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REDUCTION OF HANDOVER INTERRUPTION IN MOBILE NETWORKS

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ABSTRACT

Wired networks provide stable and high quality connection to users. However the users are limited in movement since the wired networks do not enable mobility. It gives rise to mobile wireless networks. The history of mobile networks is dated to 80's of 20th century when several analog mobile networks were developed around Europe.

Former analog mobile networks were replaced by digital ones. Currently deployed and utilized mobile networks in Europe are usually based on UMTS (Universal Mobile Telecommunication System), denoted as third generation (3G) of mobile communications systems. Concurrently, wireless networks based on IEEE 802.16 standards were developed at the end of 20th century. The networks based on these standards are known as WiMAX. The first versions of IEEE 802.16 standard describe wireless networks without support of mobility. The mobility was introduced in version IEEE 802.16e, issued in 2006. Considering rising demands of users for higher quality of service, the next versions of WiMAX are still developed. Developed versions have defined particular system parts and features and minimum limit for its performance.

This thesis investigates a handover procedure. The handover enables full mobility of users along area covered by a system due to automatic (without user's intervention) change of serving base station. The handover procedure consists of several steps. The first one is a monitoring of quality of physical channel parameters between user and all neighboring base stations. Based on the observed parameters, the potential new serving base station is selected. Consequently the connection to the new station is set up.

According to the time when the connection with current serving BS is closed, the handover can be divided in two groups: i) hard handover and ii) soft handover. If the hard handover is performed, the connection with the serving station is closed before an establishment of connection with new base station noted as target base station. In case of the soft handover, simultaneous connections with more than one base station can be maintained. An advantage of the hard handover is higher simplicity of implementation in comparison to the soft handover. Hence the hard handover is only mandatory type of handover in WiMAX networks. In both cases, the handover procedure is controlled and managed by medium access control layer in WiMAX. The handover management procedure is defined by a sequence of management messages exchanged between the mobile station and serving base station. An individual set of messages is utilized for each of the handover stages. As the messages are exchanged consequently, a short interval during which the mobile station can not receive and/or transmit data occurs in case of the hard handover. This interval is called handover interruption or handover delay. An evaluation of the handover interruption duration is faced in the thesis.

As no data transmission is enabled during the handover, a quality of service provided to users is temporarily impaired. It leads to a dissatisfaction of users with connection. The impact of handover interruption duration on the quality of service is also investigated in this thesis in form of voice over IP communication quality assessment.

As the mobile wireless networks enable to monitor a huge set of parameters of communication channel between a mobile station and neighboring base stations, the evolution of this parameters can be utilized to predict in advance next station that will serve the mobile station after accomplishing the handover. The prediction with high ratio of correctly predicted target base stations enables to introduce novel handover procedure that results into significant reduction of the handover interruption. With this purpose, techniques for the handover prediction are investigated. Moreover, possible improvement that leads to increase of the prediction efficiency is proposed.

Exploiting results of the target base station prediction, the handover procedure that allows to reach low handover interruption is proposed. The proposal contains definition of a flow of management messages together with a description of content of new management messages. The novel procedure is analyzed not only from the handover interruption point of view however its impact on a management overhead and user's throughput are also discussed in the thesis.

ABSTRAKT

Pevné kabelové nebo optické sítě poskytují uživatelům stabilní a vysoce kvalitní připojení. Nevýhodou tohoto typu připojení je omezení pohybu uživatelů. To bylo důvodem vzniku mobilních bezdrátových sítí. Jejich historie se datuje do 80-tých let dvacátého století, kdy v Evropě vznikaly první analogové mobilní systémy.

Původní analogové sítě byly postupně nahrazeny digitálními. V současné době jsou v Evropě budovány a využívány sítě založené zpravidla na technologii UMTS (Universal Mobile Telecommunication System). Tyto sítě jsou často označovány jako sítě třetí generace neboli 3G. Na konci dvacátého století se začaly vyvíjet také bezdrátové sítě založené na standardech IEEE 802.16. Tyto sítě jsou dnes označovány jako sítě WiMAX. První verze standardu WiMAX byly navrhovány jako bezdrátové sítě bez podpory mobility uživatelů. Ta byla umožněna až v roce 2006, kdy byl vydán standard IEEE 802.16e. Jelikož nároky uživatelů na kvalitu služeb se stále zvyšují, tak je i technologie WiMAX neustále vyvíjena. Nově připravované verze standardu však musí zohledňovat požadavky uživatelů a definovat pro ně nároky na jednotlivé části systému a limity pro jejich výkonnost.

Tato disertační práce je zaměřena na tzv. proceduru handover, která zajišťuje plnou mobilitu uživatelů v oblasti pokryté danou technologií. Handover zajišťuje automatickou změnu základnové stanice bez zásahu uživatele. Celá procedura je rozdělena do několika kroků. Prvním krokem je monitorování kvality komunikačního kanálu mezi uživatelem a okolními základnovými stanicemi. Na základě tohoto pozorování je pak vybrána potenciální nová obsluhující stanice, se kterou je později navázáno nové spojení.

Podle toho kdy dojde k ukončení spojení s obsluhující základnovou stanicí je možné rozlišit dva typy handoveru: i) tvrdý handover a ii) měkký handover. V případě, že se jedná o tvrdý handover, je spojení s obsluhující stanicí ukončeno ještě předtím, než je navázáno spojení s novou základnovou stanicí. Ta je nazývána stanicí cílovou. V případě měkkého handoveru je udržováno spojení s více základnovými stanicemi současně. Výhodou tvrdého handoveru je jednodušší implementace oproti handoveru měkkému. Zejména z toho důvodu je pouze tvrdý handover povinně implementován do všech mobilních sítí WiMAX. U obou typů je celý proces handoveru v mobilních sítích WiMAX řízen a kontrolován vrstvou pro řízení přístupu k médiu. Celá procedura je definována jako sekvence řídících zpráv vyměňovaných mezi základnovou a mobilní stanicí. Každá z fází handoveru využívá jiné zprávy. Jelikož řídící zprávy jsou vysílány postupně, objevuje se v případě tvrdého handoveru krátký časový interval během něhož nemůže mobilní stanice vysílat ani přijímat data. Tento interval se nazývá přerušení v důsledku handoveru. Výpočet doby trvání přerušení je prezentován a analyzován v disertační práci.

Jelikož během tvrdého handoveru nejsou přenášena data, tak dochází k dočasnému snížení kvality služeb poskytovaných uživateli. To může mít za následek vyšší nespokojenost uživatelů s nabízeným spojením. Vliv handoveru na kvalitu služby je proto v disertační práce také vyhodnocován formou hodnocení vlivu handoveru na kvalitu hovoru při přenosu hlasu přes IP sítě.

Mobilní sítě průběžně monitorují velké množství parametrů komunikačních kanálů mezi mobilní stanicí a sousedními základnovými stanicemi. Vývoj těchto parametrů je možné využít k předpovědi následující základnové stanice, která bude obsluhovat danou mobilní stanici po vykonání handoveru. Pokud je dosaženo dostatečné úspěšnosti predikce, tak je možné provést úpravy procedury handoveru, které povedou k výraznému zkrácení doby přerušení v důsledku handoveru. Za tímto účelem jsou v disertační práci zkoumány některé metody predikce a zároveň je popsán a analyzován nový způsob pro zvýšení úspěšnosti predikce cílové stanice.

Dále je navržena metoda handoveru, která díky využití výsledků predikce dosahuje podstatně nižšího zpoždění v důsledku handoveru. Návrh se skládá z popisu nových řídících zpráv a zároveň jejich výměny mezi zařízeními. Inovovaná procedura je zkoumána z hlediska zkrácení přerušení v důsledku handoveru a zároveň z hlediska vlivu na velikost záhlaví generovaného řídícími zprávami a vlivu na propustnost mobilní stanice.

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LIST OF ABBREVIATIONS

AC	Access Code
ACD	Average Call Duration
ARQ	Automatic Repeat reQuest
BS	Base Station
CA	Complete re-Authentication
CINR	Carrier to Interface plus Noise Ratio
CID	Connection IDentifier
CV	Channel Variation
DCD	Downlink Channel Descriptor
DPS	Data Per Subchannel
ELT	Expected Link Throughput
FBSS	Fast Base Station Switching
FPHO	Fast Predicted HandOver
GPS	Global Positioning System
HDT	Handover Delay Timer
HM	Hysteresis Margin
HR	Hit Ratio
IEEE	Institute of Electrical and Electronics Engineers
IMT	International Mobile Telecommunications
LOS	Line Of Sight
LPM	Last Packet Marking
LTE	Long Term Evolution
LTE-A	LTE-Advanced
MAC	Medium Access Control
MCS	Modulation Coding Scheme
MDHO	Macro Diversity Handover
MMM	Manhattan Mobility Model
MOS	Mean Opinion Score
MS	Mobile Station
NA	Not re-Authenticated
NLOS	Non-Line Of Sight
NPR	Not Predicted handovers Ratio

NS	Neighboring Set
OFDMA	Orthogonal Frequency Division Multiple Access
PESQ	Perceptual Evaluation of Speech Quality
PHO	Passport HandOver
PHY	Physical layer
PLC	Packet Loss Concealment
PRWMM	Probabilistic Random Walk Mobility Model
PUSC	Partial Usage of Sub-Channels
QoS	Quality of Service
RAHO	Relay Assisted Hard handover
RASH	Relay Assisted Soft Handover
RRC	Radio Resource Cost
RS	Relay Station
RSSI	Receive Signal Strength Indicator
RTD	Round Trip Delay
SA	Security Association
SF	Shadowing Factor
SNR	Signal to Noise Ratio
TLV	Type-Length-Value
ToCL	Time of Code Life
TP	Turn Probability
TTT	Time To Trigger
UCD	Uplink Channel Descriptor
UL-MAP	UpLink map
UMTS	Universal Mobile Telecommunication System
VoIP	Voice over Internet Protocol
WiMAX	Worldwide Interoperability for Microwave Access
WPR	Wrong Predicted handovers Ratio
WS	Window Size

1 INTRODUCTION

WiMAX (Worldwide Interoperability for Microwave Access) is a broadband wireless technology based on standards developed by IEEE 802.16 working group. The first completed version of WiMAX is described in standard IEEE 802.16-2004 [1]. This version was published in October 2004. Standard IEEE 802.16-2004 does not support full mobility of users. To enable full mobility of users, a handover procedure is introduced in consequent version of WiMAX defined by standard IEEE 802.16e [2] issued in 2006. Following version of standard, IEEE 802.16j [3], introduces RSs into network topology [4]. The next version, IEEE 802.16m [5], is currently under development. The main objective of this version is to provide higher quality of service and higher throughput. Both versions, IEEE 802.16j and IEEE 802.16m, assume full mobility of users. Thus IEEE 802.16m defines among others strict requirements on the handover procedure.

1.1 HANDOVER PROCEDURE

General purpose of the handover procedure is to ensure continuous connection to a Mobile Station (MS) while it is moving among an area of several Base Stations (BSs). Therefore, the BS that provides connection to the MS, denoted as serving BS, must be updated. The new BS that will serve the MS after the handover is called target BS.

Following subsections provide an overview on the handover according to all versions of WiMAX standards with support of full mobility.

1.1.1 HANDOVER ACCORDING TO IEEE 802.16E

This standard defines three basic types of handover [2]: hard handover, Macro Diversity Handover (MDHO) and Fast Base Station Switching (FBSS). Hard handover is mandatory in WiMAX systems. Other two types of handover are optional.

1.1.1.1 HARD HANDOVER

During the hard handover, a MS communicates with just one BS in each time. All connections with a serving BS are broken before new connections to a target BS is established. It means that there is a very short time interval when the MS is not connected to any BS. Handover is executed after an observed channel parameter (e.g.

signal strength) from a neighboring BS exceeds the same parameters from the serving BS. This situation is shown in Figure 1.

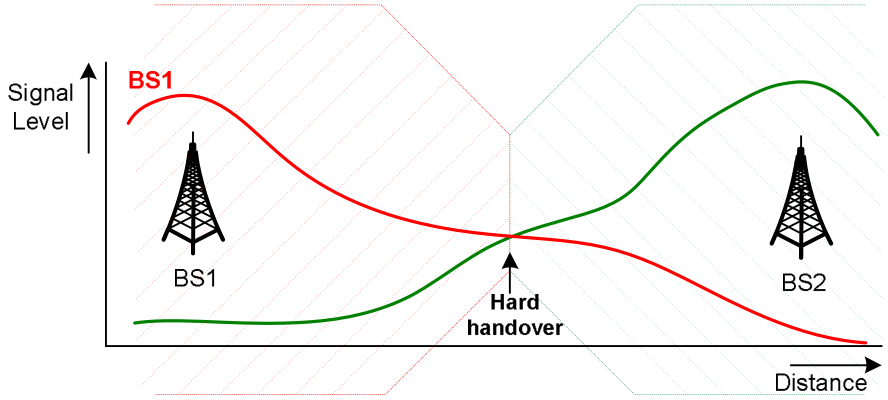


Figure 1. Hard handover

This type of handover is less complex and fairly simple. However it causes higher delay of packet [7].

1.1.1.2 MACRO DIVERSITY HANDOVER

When the MDHO is supported by a MS as well as by BSs, a diversity set (in some publications, usually focused on UMTS or LTE (Long Term Evolution) [8], noted as active set [9] [10]) is maintained by the MS and BSs. The diversity set is a list of BSs, which are involved in the handover procedure. The diversity set is maintained by the MS and by the BSs. It is updated via MAC (Medium Access Control) management messages [2]. A transmission of these messages is usually based on a CINR (Carrier to Noise plus Interface Ratio) level of BSs and it depends on two thresholds defined for addition and deletion of a BS from the diversity set: Add Threshold and Delete Threshold [2]. Threshold values are broadcasted in DCD (Downlink Channel Descriptor) message [2]. The diversity set is defined for each MS in the network. The MS continuously monitors all BSs in the diversity set and selects an anchor BS. The anchor BS is one of the BSs from diversity set. The MS is synchronized, authorized and registered to the anchor BS. Furthermore, the MS performs ranging and monitors a downlink channel of anchor BS for control information. The MS communicates simultaneously (including user traffic) with the anchor BS and with all active BSs in the diversity set (see Figure 2).

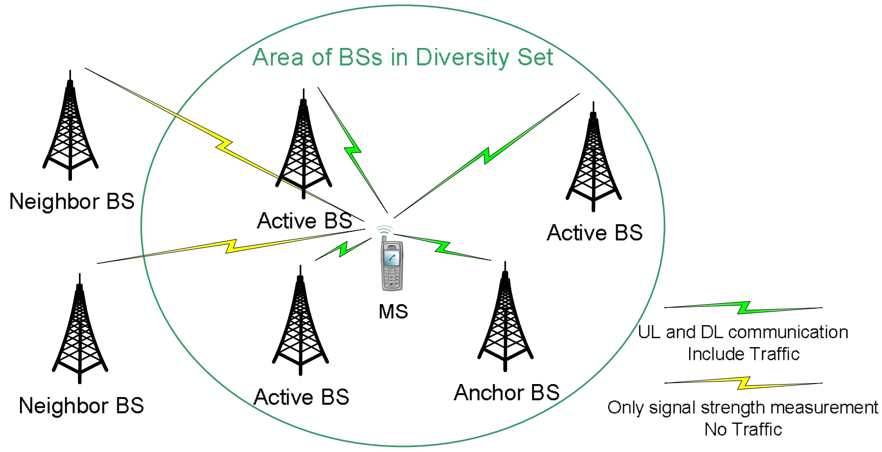


Figure 2. Macro Diversity Handover

In downlink direction, two or more BSs transmit data to the MS such that diversity combining can be performed by the MS [11]. In uplink direction, the MS transmission is received by multiple BSs. Consequently, a selection diversity of received information is performed [12]. The BS, noted as neighbor BS, can receive communication among the MS and other BSs, however signal level received by the MS from this BS is not sufficient to add the neighbor BS to the diversity set.

1.1.1.3 Fast Base Station Switching

In FBSS, the diversity set is maintained by a MS and by BSs exactly as in the case of MDHO. Opposite to the MDHO, the MS communicates only with anchor BS for all types of uplink and downlink traffic including management messages (see Figure 3). When the MS is connected to just one BS, thus the diversity set contains only this one BS that must be termed the anchor BS. The anchor BS can be changed on frame to frame basis depending on a BS selection scheme. This means that every frame can be sent via different BS in the diversity set. The anchor BS updating procedure is based on the same principle as the diversity set update.

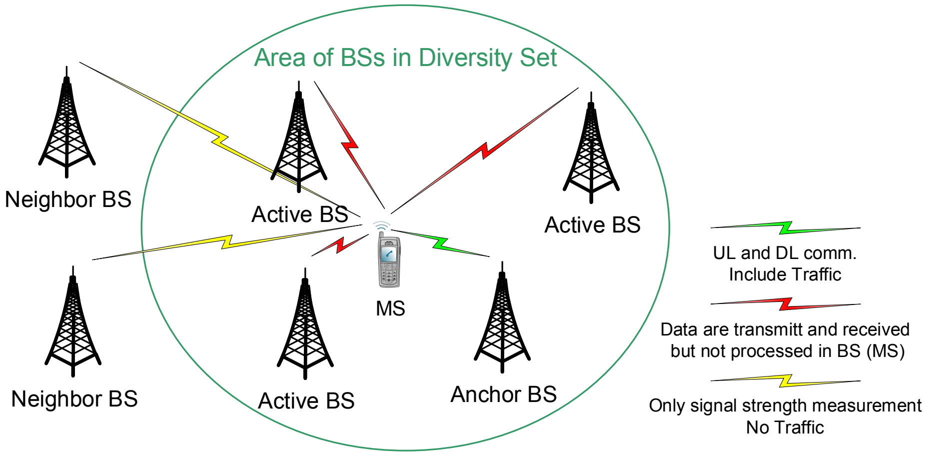


Figure 3. Fast Base Station Switching

1.1.2 HANDOVER ACCORDING TO IEEE 802.16J

The IEEE 802.16e standard defines a handover only among BSs. It does not consider RS (Relay Station). The implementation of RSs into WiMAX networks is the objective of standard IEEE 802.16j. This standard was released in June 2009. The RSs are generally simplified BSs and may be used either to extend coverage of a BS or to increase capacity in specific area [13]. There are two types of RS: fixed and mobile. The fixed RS is permanently installed at the same place whereas the mobile RS is supposed to be implemented into moving vehicles (e.g. bus, train, etc.) [14]. The RSs are connected to a network via radio interface, i.e. there is no wired connection to the backbone. Two systems from relay capability point of view can be distinguished: centralized and decentralized relaying [15].

As mentioned in [16], several scenarios of the handover should be distinguished according to serving BS, access station, target serving BS and target access station (see Figure 4). Moreover, two types of handovers (according to the change of serving BS) must be distinguished: i) inter BS handover and ii) intra BS handover.

The inter BS handover means the handover between cells of different BS (MS1 in Figure 4). Contrary, the intra BS handover represents scenario in which a MS performs handover in the area of one BS (MS2 in Figure 4).

The connection of a MS to a networks via a RS leads to so called multi-hop communication, where the "multi" represents a number of parts of path between a MS and its serving BS (e.g. two hop communication is depicted in Figure 4 between MS1 and BS1 or BS2 via RS2 or RS3 respectively). One hop communication is performed all the time in networks according to IEEE 802.16e.

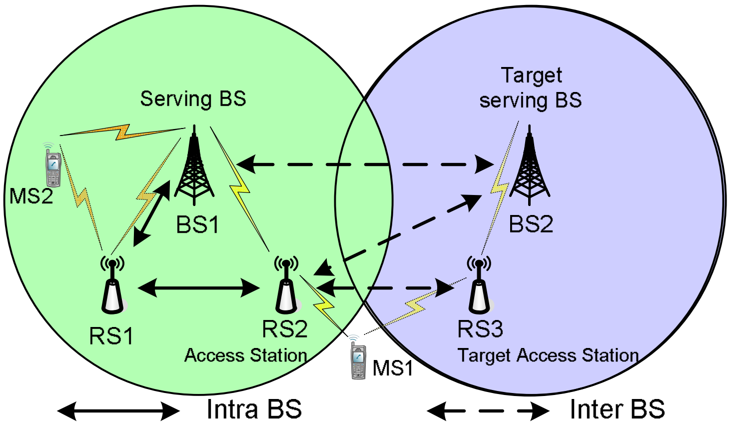


Figure 4. Inter BS vs. intra BS handover

1.1.3 HANDOVER ACCORDING TO IEEE 802.16M

The IEEE 802.16m standard is in the middle stage of standardization process. At this time, several documents that define requirements on a target system [17], methodology for an evaluation of simulation for proposed techniques [18], system description [19] and a working version of final standard [20] are available. Also the first draft [21] is completed and works on the second draft [22] are in progress.

Generally, a goal of this version is to provide an advanced air interface for operation in licensed bands. The standard should design a system with performance improvements necessary to support future services and applications specified by IMT-Advanced [23].

In target IEEE 802.16m system, the handover procedure shall be compatible with all previous IEEE 802.16 standards (see [19]). The handover procedure has to be improved (in comparison to IEEE 802.16e) especially in the meaning of minimization of a handover interruption time, sometimes called handover latency or handover delay (this is further addressed in section 2).

1.2 RELATED WORKS

Related works are divided into several subsections to address particular topics investigated further in this thesis.

1.2.1 HANDOVER IN WIMAX

A general principle of handover in mobile WiMAX networks is described e.g. in [2],[24].

An implementation of RSs leads to the modification of handover procedure as no wired connection is between BSs and RSs. The handover decision should be based on a new metric that consider specifics of RSs. An example of such a modification is a relay path and access station selection based on algorithm considering multiple QoS (Quality of Service) parameters [25], Radio Resource Cost (RRC) [26] or Expected Link Throughput (ELT) [27].

The new concept of hybrid handover in multihop radio access networks with RSs is addressed in [28]. This paper compares reactive and proactive handover approach from the overhead point of view. Complex modifications of handover procedure considering RSs are presented e.g. in [16] [29] [30] [31] [32]. Paper [16] introduces and describes the handover procedure for networks with RSs. This paper is further extended by proposal on all particular stages of handover in [29] – [32].

Some of these proposals are further exploited e.g. in [33] [34] where scanning overhead is reduced. The scanning overhead reduction is achieved by joint transmission of scanning requests and reporting messages from all MSs by an access station.

Other proposals of an implementation of RS and its support for handover procedure are described e.g. in [35] [36] [37].

1.2.2 PREDICTION OF TARGET BS

During the hard handover process a MS that is performing handover closes all connections with the serving BS and subsequently initiates a negotiation with the purpose of establishment of new connections with a target BS. After the connections with the serving BS are closed the MS is disconnected from the network until new connections to the target BS are setup. This short time break, known as handover interruption, handover delay or handover latency [18], should be minimized since it decreases QoS (see section 2.1.1). The handover interruption occurs if a hard handover is executed, i.e. when the MS always communicates with just one BS. However, the MS can be simultaneously connected to more then one BS. This type of communication during handover is generally called soft handover. In WiMAX, the soft handover is presented by MDHO and FBSS. To ensure an optimum performance of a network, the size of diversity set (a number of BSs in the diversity set) needs to be optimized according to the network conditions and signal quality in case of soft handover [38].

The minimization of interruption during hard handover as well as optimal size of diversity set in case of soft handover can be achieved by handover prediction.

Moreover, if proper and efficient handover prediction is performed, a number of unnecessary handovers can be reduced. The unnecessary handovers can be caused by so called "ping-pong" effect when the MS is continuously switched between two neighboring BSs since it is moving along the edge of cells' boundaries [39].

Another purpose of the handover prediction is to optimize an admission control as presented in [40] [41]. The utilization of the handover prediction for resource reservation for an admission control is also presented in [42]. The paper proposes two schemes of the admission control to optimize a utilization of dedicated bandwidth.

In [43], authors analyze effectiveness of the prediction with power consumption reduction purpose in adhoc networks. The reduction of power consumption is achieved by delaying of communication until a MS becomes closer to a target BS. The prediction is based on movement history of the MS. General approach of the prediction based on movement history is to determine consequent positions of user's based on its movement in the past as it is addressed e.g. in [44]. It assumes that user's are moving in compliance with a specific pattern of movement. However, the behavior of users is very variable and moreover this approach needs some time to adaptation to the individual user's. Several approaches that should improve the efficiency of this technique are described in literature. One of proposed schemes, presented in [45], determines a MS's location based on its quasi-domestic mobility behavior stored in the MS's profile. Consequent extension of this technique, described in [46], utilizes a modeling of a MS's movement according to the MS's previous elementary motion patterns. The prediction of user's position is further exploited for example in [47]. The paper presents advanced algorithms for a prediction of user's location. In [48], a multilayer neural network is utilized for a motion prediction. The prediction efficiency of this technique is over 90% for uniform user's movement. Novel mobility prediction utilizing real-word maps and user's available information (e.g. user's profile, schedule, etc.) is presented in [49].

Several filtering methods for the handover prediction are evaluated in [50]. Authors compare efficiency of the handover prediction for Grey [51], Kalman [52], Fourier [53] and Particle [54] filtering of RSSI (Receive Signal Strength Indication) values. The results show the best performance (roughly 80% of successful handover prediction) for no filtering and Grey filtering techniques. The Grey filtering technique is also analyzed in [55]. The paper evaluates and proofs positive impact of Grey prediction on the reduction of number of executed handovers. In [56], several techniques for handover prediction such as handover history, mobility pattern, movement extrapolation or distance are compared. The paper shows that the best performance (highest ratio of correct predictions) can be achieved by the prediction based on a mobility pattern or movement extrapolation for road mobility model or random waypoint mobility model [56] respectively. The prediction efficiency is approximately 60% in both cases.

Authors in [57] investigate two approaches of the handover prediction: cell and user. The cell approach predicts a number of users in the cell whereas the user approach utilizes a mobility prediction to determine information on next handover. The paper summarizes advantages of both approaches and their suitability for utilization in different scenarios. An extension of previous paper is presented in [58]. The authors propose new resource allocation mechanism that shows better performance for users approach if it dealing with reduction of the handover failures. On the other hand, the cell approach together with proposed resource allocation mechanism improves cell blocking probability.

In [59], authors describe a handover prediction based on a weighted combination of several network parameters such as bit rate, latency or power consumption.

1.2.3 Reduction of handover Interruption

An analytical analysis of handover is provided in [60]. It defines the impact of signal averaging and handover hysteresis margin on the handover interruption and handover overhead. The evaluation of handover interruption is presented in [61]. This paper evaluates the handover interruption over cell load ratio in several scenarios defined in IEEE 802.16e including e.g. fast ranging. The minimum achievable handover delay according to [61] is approximately 60 ms while fast ranging is considered. The analogical evaluation is described in [62] as well. It also provides further optimization based on the similar principle as [61]. Proposed method reduces handover interruption to 34 ms. Proposal of three different algorithms for the reduction of handover interruption and minimization of an amount of scanning processes are described in [63]. The reduction of handover interruption is accomplished by utilization of a target BS prediction. The evaluation is performed for two cell loads, i.e. 0% and 50%. The minimum handover interruption duration is about 175 ms.

The handover overhead and handover interruption is analyzed in [64]. The paper proposes relay assisted hard and soft handover (RAHO, RASH) procedures. According to presented results, both techniques bring significant reduction of the handover interruption (RAHO and RASH roughly 140 ms and 25 ms respectively). On the other hand, RASH and RSHO leads to the significant rise of overhead.

Different approach is defined in [65]. This paper proposes Last Packet Marking (LPM) procedure to reduce handover interruption. However, the LPM merges MAC layer and network layer features; therefore it is out of scope of IEEE 802.16 standards. In [66], authors present three schemes for handover with reduced handover interruption time. The most effective scheme, based on the reduction of time for uplink reservation for initial ranging, enables minimum handover interruption of 34 ms.

The hard handover modification that allows receiving downlink data just after synchronization with downlink channel of a target BS is proposed in [67]. The reduction of interruption is accomplished by introduction of new MAC management message called FastDL_MAP_IE. This message is used for the transmission of high priority packets (packets with payload of delay sensitive services) by the target BS to MS. This transmission can be used for sending of downlink packets before the MS finishes uplink synchronization with the target BS and before CID (Connection Identifier) update is completed. The CID assigned by a serving BS is used for a communication with the target BS. Therefore, a collision with existing CID can occur in the cell of the target BS.

The collision of CIDs can be solved by transport CID mapping scheme proposed in [68]. The transport CID assignment uses 3 bits of CID to differentiate neighboring BSs. The same paper introduces Passport Handover (PHO) to decrease the hard handover interruption. The most significant difference between PHO procedure and conventional IEEE 802.16e handover is in the ranging process. In the downlink, a serving BS sends QoS parameters to a target BS via backbone. After receiving these parameters, the target BS starts transmission using the same QoS parameters and the same CID (assigned by using of transport CID mapping) as were used by the serving BS. In the uplink direction, the MS checks the QoS level and starts data transmission immediately after receiving RNG-RSP. Re-authorization and re-registration steps are performed after the start of communication with the target BS. Hence, there is short interval within the MS communicates without updated authorization and registration with target BS. Another drawback is a significant reduction (eight times) of an amount of CID in frame of one cell. Also an assumption of only 6 neighboring BSs can be limiting especially in networks with RSs. The PHO reduces the handover interruption to 25 ms for downlink transmission and 80 ms for uplink for frame duration of 5 ms.

1.3 MOTIVATION, OBJECTIVES AND METHODS

The area of user's mobility support in form of the handover procedure is heavily investigated as described in previous section. One of the most challenging problem is an optimization of the handover procedure to minimize its negative impact on a QoS and thus fulfill the requirements on fourth generation of mobile wireless networks (4G) defined by IMT-Advanced [23]. The minimization of this interruption is the main goal of this thesis. To reach this general objective, three sub-objectives have to be addressed.

The first one is an analysis of dependence between duration of handover interruption and physical layer frame duration. For this purpose, the duration of handover procedure have to be determined. It enables to evaluate the impact of handover interruption on a speech quality of Voice over IP (VoIP). The VoIP speech quality represents a qualitative parameter of connection between a MS and a BS. This parameter is selected since the VoIP is largely utilized technique for voice communication nowadays.

Another sub-objective is a design of target BS prediction techniques that provides maximum efficiency of consequent serving BS prediction. The maximum efficiency means the as high as possible ratio of successfully predicted target BSs. The prediction technique must be applicable on current mobile wireless networks as well as on emerging ones.

The proposed target BS prediction technique has to enable to introduce novel handover procedure with reduced duration of handover interruption to fulfill IEEE 802.16m requirements (see section 4.1 or [17]). The novel handover could support conventional mobile networks as well as the networks with implemented RS. The design of novel handover should consist of proposal on new MAC management messages required for exchange of results of target BS prediction between BS and MS. Moreover, it should include also and exploitation of the target BS prediction results by modification of the conventional handover procedure. Besides the reduction of handover interruption, the thesis is focused on the analysis of proposed technique on overall overhead generated during a handover procedure. Furthermore, an analysis of new target BS prediction method on the throughput of a single MS should be considered. The proposed novel handover procedure should not increase the handover overhead and its impact on the throughput of MS's should be minimal.

The above mentioned goals have to be reached by conventionaly used working methods in the area of mobile wireless research. Therefore, all models for simulations are selected form the generally used models for evaluation. Moreover, most of them are based on document that defines evaluation methodology for just developing standard IEEE 802.16m [18]. This document is also in line with description of IMT-Advanced systems [23].

All evaluations of proposed techniques are done via simulations in MATLAB since it is common and universal simulation tool used for mobile networks. The following models are used in evaluations and simulations:

- Path loss model
  - Urban macrocell (see Appendix B1)
  - Urban microcell (see Appendix B2)
  - Models for shadowing (see Appendix B3) and channel variation (see Appendix B4)
- Mobility models
  - Probabilistic random waypoint/walk mobility model (see Appendix C1)
  - Manhattan mobility model (see Appendix C2)

1.4 STRUCTURE OF THESIS

The thesis is separated into five chapters. Each of them is focused on a main particular objective of this thesis. The rest of thesis is organized as follows.

The ***second chapter*** describes and analyzes the problem of handover interruption occurrence. Moreover, it also defines parameters of a model for evaluation of the handover interruption duration. This model is later utilized for analysis of the impact of handover interruption on speech quality in VoIP communication and on the throughput of single MS. The impact of techniques utilized for a reduction a number of handovers is also investigated in this subsection.

The ***third chapter*** focuses on the prediction of target BS that will serve a MS after the handover. Several techniques are discussed and than, prediction based on history of handovers and based on channel characteristics are evaluated from the prediction efficiency point of view. Novel approach to prediction using channel characteristics is outlined. The description of scenarios for simulations is contained in this chapter. Consequently, the simulation results are presented and analyzed. Further, efficiency of the prediction based on channel characteristics is improved by utilization of techniques originally designed for a reduction of amount of handovers.

The ***fourth chapter*** deals with the implementation of the prediction, investigated in third section, to the handover procedure to achieve minimal handover interruption. The complete handover procedure that enables to exploit the results of prediction is presented. The proposed handover technique includes the design of new MAC management messages as well as the complete flow of MAC management messages exchanged during the handover. Subsequently, evaluation of the duration of handover interruption for proposed technique is performed. The results are compared to the conventional IEEE 802.16e handover procedure. Moreover, the results for scenarios with RSs are also included in this chapter. As the novel handover procedure leads to several changes in MAC management message exchange in comparison to conventional one, the impact of proposal on handover overhead and MS's throughput is discussed.

The ***fifth chapter*** presents general conclusions of the whole thesis. Furthermore, it outlines possible ways of future investigation and future objectives of research work in the field of handover optimization.

2 ANALYSIS OF HANDOVER INTERRUPTION

2.1 DURATION OF HANDOVER INTERRUPTION

A handover interruption in mobile wireless systems is caused by switching of a MS from a serving BS to a target BS. Explanation of the interruption caused by the hard handover is presented in Figure 5. Before handover, the MS communicates with the serving BS (Phase 1 in Figure 5). All connections with the serving BS are terminated if the MS crosses a border of cells between the serving and target BSs (Phase 2 in Figure 5) and the MS has no connection to the network. Subsequently, new connections with the target BS are established (Phase 3 in Figure 5). The short interruption in connection begins when the MS gets disconnected from the serving BS and it lasts until the MS sets up new connections with the target BS. During interruption, all packets must be forwarded from the serving BS to the target BS via backbone. When the connections between the MS and target BS are established, the packets are transmitted to the MS.

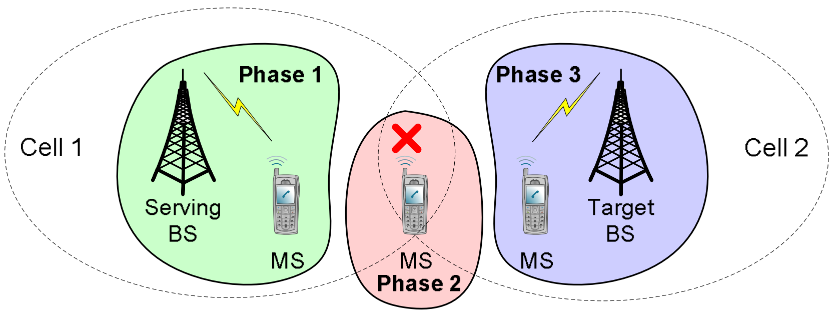


Figure 5. Interruption within hard handover

According to [2], the handover procedure can be separated into several stages: network topology advertisement, scanning of MS's neighborhood, cell reselection, handover decision and initiation, synchronization and network re-entry (see Figure 6).

First two stages, network topology advertisement and scanning of MS's neighborhood, are performed before the start of handover process. These stages enable the MS to investigate and collect information on all neighboring BSs. Within scanning process, the MS seeks for a suitable target BS or BSs that are appropriate to be added to a diversity set. The scanning is accomplished in so called scanning intervals which interleave normal operation of the MS. Once the scanning is finished, the MS sends results back to the serving BS. The scanning results can be delivered to the serving BS by two types of reporting. The first one is Event trigger reporting as the MS sends reports when certain triggering condition is met (e.g. if CINR drops below or rise above certain threshold). In the second type, Periodic reporting, the MS sends reports at regular intervals.

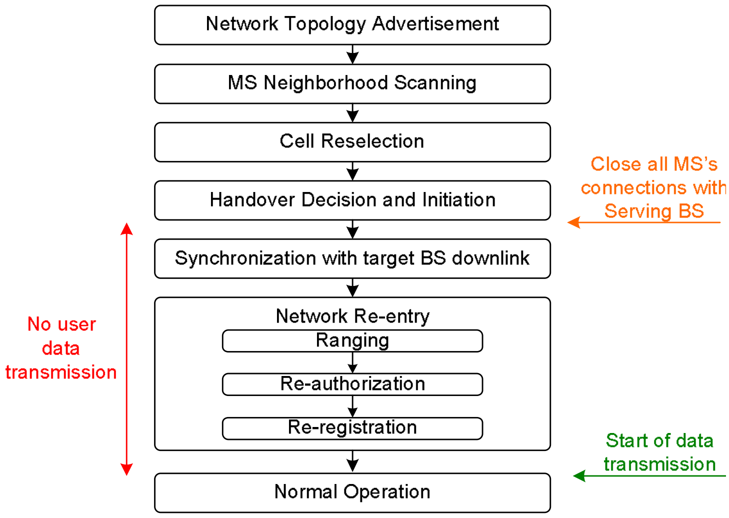


Figure 6. Phases of handover procedure

The results obtained during the scanning process are used in the next step of the handover procedure, i.e. cell reselection. In this step, a possible target BS is selected based on channel parameters and/or offered QoS. Afterward, handover decision and initiation phase is performed if all conditions and requirements for the handover are fulfilled. The first step after the handover initiation is MS's synchronization to the downlink channel of target BS. Before the synchronization is completed, all connections with the serving BS are closed and the MS cannot neither receive nor transmit data. This time corresponds to the beginning of handover interruption.

As soon as the synchronization with the downlink channel of target BS is finished, the MS can start next stage of handover – network re-entry procedure. The network re-entry consists of three substages: ranging, re-authorization and re-registration. At the beginning of the ranging process, the MS obtains information on an uplink channel through UCD (Uplink Channel Descriptor) message and information on resource allocation by means of UL-MAP (Uplink MAP) message. Consequently, ranging parameters (such as transmitting power, timing information or frequency offset) are exchanged. The ranging process is followed by the re-authorization and re-registration of the MS to the target BS. After the successful authorization and registration, the MS can start with normal operation. It means that the MS can resume data exchange since the handover interruption is over. All time intervals spent by interactions with the core network (i.e. with all network entities beyond the radio access network) are assumed to be zero according to [17].

The principle of both types of soft handovers (MDHO and FBSS) is based on a simultaneous communication with more than one BS (see [2] for more details). Therefore the duration of handover interruption is different in comparison to the hard handover.

In case of the MDHO, a MS and BSs have to maintain a diversity set. The MS communicates (including user traffic) simultaneously with all BSs in a diversity set. Therefore, if the diversity set contains more than one BS, no delay of data packets is introduced by the MDHO. The delay is similar as in case of the hard handover if just one BS is included in the diversity set since the same MAC management messages as during the hard handover are exchanged. Only the content of these messages is slightly modified.

In case of the FBSS, a situation is analogical as in the MDHO. A MS and BSs also have to maintain a diversity set. The MS transmits/receives data to/from all BSs in the diversity set, however only a communication with anchor BS is performed. The FBSS actually means just a switching of the anchor BS. This introduces no delay if the diversity set contains two or more BSs. If the diversity set includes just one BS, the handover interruption duration is the same as in the hard handover scenario.

A minimization of only hard handover interruption is considered for further analysis and investigation in this thesis since no handover interruption is introduced by the MDHO and FBSS with more than one BS contained in the diversity set. Moreover, as the hard handover is only mandatory in WiMAX systems, then assumption that the hard handover will be most often integrated type of handover in real networks must be accepted.

The impact of handover interruption on speech quality in VoIP is considered for further analysis in this thesis. The two facts support the selection of VoIP speech quality as qualitative parameter. Firstly, the VoIP communication is becoming more and more utilized and it is going to be the most profitable type of communication for users as well as for operators [69]. Secondly, the VoIP is generally accepted and generally supposed to be a major type of voice communication for 4G mobile networks [18].

2.1.1 IMPACT OF HANDOVER ON SPEECH QUALITY IN VOIP

As the hard handover procedure causes an interruption in connection, data flow of packets with voice are delivered to a user with increased delay.

Generally, three types of degradation can occur in VoIP: packet delay, jitter and packet loss. The packet loss can be solved by transmission of lost packet using ARQ (Automatic Repeat reQuest) mechanism. On the other side, ARQ increases a packet delay and a jitter. Thus, it is not usually used in VoIP communication. Hence, the communication with disabled ARQ is assumed. All of three types of VoIP speech quality degradation can be minimized or eliminated by some of Packet Loss Concealment (PLC) techniques (see e.g. [70] [71] [72] [73] [74]).

The packet losses and jitter caused by network as well as PLC techniques are not considered in consequent evaluation to separate only the impact of handover from other negative effects.

2.1.1.1 DEGRADATION OF SPEECH BY HANDOVER

A packet delay is affected by many factors (e.g. routing, signal propagation, etc.). Overall delay of packets during the handover can be defined by the following equation:

$$D_{TOT} = D_{HO} + D_{NET} \tag{1}$$

Overall delay consists of a delay caused by handover D_{HO} (handover interruption time) and a delay caused by transport network D_{NET}.

Duration of the handover interruption depends on a length of frames (frame duration) used in network on physical layer (PHY) since it corresponds to the time required for exchange of MAC management messages between a MS and a BS. Every phase of handover lasts certain time interval. Therefore every stage can increase the delay of packets. The overall delay caused by the hard handover can be expressed (according to [68]) by the next formula:

$$D_{HO} = T_{sync} + T_{cont\_res} + T_{rng} + T_{auth} + T_{reg} \tag{2}$$

where T_{sync} is a synchronization time; T_{cont_res} corresponds to a time dedicated for contention resolution procedure; T_{rng} represents a time spent by ranging process; T_{auth} expresses a time requested for re-authorization and T_{reg} is a time used for MS's re-registration.

The duration of every handover depends on a flow of MAC management messages and on the length of PHY frame. Hence the duration of each stage is equal to a sum of durations of particular messages exchanged within the stage. The length of PHY frame is invariable within the communication [2]. Thus the lasting of each stage (T_{stage}) is a multiplication of a number of frames utilized for the MAC message exchange during particular stage and the frame duration. Therefore, the general duration of a stage can be determined by the following formula:

$$T_{stage} = \sum_{i=1}^{n_{stage}} MMD_i = n_{stage} \times MMD_i = n_{stage} \times FD \tag{3}$$

where MMD_i is a duration of particular MAC management message; n_{stage} represents a number of frames required for exchange of all messages in the stage; FD is a frame duration. The FD is equal to the MMD_i since each MAC management message consumes just one frame [2]. Minimal and typical durations of all stages are summarized in Table 1.

Table 1. Minimal and typical values of components of handover interruption

Delay	Minimal value	Typical value
T_{sync}	1 frame	1-2 frames
T_{cont_res}	0 ms (dedicated ranging slot)	tens ms
T_{rng}	5 frames	6-9 frames
T_{auth}	3 frames + 2 frames for an SA	5 frames for 1 SA
T_{reg}	2 frames	2 frames

The ranging and re-authorization processes have the most significant impact on the overall handover delay (see Table 1). The length of re-authorization depends on a number of Security Associations (SAs) that have to be exchange. The SA defines a set of security parameters describing each data connection. Note that several data connections can use the same SA (for more details on SA see [2]).

The length of frame has to be considered in evaluation of the handover interruption time. The WiMAX enables to use following frame lengths: 2; 2.5; 4; 5; 8; 10; 12.5 and 20 ms [2]. In this thesis, three scenarios are defined for the analytical calculation of handover interruption (Table 2). *Scenario A* corresponds to the "typical handover without dedicated ranging slot". This means the numbers of frames are selected with respect to the values usually assumed in practice or simulations [68] if the utilization of dedicated ranging slot is not considered. The second scenario, *scenario B*, is analogical to scenario A. Both scenarios differ just in consideration of the dedicated ranging slot. The last scenario, *Scenario C*, represents "optimal handover" when all values are set to minimal levels that can be theoretically reach in real WiMAX systems.

Table 2. Parameters for calculation of handover duration

Delay	Duration of stage [frames]		
	Scenario A	Scenario B	Scenario C
T_{sync}	2	2	1
T_{cont_res}	2	0	0
T_{rng}	7	7	5
T_{auth}	3 + 2 per SA	3 + 2 per SA	3 + 2 per SA
T_{reg}	2	2	2

The delay of each packet due to the network, D_{NET}, is calculated based on the next equation (according to [75]):

$$D_{NET} = T_{EndTr} + T_{EndRc} + 2 \times T_{AcNet} + T_{CoreNet} \tag{4}$$

where T_{EndTr} represents a delay caused by end-device at transmitting side (incl. signal processing, packetization and serialization). Time T_{EndRc} is a delay incurred by receiving end-device. It is composed of speech processing and jitter buffer delay (no jitter buffer delay is assumed since the jitter is not considered to separate out the impact of handover). Parameter T_{AcNet} is a delay originated in access networks (two access networks are included in telecommunication chain, one access network at each communicating side). Finally, $T_{CoreNet}$ represents a delay introduced by core network. Parameters T_{AcNet} and $T_{CoreNet}$ include signal propagation, data serialization and queering. The parameters used for network delay evaluation are shown in Table 3. The values are defined with respect to [75].

Table 3. Parameters for network delay calculation

Delay	Duration
T_{EndTr}	25 ms
T_{EndRc}	5 ms
T_{AcNet}	30 ms
$T_{CoreNet}$	50 ms

The modification of speeches for evaluation of the handover impact on the speech quality and its assessment are presented in Appendix A.

2.1.1.2 RESULTS

Relation between the PHY layer frame duration and the handover interruption duration is shown in Figure 7. It is presented for 3 different values of SAs. The figure shows a linear dependence between the frame duration and the handover interruption. The minimal achievable value of handover interruption is 26 ms. This delay is caused by the optimal handover (*Scenario A*) with 1 SA. This is only value that fulfills requirements on IEEE 802.16m networks [17]. Significant impact of the number of SA on the handover interruption can be also observed from Figure 7.

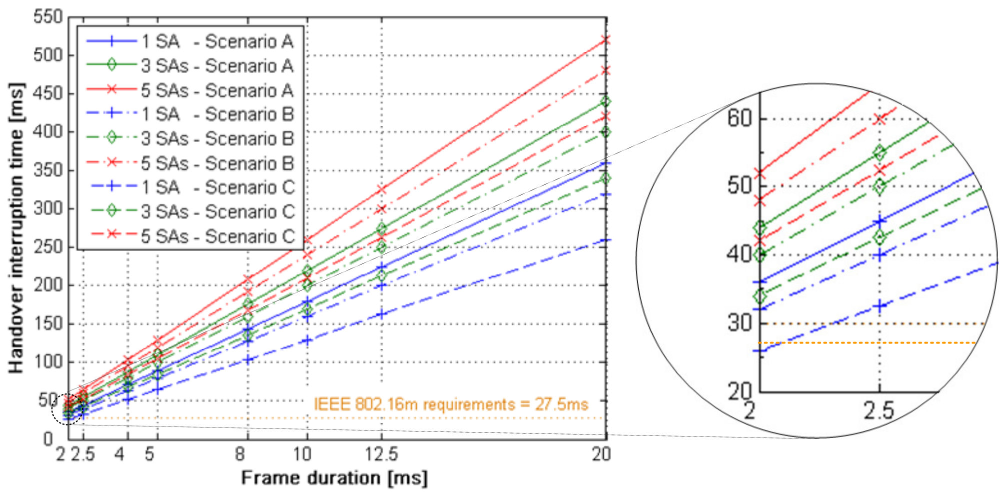


Figure 7. Duration of handover interruption over frame duration

The values of packet delay for speech quality evaluation are determined based on the results presented in Figure 7. Therefore, durations of 25; 50; 75; 150; 250 and 350 ms are selected for further investigation.

The results of the handover impact on the speech quality in VoIP are depicted in Figure 8. It presents the behavior of VoIP speech quality over duration of call with just one handover occurrence (*x* axis). In other words, it expresses an average time interval between two handovers that are results of multiple MS mobility model over 19 cell scenario (see [18]). The values considered on *x* axis are with regard to a distribution of Average Call Duration (ACD) presented in [76] and with taking into account a fact that just one handover can occurs during one call. According to [76], the ACD observed form one million calls is approximately 107 s. Moreover, roughly 35% of calls are shorter than 50 s and the most of call durations is under 100 s.

As can be observed from Figure 8, the handover interruption has negative impact on the speech quality. It is apparent especially in case of short calls (or short intervals between two handovers) when the decrease of speech quality is between 0.2 and 0.9 MOS depending on the duration of handover interruption. Marked fall of the speech quality is more noticeable with lengthening the duration of handover interruption. Lowering speech quality is noticeable even for short durations of the handover interruption simultaneously with long call durations (maximum speech quality of non degraded speech in plotted by dash line in Figure 8). Nevertheless, average speech quality for the handover interruption equal to 25 ms shows expressively lower drop of the speech quality (between 0.2 and 0.05 MOS) over all call durations. The speech quality degradation is significant for all longer duration of the handover interruption.

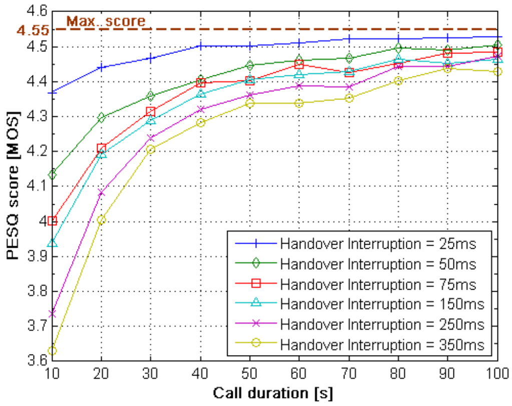


Figure 8. Dependence of VoIP speech quality over duration of handover interruption and call duration

Considering results presented in Figure 7 and Figure 8, the negative impact of frame duration on speech quality should be assumed. The exact evaluation of this effect is presented in Figure 9. Five call durations (20, 40, 60, 80 and 100 s) and *scenario B* are considered for evaluation.

The significant impact of frame duration is noticeable for short calls (20 s). The difference in the speech quality between 2 ms and 20 ms frame durations is up to approximately 0.25 MOS for this call duration. Increase of the call duration leads to the less significant drop of the speech quality over the PHY layer frame duration. It is less than 0.1 MOS in all other cases.

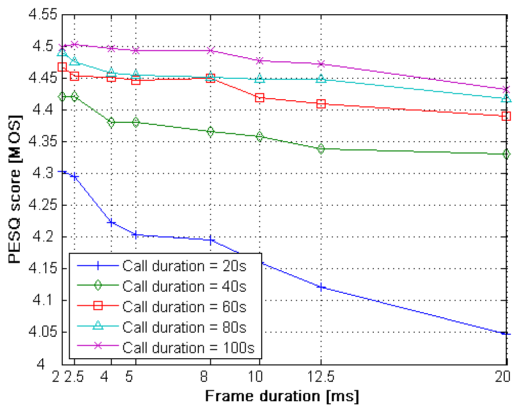


Figure 9. Impact of handovers on speech quality over frame duration

Results presented in Figure 9 leads to the conclusion that the improvement of speech quality can be achieved by utilizing shorter frames on PHY layer in all cases. However, the impact of frame duration on speech quality is marginal in case of calls with duration over 20 s.

2.2 SUPPRESSION OF NEGATIVE IMPACT OF HANDOVER

The handover interruption and its negative impact on the VoIP speech quality can be minimized by two ways:

- reduce a number of redundant handovers
- modification of MAC management procedure of handover

To improve the VoIP speech quality, PLC techniques, can be considered. However these techniques are focused on a signal processing of speech and not on the improvement of the handover procedure. Hence, no PLC techniques are further considered in this thesis.

2.2.1 REDUCTION OF NUMBER OF REDUNDANT HANDOVERS

Redundant handovers (or unnecessary handovers) represent a case when the handover is executed, however it is not finished before time when a next handover decision is made. Also a handover that is repeated several times between two adjacent cells can be considered as the redundant handover. Several techniques can be utilized for minimization of the number of redundant handovers caused by short time channel variation (e.g. fast fading or shadowing) or by movement of MSs along the edge of the two neighboring cells. Standard IEEE 802.16e defines Hysteresis Margin (HM) [2] [60] and Time-To-Trigger (TTT) [2] for elimination of the redundant handovers. Another commonly used technique is windowing (known also as signal averaging) [60]. Last method that will be considered is based on the similar principle as TTT. It is called Handover Delay Timer (HDT) [77] [78]. All methods are based on delaying of the handover for some time interval. During this interval, the MS is not connected to the station providing the best quality of communication channel. Therefore, it has negative impact on QoS provided to the MS due to the utilization of worse quality of the channel than a quality available from other BS. On the other hand, each stand alone method reduces the amount of redundant handover initiations.

2.2.1.1 TECHNIQUES FOR REDUCTION OF REDUNDANT HANDOVERS

The principle of all four techniques for reduction of amount of redundant handovers is briefly introduced in following subsections.

2.2.1.1.1 HYSTERESIS MARGIN

The handover decision and initiation is based on a comparison of one or several signal parameters (CINR, RSSI, Round Trip Delay (RTD) or relative delay) of a serving and target BS. The handover is initiated if the signal parameter of target BS exceeds the signal parameter of serving BS plus *HM*.

$$S_i^{Tar} > S_i^{Ser} + HM \tag{5}$$

where S_t^{Ser} and S_t^{Tar} represents a signal quality parameter of the serving and target BS respectively.

The disadvantage of this principle is that it cannot eliminate rapid variation in observed parameter (e.g. fast fading [79]). Moreover, it cannot cope with short time shadowing with decrease of signal higher than *HM* as it compares only current values of observed parameter.

2.2.1.1.2 TIME TO TRIGGER

The handover initiation is accomplished after short period within the signal parameters from a target BS are higher than parameters of a serving BS. It can be described by the following equation:

$$S_t^{Tar} > S_t^{Ser} \big| t \in (t_{HO}, t_{HO} + TTT) \tag{6}$$

where t_{HO} corresponds to a time when the handover decision would be done if no other technique for handover elimination is considered, and *TTT* is a duration of Time-To-Trigger timer. Standard [2], enables to use TTT duration with following values: TTT$\in$ (0, 255 ms).

In comparison to the HM, this technique monitors signal parameters for a short time interval. Therefore, it enables to deal with fast fading. On the other hand, a MS has to monitor signal parameters for a whole duration of TTT. It leads to the reduction of throughput during TTT. Furthermore, very low level of maximum duration of TTT limits the effect of this technique (e.g. it cannot fully eliminate ping-pong effect or shadowing with duration over 255 ms).

2.2.1.1.3 WINDOWING

The handover decision is done if an average value of observed signal parameter from target BS drops under an average level of the same parameter at serving BS (see formula (7)). The average value is calculated over a number of samples, denoted as Window Size (*WS*).

$$\frac{\sum_{i=1}^{WS} S_i^{Tar}}{WS} > \frac{\sum_{i=1}^{WS} S_i^{Ser}}{WS} \tag{7}$$

The efficiency of elimination of redundant handovers that are result of ping-pong effect, shadowing or fast fading depends heavily on the value of WS.

2.2.1.1.4 HANDOVER DELAY TIMER

Technique HDT is developed with purpose to cope especially with temporary drop of a signal level due to fast fading or when a user is located on shadowed places for a short time interval (longer than Reporting Period (RP)) [78]. Additionally, it enables a reduction of ping-pong effect.

According to the IEEE 802.16e version of WiMAX [2], the handover starts immediately after the channel conditions (e.g. signal levels) reach a threshold level. However the handover must be canceled (if it has not finished yet) or must be performed again (if it has finished) when a MS moves from the shadowed place.

Implementation of the HDT results into insertion of a short delay between the time when handover conditions are met and the time when the handover initiation is carried out (see Figure 10). This delay is noted as HDT (HDT=2×RP in Figure 10).

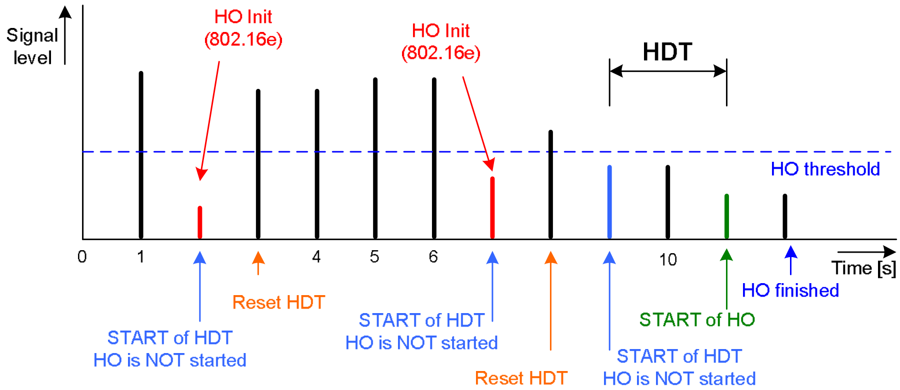


Figure 10. Handover initiation with HDT

These conditions for the handover have to be fulfilled over the whole duration of HDT to execute handover initiation. Generally, the handover is performed only if:

$$S_t^{Ser} < S_t^{Tar} \Big| t \in (t_{HO}, t_{HO} + HDT) \tag{8}$$

where *HDT* represents a duration of the handover delay timer.

As the signal level is measured and reported to the serving BS in discrete time interval (not continuously), the handover decision is done if exact number of consequent samples ($n_{samples}$) fulfills handover conditions as expresses the next equation:

$$S_i^{Ser} < S_i^{Tar} \Big| i \in (1, n_{samples}) \tag{9}$$

The $n_{samples}$ is equal to an amount of a channel quality reports sent during HDT from the MS to the BS (n_{rep}) as it is defined by the following formula:

$$n_{samples} = \frac{HDT}{n_{rep}} \tag{10}$$

If the periodic reporting is considered, the reports are transmitted in regular time intervals (equal to the reporting period *RP*). Then the $n_{samples}$ can be derived as:

$$n_{samples} = \frac{HDT}{RP} \tag{11}$$

As the HDT is based on the TTT, only HDT is considered for further evaluations.

2.2.1.2 IMPACT OF HM, HDT AND WINDOWING ON MS'S THROUGHPUT

All above mentioned techniques enable to reduce a number of handovers [80], however it is at the cost of decrease of throughput since all of them results in a postponement of handover execution. The delay of handover execution leads to the utilization of lower Modulation and Coding Scheme (MCS) [2] for communication between a MS and its serving BS. The impact of HM, HDT and windowing on the user's throughput is investigated for scenario with single MS (see Figure 11).

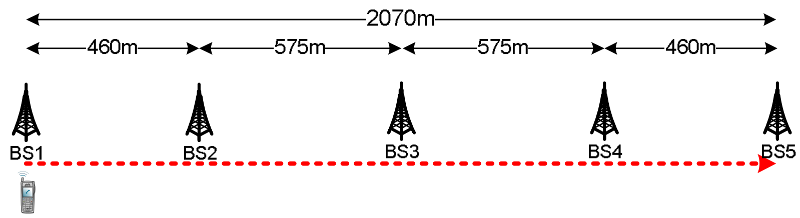


Figure 11. Scenario for evaluation of impact of all techniques on throughput

The MS moves along the straight line crossing 5 BSs. Speed of the MS is 10 m/s. The scanning reporting period is setup to 0.5 s since this value is close to an optimum scanning interval for maximization of throughput [81]. All parameters for evaluation are summarized in Table 4.

Table 4. Simulation parameters for evaluation of throughput of single MS

Parameter	Value
Number of BS [-]	5
Number of MS [-]	1
BS transmitting power [dB]	36
BS height [m]	32
MS height [m]	2
MS speed [m/s]	10
Frequency band [GHz]	2.5
Frame duration [ms]	20
Data subcarriers per sub-channel	48
OFDMA symbols per frame	198
Bandwidth [MHz]	20
Hysteresis Margin [dB]	0-20
Window Size [samples]	1-20
Duration of HDT [s]	0-5
Scanning reporting period [s]	0.5
Path loss model	802.16m Urban Macrocell

The throughput of single MS is calculated based on the received signal quality. The full buffer traffic model is assumed. It means that the MS has always a full queue of data to transmission. This model is often used in simulation as it enables to evaluate maximum efficiency of a system [82]. The strength of received signal is used for a calculation of number of data carried per one downlink sub-channel (Data Per Sub-channel – *DPS*).

$$DPS = NoDSC \times N_{ob} \times CR \tag{12}$$

where *NoDSC* express a number of data subcarriers per a sub-channel in PUSC (Partial Usage of Sub-Channels) OFDMA (Orthogonal Frequency Division Multiple Access), *CR* represents Code Rate and N_{ob} is a number of bits carried per one subcarrier (depending on used modulation scheme). The N_{ob} is derived from the next equation:

$$N_{ob} = log_2(N_{States}) \tag{13}$$

where N_{states} is a number of modulation states.

Every frame can be divided into sub-channels and a number of frames transmitted per second depends on the frame duration (*FD*). Therefore, the final bit rate in each step (BR_{step}) can be evaluated by the following way:

$$BR_{step} = \frac{DPS \times SPF}{FD} \tag{14}$$

where *SPF* represents number of sub-channels per frame.

The average throughput over the simulation duration (*AvgBR*) is calculated as a weighted average of the all throughput obtained during simulation (see consequent formula).

$$AvgBR = \frac{\sum_{Step=1}^{N_{steps}} \left(BR_{Step} \times StD\right)}{SimD} \tag{15}$$

where *StD* is a duration of a simulation step, *SimD* is a duration of whole simulation and N_{steps} represent overall number of steps during simulation. It can be calculated according to the following equation:

$$N_{steps} = SimD / StD \tag{16}$$

Figure 12 and Figure 13 show results of the impact of HDT and windowing on the throughput of single MS. The evaluation considers several levels of HM. Significant decrease of the MS's throughput is noticeable form both figures especially with increasing HM. In both cases, the reduction of throughput for low HM is not so rapid if shorter duration of HDT or lower amount of averaged samples are considered. On the other hand, the fall of throughput over the duration of HDT or WS is getting more linear for higher level of HM.

The impact of only HM is depicted in Figure 14. The HM leads to the minor drop of throughput at lower levels of HM. Then the reduction of MS's throughput is more significant.

Note that the impact of all techniques on throughput depends heavily on a deployment of BSs described by function Θ_{BS} and on a time interval between two handovers (HO_{per}). The HO_{per} is influent by a speed of user (υ_{MS}) and a movement of user (χ_{MS}). Therefore, the average bit rate can be expressed as the following function:

$$AvgBR = f(\Theta_{BS}, HO_{per}) = f(\Theta_{BS}, \nu_{MS}, \chi_{MS}) \tag{17}$$

Higher υ_{MS}, higher density of BSs and more direct movement of MSs among BSs leads to the decrease of the negative impact of all these techniques on the throughput of single MS.

Combined efect of HDT and WS over HM on the throughput is depicted in Figure 15. The figure shows that the incresing of one of parameter while the others are constant leads to nearly linear reduction of the throughput (compare the spacing between lines with same color or with same marker at the constant HM).

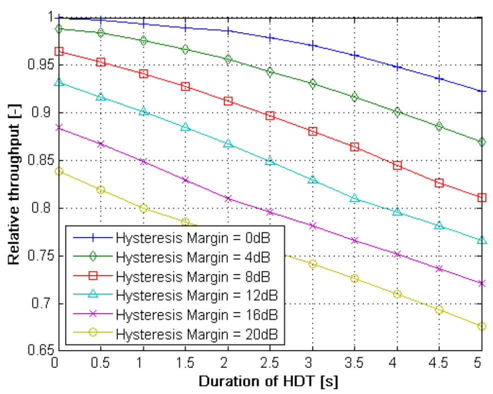


Figure 12. Impact of HDT duration on throughput of single MS

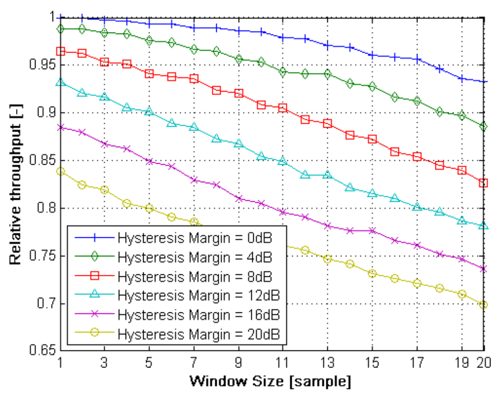
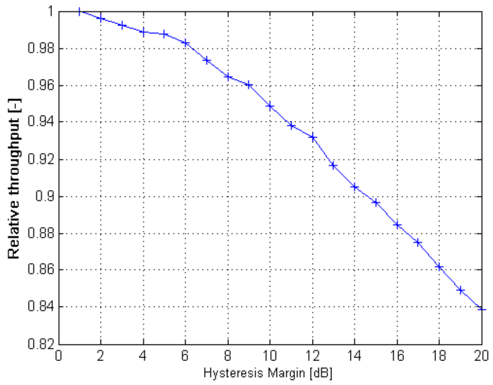


Figure 13. Impact of Window Size on throughput of single MS



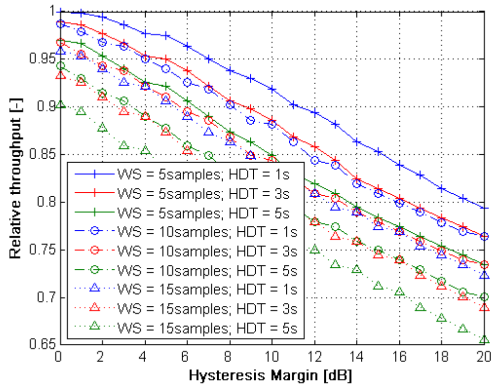


Figure 14. Impact of HM duration on throughput of single MS

Figure 15. Joint impact of HDT and WS on throughput of single MS

2.2.2 MODIFICATIONS OF HANDOVER MAC MANAGEMENT PROCEDURE

The second way of the reduction of handover interruption is to modify a flow of MAC management message during the handover procedure. A lot of papers deal with this idea as stated in section 1.2.3. Generally, minimization of the handover interruption can be achieved either by a reduction of number of MAC management messages exchanged during the handover or by a shifting of a part of the handover before time when a MS close connections with serving BS. The way based on performing of a part of the handover procedure before the MS closes all connections with serving BS can exploits results of prediction of target BS (see section 1.2.2).

The prediction of the handover as well as the approach based on utilization of prediction results with purpose to reduce the handover interruption are further analyzed in the followings chapters of this thesis (chapter 3 and 4).

2.3 CONCLUSION

The duration of interruption caused by hard handover depends on the duration of following stages: synchronization of a MS to a target BS, contention resolution and network re-entry. The duration of each particular stage depends on the number of MAC management messages exchanged between both stations and on the frame duration utilized on PHY layer of WiMAX system. Moreover, another factor that influents the handover interruption is a number of SA. The duration of handover interruption linearly increases as rises the number of SA. Conventional handover procedure according to IEEE 802.16e can fulfill requirements on the handover procedure defined in [17] only for scenario with assumption of “optimal handover” and with the frame duration equal to 2 ms.

Not only the handover interruption but also network delay must be considered to determine the impact of handover interruption on VoIP speech quality. The speech quality is decreased by the handover interruption. The level of speech quality degradation depends on the duration of handover interruption as well as on the duration of call within just one handover occurs. The result shows that the handover interruption has negative impact on the speech quality even for its short durations and long-lasting calls. Nevertheless, the average speech quality is increasing with shortening the handover interruption.

The abovementioned results lead to the assumable conclusion that the speech quality is influenced by the frame duration. The simulation results of the impact of frame duration on the speech quality confirm this assumption. Rise of the speech quality by utilization of shorter frames is reach over all call durations. Additionally, decreasing of the positive impact of frame duration on speech quality with prolonging call duration can be observed from presented results. However, the significant speech quality improvement (0.25 MOS) is achieved only for call duration of 20 s. The calls with duration over 20 s show only marginal increase of the speech quality (less than 0.1 MOS).

Moreover, the negative impact of handover can be minimized by reduction of a number of handovers or by modification of MAC management message flow. The first approach utilizes techniques developed exactly for this purpose. All of these techniques lead to the postponement of the handover execution time instant. Hence, it results to the decrease of MS’s throughput.

Following parts of the thesis are focused on the reduction of handover interruption by approach based on the possible modification of handover procedure. A prediction of target BS is considered as technique that can enable to cope with the handover interruption.

3 HANDOVER PREDICTION

The minimization of handover interruption or minimization of overhead generated due to the handover procedure can be accomplished by a prediction of handover.

The prediction of handover is a process of estimation of next target BS for a moving MS. Knowing in advance the target BS allows modifications in a handover management that lead to a fast handover with minimal interruption. Besides, it allows a reduction of MAC management overhead due to optimization of a list of neighboring stations for scanning.

3.1 PRINCIPLE OF PREDICTION

Generally, the prediction of target BS can be based on several approaches. The first one is a monitoring of pairs of serving and corresponding target BSs of MS's handover in the past. The second approach is a handover prediction based on characteristics of communication channel. The last way of the prediction is based on knowledge of user's position and position of all neighboring BSs. The objective of the last type of prediction is usually to forecast future motion of MS.

3.1.1 HANDOVER HISTORY

The prediction based on handover history utilizes results of a monitoring of pairs of serving and corresponding target BSs of MS's handover procedure in the past. This method requires to monitor and register all updates of the serving and target BSs of all MSs in network. It means, if a MS performs handover, identification of the serving and target BSs are stored into memory.

Handover history based prediction can be managed either by a MS or by BSs (by network). However, as the MS has no access to up to date information on the handover of another MSs, it is more efficient to manage it by BSs. Furthermore, the management of prediction by the MS leads to the prolonging of time necessary for collection of enough information in order to guarantee high probability of successful prediction.

Amounts of the handovers among BSs are represented by a matrix (see example in Table 5). The matrix has the same number of rows (x) and columns (y). The number of rows and columns corresponds to a number of neighboring BSs. Every field of the matrix represents a count of handovers between *BSx* (serving BS) and *BSy* (target BS) within observed time interval. For example, the field in the third row and sixth column represents the amount of handovers from BS3 to BS6 (79 handovers). A number of handovers performed in the opposite direction (from BS6 to BS3) is presented by field in the sixth row and third column (21 handovers).

Table 5. Example of matrix representing number of handovers among BSs

HO count.	BS1	BS2	BS3	BS4	BS5	BS6	BS7
BS1	x	31	0	35	34	0	0
BS2	16	x	51	33	0	0	0
BS3	0	18	x	3	0	79	0
BS4	28	41	13	x	8	6	4
BS5	11	77	0	0	x	0	12
BS6	0	0	21	13	0	x	66
BS7	0	0	0	20	23	57	x

The situation that corresponds to Table 5 is depicted in Figure 16 for scenario with 7 BSs.

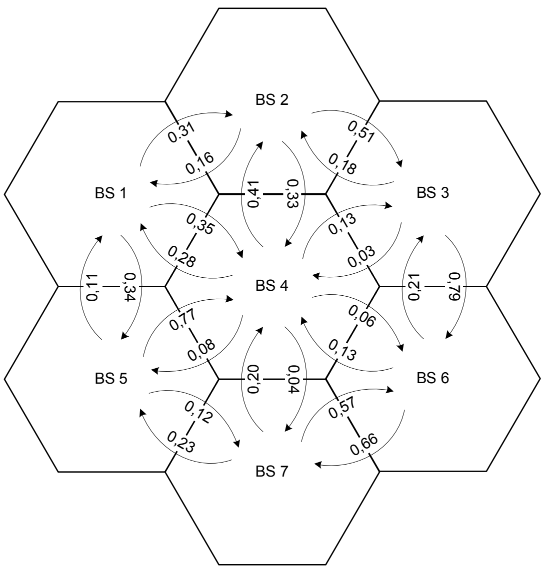


Figure 16. The probabilities of handover among neighboring BSs

The matrix of number of handovers is recalculated to the probability of handover from a serving BSx to a target BSy ($P_{x,y}$) according to the following formula:

$$P_{x,y} = \frac{HO_{x,y}}{\sum_{y \in NS_x} HO_{x,y}} \tag{18}$$

where $HO_{x,y}$ represents a number of handovers form BSx to BSy; all neighboring BSs of BSx are included in so called Neighbor Set of BSx denoted in thesis as *NSx*. Based on the NS, the probability of handover from BSx to BSy can be rewritten as:

$$P_{x,y} = \begin{cases} \alpha, & y \in NS_x \\ 0, & y \notin NS_x \end{cases} \tag{19}$$

where $0 \leq \alpha \leq 1$. The α depends not only on the number of BSs in the NS but also on the layout of area where the situation is analyzed and monitored.

An example of the matrix of probabilities calculated according to Table 5 and equation (18) is shown in Table 6.

Table 6. Matrix of handover probabilities among BSs

HO prob.	BS1	BS2	BS3	BS4	BS5	BS6	BS7
BS1	x	0.31	0	0.35	0.34	0	0
BS2	0.16	x	0.51	0.33	0	0	0
BS3	0	0.18	x	0.03	0	0.79	0
BS4	0.28	0.41	0.13	x	0.08	0.06	0.04
BS5	0.11	0.77	0	0	x	0	0.12
BS6	0	0	0.21	0.13	0	x	0.66
BS7	0	0	0	0.20	0.23	0.57	x

Figure 16 assumes the close area with no possibility of accomplishing the handover to another BS except BS1 – BS7. The situation when all MSs performing handover during monitored time interval is assumed. Thus, the sum of all probabilities in rows of Table 6 is always equal to 1 as expresses the next equation:

$$P_{x,y|x=const} = \sum_{\forall y | y \in NS_x} P_{x,y} = 1 \tag{20}$$

If the real system in which every BS has defined NS is considered, sum of the probabilities of MS's handover from BSx to one of all neighboring BSs tends to 1 since the handover in infinite future ($t \to \infty$) is assumed. The probability can be lover than 1 if $t < \infty$ since the MS can never perform handover. Therefore, the handover probability between BSx and all neighboring BSs over finite and infinite time interval is:

$$\begin{aligned} P_{x,y|t=\infty} &= \sum_{\forall y|\, y\in NSx} P_{x,y} = 1 \\ P_{x,y|t<\infty} &= \sum_{\forall y|\, y\in NSx} P_{x,y}^{t} \in \langle 0,1 \rangle \end{aligned} \tag{21}$$

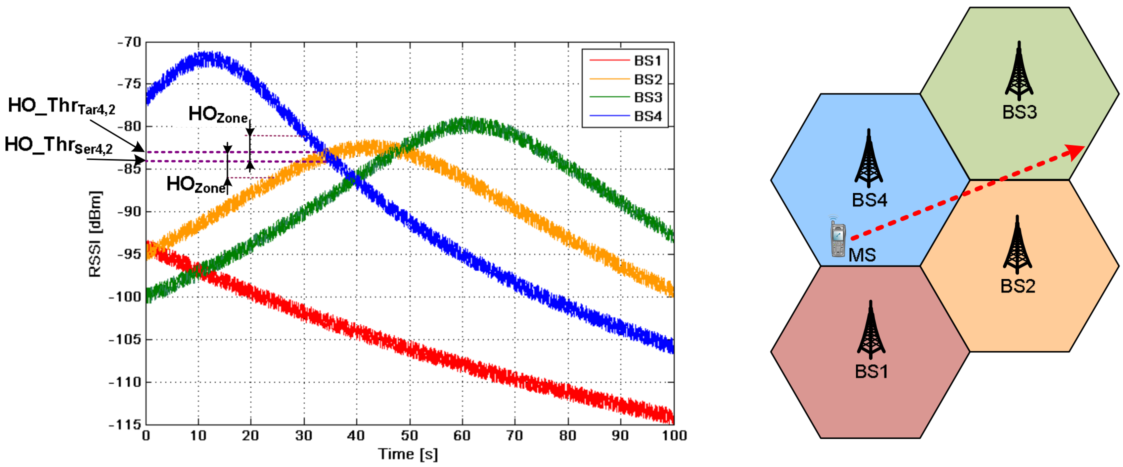
where $P_{x,y}^{t}$ is a function of time t and *NNSx* represents a number of BSs in the NS of BSx.

3.1.2 CHANNEL CHARACTERISTICS

The approach that uses channel parameters or characteristics to predict a target BS assumes to store results of scanning of MSs' neighborhood. Based on the stored information, typical thresholds for the handover between two neighboring BSs are derived. The results can be stored either in MSs or in BSs. Therefore, the prediction can be done either by the MSs or by the BSs. Nevertheless, since a lot of data have to be kept in station's memory, it is preferred to perform prediction by the BSs due to a memory limitation of MS. Further, the BSs can collect information from all MSs (BSs has knowledge about all handovers in the network) whereas the MS can collect only information on itself. The collection of information by the MS results in significant prolonging of time necessary for collection of enough information in order to guarantee high probability of successful prediction as in case of the prediction based on handover history.

The typical thresholds are based on the observation of channel characteristics corresponding to the signal level that triggers handover initiation. Figure 17a depicts RSSI between a MS and several BSs during the MS's movement along a straight line (red dashed trajectory in Figure 17b). The speed of MS is 20 m/s and observation time is 100 s, i.e. the distance covered by the MS within one observation cycle is 2000 m. Each curve in Figure 17a represents the set of RSSIs received by the MS from all BSs obtained within several movements of the MS. Minor fluctuation of RSSI is caused by variations of channel parameters among all runs of the MS. Simulation parameters and channel model are taken from [18].

Usually, only one threshold for handover decision is defined as described e.g. in [2] [26] [59]. The proposed improvement of this technique is based on definition of two thresholds as depicted in Figure 17a.



(a) (b)

Figure 17. Definition of handover threshold (a) based on movement of MS along the same direction (b)

Following two thresholds are defined: $HO\_Thr_{SerX,Y}$ and $HO\_Thr_{TarX,Y}$. The first one represents a typical RSSI level of serving BSx when a handover to target BSy is initialized (see Figure 17a where the serving BS is BS4 and the target BS is BS2 at the beginning phase of the MS's movement). The second threshold corresponds to a typical RSSI level of the predicted target BSy when the MS initiates the handover from serving BSx. Both threshold levels are usually very close in the most cases in praxis. In principle, the level of $HO\_Thr_{TarX,Y}$ is slightly higher as the handover decision is generally done only if the target BS can provide higher quality of connection than the serving BS. Moreover, both thresholds also differ due to non-stationary signal levels. The monitoring and evaluation of signal evolution from all neighboring BSs is performed in the following manner:

- If RSSI from the serving BS decreases and draws near to the $HO\_Thr_{SerX,Y}$, the probability of handover from BSx to BSy increases (in Figure 17a BSx is BS4, BSy is BS2 and $HO\_Thr_{SerX,Y}$ is $HO\_Thr_{Ser4,2}$).
- If RSSI from one of neighboring BSs increases and draws near to the $HO\_Thr_{TarX,Y}$, the probability of handover from BSx to BSy increases (in Figure 17a $HO\_Thr_{TarX,Y}$ is $HO\_Thr_{Tar4,2}$).

- If RSSI of the serving BS decreases under $HO\_Thr_{SerX,Y}$ and simultaneously RSSI of a neighboring BS exceeds $HO\_Thr_{ThrX,Y}$, the result of prediction is an estimation of MS's handover from the BSx to BSy. It means, the BSy is identified as "predicted target BS".

Both groups of thresholds for each pair of BSx and BSy (thresholds $HO\_Thr_{SerX,Y}$ and thresholds $HO\_Thr_{ThrX,Y}$) are averaged out in order to find one specific value for the serving BS threshold as well as one value for the target BS threshold. The mean values of typical thresholds for handover are calculated as an average of several previous signal levels that leads to the handover initiation. Determination of sufficient number of signal level samples is an object of investigation addressed further in this thesis. The values of samples' signal level are obtained within MS's scanning of its neighborhood. The mean thresholds can be described by the next equations:

$$AvgHO\_Thr_{SerX,Y} = \frac{\sum_{i=1}^{HOCount_{BSX,BSY}} RSSI_{MS,BSX}}{HOCount_{BSX,BSY}} \quad (22)$$

$$AvgHO\_Thr_{TarX,Y} = \frac{\sum_{i=1}^{HOCount_{BSX,BSY}} RSSI_{MS,BSY}}{HOCount_{BSX,BSY}} \quad (23)$$

where $HO_{CountBSX,BSY}$ represents a number of handovers occurred between the current serving BS and potential target BS during observed time interval.

The target BS cannot be predicted according to an exact level of typical thresholds since the prediction would be done too late (not enough time in advance to exploit results of prediction). Therefore, the handover should be predicted if the parameters of channel between the MS and the serving and target BS are within intervals of HO_{Zone} defined by following formulas:

$$HO\_Thr_{SerX,Y} + HO_{Zone} < RSSI_{MS,BSX} \quad (24)$$

$$HO\_Thr_{TarX,Y} - HO_{Zone} > RSSI_{MS,BSY} \quad (25)$$

where HO_{Zone} represents an interval where the *BSy* is marked as the predicted target BS (see Figure 17a), $RSSI_{MS,BSX}$ and $RSSI_{MS,BSY}$ correspond to a signal level received by the MS from the serving and target BS respectively.

According to the *NS*, the probability of handover from BSx to BSy can be formulated, likewise in the case of prediction based on the handover history, by the consequent equation:

$$P_{x,y} = \begin{cases} \alpha, & BSy \in NS_x \\ 0, & BSy \notin NS_x \end{cases} \tag{26}$$

where $0 \le \alpha \le 1$. The α depends not only on the number of BSs in *NS* but also on the layout of area where the prediction is analyzed and monitored (e.g. layout of streets, deployment of buildings and transmitters, etc.).

The $P_{x,y}$ is a function of HO_{Zone} as well as a function of current values of $RSSI_{MS,BSX}$ and $RSSI_{MS,BSY}$. If the α represents a value of probability $P_{x,y}$, the probability of handover from BSx to all BSs excluding BSy ($P_{x,\bar{y}}$) is 1-α. The probability of case when the MS will not handover to the predicted BSy ($P_{x,\bar{y}}$) in spite of the fact that the BSy fulfills the conditions according to (24) and (25), is expressed by ξ function. This can happen e.g. when the MS randomly turns away from the assumed direction. Then the probability of handover from BSx to BSy could be described by a function formulated in the owing way:

$$P_{x,y} = f(RSSI_{MS,BSX}, RSSI_{MS,BSX}, HO_{Zone}, \xi) \tag{27}$$

The ξ is directly proportional to the probability that the MS will not change direction in subsequent time interval τ so significantly that the MS would perform handover to the different BS. This probability is labeled as P_{MS}^{SD} . Thus, the ξ can be formulated as:

$$\xi = f(\tau, P_{MS}^{SD}) \tag{28}$$

and τ is a function of speed and distance as imply the next formula:

$$\tau = f(v_{MS}, Dist_{MS,cell}) \tag{29}$$

where v_{MS} is a velocity of the MS and $Dist_{MS,cell}$ is a distance of the MS from the place where the handover should be executed. Then the equation (27) can be rewrite as follows:

$$P_{x,y} = f(\, RSSI_{MS,BSX}, RSSI_{MS,BSX}, HO_{Zone}, v_{MS}, Dist_{MS,cell}, P_{MS}^{SD}\,) \tag{30}$$

The previous formula can be expressed in other way as the probability of successful prediction of target BSy if the MS is moving out of the coverage area of serving BSx. It is obvious that the probability of successful prediction is increasing either with lowering $Dist_{MS,cell}$ or with decreasing of difference between actual values of $RSSI_{MS,BSx}$ and $AvgHO\_Thr_{SerX,Y}$ as well as $RSSI_{MS,BSy}$ and $AvgHO\_Thr_{TarX,Y}$. On the other hand, drop of P_{MS}^{SD} or v_{MS} can lower the probability of successful handover prediction. The impact of HO_{Zone} is investigated later in this thesis.

Since the MS can arrive into the area in which more than one potential target BS fulfill the conditions for prediction, a mechanism for selection of just one predicted target BS should be defined. This mechanism is based on the calculation of minimum difference of both thresholds ($AvgHO\_Thr_{SerX,Y}$ and $AvgHO\_Thr_{SerX,Y}$) and current RSSIs between the MS and serving and target BS. This is done for all possible target BSs with RSSI values in the range defined by (24) and (25). These stations are listed in *ListOfTargetBS*. The decision about target BS that will be labeled as the predicted target BS is done according to the results of next equation:

$$Diff_{BSX,BSY} = \left| AvgHO\_Thr_{SerX,Y} - RSSI_{MS,BSX} \right| + \left| AvgHO\_Thr_{TarX,Y} - RSSI_{MS,BSY} \right| \tag{31}$$

The differences of RSSI levels and thresholds for each of potential target BS are compared afterwards and as the predicted target BS is selected the BS with minimum $Diff_{BSX,BSY}$ (see next formula).

$$PredictedTargetBS = \{Y\} \big| ListOfTargetBS_{Y} = min(\, ListOfTargetBS\,) \tag{32}$$

Other signal quality parameters such as CINR or SNR (Signal to Noise Ratio) or other parameters that express the quality of channel between a MS and BS can be used for prediction instead of RSSI. The parameter utilized for prediction has to correspond to the metric used in scanning reports sent by the MS to serving BS (see [2]).

To achieve as high prediction efficiency as possible, the techniques for reduction of redundant handovers (HM, HDT and Windowing) are considered and their impact is analyzed later in the thesis.

3.1.3 MOTION OF MS

This type of prediction is based on the monitoring of users geographic location. The general approach is to determine consequent positions of MSs based on its movement in past as it is addressed e.g. in [44].

Three general assumptions must be fulfilled for efficient motion prediction of MS [49]:

- knowledge of current and previous exact positions of the MS
- knowledge of users profile
- knowledge of profile of area in which the prediction is performed

The knowledge of exact position implies a utilization of localization equipment such as GPS (Global Positioning System) [83] to ensure exact determination of MS's position.

The second bullet results to requirement of acquisition of user's information such as area of interests, favorite places, time schedule, etc. It means that each user has to fill up this information and keep it updated.

Last assumption requires the knowledge of real maps of area in which the prediction is executed. This information can be acquired by a provider of network connection and must be also keep updated. This principle of utilization of real maps is depicted in Figure 18. The MS is connected to BS3, when it comes to the cross, it should turn to the left or right. Then the knowledge of MS's direction enables to predict target BS (turn to right or left leads to the prediction of BS2 or BS1 respectively).

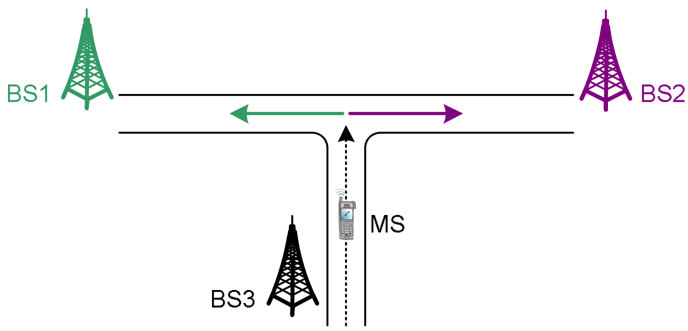


Figure 18. Utilization of map's knowledge to target BS prediction

Fulfillment of all above mentioned assumptions lead to very high demands on user, MS and network and it also makes the prediction not much comfortable for users. Therefore, the prediction of MS's movement is not furthermore an object of investigation in this thesis.

3.2 Scenarios for Evaluation of Handover Prediction Efficiency

The efficiency of target BS prediction is evaluated for specific scenarios depending on the type of prediction.

3.2.1 Handover History

A street scenario corresponding to the Manhattan Mobility Model (MMM) [84] (see Appendix C2) is used for the simulation of efficiency of handover prediction based on the handover history. The parameters of simulation scenario are presented in Table 7.

Table 7. Simulation parameters and scenario definition for handover history

Parameter	Value
Number of BS [-]	15
Number of MS [-]	48
BS transmitting power [dB]	46
BS height [m]	32
MS height [m]	2
MS speed [m/s]	15
Frequency band [GHz]	2.5
Frame duration [ms]	10
Scanning reporting period [s]	1
Simulation duration [s]	10800
Hysteresis margin [dB]	1
Path loss model	802.16m LOS/NLOS Urban Microcell
Mobility model	Manhattan
Turn Probability [-]	0.5 / 0.75 / 0.9 / 1
Number of vertical streets [-]	10
Number of horizontal streets [-]	11
Street width [m]	30
Size of block of building [m]	200 x 200
Size of simulated area [m]	2330 x 2100

The BSs are deployed regularly as presented in Figure 19. All BSs are placed in streets, not on roofs of buildings. The 48 MSs is randomly dropped in the streets at initiation of simulation. The signal strength among all MSs and BSs is calculated using urban microcell path loss model defined by [18] (see Appendix B2). This model distinguishes if a line of sight (LOS) between a MS and BS is enabled or not (Non-LOS communication denoted as NLOS) [85]. The signal levels are calculated at each interval of reporting of scanning results. It corresponds to the determination of parameters at each second in abovementioned scenario.

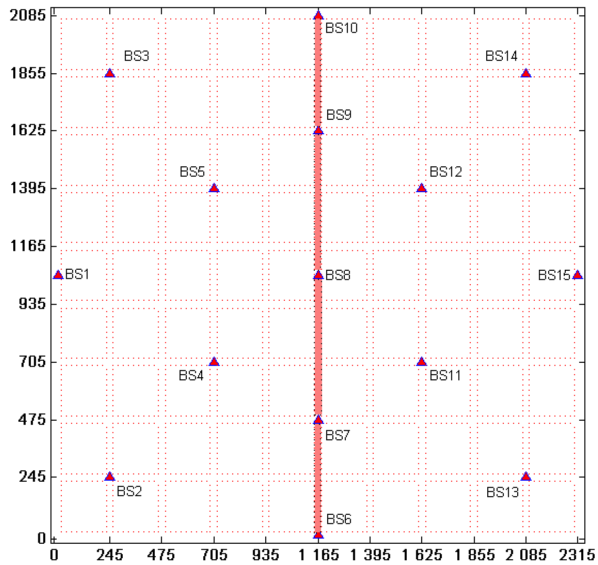


Figure 19. Simulation scenario for handover prediction evaluation

Four different varieties of mobility model are defined to compare results of the prediction. The first one represents conventional MMM with turn probability (TP) of 0.5. It means that each MS selects its next direction at each cross with 1−TP (= 0.5) probability of direct movement and TP/2 (= 0.25) probability of turn to right and same probability of turn to left. Next three scenarios define so called “Main Street” (red vertical street *no. 6* with *x* coordinate equal to 1165 m in Figure 19). The TP of a MS is 0.5 at all streets excluding the Main Street. While the MS comes to the cross with the Main Street, the turn probability of belonging MS temporarily increases to TP_{MS} = 0.75, 0.9 or 1 for three scenarios respectively. If the MS is moving on the Main Street, the probability of direct movement is increased to TP_{MS} (Turn Probability – Main Street) and the probability of turn out of the Main Street is equal to 1-TP_{MS}. The turn probability is set back to TP while the MS reach end of the Main Street or if the MS leaves the Main Street. This situation corresponds much more to the real movement in the city centre with one busy street or with a main square.

3.2.2 CHANNEL CHARACTERISTICS

The prediction of target BS based on channel characteristics is evaluated for scenario where RSSI parameter is observed (as described in theoretical analysis in section 3.1.2). The simulation parameters are summarized in Table 8.

Table 8. Simulation parameters and scenario definition for channel characteristics

Parameter	Value
Number of BS [-]	15
Number of MS [-]	48
BS transmitting power [dB]	46
BS height [m]	32
MS height [m]	2
MS speed [m/s]	15
Frequency band [GHz]	2.5
Frame duration [ms]	10
Scanning reporting period [s]	1
Simulation duration [s]	10800
HO_{Zone}[dB]	0–16
Hysteresis margin [dB]	4/8/12/16/20
HDT [s]	1/2/3/4/5
Window Size [samples]	5/8/10/15/20
LOS/NLOS path loss model	802.16m Urban Macrocell
Shadowing and channel variation	802.16m; σ = 0/0.3/0.6/0.8
Mobility model	PRWMM
Size of simulated area [m]	2330 x 2100 (4.89km^2)

All BSs are deployed in symmetric manner as presented in Figure 20. All BSs are placed in the same height and all transmits wit the same power level. At the beginning of simulation, the position of all 48 MSs is randomly generated. Probabilistic Random Walk Mobility Model (PRWMM) [86] (see Appendix C1) is considered for the MSs' movement. Signal strengths among all MSs and BSs are calculated using urban macrocell path loss model defined in [18] (see Appendix B1). The signal parameters are evaluated at each scanning reporting period which corresponds to the enumeration of parameters every second in used scenario.

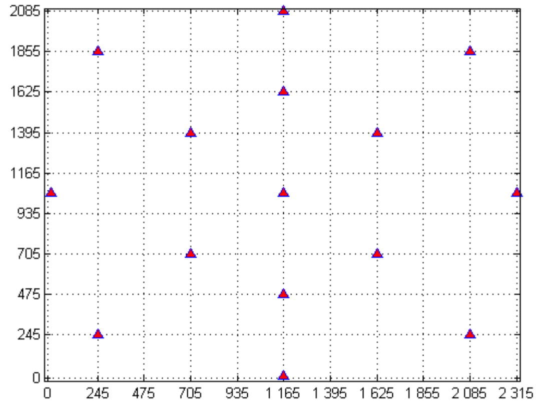


Figure 20. Deployment of BSs in the simulation

Several sets of simulations are performed considering different kind of scenarios. The first one analyzes the impact of HO_{Zone} on the prediction efficiency if no other technique for efficiency improvement is considered (i.e. HM, HDT or Windowing). Next three sets of simulations are focused on analysis of the impact of individual techniques on prediction efficiency. The last one determines an optimum setting of all three techniques in cooperating mode to obtain maximum prediction efficiency.

Three parameters, from the analysis of results point of view, are considered: ratio of successfully predicted handover *HR* (Hit Ratio), ratio of not predicted handovers (*NPR*) and ratio of wrong prediction (*WPR*). The handover prediction is assumed to be successful if the MS executes the handover to the predicted target BS. The number of successfully predicted handover (SP_{HO}) is used for calculation of prediction hit ratio according to the following formula:

$$HR = \frac{SP_{HO}}{HOCount} \tag{33}$$

where *HOCount* represents a total number of all handovers in network. It can be calculated as:

$$HOCount = \sum_{BSX=1}^{BSCount} \sum_{BSY=1}^{BSCount} HOCount_{BSX,BSY}, \qquad BSX \neq BSY \tag{34}$$

where *BSCount* is a total number of BSs in the network.

Not predicted handover occurs when the handover is done despite of the fact that no target BS has been predicted since a time when the MS have performed previous handover. This situation happens especially at the beginning of simulation as not enough data is collected to successfully predict target BS (typical thresholds cannot be setup since there are still either no information on previous handovers at all or the gathered information are insufficient). The ratio of not predicted handovers is calculated by the subsequent way:

$$NPR = \frac{NP_{HO}}{HOCount} \tag{35}$$

where NP_{HO} is a total number of not predicted handovers.

An error in prediction (wrong prediction) occurs if the predicted target BS differs from the real target BS of the MS. The ratio of wrong predictions can be expressed by the following equation:

$$WPR = \frac{WP_{HO}}{HOCount} \tag{36}$$

where WP_{HO} is a total number of wrong predictions of handover.

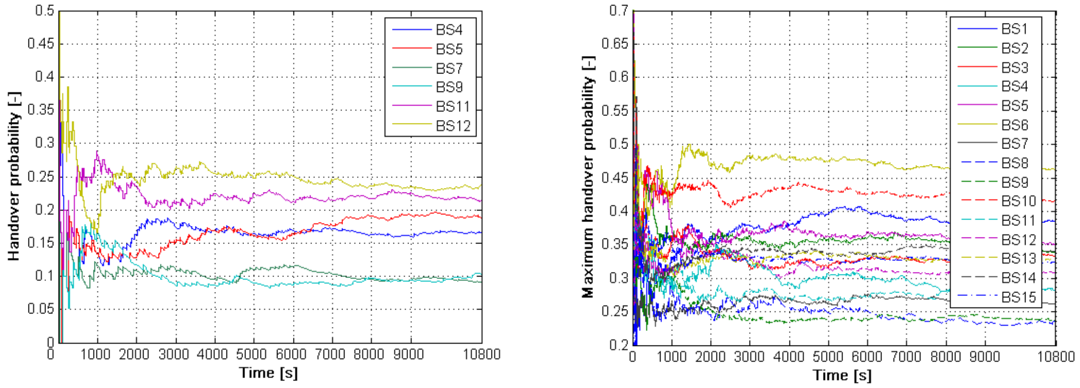
3.3 RESULTS

3.3.1 HO HISTORY

Simulation results are presented in Figure 21 – Figure 24. The probability of MS's handover from the BS8 to all neighboring BSs (BS4, 5, 7, 9, 11 and 12) is presented in Figure 21a – Figure 24a.

The scenario with no Main Street (Figure 21a) shows that time required to collect enough information on handovers is approximately 2000 s Then the probability is getting stable and do not vary rapidly. Totally, about 2000 handovers occur per 1000 s of simulation. The maximum efficiency of successful prediction corresponds to the probability of handover as no other criterion is considered for the prediction. Therefore, it can be assumed equal after the probability curves get stable. Based on Figure 21a can be observed that BS12 is the most probable target BS for MS's handover from BS8. However the handover probability from BS8 to BS12 is only approximately 25%. The lowest probability of the handover from the BS8 is to BS7 and BS9 (roughly 10%). That low probability results to very low efficiency of the prediction.

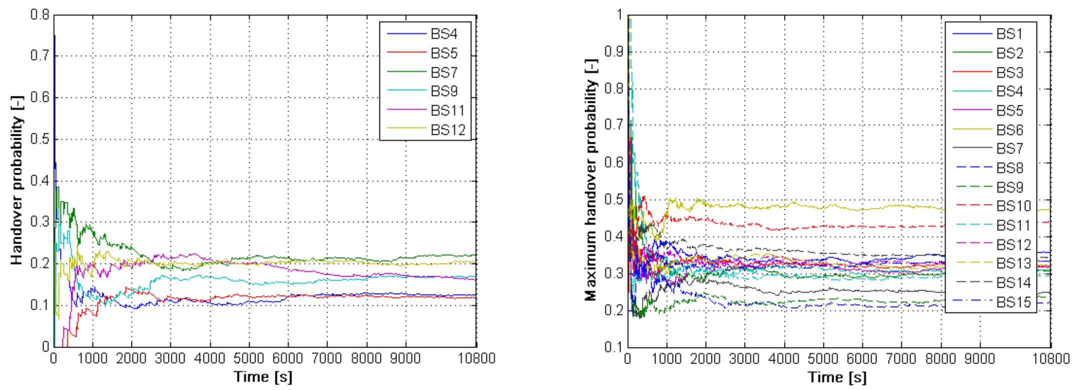
The maximum efficiency of target BSs prediction for all serving BSs are presented in Figure 21b. It expresses a ratio between efficient and failed target BS prediction. Each curve corresponds to the one serving BS. Hence every curve copies the maximum value of prediction probability for appropriate BS (e.g. maximum of all curves in Figure 21a corresponds to the blue dash line in Figure 21b).



(a) (b)

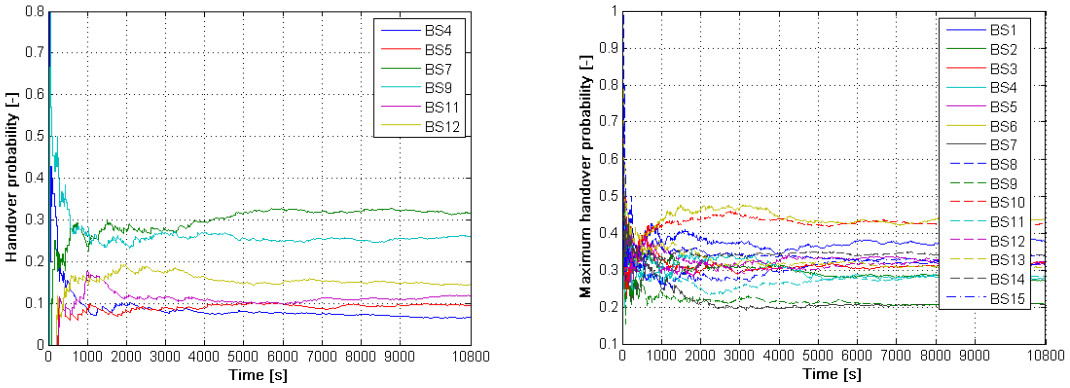
Figure 21. Results of handover prediction for no Main Street
(a) Handover probability over duration of handover history collection for BS8
(b) Maximum probability of handover from all serving BS over observation time

The presence of Main Street results into an increase of ratio of correct target BS prediction for a MS moving from the serving BS8 (see Figure 21b – Figure 24b) from 25% with no Main Street to nearly 45% with mostly visited Main Street (TP_{MS} = 1). The example of handover probability from BS8 shows an increase of handover probability to BS7 or BS9 since both are neighbor BSs of BS8 deployed on the Main Street. The probabilities of handover from BS8 to BS7/BS9 rise from 10% for no Main Street to 40% for the mostly visited Main Street.



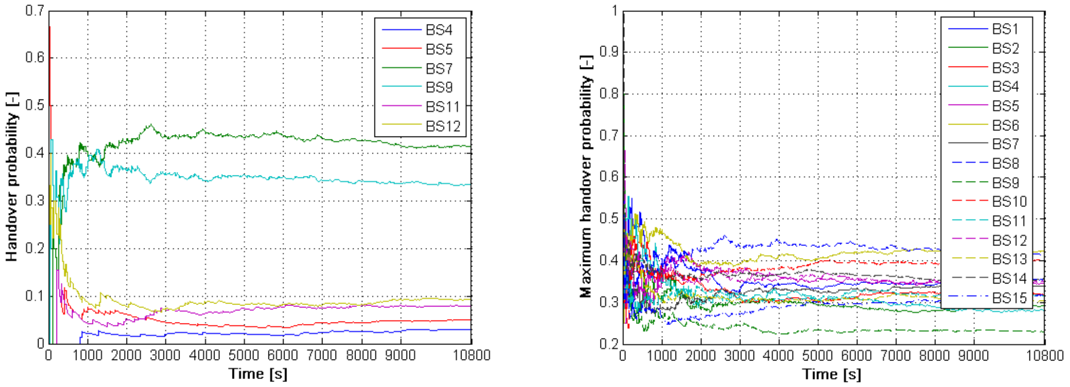
(a) (b)

Figure 22. Results of handover for Main Street TP = 0.75
(a) Handover probability over duration of handover history collection for BS8
(b) Maximum probability of handover from all serving BS over observation time



(a) (b)

Figure 23. Results of handover for Main Street TP= 0.9
(a) Handover probability over duration of handover history collection for BS8
(b) Maximum probability of handover from all serving BS over observation time



(a) (b)

Figure 24. Results of handover prediction for Main Street TP = 1
(a) Handover probability over duration of handover history collection for BS8
(b) Maximum probability of handover from all serving BS over observation time

From Figure 21b – Figure 24b can be further observed that the maximum ratio of successful target BS prediction is achieved for the prediction of handover from BS6 (about 47%) and BS10 (about 42%). The lowest ratio is attained while the MS is connected to the serving BS8 and BS9 (both about 25%). It leads to the conclusion that the efficiency of target BS prediction depends on the number of neighboring BS (BS6 and BS10 have only 3 neighboring BSs however BS8 has 6 neighboring BSs).

The impact of number of neighboring BSs on the target BS prediction efficiency depicted in Figure 25. The efficiency is significantly decreasing if the number of neighboring BSs is increasing for scenario without Main Street (TP = 0.5) and with Main Street with $TP_{MS} = 0.75$ and $TP_{MS} = 0.9$. In these scenarios, the ratio of successfully predicted target BS decreases from about 45% to 24% while the number of neighboring BSs rises from 3 to 6. In case of the Main Street with $TP_{MS} = 1$, the handover probability decreases only up to 4 neighboring BSs and then the increasing number of neighbor BSs does not influent the efficiency of prediction. All evaluations of the impact of neighboring BS count on the prediction efficiency are performed with the same deployment of BSs as presented in Figure 19.

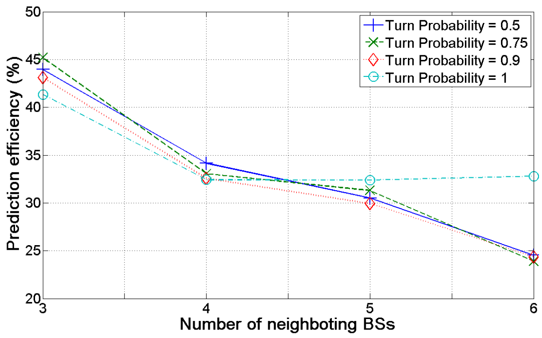


Figure 25. Efficiency of target BS prediction over number of neighboring BSs

3.3.2 Channel Characteristics

Results obtained by simulations are separated into subsections according to the techniques that are investigated.

3.3.2.1 Determination of Optimal HO_{ZONE} for RSSI based prediction

Firstly, the impact of HO_{Zone} on prediction efficiency is analyzed. The results of simulation are shown in Figure 26 and Figure 27. Both figures present the results of handover prediction hit ratio, ratio of not predicted handovers anf ratio of wrong prediction.

It is evident that if the ratio of wrong prediction together with the ratio of not predicted handovers is decreasing, the overall ratio of successful prediction is increasing proportionally. The time instant when sufficient number of data is collected is approximately 1000 s (at that time, roughly 2000 handovers is already performed within the simulation; it is roughly 10 handovers between each pair of neighboring BSs). At this stage, the ratio of successful handover prediction (handover prediction hit ratio) gets stable. The "get stable" means that the prediction ratio does not vary more than ±5% during following 3600 s.

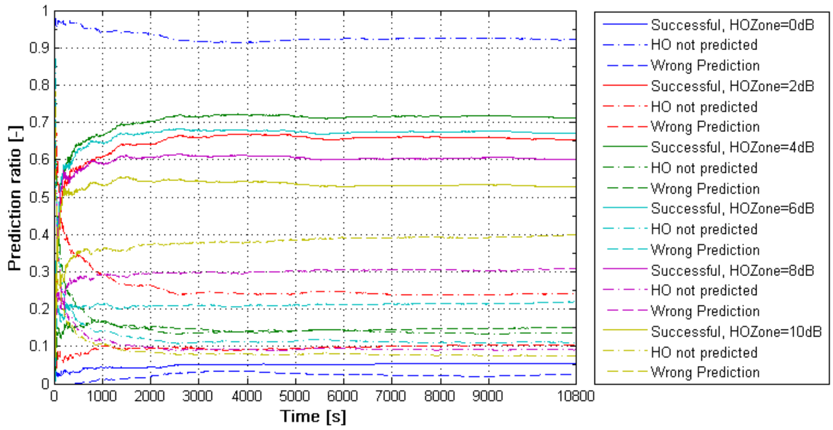


Figure 26. Results of handover prediction based on the RSSI evolution, no channel variation

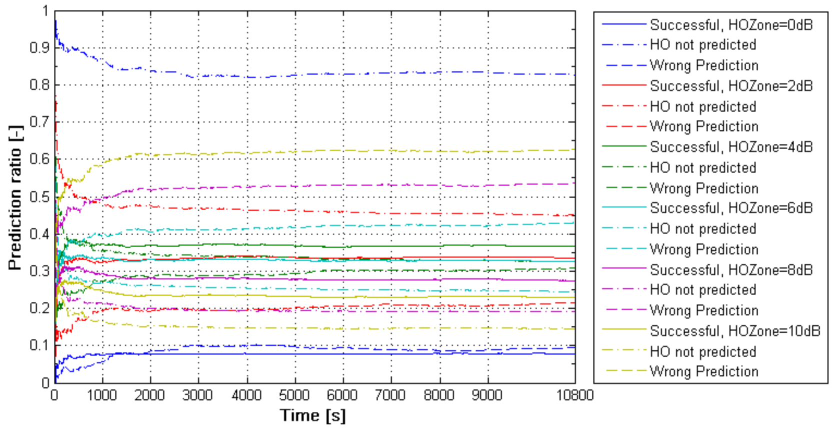


Figure 27. Results of handover prediction based on the RSSI evolution, with shadowing and channel variation (σ = 0.8)

Individual presented figures differ in diverse parameters setting of channel model. While Figure 26 assumes that the handover prediction is done only according RSSI evolution, Figure 27 takes further into consideration another factor, i.e. channel variation with standard deviation, σ = 0.8 [18] and shadowing, (see Appendix B3 and B4). Windowing, HDT and HM are disabled in Figure 26 and Figure 27.

Overall results of the prediction hit ratio acquired during 3 hours (10 800 s) data collection are summarized in Figure 28. This figure considers also other levels of channel variation, i.e. σ = 0.3 and σ = 0.6. The maximum prediction hit ratio (approximately 71%) is achieved if the channel variation is not considered. The higher channel variation degrades maximum prediction hit ratio to 45%/41%/37% for σ = 0.3/0.6/0.8 respectively. Additionally, Figure 28 demonstrates that the prediction and its effectiveness significantly depend on the HO_{Zone} size. The prediction efficiency is under 10% if no HO_{Zone} is taken into account. The optimal value of HO_{Zone} observed from Figure 28 is 4 dB for all levels of channel variation. At this HO_{Zone} level, the prediction mechanism shows highest ratio of successfully predicted target BS (71%) and low ratio of not predicted (14%) and wrongly predicted handovers (15%) if no channel variation is introduced. Figure 29 depicts that the channel variation increases the ratio of not predicted handovers slightly over 30% for all levels of σ at HO_{Zone} = 4 dB. The impact of channel variation level on ratio of not predicted handovers is negligible. The ratio of the wrong prediction also rises to 25%/28%/31% for σ = 0.3/0.6/0.8 respectively at HO_{Zone} = 4 dB (see Figure 30).

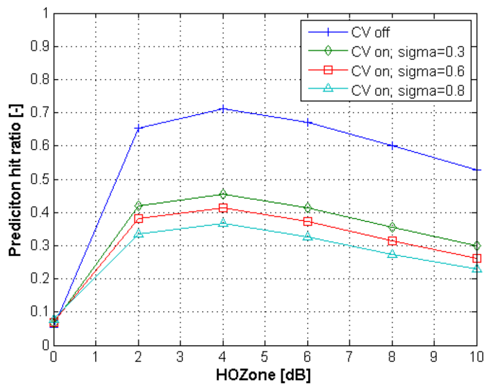


Figure 28. Target BS prediction hit ratio over HO_{Zone}

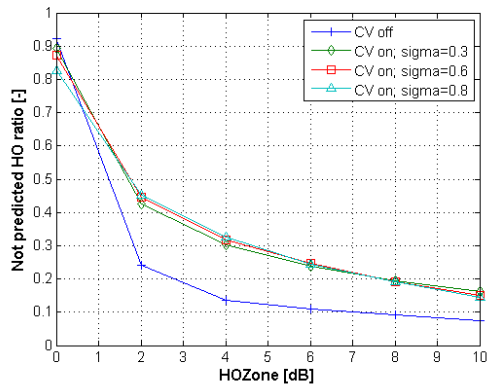


Figure 29. Ratio of not predicted handover over HO_{Zone}

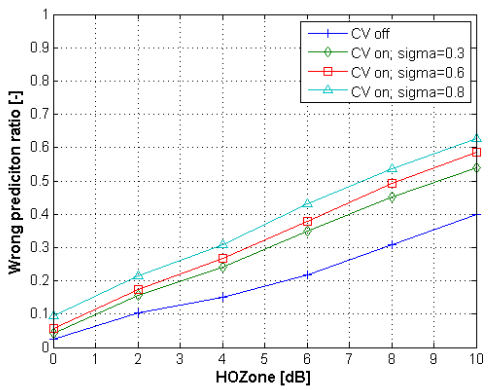


Figure 30. Ratio of wrongly predicted target BS over HO_{Zone}

In the rest of simulations where a comparison of impact of windowing, MH and HDT is addressed, the channel variation with σ = 0.8 will be taken into account (in figures denoted as “CV on”). The results of scenario with not considered channel variation (in figures denoted as “CV off”) are also depicted in following figures to enable a comparison of results. The detailed behaviors of prediction ratios over simulation time, as depicted in Figure 26 – Figure 27, are not presented in the following subsections since the overall results are presented in form of separated results for hit ratio, not predicted handover ratio and ratio of wrong prediction. This form of figures is more transparent. If not stated otherwise, the ratio of successful handover prediction gets stable approximately after 1000 s of simulation.

3.3.2.2 IMPACT OF WINDOWING ON PREDICTION EFFICIENCY

In order to suppress the negative impact of channel variation on prediction or to increase the prediction efficiency, technique based on averaging of several samples of RSSI (Windowing) can be used. Figure 31 illustrates that the prediction hit ratio is distinguishable increased by windowing even if the channel model including channel variation ($\sigma = 0.8$) is assumed (see appendix B). The WS is set to 5, 8, 10, 15 and 20 samples. The utilization of windowing technique with WS = 5 results in a rise of hit ratio from 37% to 65% (38% increase of hit ratio) in comparison to the case with no windowing. In the rest of this chapter, if the gain is for example 10%, it means that the hit ratio increases e.g. from 50% to 60%.

Additional gain of 6% is introduced when the size o WS is further increased to 8 samples. In this case, the impact of channel variation is eliminated and the prediction hit ratio for channel variation with $\sigma = 0.8$ and WS = 8 is equal to scenario with no channel variation and no windowing. In both cases, maximum prediction hit ratio is 71%. Another gain of 2% and 5% is observed by increasing WS to 10 and 15 samples respectively. The results for WS = 20 shows only marginal increase of the prediction hit ratio. This indicates that further increase of WS is useless.

Analogical conclusions can be derived from Figure 32 and Figure 33. Figure 32 shows no noticeable decrease of not predicted handovers ratio for WS above 15 samples. Similarly, Figure 33 presents only marginal decrease of wrong prediction ratio while higher values of WS than 15 samples are utilized. As it can be observed, the number of not predicted handovers is decreasing while the level of HO_{Zone} is rising since larger HO_{Zone} enables to perform the prediction earlier. However, sooner prediction leads to the increase of ratio of wrong prediction as well. It is due to early prediction that results to higher probability of MS's turn away from current direction (see equation (30)).

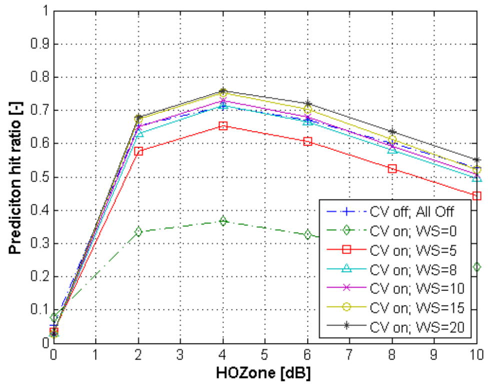


Figure 31. Target BS prediction hit ratio over HO_{Zone} for a set of WS

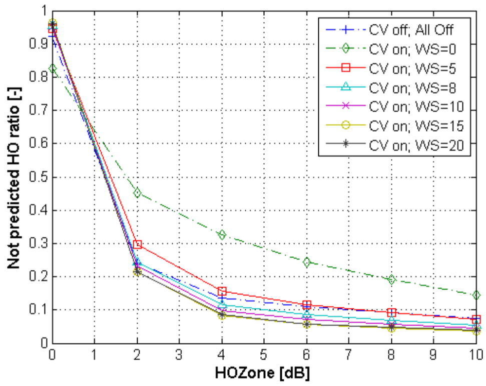


Figure 32. Ratio of not predicted handover over HO_{Zone} for set of WS

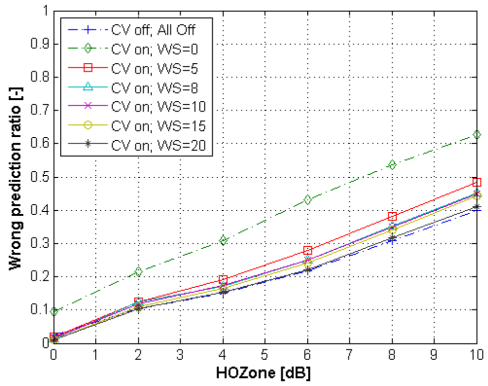


Figure 33. Ratio of wrongly predicted target BS over HO_{Zone} for set of WS

3.3.2.3 IMPACT OF HYSTERESIS MARGIN ON PREDICTION EFFICIENCY

Another way of suppressing the negative impact of channel variation is to use HM. The results of this investigation are presented in Figure 34 – Figure 36 with utilization of following values of HM; HM = 4, 8, 12, 16, 20 dB.

The scenario with HM = 4 dB improves the prediction hit ratio by 41% (at HO_{Zone} = 4 dB) in comparison to the scenario where HM is not considered. Note that the ratio of successful prediction for HM = 4 dB at HO_{Zone} = 4 dB and channel variation with $\sigma = 0.8$ is higher (roughly for 7%) in comparison to the scenario with no channel variation. Additional gain is further reached by increase of HM up to 12 dB at HO_{Zone} = 4 dB. The prediction efficiency gets as high as approximately 90%. If HM is set either to 16 dB or to 20 dB no significant gain in prediction hit ratio is acquired at $HO_{Zone} = 4$ dB in comparison to HM = 12 dB. The prediction hit ratio for HM over 12 dB reach its maximum at higher HO_{Zone}. The maximum for HM = 16 dB and HM = 20 dB is achieved at $HO_{Zone} = 6$ dB (91%) and $HO_{Zone} = 8$ dB (92%).

Besides, Figure 35 and Figure 36 show that an increase of the HM above 12 dB leads neither into significant improvement in the reduction of ratio of not predicted handovers nor into considerable minimization of ratio of wrong predictions. The ratio of not predicted handovers is even slightly higher for HM = 16 dB and HM= 20 dB than for HM = 12 dB.

Another reason for utilization of lower level of HM is that the increasing the HM can lead to major decrease of network throughput since it results in utilization of more robust MCS for communication between MS and BS (see an explanation of this fact described in section 2.2.1.2).

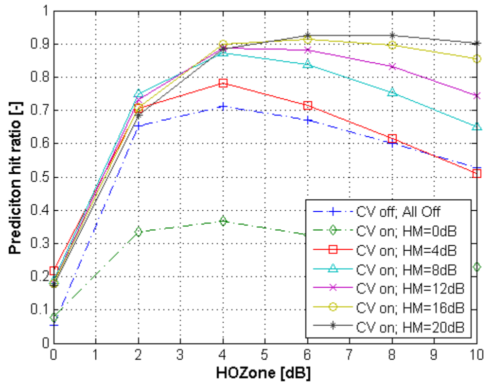


Figure 34. Target BS prediction hit ratio over HO_{Zone} for set of HM

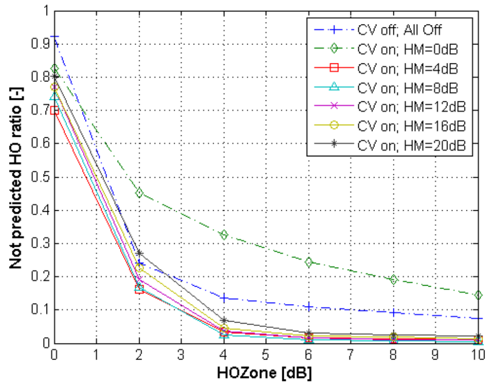


Figure 35. Ratio of not predicted handover over HO_{Zone} for set of HM

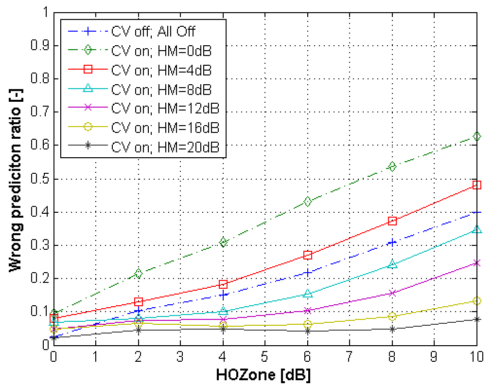


Figure 36. Ratio of wrongly predicted target BS over HO_{Zone} for set of HM

The important aspect which should be considered is prolongation of the period necessary for collection of enough handover information due to utilization of HM higher than 4 dB. This period is extended roughly by 1000 s per 4 dB of HM for HM > 4 dB in comparison to scenario with HM, HDT or windowing turned off.

3.3.2.4 IMPACT OF HANDOVER DELAY TIMER ON PREDICTION EFFICIENCY

The third way how to mitigate negative impact of the channel variation on prediction efficiency is an application of HDT. The results of simulation are depicted in Figure 37 – Figure 39 with channel model including channel variation with $\sigma = 0.8$ and utilization of HDT with duration of 1, 2, 3, 4 and 5 s. As Figure 37 indicates, HDT cannot fully eliminate the negative impact of channel variation. If the HDT is set to 1 s, the prediction hit ratio is improved only by 2% at $HO_{Zone} = 4$ dB in comparison to the scenario without HDT. Further increase of HDT duration up to HDT = 3 s brings rise of the prediction hit ratio for 2% per 1 s of HDT duration. Then the improvement of prediction hit ratio is negligible (less than 1% per 1 s of HDT duration). Figure 37 demonstrates minor continuous increase of the prediction hit ratio with rise of the HO_{Zone} for HDT = 1 s.

The HDT increases ratio of not predicted handovers up to roughly 10% (depending on HO_{Zone}) in comparison to scenario with no HDT, HM or windowing. The duration of HDT has only minor impact on the ratio of not predicted handovers (see Figure 38). On the other hand, the ratio of wrong prediction is improved approximately by 15% (see Figure 39) by implementation of HDT. The wrong prediction ratio is also influenced only marginally by HDT.

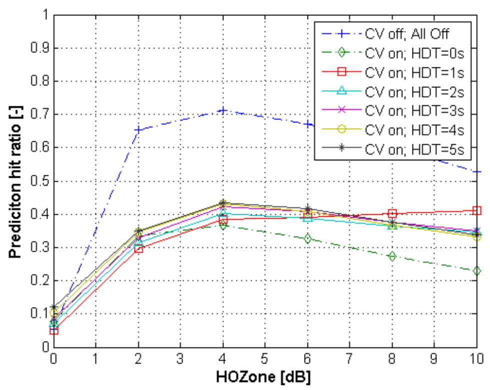


Figure 37. Target BS prediction hit ratio over HO_{Zone} for set of HDT

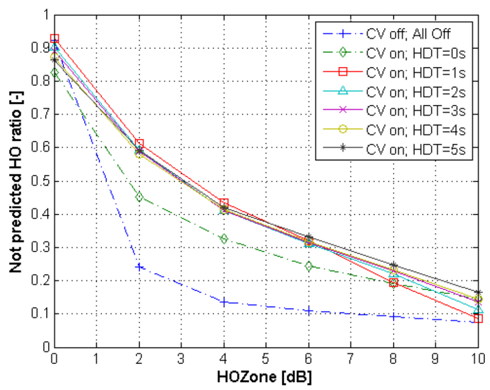


Figure 38. Ratio of not predicted handover over HO_{Zone} for set of HDT

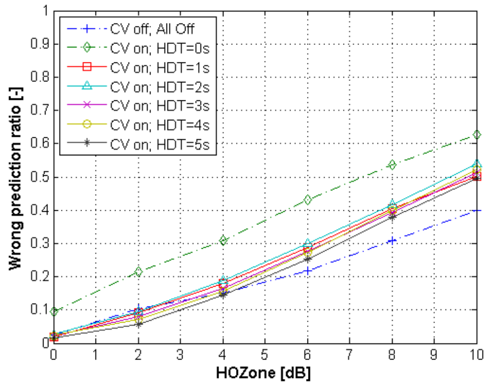


Figure 39. Ratio of wrongly predicted target BS over HO_{Zone} for set of HDT

Utilization of HDT shortens the period required for collection of enough handover information roughly to 500 s for all durations of HDT.

3.3.2.5 MAXIMIZATION OF HANDOVER PREDICTION EFFICIENCY

This section presents the results when all three methods cooperate in order to improve overall prediction efficiency. A very high number of combinations of all techniques can be set. Therefore, only the combinations which will potentially guarantee best results are assumed. The selection of optimal parameters is based on the results of individual techniques obtained in sections 3.3.2.2 – 3.3.2.4. If the system performs similarly for two different values, the lower one is selected as optimal since lower negative impact on the throughput can be assumed (see section 2.2.1.2). The negative impact on throughput is a result of postponing of the handover by all three techniques. Therefore, the MS can communicate with the BS that does not provide the best signal quality (as explained in section 2.2.1.2). To find an optimal setting of all parameters, the values of individual methods that show best performance of prediction are summarized in the Table 9. The fist line in Table 9 presents values considered as optimal.

Table 9. Best performing parameters of particular techniques

WS [samples]	HM [dB]	HDT [seconds]
15 @ HO_{Zone} = 4 dB	12 @ HO_{Zone} = 4 dB	1 @ HO_{Zone} = 10 dB
10 @ HO_{Zone} = 4 dB	16 @ HO_{Zone} = 6 dB	2 @ HO_{Zone} = 4 dB
20 @ HO_{Zone} = 4 dB	20 @ HO_{Zone} = 8 dB	3 @ HO_{Zone} = 4 dB

The complete list of all investigated scenarios is introduced in Table 10. Scenarios are defined with respect the impact of each particular technique on overall cooperation of more of them. The value of HM higher than 12 dB are not considered in the following scenarios as this values results into significant decrease of throughput (see Figure 15). Note that the minimal value for WS is 1 sample, for HM is 0 dB and for HM is 0 s.

Table 10. List of all simulation scenarios

Scenario	WS [samples]	HM [dB]	HDT [seconds]
A	15	12	3
B	15	12	1
C	15	12	0
D	1	12	3
E	1	12	1
F	1	12	0
G	10	12	1
H	2	12	1
I	2	1	1
J	15	1	1
K	8	4	0
L	3	2	0

In total, 12 scenarios are defined for an investigation of maximum prediction efficiency. All results are distributed in two figures (the first one contains results of scenario A–F, the second one presents scenario G–L) due to the higher clarity of plotted curves.

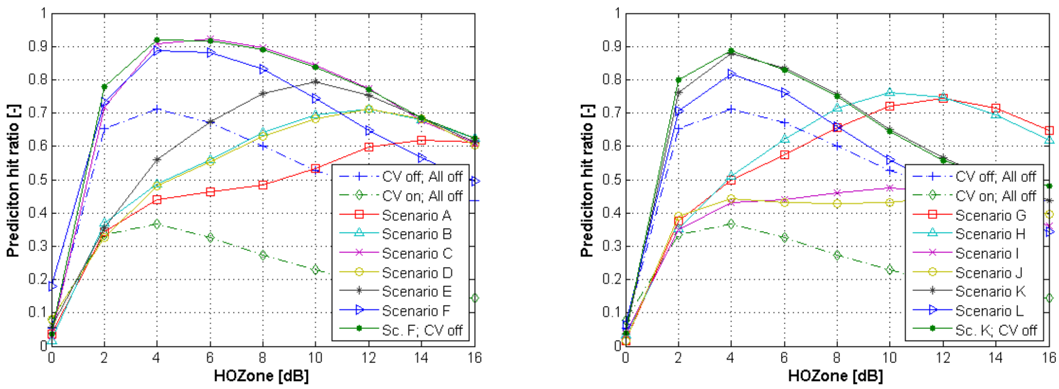
As can be observed from Figure 40, the highest hit ratio can be achieved by scenario C (prediction hit ratio is 93% for HO_{Zone} = 6 dB). This scenario corresponds to the case when the optimum parameters of HM and WS are set up (see Table 9) while HDT is disabled.

Furthermore, rising of optimal HO_{Zone} with rising HDT is noticeable from figure. However, the maximum hit ratio is decreasing with increasing duration of HDT (compare scenarios A, B and C or D, E and F). Another conclusion can be obtain by comparison of scenario C with F, B with E, A with D and G with H. These scenarios show that higher WS leads to the improvement of hit ratio only while the HDT is turn off. Otherwise, it is better to utilize lower WS. From the scenarios I and J in Figure 40 can be observed that utilization of low HM together with enabled HDT leads to the hit ratio under 50% and moreover hit ratio is nearly constant for HO_{Zone} higher than 4 dB. Additionally, scenarios K and L reach very high hit ratio (89% and 82%) in spite of fact that both utilize very low values of HM and WS simultaneously with HDT equal to zero.

All combinations of techniques lead to the improvement of prediction efficiency over all HO_{Zone} (compare all scenarios with green dash-dot line) and only very low values of HM and WS enable to fully eliminate the impact of channel variation.

While the results presented in section 2.2.1.2 are considered, the negative impact of scenario C (highest hit ratio) on throughput of a single MS is over 18%. The next scenarios with very high hit ratio (roughly 89%), scenarios F and K, lead to the significantly reduced drop of throughput (approximately 7% and 3.5% respectively). Furthermore, scenario L utilize only very low values of all techniques (decrease of throughput of single MS is roughly 1%) however it enables to reach hit ratio of 82%.

With relation to the prediction hit ratio and impact on the throughput of a single MS, scenario K is considered as the best one of the evaluated scenarios. This scenario and scenario F are evaluated also for disabled channel variation. The results show insignificant impact of the channel variation on hit ratio. The improvement of hit ratio is only approximately 3% for scenario F and 0.5 % for scenario K at HO_{Zone} = 4 dB.



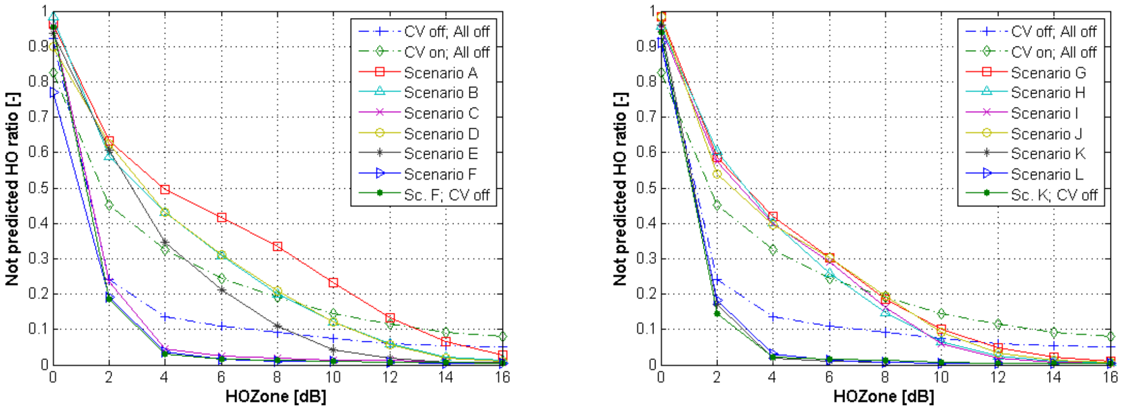
(a) (b)

Figure 40. Target BS prediction hit ratio over HO_{Zone} for set of combination of HDT, HM and WS, (a) Scenario A–F, (b) Scenario G–L

Note, that the impact of throughput is evaluated for a single MS with assumption of full buffer traffic model and fixed amount of OFDMA symbols assigned to the MS.

The impact on overall network throughput is significantly lower as the radio resources that are not utilized by the MS that is performing handover can be temporarily utilized by another MSs. Moreover, the scenario considered for the evaluation of throughput of single MS is set up to corresponds to the worse case scenario with very often handover execution.

Figure 41 illustrates the ratio of not predicted handovers over HO_{Zone} for all scenarios. As can be observed form Figure 41, the NPR is always decreasing with rising HO_{Zone}. All scenarios with disabled HDT show only negligible level of ratio of not predicted handovers for HO_{Zone} higher than 4 dB (it is nearly 0%). The scenarios with enabled HDT indicate gradual lowering of NPR over HO_{Zone}. The NPR is getting higher while the duration of HDT is increasing. The impact of channel variation on NPR is insignificant in the case of best performing scenarios (F and K).

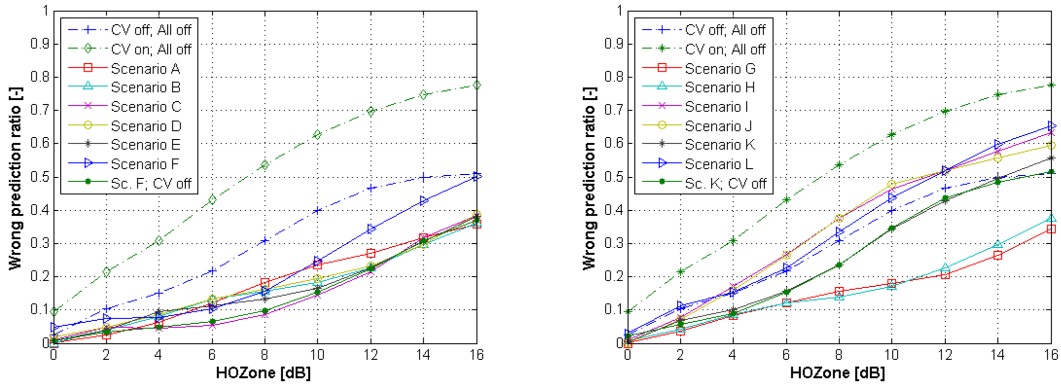


(a) (b)

Figure 41. Ratio of not predicted handover over HO_{Zone} for set of combination of HDT, HM and WS, (a) Scenario A–F, (b) Scenario G–L

Exactly opposite behavior in comparison to Figure 41 can be derived from Figure 42. This figure presents the dependence of wrong prediction ratio over HO_{Zone}. All scenarios perform significantly better than scenario with all techniques turn off. The ratio of wrong prediction is between 4% and 17% at $HO_{Zone} = 4$ dB for all scenarios. All scenarios with higher value of HM (HM = 12 dB) show the WPR under 9%. Therefore, lower values of HM lead to the higher WPR. Figure 42 also shows decreasing ratio of wrong prediction for increasing HM.

Turn off the channel variation results into slight decrease (up to 3% at $HO_{Zone} = 4$ dB) of wrong prediction ratio in scenario F however in case of scenario K the channel variation do not influence the ratio of wrong prediction.



(a) (b)

Figure 42. Ratio of wrongly predicted target BS over HO_{Zone} for set of combination of HDT, HM and WS, (a) Scenario A–F, (b) Scenario G–L

3.4 Conclusion

The performance of two approaches is investigated in this chapter: i) Prediction based on handover history and ii) Prediction based on channel characteristics. The third approach, prediction based on knowledge of MS motion, is not evaluated as it assumes to know exact position of user, user's profile or profile of user's location.

3.4.1 HO History

The advantage of prediction using handover history is the very simple implementation. Only a simple modification of a control mechanism of BSs is required. On the other hand, this method needs some time to adapt to the changes in environment (e.g. installation of new BS, close a road, etc.) or changes of channel characteristics (transmitting power, etc.) since the enough number of information must be collected. It is approximately 2000 s that corresponds to 4000 handovers per 15 BSs in total (roughly 19 handovers per each pair of BSs) in our simulation.

The prediction efficiency is strongly influent by the number of neighboring BSs. The twice increase of neighboring BSs (from 3 to 6) leads to the drop of prediction efficiency from about 45% to 24%.

In all cases, the prediction hit ratio varies between 20% and 47%. Hence this way of prediction can be generally utilized as a supporting method for other techniques; however its utilization as a stand alone method is very inefficient due to low prediction efficiency.

3.4.2 CHANNEL CHARACTERISTICS

This kind of prediction can be also implemented in a very simple manner into network since no modifications at MAC layer or hardware are required. This method needs some time to adapt to the changes in environment and to the fluctuations of channel characteristics. The time necessary for adapting is approximately 1000 s (roughly 2000 handovers in total and 10 handovers per each pair of BSs) for the scenario excluding HM, HDT and windowing. This time is significantly (approximately two times) lower in comparison to previous technique.

The maximum prediction hit ratio depends on the variation of channel characteristics and on the level of HO_{Zone}. The best performance of prediction is usually achieved for HO_{Zone} equal to 4 dB while HM, HDT and windowing prediction techniques are disabled.

As the results of analysis of prediction efficiency improvement indicate, the level of prediction hit ratio can be positively influenced by using of all these techniques (HM, HDT and windowing). The most efficient one is HM since its utilization can increase the prediction hit ratio from 37% to more than 90% if channel variation is considered. However this technique prolongs the time for collection of information regarding to the previous handover. Even the worst performing method, i.e. HDT, enables to increase the prediction hit ratio to 45%. On the other hand, the advantage of HDT is in reduction of the time interval necessary for collection of sufficient amount of data for efficient prediction approximately to 500 s.

The maximum prediction hit ratio when individual techniques are combined together is 93% at HO_{Zone} = 6 dB while WS = 15 samples, HM = 12 dB and HDT = 0 s. Generally, enabling the HDT leads to the increasing of optimum HO_{Zone} and simultaneously it lower the prediction efficiency. Therefore, more efficient is a combination of only HM and windowing without using HDT. Lower efficiency of the prediction while HDT is utilized is the result of higher ratio of not predicted handovers. From the wrong prediction ratio point of view, the higher values of HM allow to reach lower ratio of wrong prediction.

Considering the impact on throughput between a MS and BS, the scenarios with lower values of HM, WS and HDT are preferred. Therefore, scenarios K and L can be selected as the optimal ones. These scenarios provide prediction hit ratio 89% and 82% while throughput of a single MS is decreased roughly for 3.5% and 1%.

4 FAST PREDICTED HANDOVER

4.1 INTRODUCTION

The modifications of MAC management messages exchange flow is a possible approach for handover optimization in term of handover interruption time point of view. The duration of current hard handover interruption depends on frame duration. Its minimum is approximately 32 ms for the real systems while frames with duration of 2 ms are utilized (see section 2.1.1.2). According to IEEE 802.16m requirements, it has to be decreased below to 27.5 ms for intra-frequency handover [17]. Maximum target values defined by [17] for inter and intra frequency handovers are presented in Table 11. Target handover interruption time is 27.5 ms for this thesis as the goal of the thesis is to design universal procedure with no limitation to only one of the specific type of handover (inter or intra frequency). Another reason is that the times for tune up receivers and transceivers to the communication frequency are not considered.

Table 11. Maximum interruption times for hard handover

Handover type		Maximum interruption time [ms]
Intra-frequency		27.5
Inter-frequency	within a spectrum band	40
	between spectrum bands	60

4.2 DESIGN OF FAST PREDICTED HANDOVER

The handover interruption origins while a synchronization, contention resolution and network re-entry are executed (see section 2.1). It is apparent that the time for synchronization with BS's downlink channel can not be reduced by a modification of handover MAC message flow. According to IEEE 802.16e, the time for contention resolution can be eliminated (reduced to 0 ms) by utilization of dedicated ranging slot. Additional reduction of the handover interruption can be accomplished via a modification of network re-entry procedure (ranging, authorization and registration stages). To purpose of design a fast handover with minimum interruption, the results of target BS prediction presented in chapter 3 are exploited. The novel handover procedure is called Fast Predicted Handover (FPHO).

The overall MAC management message flow and a principle of FPHO are shown in Figure 43. The red parts in figure highlight the modifications and new parts in comparison to IEEE 802.16e procedure.

Since the principle of handover in network with and without RSs is the same, this section is focused on the description of FPHO for networks without RSs. However the FPHO can be performed in the same manner even if the serving/target BS is replaced by access/target access RS.

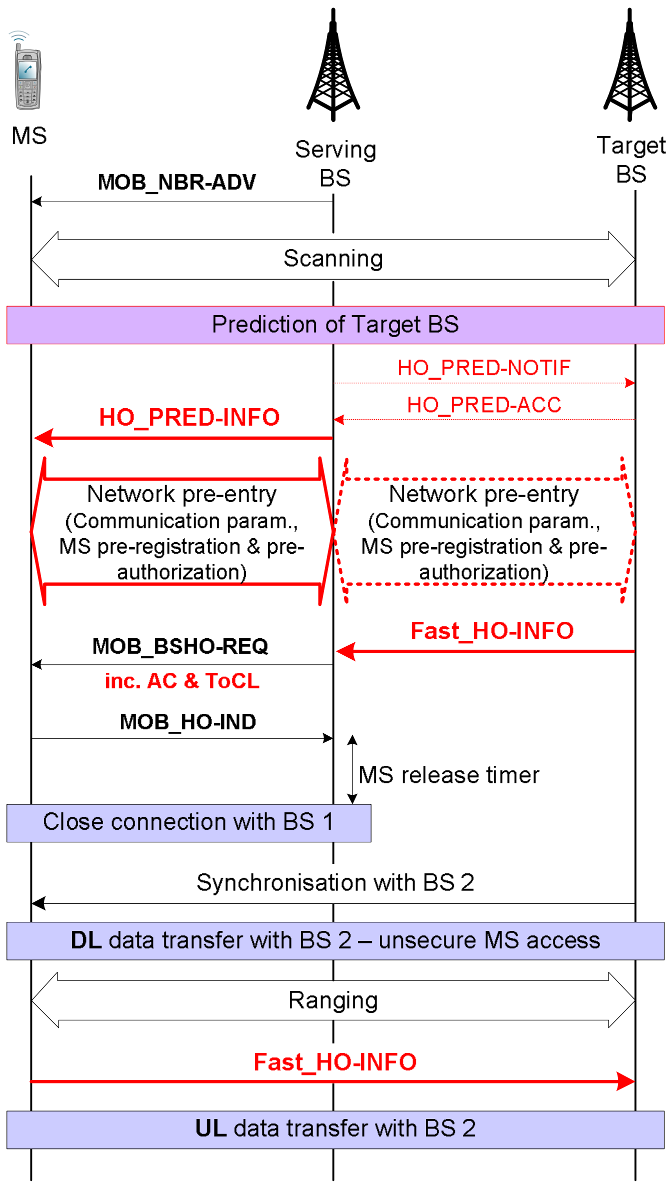


Figure 43. Flow of management messages during FPHO

All steps of FPHO are consequently executed as follows. During a normal operation, a MS continuously scans its neighborhood as in conventional IEEE 802.16e. Based on the results of scanning, a serving BS can predict most likely target BS using some of the methods for target BS prediction (e.g. prediction based on the channel characteristics as proposed in section 3). Consequently, the serving BS checks a possibility of MS's handover to the predicted target BS via messages HO_PRED-NOTIF and HO-PRED-ACC for notification of handover and acceptance/rejection of the MS respectively. The notification message contains identification of the MS and information on requested resources. The second message consists of information on acceptance or rejection of the handover request and information on time interval within the MS will be accepted. The target BS has to ensure the reservation of resources for the MS during whole time interval within the MS will be accepted at the target BS. The communication between serving and target BS is performed over backbone. Thus, it do not affect a network throughput. The backbone communication is out of scope of this thesis since it is not defined by standards and it is manufacturer dependent. If the target BS can accept the MS, the serving BS transmits message HO_PRED-INFO containing identification of the predicted target BS and time within the predicated target BS will enable the handover of MS (see Table 12). Moreover, the message can contain additional information on resource or service availability in form of TLV (Type-Length-Value) coded information.

Table 12. Structure of HO_PRED-INFO message

Syntax	Size (bites)	Notes
HO_PRED-INFO{		
Management Message Type = TBD	8	
Target BSID	48	Identification of the target BS
TimeOfRR	16	Time of resource reservation. Time within the target BS will reserve the requested resources
TLV	variable	Additional information on resources can be included
}		

Consequently, the MS initiates network re-entry procedure with the predicted target BS. It is performed via current serving BS. In the thesis, this stage is entitled as network pre-entry since it is executed before the conventional handover initiation stage is accomplished (transmission of MOB_BSHO-REQ or MOB_MSHO-REQ by BS or MS respectively). In the course of the network pre-entry, the MS performs an authorization and registration to the target BS through the serving BS. Besides, the MS obtains CID used for communication with the target BS. Furthermore, the MS can obtain some of the ranging parameters like a predicted transmitting power, timing information or frequency offset. These parameters can help to reduce duration of conventional ranging procedure with the target BS.

The important issue is how to initiate the pre-entry procedure. Both, the serving BS or MS can initiate the network pre-entry procedure by transmission of RNG_RSP or RNG-REQ respectively. One of the eight reserved bits of these messages (see [2]) can indicate if the message belongs to pre-registration procedure or to conventional IEEE 802.16e re-registration. Then the RNG-REG/RSP messages contain: 8-bits used for information on message type, 1-bit for indication of conventional registration or fast handover pre-registration (if bit = 0 then the conventional procedure takes place, if bit = 1 then FPHO pre-registration procedure is going to be performed), 7 reserved bits and finally TLV information. The pre-registration and pre-authorization procedures should utilize new messages (PREG_REQ/RPS and PPKM_REQ/RPS respectively). Except of “Management Message Type” field, the content of the mew messages should be identical as the content of authorization and registration messages defined in conventional IEEE802.16e. The new management messages have to be designed to keep backward compatibility with previous versions of standard. The new messages for authorization and registration have to be created due to no available bit for distinguishing of the conventional entry and pre-entry procedure as no unused bit (e.g. reserved bit) is available. The above described method does not increase overhead of the handover procedure since proposed ranging, registration and authorization messages have the same length as former IEEE 802.16e messages.

Another way of network pre-entry initiation is to design new messages that inform the MS or serving BS that the following messages (up to Fast_HO-INFO) belong to network-pre-entry procedure. This solution needs no modification in current IEEE 802.16e MAC management messages. However, it increases overhead as additional two messages must be sent (info about beginning of pre-entry procedure and confirmation of this message). Since the MS is still connected to the serving BS, both solutions do not prolonging the handover interruption.

The network pre-entry process is finished by the reception of Fast_HO-INFO by the MS. The structure of this message is presented in Table 13.

Table 13. Structure of Fast_HO-INFO message

Syntax	Size (bites)	Notes
Fast_ HO-INFO{		
Management Message Type = TBD	8	
Access Code	24	Code for the fast access to the target BS
Originator	1	0... Target BS – assignment of the AC 1... MS –verification of the AC
If (Originator = 0) {		
Time of Code Life	14	Indication of the time when the AC is valid. It is expressed as a number of frames up to 2^{14}-1.
}		
If (Originator = 1) {		
MS Address	48	MS MAC address
}		
Reserved	1 or 7	Alignment to byte; Length is 1 bit if Originator = 0; and length is 7 bits if Originator =1.
}		

The Fast_HO-INFO consists of randomly generated Access Code (AC). The target BS generates AC in order to verify the MS. Furthermore, if the Fast_HO-INFO is transmitted by the MS, the MS ID is included to facilitate easier check of the validity of AC by the target BS. If the Fast_HO-INFO is transmitted by the target BS, it contains information on the duration of AC's validity represented by Time of Code Life (ToCL). The ToCL indicates a number of frames when the AC is valid and within it can be used for the MS's verification. The duration of AC's validity is equal to $ToCL \times FD$. After an expiration of ToCL, the MS has to ask for new AC or perform full (conventional) handover management message exchange.

The maximum duration of AC's validity depends on the PHY frame duration, e.g. for the frame duration of 2 ms, the AC is valid maximal 32.7 s ($(2^{14}-1)\times 2\text{ms}$). The minimum value has to consider a number of frames between the delivery of AC to the MS and the beginning of verification procedure (transmission of AC by MS to target BS). The ToCL timer ensures that the AC cannot be later exploited for unauthorized network re-entry by another MS and it reduces the requirements on a storage capacity of BS dedicated for ACs of all MSs.

The Fast_HO-INFO is transmitted to the MS via serving BS. The serving BS forwards this message either appended to the end of MOB_BSHO-REQ message or as a standalone message. The transmission of Fast_HO-INFO message embedded in the MOB_BSHO-REQ requires two modifications in MOB_BSHO-REQ. The first one is a utilization of one padding (reserved) bit to indicate the presence of Fast_HO-INFO information and the second one is a copying of 2 fields from Fast_HO-INFO (AC and ToCL) at the end of MOB_BSHO-REQ. As one of the reserved bits is used for indication of AC and ToLC presence, the size of message is not increased if the conventional IEEE 802.16e handover is performed instead of FPHO. However the size of message is extended by 38 bits while FPHO is executed since AC and ToLC are included in this message (see Table 13).

Although the first option (appended information to the MOB_BSHO-REQ) increases overall duration of handover procedure, the length of handover interruption is still the same since the MS is still connected to the serving BS. The second approach (standalone transmission of Fast_HO-INFO) results in additional increase of the overhead about 8 bits due to the introduction of new messages (the field “Management Message Type” is sent redundantly). Therefore the first approach is considered in the rest of paper.

After reception of MOB_BSHO-REQ with AC and ToCL, the MS releases all connections with the serving BS (using MOB-HO-IND message) and executes a synchronization with the target BS downlink. At this time, transmitted data in downlink are stored in a buffer of new serving BS (previous target BS). When the synchronization is completed, the ranging is performed. Simultaneously, the MS can start receiving the data in downlink. The data are ciphered by new keys negotiated during pre-entry procedure. Also new registration and communication parameters (e.g. CID) are utilized for communication. During this communication, authentication of the MS is not completed as the Fast_HO-INFO is not received by the target BS. The ranging process is followed by the transmission of Fast_HO-INFO from the MS to the new serving BS (former target BS). When the verification is successful, the new serving BS starts transmission of the data stored in buffer. The MS is informed about successful verification by the reception of first packet from the new serving BS. To guarantee that the Fast_HO-INFO was not lost during transmission, which would increase the interruption, the MS sends the Fast_HO-INFO repeatedly in every frame until it receives the data or until the ToCL timer does not expire.

If the MS or BS needs an update of pre-registration and pre-authorization parameters (e.g. due to change of cell load), the conventional re-registration and re-authorization procedures can be called at any later time.

4.3 EVALUATION OF FPHO INTERRUPTION

The proposed technique can be applied on all handover scenarios according to IEEE 802.16e and IEEE 802.16j [16]. Thus the evaluation considers inter BS as well as intra BS handover (see section 1.1.2).

The minimization of handover interruption cannot be done for the major part of handover stages due to no update of serving BS in all intra BS handover scenarios since serving BS manages all MSs and RSs in its cell. As the serving BS is not updated in intra BS handover, the MS need not accomplished whole network re-entry procedure. All other scenarios of handover according to IEEE 802.16e or IEEE 802.16j introduce the interruption composed of all handover stages subsequent to the handover decision and initiation phase.

The same frame length and scenarios based on the same assumptions as in section 2.1.1.1 are considered for evaluation of the handover interruption. Exact values of each stage of each scenario are summarized in Table 14.

Table 14. Parameters for evaluation of handover interruption duration

Delay	Duration of stage [frames]		
	Scenario A	Scenario B	Scenario C
T_{sync}	2	2	1
T_{cont_res}	2	0	0
T_{rng}	7	7	5
T_{auth}	5	5	5
T_{reg}	2	2	2
T_{fastHO}	1	1	1

The overall interruption as a result of FPHO can be derived from the exchange of management messages presented in Figure 43. The FPHO interruption is defined by the following equation:

$$D_{FPHO} = T_{sync} + T_{cont\_res} + T_{rng} + T_{fastHO} \tag{37}$$

where T_{fastHO} represents a delay caused by the verification of MS by target BS (delivery of Fast_HO-INFO). This value corresponds to the duration of just one frame.

Equation (37) is generally valid for both, uplink and downlink directions with completely re-authenticated communication (i.e., the MS is completely authorized and registered at the target BS). Nonetheless, sometimes the fast communication without completed re-authentication in downlink is preferred. In this case, new serving BS can begin data transmission to the MS immediately after the successful synchronization. The ranging process is done later since it is related to the communication in uplink channel. Therefore, the FPHO interruption in downlink direction without re-authenticated communication (noted later in thesis as NA - Not re-Authenticated) is defined by the following formula:

$$D_{FPHO\_NA} = T_{sync} + T_{cont\_res} + T_{fastHO} \tag{38}$$

Equations (37) and (38) express the situation in inter BS handover scenario. As noted above, the intra BS handover scenario does not require complete network re-entry procedure as the serving BS stays identical. Thereupon the last term in (37) and (38) can be neglected since network pre-entry procedure is not performed and hence Fast_HO-INFO is not transmitted.

Figure 44 – Figure 46 show the behavior of handover interruption time over the frame duration of conventional IEEE 802.16e and proposed FPHO procedures for all individual scenarios. All scenarios are designed to be in line with typical durations of all stages while no lost of message is assumed and no additional delay (e.g. due to network overload) of management message is considered. This assumption enables to obtain results not biased by effect of random packet losses. The dotted lines with circle mark represent the handover interruption when handover procedure is performed according conventional IEEE 802.16e. The difference in delay between IEEE 802.16e and IEEE 802.16j inter BS handover with two hops is equal to the sum of signal processing in RSs. However, the signal processing delay is insignificant as the processing time of management messages is negligible in comparison to the frame duration (see e.g. [87]). The results of optimized procedure for the downlink with Completed re-Authentication (CA) and for the uplink are presented by solid lines with *x* mark and dash-dotted lines

with plus mark for the intra BS handover and the inter BS handover respectively. The dashed lines with diamond mark represent the results for inter and intra BS handover while NA communication in the downlink direction between the MS and BS is enabled.

The results of scenario A and scenario B are illustrated in Figure 44 and Figure 45 respectively. Scenario B takes advantage over scenario A from the utilization of dedicated ranging slot. As Figure 44 and Figure 45 depict, the conventional IEEE 802.16e handover can not be performed fast enough to achieve the handover interruption under the level of IEEE 802.16m requirement (27.5 ms; depicted by orange dashed line in all figures) independently on the frame duration. On the other hand, FPHO enables the handover interruption less than 27.5 ms for frames with duration of 2 ms in downlink and uplink CA communication in scenario A. Moreover, Scenario B leads to the handover interruption less than 27.5 ms for the frame duration of 2 and 2.5 ms for CA communication in both directions. The interruption achieved by FPHO fulfills the IEEE 802.16m requirements up to 8 ms and 12.5 ms frame duration for the NA downlink communication in scenario A and scenario B respectively.

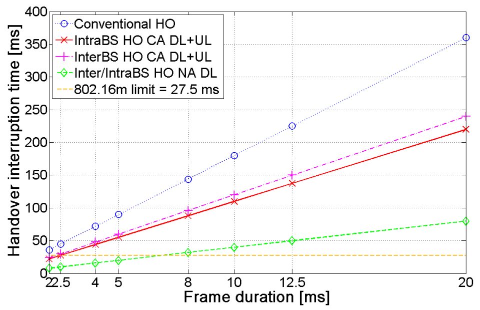


Figure 44. Handover interruption time over frame duration – scenario A

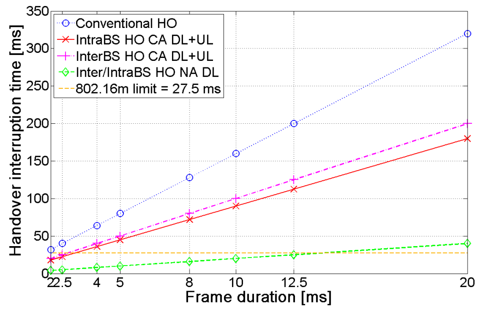


Figure 45. Handover interruption time over frame duration – scenario B

From the results reached by Scenario C (Figure 46) can be observed that the IEEE 802.16m requirements are fulfilled for all frame durations if NA communication in downlink channel is performed. The handover interruption for inter and intra BS handover with CA downlink and uplink communication enables to meet the IEEE 802.16m requirements for the frames with duration up to 4 ms. If the conventional handover is utilized, the IEEE 802.16m requirements are met only for frames with length of 2 ms. Since the scenario C represents optimal scenario, the values of handover interruption are minimum values that can be achieved in praxis only if ideal conditions are attained.

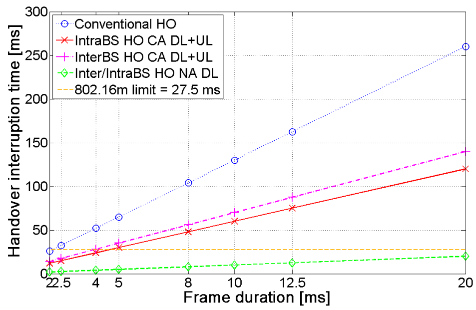


Figure 46. Handover interruption time over frame duration – scenario C

The overall reduction of handover interruption in comparison to conventional IEEE 802.16e handover is presented in Figure 8. The figure illustrates achieved reduction of the handover interruption as a percentage of conventional IEEE 802.16e handover interruption for all investigated scenarios. Since the dependence of handover interruption duration over frame duration is linear, the percentage reduction is identical for all frame durations. The FPHO enables reduction between 39% and 53% for intra BS handover with the CA communication in the downlink and uplink direction and between 32% and 46% for inter BS handover with the CA communication also in both directions. The highest reduction, i.e. between 78% and 92%, is reached in the downlink direction while using NA communication. The handover interruption reduction in percentage rising with using more optimized scenario (Scenario A shows lowest reduction whereas scenario C enables highest ratio of reduction).

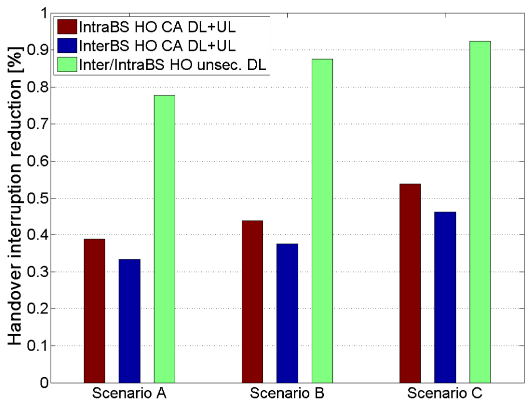


Figure 47. Handover interruption reduction by FPHO

The duration of ranging procedure is affected by initial set up of ranging parameters. Therefore, it can take more time than presented in Table 14. It leads to the assumption that the conventional IEEE 802.16e handover interruption in real networks will be more probable between results of scenario A and scenario B. On the other hand, the FPHO enables to more precise set up of these parameters during network pre-entry. Hence the results of the FPHO will be more likely between results of scenario C and scenario B in practical cases.

The complete list of frame durations that fulfill the IEEE 802.16m requirements is summarized in Table 15. Bold text highlights the improvements achieved by FPHO in comparison to the conventional IEEE 802.16e handover. Note that the intra BS handover is not defined in IEEE 802.16e since RSs are not considered. Therefore the values in appropriate cells of table express the duration of handover according to IEEE 802.16j while processing of signal by RS is neglected.

Table 15. List of frame durations that fulfil IEEE 802.16m requirements

HO Type	Scenario A		Scenario B		Scenario C	
	Inter BS	Intra BS	Inter BS	Intra BS	Inter BS	Intra BS
Conventional HO	NO	2; 2.5; ms (802.16j)	NO	2; 2.5; ms (802.16j)	2 ms	2; 2.5; 4 ms (802.16j)
FPHO CA DL+UL	**2** ms	2; 2.5 ms	**2**; **2.5** ms	2; 2.5 ms	2; **2.5**; **4** ms	2; 2.5; 4 ms
FPHO NA DL	**2**; **2.5**; **4**; **5** ms	2; 2.5; **4**; **5** ms	**2**; **2.5**; **4**; **5**; **8**; **10**; **12.5** ms	2; 2.5; **4**; **5**; **8**; **10**; **12.5** ms	2; **2.5**; **4**; **5**; **8**; **10**; **12.5**; **20** ms	2; 2.5; 4; **5**; **8**; **10**; **12.5**; **20** ms

The results presented in Figure 44 – Figure 46 consider exchange of only one SA. The proposed FPHO scheme is not influenced by a number of SAs as in case of conventional handover (see section 2.1.1.2) since the number of SAs influences only the duration of authorization phase. The authorization can be done for unlimited number of SAs via serving BS in advance to the handover interruption in FPHO.

4.4 Analysis of Impact of FPHO on Handover Overhead

The overall overhead of a handover procedure is equal to the sum of overhead generated during each stage (OH_i) as described in the following equation:

$$OH_{HO} = \sum_{i=1}^{n_{HOstage}} OH_i \tag{39}$$

where $n_{HOstage}$ represents a number of all stages that should be performed during the handover procedure. Following messages are exchanged subsequently to the time when the handover decision is done: MOB_BSHO-REQ (only if the handover is initialized by BS), MOB_MSHO-REQ, MOB_BSHO-RSP, MOB_HO-IND, UL-MAP, RNG-REQ, RNG-RSP, PKM-REQ, PKM-RSP, REG-REQ, REG-RSP. As can be observed from [2], actual size of handover overhead heavily depends on configuration of network, MS's requirements, number of neighboring stations and number of parameters that are exchanged between a MS and a serving BS.

The only additional messages transmitted over wireless interface between the MS and BS are Fast_HO-INFO and HO_PRED-INFO. The size of first message is 48 bits or 88 bits if it is transmitted by the BS or by the MS respectively. The size of second message is 72 bits. It results to the overall increase of overhead for 208 bits (48 + 88 + 72). The overall overhead created during handover according to [2] is in order of tens kbits. Hence, the increase of overhead by both messages is negligible.

Another factor that influents the handover overhead of FPHO is efficiency of a target BS prediction. The ratio of handover overhead produced by FPHO (OH_{FPHO}) and overhead of conventional IEEE 802.16e handover (OH_{convHO}) can be defined by next equation:

$$\frac{OH_{FPHO}}{OH_{convHO}} = WPR \times \frac{OH_{PE} + OH_{FastHO}}{OH_{convHO}} = WPR \times RSO \tag{40}$$

where WPR represents ratio of wrongly predicted target BSs since only wrong prediction affects additional overhead. If no prediction is performed, the FPHO cannot be executed. Next part of (40) expresses a ratio between the overhead generated before handover decision as the result of modification of handover procedure and the overhead of conventional handover. The overhead generated before handover decision due to the FPHO consists of the overhead of network pre-entry (OH_{PE}) and overhead caused by exchange of Fast_HO-INFO and HO-PRED-INFO (OH_{FastHO}). This part of equation is denoted as RSO (Ratio of Shifted Overhead).

Note that not all users need to use the FPHO since it is profitable especially for delay sensitive services. To compare the impact of FPHO on handover overhead from the system point of view, the ratio of users that execute FPHO should be considered as shown in the following formula:

$$\frac{OH_{FPHO}}{OH_{convHO}} = WPR \times RSO \times \frac{NU_{FPHO}}{NU_{All}} = WPR \times RSO \times RU \tag{41}$$

Last term, noted RU (Ratio of Users) represents a ratio between number of users that accomplish FPHO (NU_{FPHO}) and number of all users in the system (NU_{All}).

The above mentioned facts lead to the conclusion that the utilization of target BS prediction scenario with minimum WPR is preferred. Than the utilization of low HO_{Zone} seems more effective (see Figure 42). However, low level of HO_{Zone} results into significant increase of not predicted handover that forbids the utilization of FPHO. Therefore, the tradeoff between these statements must be considered.

Exact evaluation of the overhead can be easily expressed by example. If we assume the utilization of scenario F and K from section 3 (see Table 10) with WPR = 8% and 10% at HO_{Zone} = 4 dB, RSO = 2/3 and RO = 1/2, the rise of handover overhead due to utilization of the FPHO is 2,66% (8% x 2/3 x 1/2) and 3.33% (10% x 2/3 x 1/2) respectively. This rise of overhead is marginal and can be neglected. The insignificance of overhead increase is more obvious if the user's throughput in order of low level Mbps [88] [89] and overhead generated during a handover in order of tens kbits per handover is assumed [2]. Moreover, handover occurrence is not often than few times per minute (see e.g. section 3.3.2.1). Consequently, the impact of FPHO on user's throughput is not perceptible.

4.5 CONCLUSION

The proposed FPHO procedure enables to reach IEEE 802.16m requirements on the duration of handover interruption for longer frame duration in comparison to conventional IEEE 802.16e handover procedure.

The decrease of handover interruption duration is accomplished by the exploitation of results of proposed target BS prediction. This prediction is performed by serving BS. The results of prediction are delivered to the MS in form of the new MAC management message that contains information on the target BS. It allows a negotiation of parameters of connection, re-registration and re-authorization with the predicted target BS before the MS is disconnected from the current serving BS. The negotiation of parameters is performed via serving BS. The negotiation is completed by transmission of the fast handover information (Fast_HO-INFO) from the target BS to MS. This message can be delivered either as a stand alone message or it can be appended to the

MOB_BSHO-REQ that is transmitted in all handover cases. The content of Fast_HO-INFO is later utilized to perform faster re-authentication of the MS.

The FPHO results into reduction of the handover interruption between 77% and 92% (depending on scenario) in downlink if not re-authenticated communication is enabled. If the communication with completed re-authentication in downlink and uplink is required, the FPHO decreases the handover interruption for 33% – 47% in case of inter BS handover scenario. The interruption caused by intra BS handover is lowered for 40% – 55% in both directions of communication. Fast handover that fulfils IEEE 802.16m requirements on the handover interruption is achieved for intra BS as well as for intra BS handover in all considered scenarios. Moreover, the FPHO enables to satisfy the requirements also for longer frame durations in comparison to the conventional IEEE 802.16e standard. Considering the results obtained in 2.1.1.2, the speech quality can be increased up to roughly 0.7 MOS. The exact level of improvement varies with the frame and call durations.

The novel handover procedure leads to the transmission of several additional information that increase overall overhead generated by the handover. The impact of FPHO on the overhead depends mainly on the ratio of wrong predictions of target BS. If the results of chapter 3 are considered, the rise of handover overhead is in order of ones percents. Therefore it can be completely neglected as it absolutely does not influence the user’s throughput.

The general principle of FPHO can be applied to other mobile wireless technologies such as LTE. Moreover, the implementation of FPHO to existing networks based on IEEE 802.16e standard need no modification of hardware of currently provided equipment. An implementation of FPHO is only a question of update of software related to the MAC management layer.

5 CONCLUSIONS AND FUTURE WORK

This thesis addresses the problem of users' mobility in relation to the quality of service known as the handover interruption.

5.1 GENERAL CONCLUSIONS

Important factors influencing the duration of handover interruption are i) number of SA and ii) duration of frame utilized for communication on physical layer. Both parameters influence the duration of handover interruption linearly. In optimal case, the IEEE 802.16m requirements on handover (it means to reach the handover interruption less than 27.5 ms) can be met only by using frames with duration of 2 ms and assuming optimal handover procedure. However, these requirements cannot be fulfilled in any more realistic scenarios. It results into the decrease of speech quality of VoIP communication. The impact of handover interruption on VoIP speech quality depends on the frame duration as well as on the intervals between handovers. Therefore, the utilization of shorter frames and the reduction of number of handovers lead to the improvement of speech quality.

The target BS prediction techniques are utilized to cope with the negative impact of handover interruption. The efficiency of target BS prediction by two approaches (utilization of handover history as well as exploiting of channel characteristics) is investigated in this thesis. The implementation of both techniques into real network is very simple and requires no modification of hardware. Only a simple modification of BS's control mechanism is required. Both techniques need some time to adapt to modifications in surrounding environment or to changes of channel characteristics since enough number of information related to handovers must be collected. The first technique (handover history based prediction) requires to perform roughly 19 handovers per a BS to achieve stable prediction efficiency. The second technique enables to reduce this amount to approximately 10 handovers per BS.

The efficiency of handover history based prediction depends strongly on the number of neighboring BSs. The efficiency is up to roughly 47% for three neighboring BSs. Therefore, this way of prediction does not achieve enough high level of efficiency to utilize it for the prediction of handover as a standalone method.

The maximum hit ratio of prediction based on channel characteristics significantly depends on the variation of channel characteristics and on the level of HO_{Zone}. As the results of the analysis of prediction efficiency improvement indicate, the level of prediction hit ratio can be positively influenced by using HM, HDT as well as windowing techniques. The best performance is achieved by HM. On the other hand, this technique prolongs the time for collection of information on previous handover. Even the worst performing method, i.e. HDT, enables to increase the prediction hit ratio in comparison to the case with no technique considered. Moreover, the advantage of HDT is in a decrease of the time interval necessary for collection of sufficient amount of data for efficient prediction. The maximum prediction hit ratio when individual techniques are combined together is 93%. The utilization of only HM and windowing results to the highest prediction efficiency as the HDT leads to the increase of ratio of not predicted handovers.

The ratio of successfully predicted handover achieved by proposed technique enables to design handover procedure with minimized interruption and with negligible impact on the handover MAC management overhead. The proposed procedure, called FPHO, allows to reach the IEEE 802.16m requirements on the duration of handover interruption for higher frame duration in comparison to the conventional IEEE 802.16e handover. The FPHO exploits the results of prediction to enable a negotiation of connection parameters, re-registration and re-authorization with the predicted target BS before the MS is disconnected from the current serving BS. The negotiation of connection parameters, re-registration and re-authorization are performed via serving BS. After the disconnection from the serving BS, the MS executes only accelerated re-authorization that requires only a transmission of just one MAC management message. Faster handover that fulfils IEEE 802.16m requirements on the handover interruption is achieved for scenarios without RSs as well as for scenario with RSs (for both intra BS handover and intra BS handover). The impact of FPHO on the management overhead depends on several parameters such as the ratio of wrong predicted target BSs or ratio between overhead generated during pre-registration and overall handover overhead. The rise of overhead due to the FPHO is insignificant.

The general principles of FPHO as well as target BS prediction are not limited to the utilization only in WiMAX networks. Both are applicable on the other mobile wireless technologies such as LTE or LTE-A. The prediction technique is designed completely with no relation to some network technology. Therefore, it is independent on a type of utilized mobile wireless network. The FPHO is developed, described and analyzed for MAC and PHY layers corresponding to the WiMAX technology, however the general principle and exploitation of prediction results can be easily transferred to other technology.

The proposed solution is designed with respect to possible adoption to the currently developed IEEE 802.16m standard. Therefore, it is designed with no requirements on modification of hardware of currently used WiMAX devices and equipments manufactured according to IEEE802.16e standard. Only software modifications of handover control procedure must be implemented to utilize the FPHO in real mobile wireless networks.

5.2 FUTURE WORK

Future investigations in the area of handovers or general support of user's mobility can be divided into three groups. The first one is to further improve the prediction efficiency by all three techniques: handover history, channel characteristics and MS's movement. Especially, the third one provides a lot of areas to further research (e.g. utilization of advanced algorithms for prediction of user's movement together with prediction of users profile evolution). Also the combination of those three methods can lead to the improvement of prediction efficiency. Moreover, the results of prediction can be exploited e.g. for optimization of resource allocation or admission control. Next way of future investigation is related to the implementation of RSs into network since it requires a new approach to the handover procedure. In this scenario, the handover is more related to the selection of the best routing path. Therefore, the investigation of innovative techniques to enable the handover initiation based on conditions on individual hops between a MS and its serving BS can bring significantly more effective assignment of radio resources. Another way of investigation can tackle the minimization of MAC management overhead as it influence overall throughput of users. This topics gets more relevancy for networks with RSs since in this scenario, each message is transmitted repeatedly over each hop. It leads to the rapid rise of total overhead transmitted over the wireless part of communication chain. Very promising approach is a joint transmission of individual management messages, produced by MSs connected to the network via the same RS, by the access RS.

RESEARCH CONTRIBUTIONS

The thesis is focused on the innovation of handover procedure in mobile wireless networks. The summarized contributions of the thesis into the area of handovers are following:

Chapter 2

- exact evaluation of the handover interruption in WiMAX networks over the frame duration
- proof and evaluation of the negative impact of handover interruption on the VoIP speech quality
- analysis and evaluation of the impact of techniques originally proposed with purpose of the reduction of handover amount on the throughput of a single MS

Chapter 3

- analysis of the efficiency of handover history based prediction over a number of neighboring BSs
- design of the channel characteristics based prediction technique for determination of the next target BS
    - this technique is applicable to the majority of handover related parameters such as CINR, RSSI, RTD, etc.
    - utilization of this technique is not limited only on WiMAX networks, the same principle is applicable to other standards for mobile wireless networks such as LTE, LTE-A
    - the introduction of this technique into real systems need no hardware modification of current IEEE 802.16e equipment
- improvement of the target BS prediction efficiency by techniques for reduction an amount of redundant handovers
    - generally, this is also applicable to other mobile wireless standards and it needs no hardware modifications of current IEEE 802.16e equipment

Chapter 4

- design of the fast handover procedure with reduced handover interruption by exploitation of the results of target BS prediction
    - it contains designs of:
        - new MAC management messages

        - complete exchange of MAC management message flow during handover procedure
    - general principle and the basic idea of novel handover procedure is also applicable to the other mobile wireless standards
    - novel technique requires no modification of IEEE 802.16e equipment
- evaluation and analysis of the impact of proposed fast handover procedure on:
    - the duration of handover interruption including scenarios corresponding to WiMAX networks with RSs
    - MAC management overhead produced due to handover procedure
    - throughput of a single MS

The topics investigated within the thesis result into several publications in conferences and journals. The details of each paper are as follows.

All results are included in deliverables of FIREWORKS [90] and ROCKET [91] projects funded by European Commission.

- Hoyman, C., et. al., "Advanced Radio Resource Management Algorithms for Relay-based Networks," Deliverable 2D2 of IST FP6-027675 FIREWORKS project, 2007.
- Sambale, K., Bečvář, Z., Mach, P., Ulvan, A., Bourdelles, M., “Mechanisms for increasing the efficiency of MAC/PHY protocols,” Deliverable D8 of ICT-215282 STP ROCKET project, November 2009.

Results related to ***chapter 2*** are included in:

- Bečvář, Z., Zelenka, J., "Implementation of Handover Delay Timer into WiMAX," 6th Conference on Telecommunications, Lisboa: Instituto de Telecomunicaçoes, 2007, pp. 401-404.
- Bečvář, Z., Mach, P., Bešťák, R., "Impact of Handover on VoIP Speech Quality in WiMAX Networks," The Eighth International Conference on Networks (ICN 2009), Los Alamitos: IEEE Computer Society, 2009, pp. 281-286.

Results related to ***chapter 3*** are included in:

- Bečvář, Z., "Efficiency of Handover Prediction Based on Handover History," Journal of Convergence Information Technology JCIT, Vol. 4, No. 4, 2009.

- Bečvář, Z., Mach, P., Šimák, B., "Improvement of Handover Prediction in Mobile WiMAX by Using Two Thresholds," Computer Networks, Elsevier, Vol. 55, No. 16, November 2011.

Results related to ***chapter 4*** are included in:

- Bečvář, Z., Mach, P., "Fast Predicted Handover in IEEE 802.16 Networks," European Transactions on Telecommunications Journal (ETT), John Wiley & Sons, Vol. 22, No. 2, March 2011.
- Bečvář, Z., Mach, P., "Fast Predicted Handover," Contribution to IEEE 802.16m standard no. IEEE C80216m-09_2308, November 2009, available at: < http://dot16.org/ul//upload/TGm_db/C80216m-09_2308.pdf >.

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APPENDIX A

A1 – SPEECH QUALITY DEGRADATION END EVALUATION

During a VoIP communication, several complications such as a packet loss, packet delay or jitter may be recognized. The most of these problems generally lead to losses of data flow continuity and further to the losses of signal information elements. From the point of human perception system view, these losses are represented as dropouts in speech.

The speech quality can be evaluated either by subjective tests or by objective methods [92]. Both, subjective and objective ways, usually use a parameter MOS (Mean Opinion Score) [93] to a speech quality assessment. A MOS scale range used in subjective tests is from 5 to 1 (Excellent = 5; Good = 4; Fair =3; Poor = 2; Bad = 1). In practice, objective methods are usually used due to easier implementation than the subjective ones. In this thesis, the objective method PESQ (Perceptual Evaluation of Speech Quality) [94] is used for an evaluation of a quality of speeches affected by the handover procedure. The PESQ is one of the most spread objective methods developed for end-to-end speech quality assessment in a conversational voice communication.

The principle of PESQ is depicted in Figure 48. The PESQ method is based on the comparison of original (non-degraded) signal *X(t)* with degraded signal *Y(t)*. The signal *Y(t)* is result of transmission of signal *X(t)* through a communication system. The PESQ method generates a prediction of quality which would be given to signal *Y(t)* in subjective listening test.

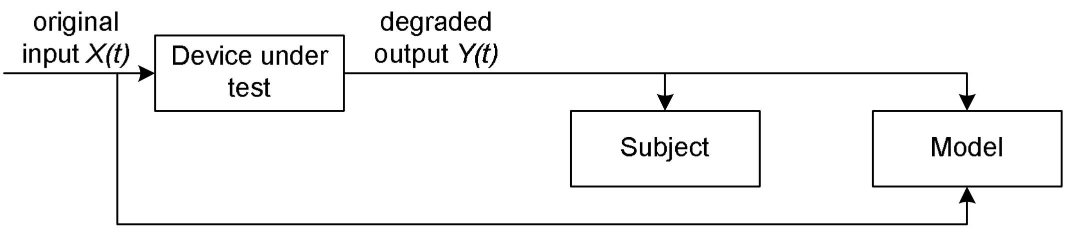

Figure 48. PESQ principle

The range of the PESQ MOS score (according to ITU-T P.862) is between -0.5 and 4.5. Since this range does not correspond to a scale used for the subjective test, the ITU-T P.862.1 recommendation [95] enables to recalculate the PESQ MOS according to formula (42) to better comport the results of subjective test (range from 1 to 4.55).

$$y = 0.999 + \frac{4.999 - 0.999}{1 + e^{-1.4945 \times x + 4.6607}} \tag{42}$$

where x is an objective PESQ MOS score and y is a matching ITU-T P.862.1 MOS score.

ITU-T P.862 recommendation describes all requirements on the tested speech signal (e.g. the character of speech signals, duration of speech, etc.). Frequency characteristics of the speech signal and signal level alignment must be in accordance with recommendation ITU-T P.830 [96].

A2 – MODIFICATION OF SPEECHES FOR OBJECTIVE QUALITY ASSESSMENT

Speeches for the evaluation of handover impact are modified according to the voice signal degradation caused by the handover in real mobile wireless network. Since separated impact of only handover is investigated, core network packet losses and jitter are neglected. All speech processing is done in MATLAB. Individual parts of speech modification and evaluation are depicted in Figure 49.

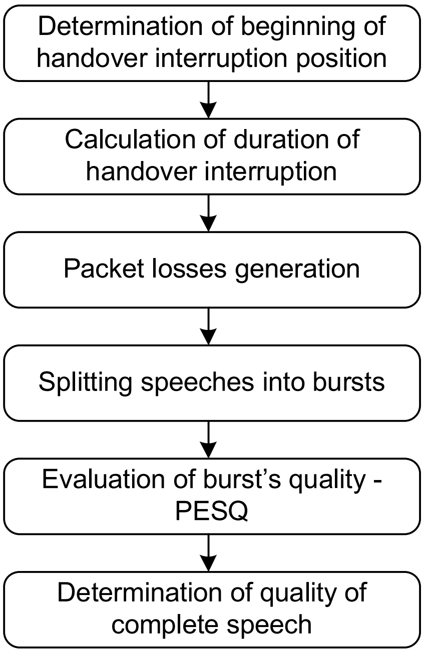

Figure 49. Process of calculation of handover impact on speech quality

The first step is a determination of all positions of handover interruptions positions. This task corresponds to the determination of a time instant when the handover occurs. Based on the mobility model defined for handover evaluation according to IEEE 802.16m evaluation methodology [17], handover in periodic intervals can be assumed. This simplification has only a negligible impact on the results as a high number of speeches for the speech quality evaluation is used. This mobility model assumes a direct movement of users with a constant speed among the regular hexagonal cells with same radius. The calculation of duration of handover interruption is based on the principle described in Appendix A1. Overall delay is converted to the amount of lost VoIP packets (LP):

$$LP = \frac{D_{TOT}}{PL_{VoIP}} \tag{43}$$

where PL_{VoIP} is the length of VoIP data packet. The parameter PL_{VoIP} is assumed to be 20 ms since this value corresponds to the typical packet length used in real VoIP communication [74]. Like the lost packet is considered each packet which delay is higher than 150 ms. This value is defined as level of high quality speech communication in ITU-T G.114 [97].

The packet losses are integrated into speeches by zeroing of speech samples at appropriate positions. The whole speech is split into several bursts with an approximate duration of 10 s to meet the requirements for evaluation by PESQ [94] [95]. The results of PESQ method are recalculated according to ITU-T P.862.1 using (42). The quality of whole speech (SQ) is calculated as the average of burst's quality (BQ) pertaining to the same speech (see following equation).

$$SQ = \frac{\sum_{n=1}^{n_{burst}} BQ_n}{n_{burst}} \tag{44}$$

where n_{burst} represents an amount of burst of the speech. In total, 250 speeches is generated for each length of intervals between two handovers and for each duration of handover interruption. The results of all 250 speeches are averaged out to avoid the impact of random drops of packet losses into silent parts of the speeches.

APPENDIX B

PATH LOSS MODELS

Two models for path loss calculation are utilized in the thesis: urban macrocell and urban microcell. Both are described in IEEE 802.16m evaluation methodology [18].

B1 – URBAN MACROCELL

This model is defined for frequencies in range *2<f<6* GHz. Actual path loss is determined according to the consequent equation:

$$PL_{urban\_macro}[dB] = 35.2 + 35 log_{10}(d) + 26 log_{10}(f/2) \quad (45)$$

where d is a distance between MS and BS.

B2 – URBAN MICROCELL

The model for urban microcell defines different calculation of path loss based on the availability of line of sight between the BS and the MS.

The path loss for scenario with *LOS* is determined by the next formula:

$$\begin{aligned} PL_{urban\_micro\_LOS}[dB] &= 32.4418 + 20 log_{10}(f) + 20 log(d) + 0.0174d \\ &+ 20 log_{10}(max(0.013d/f, 1)) \end{aligned} \quad (46)$$

If only *NLOS* communication is available, the path loss is calculated by the following way:

$$PL_{urban\_micro\_NLOS}[dB] = min(PL_{over\_the\_rooftop}, PL_{Berg}) \quad (47)$$

where $PL_{over\_the\_rooftop}$ is a path loss of communication over the rooftops. $PL_{over\_the\_rooftop}$ and PL_{Berg} are calculated according to the next equations:

$$PL_{over\_the\_rooftop} = 24 + 45 log_{10}(r_{Eu} + 20) \quad (48)$$

$$\begin{aligned} PL_{Berg} &= 32.4418 + 20 log_{10}(f) + 20 log_{10}(d_n) + 20 log_{10}(max(R/r_{bp}, 1)) \\ &+ 20 log(R) + 0.0174R \end{aligned} \quad (49)$$

where R is a distance along streets between transmitter and receiver; r_{Eu} is the Euclidean distance in meters from the transmitter to the receiver. The distance d_n is the illusory distance and it is defined by the recursive expression:

$$k_j = k_{j-1} + d_j q_{j-1}$$
$$d_j = k_j r_{j-1} + d_{j-1} \tag{50}$$

with $k_0 = 1$; $d_0 = 0$ and $q_j(\theta_j) = \left(\frac{|\theta_j|}{90}\right)^{1.5}$.

where θ_j is the angle between streets at junction j.

Further, r_{bp} and R are calculated as describe following formulas:

$$r_{bp} = min(76.67f, r_0) \tag{51}$$

$$R = \sum_{j=1}^{n} r_{j-1} \tag{52}$$

The distance r_j represents length of street between nodes j and $j+1$.

B3 – SHADOWING

Impact of shadowing is expressed by Shadowing Factor (SF). The SF has log-normal distribution with standard deviation based on [98] (see Table 16).

Table 16. Standard deviation of SF for different path loss models

Path loss model	Standard deviation [dB]
Urban microcell NLOS	4
Urban microcell LOS	3
Urban macrocell	8

The SF is obtained by interpolation by using the grid with uniform spacing is generated for each BS as shown in Figure 50.

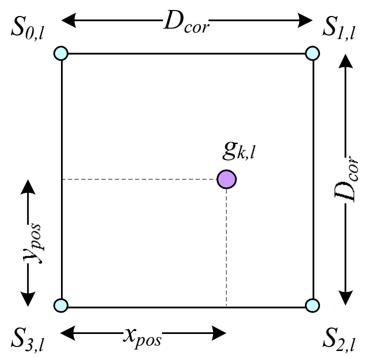

Figure 50. Interpolation of shadowing factor

In Figure 50, $S_{n,l}$ represents SF corresponding to the BS l at the geographic location n. The location is identified by coordinate (x, y). The overall number of BSs in a simulation is denoted L. The distance between two closest nodes is denoted as de-correlation distance D_{cor}. The SF from a MS at a specific location to the BS l is evaluated by interpolation of the SF of four closest nodes ($S_{0,l} - S_{3,l}$ in Figure 50). The $SF(g_{k,l})$ of BS l is calculated by the consequent equation:

$$SF(g_{k,l}) = \left(\sqrt{1-\frac{x_{pos}}{D_{cor}}}\right)\left[S_{0,l}\sqrt{\frac{y_{pos}}{D_{cor}}} + S_{3,l}\left(\sqrt{1-\frac{y_{pos}}{D_{cor}}}\right)\right] + \left[S_{1,l}\sqrt{\frac{y_{pos}}{D_{cor}}} + S_{2,l}\left(\sqrt{1-\frac{y_{pos}}{D_{cor}}}\right)\right]\sqrt{\frac{x_{pos}}{D_{cor}}} \quad (53)$$

B4 – CHANNEL VARIATION

Channel variation is represented by low signal level fluctuation [79]. The overall path loss with channel variation is equal to:

$$PL_{CV}[dB] = PL + (CV_{rand} \times S) \quad (54)$$

where PL is the path loss defined by (45), (46) or (47); CV_{rand} is a random level of fluctuation with value of CV_{rand} randomly selected according to lognormal distribution with $\mu = 0$ and $\sigma = 0.8$ (or as defined in particular scenario) [79]; and S is a sign of CV_{rand}. The S is expressed by following formula:

$$P(S_i = 1) = \begin{cases} 0.2 & S_{i-1} = -1 \\ 0.8 & S_{i-1} = 1 \end{cases} \qquad P(S_i = -1) = \begin{cases} 0.8 & S_{i-1} = -1 \\ 0.2 & S_{i-1} = 1 \end{cases} \quad (55)$$

APPENDIX C

MOBILITY MODELS

Two mobility models, PRWMM [86] or MMM [84], are utilized in simulations. Those models are not defined in [18]; however both are generally used for mobility simulations. The PRWMM is designed for simulation of movement in a free space whereas the MMM corresponds to the behavior of users in the downtown with regular deployment of streets.

C1 – PROBABILISTIC RANDOM WALK (WAYPOINT) MOBILITY MODEL

The PRWMM utilizes a probability matrix to determine the position of particular MS in the next time which is represented by three different states for position *x* and three different states for position *y* (see Figure 51).

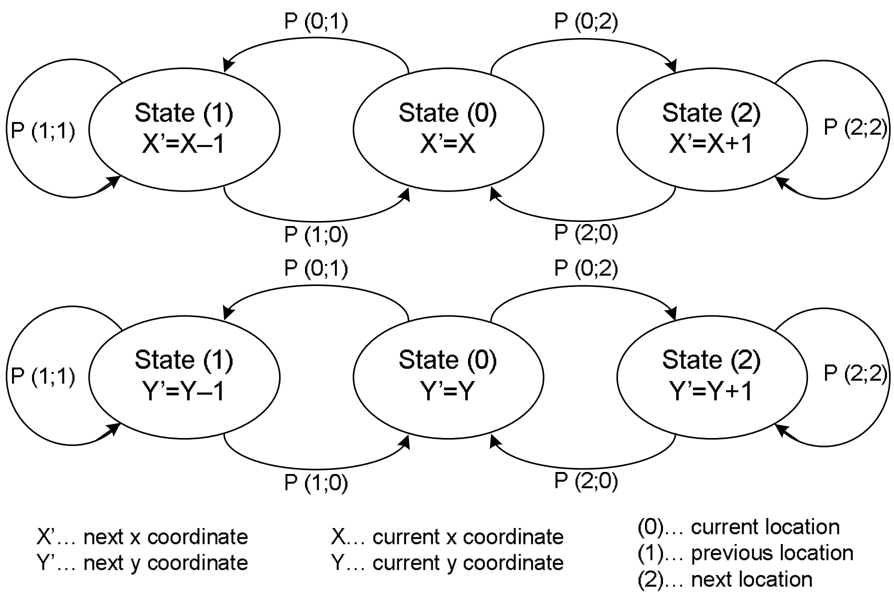

Figure 51: States in Probabilistic Random Walk Mobility Model

State 0 represents the current (*x* or *y*) position of given MS, *state 1* represents the MS's previous (*x* or *y*) position, and *state 2* represents the MS's following position. The probability of next movement direction is described by subsequent matrix:

$$P = \begin{bmatrix} P(0;0) & P(0;1) & P(0;2) \\ P(1;0) & P(1;1) & P(1;2) \\ P(2;0) & P(2;1) & P(2;2) \end{bmatrix} \tag{56}$$

The exact values set for the simulation are following:

$$P = \begin{bmatrix} 0 & 0{,}5 & 0{,}5 \\ 0{,}2 & 0{,}8 & 0 \\ 0{,}2 & 0 & 0{,}8 \end{bmatrix} \qquad (57)$$

C2 – MANHATTAN MOBILITY MODEL

The MMM models urban environment as a two-dimensional rectangular grid of streets and buildings. An example of a Manhattan grid with 5 horizontal and 7 vertical streets is shown in Figure 52.

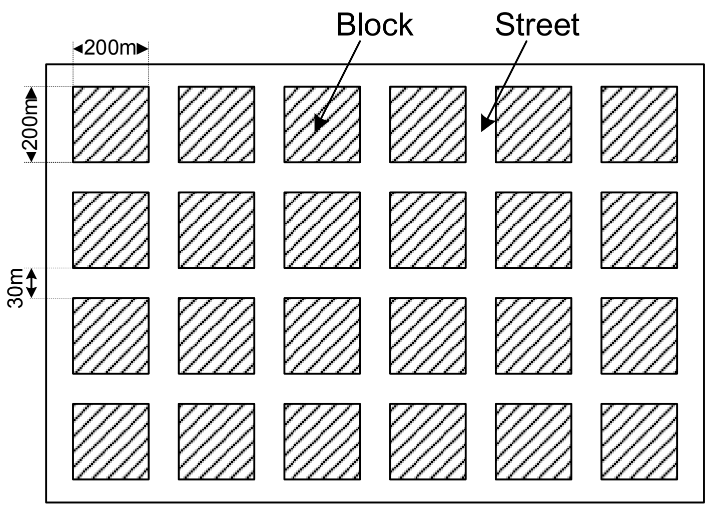

Figure 52. Street deployment for MMM with parameterization

The MMM is based on the direct movement of a MS until it reach a cross of two streets [84]. The new direction is selected at every cross. The direct movement is selected with probability 1−TP (usually TP = 0.5). The probability of turn to the right and left is TP/2. This situation is depicted in Figure 53.

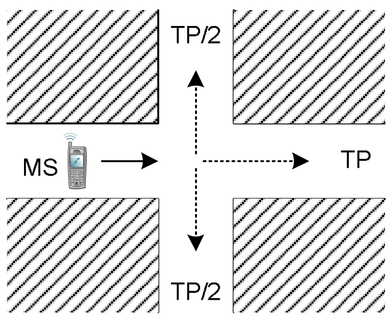

Figure 53. Turn Probability at a crossroad