	Pre-defined Service Priority
SQ/LQ	Shortest/Longest Queue
STT	Shortest Transmission Time
TCP/IP	Transport Control Protocol/Internet Protocol
TDD	Time Division Dulplex
TDMA	Time Division Multiple Access
TTI	Transmission Time Interval
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
UP	User Preference

UTRAN	UMTS Terrestrial Radio Access Network
VoIP	Voice-over-IP
WAG	WLAN Access Gateway
WFQ	Weighted Fair Queuing
WLAN	Wireless Local Area Network

Symbols

t_{ON}	Mean ON phase
t_{OFF}	Mean OFF phase
L_{ps}	Path loss
f	Carrier frequency
d	Distance between MT and BS/AP
L_O	Path loss at a distance of $1m$
n	Slop factor in a path loss model
p_{a_s}	Slow fading
σ^2	Variance
μ	Expectation value of the normal distribution
p_{a_f}	Fast fading
λ^2	Variance
P_{trans}	Transmission Power
P_{noise}	Noise Power
f_D	Dopperfrequency
f_o	Carrier frequency
v	Speed of MT
ϕ	Angle between link direction and mobility
c	Speed of light
T_c	Coherence time
$[t_0 - T_c/2; t_0 + T_c/2]$	Time interval
$f_{D,max}$	Maximum Dopplerfrequency
R	Channel coding rate
i	User i

s_i	Resource share of user i
r_i	Rate weight for the user i
$C(t)$	Total capacity at time t
$U(t)$	Number of users at time t
$W_i(t_1, t_2)$	Assigned bandwidth to user i
$(C/I)_i(t)$	Carrier-to-interference ratio for user i at time t
$T_i(t)$	Achieved data throughput for user i
$trsraterelative_i$	Relative transmission rate state for user i
$trsRate_i$	Currently achieved transmission rate for user i
$trsrateave_i$	Average transmission rate for user i
t_i	Transmission time for user i
N_i	Number of packets inside queue i
PL	Packet size
$C_i(t)$	Channel capacity at time t for user i

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