INVESTIGATION OF VERTICAL HANDOFF TECHNIQUES IN INTEGRATED WLAN/CELLULAR NETWORKS AND PERFORMANCE ANALYSIS OF HORIZONTAL HANDOFF IN WIMAX NETWORKS

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Morehead State University, 2011

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Today, advent of heterogeneous wireless networks made a huge revolution in the telecommunication systems. As IP-based wireless networking increases in popularity, handoff issue is taken into the consideration. The horizontal handoff between different cells in the network working under same technology should be managed in a way to satisfy users with high quality services. In the case that the user switches between networks under different technologies, many issues should be considered in order to increase the efficiency of the network during vertical handoff.

This thesis is divided into two parts: In the first part of this thesis, different algorithms designed for optimizing vertical handoff between a Wireless Local Area Network (WLAN) and a cellular network were compared. In the comparison part, advantages and disadvantages of those algorithms were discussed and it was mentioned what would be the effect of considering each factor on optimizing vertical handoff execution. Then, a new model for calculating the probability of vertical handoff occurring between WLAN and cellular network by taking RSS (Received Signal Strength) and application type (voice and data) into the account was proposed. In the second part of this thesis, two different scenarios for horizontal handoff in one of the most popular cellular network, WiMax, using OPNET Simulator with three types of voice applications (PCM, GSM, and IP telephony) were simulated. It was observed that PCM and GSM work better in simple networks while IP telephony is a good voice application for crowded networks.

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Chapter I. Introduction

General Background

These days, use of heterogeneous wireless networks is increasing and many fields are touched by the development of this new technology. The main purpose of heterogeneous wireless network is simulating communications. In heterogeneous wireless networks, several networks exist which work under different technologies. One of the most popular networks which is used recently for communication is the cellular network. In this section, we describe a brief history of cellular networks.

The first generation of the cellular networks (1G) began in the early 80's with commercial deployment of Advanced Mobile Phone Service (AMPS) cellular networks. For carrying out voice over channels in the 800 MHz frequency band, AMPS networks used Frequency Division Multiplexing Access (FDMA). The 1G network used analog signal for transmission (ICT, 2011).

The second generation (2G) emerged in the 90's when mobile operators deployed two competing digital voice standards, GSM and CDMA. The GSM or Global System for mobile communications, which is mainly used in the world, deployed Time Division Multiple Access (TDMA) to multiplex up to 8 calls per channel in the 900 and 1800 MHz bands. IS-95, which was used in North America, used Code Division Multiple Access (CDMA) and was able to multiplex up to 64 calls per channel in the 800 MHz band (Phifer & Lisa, 2000).

Till mid 2009, the majority GPS and CDMA was operating on the 2G network. 2G networks used digital signal for transmission. The main advantage of the 2G network is the high capability in data transmission but the problem is that the capacity is limited (Blank, 2010).

Regarding the low speed and bandwidth of the 1G and 2G networks, 3G network or third generation of mobile technology was introduced by International Telecommunications Union (ITU) in order to increase bandwidth, provide fast transmission of the mobile signal and facilitate growth in order to support a large number of applications. 3G includes EDGE (Enhanced Data rates for GSM Evolution), CDMA 2000 (Code Division Multiple Access), UMTS (Universal Mobile Telecommunications System), DECT (Digital Enhanced Cordless Telecommunications), WiMax (Worldwide Interoperability for Microwave Access) (3G Network, 2009).

By advent of the 3G networks, cell phones gained many advantages such as faster uploads and downloads. The GPS (Global Positioning System) feature was also added to some particular models, while the 2G model does not support this feature (Blank, 2010).

A 4G network is the fourth generation of wireless communication and is a step up from 3G and is currently most widespread, high-speed wireless service. The overall goal for the network is to provide a comprehensive and secure Internet Protocol solution with much faster data speeds than previous generations. In fact, a 4G wireless network such as WiMax is designed to deliver speed. On average, 4G wireless is supposed to be anywhere, anytime and can perform four to ten times faster than today's 3G networks. It supports seamless connection to a wide range of information and services, and receives a large volume of information, data, pictures, and videos. 4G networks replace the current proliferation of core mobile networks with a single worldwide core network standard, based on IP for control, video, packet data, and voice. The objective is to offer seamless multimedia services to users accessing an all IP based infrastructure through heterogeneous access technologies. IP is assumed to act as an adhesive for providing global connectivity and mobility among networks. An all IP-based 4G wireless network has inherent advantages over its predecessors. It is compatible with, and independent of the underlying radio access technology (Narisetti, 2006).

WiMax Technology

WiMax network is based on the IEEE 802.16 and it is useful for data and voice communications. It smoothes the communication as the mobile station moves between different base stations. In the WiMax technology, two kinds of handoffs occur: Soft handoff and hard handoff. Soft handoff happens when a mobile station creates a connection with the new base station before handoff occurs. Hard handoff happens when the mobile station breaks its connection with the current base station before establishing a connection with new base station. The WiMax also delivers the quality of services and it works with multiple inputs and multiple outputs (MIMO). The MIMO reduces the errors and improves data rate by using several antennas as transmitters and receivers. The WiMax also uses OFDM which is used for increasing the bandwidth by splitting the signal into smaller signal sets and modulating each of them to different subcarriers and assigning subcarriers to the base stations. The maximum data rate which can be delivered by mobile WiMax on a single channel is 70 M bits per seconds (Mbps) (Vaughan-Nicholas, 2008).

The most important factor which has been included in the MAC layer of the WiMax is the QoS. It is predicted that WiMax is totally capable of multimedia transmission such as multimedia streaming and voice over IP (VoIP). Another advantage of WiMax is that it can be used as a backbone network with long separation among the nodes because of high bandwidth transmission (Vaughan-Nicholas, 2008).

IEEE 802.16 Structure

The structure of 802.16 is shown in Figure 1. It consists of two layers: MAC and PHY layer. The MAC layer includes three sub layers: CS SAP, MAC SAP and a security sub layer. The main task of the CS (Convergence Sublayer) is transforming the external data from upper layers into proper Mac Service Data Units (SDUs) for the MAC CPS. On the other hand, the MAC CPS (MAC Common Part Sublayer) is in charge of handling QoS, system access, allocation of bandwidth, and connection establishment and maintenance. The functions such as authentication and encryption are done in the security layer. In the PHY (Physical) layer, multiple PHY specifications are supported, each of which handles a specific frequency range (Li, Qin, Low, & Gwee, 2007)

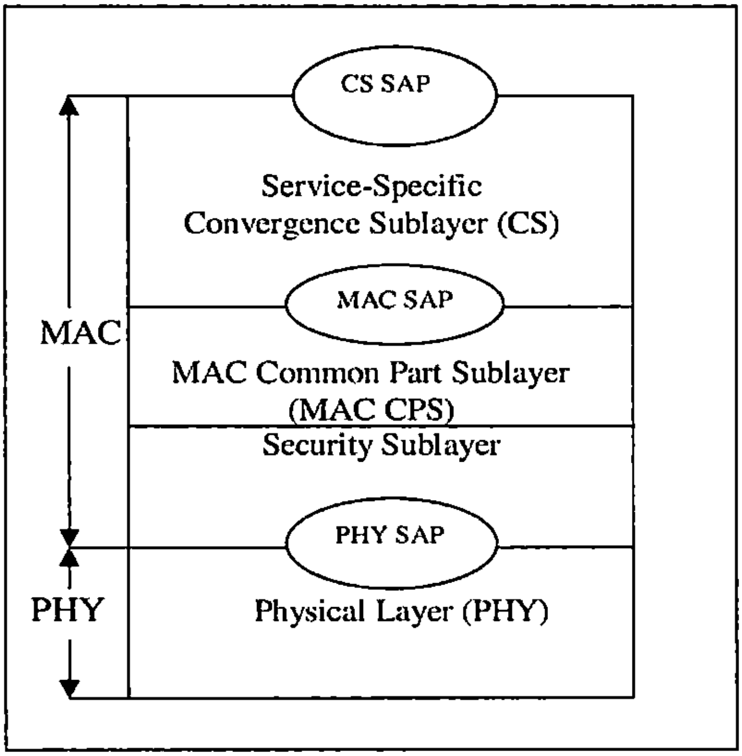

Figure1. IEEE 802.16 Structure (Li et al., 2007)

Heterogeneous Wireless Networks

Heterogeneous wireless networks consist of different Wireless Area Networks (WLAN), various cellular networks, and many other networks with different technologies. The most popular networks are WLAN and cellular networks. Recently, the use of WLAN in areas such as airports, hotels and school campuses has increased. The popularity of the WLAN is mainly laid on their low cost and their high data rate as high as 54 Mbps, as in IEEE802.11a. However, they support small area of coverage and can support users with low mobility (Narisetti, 2006).

On the other hand, although cellular networks such as 4G networks support higher degree of mobility and a wider area of coverage, they offer guaranteed quality of services in data transmission at the lower data rate. The complementary features of these two networks, WLAN and cellular network make their integration highly desirable. Their integration brings a cost-effective system, capable of providing ubiquitous data service, with high data rate service in planned locations (Wang & Kong, 2004).

Handoff Issue and Mobility Management

In a combined network consisting of WLAN and cellular networks, many issues will arise which need to be handled. One of the important issues in this regard is mobility management. In a network model, WLAN is working with their access points while cellular networks are working under their base stations. Each of these networks has a specific coverage which covers certain amount of users. Because of node mobility in the wireless networks, the nodes easily move out of their coverage and in this case cannot be connected to their main station any more. Whenever a mobile node leaves its own coverage and enters other wireless coverage, it should be able to connect to the new station in order to continue its application without being blocked (L.Chen, Sun, B.Chen, Rajendran, & Gerla, 2007). Therefore, there should be a mobility management in order to handle these kinds of issues. The event in which a mobile node from one wireless technology is connected to another technology is called handoff. Moreover, when the number of nodes which can get services from an access point in WLAN or base stations in the cellular networks exceeds what they can exactly service, the handoff will be operated and mobile nodes switch between networks with different technologies. The handoff can be divided into two groups: Vertical and horizontal.

A vertical handoff will occur when a user switches between two different network interfaces with different technologies. For example if a mobile node leaves the 802.11b network domain and enters 802.16 network domains, it is called vertical handoff. A horizontal handoff will occur when a user switches between two network access point that uses the same interface and same technologies. An example of the horizontal handoff is when a mobile users moves in various 802.16 (WiMax) network domains (Chen et al., 2007).

Handoff Methods

Handoff has several methods. The two main methods of handoff are hard handoff and soft handoff.

Hard handoff: This method which is used by Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) systems. It is called break before make which means whenever a handoff execution needs to start, the mobile node breaks the connection with the current station before switching to the other one and establishing a new connection with the new wireless technology (Narisetti, 2006). Because of nature of this method, a brief disruption of service occurs.

Soft handoff: This method is used by CDMA systems and it is called make before break. Since the user establishes a connection to the newest server before breaking the current connection with its station, there is no disruption of service.

Along with inter-network movements, seamless handoff execution should be considered as well. Seamless handoff aims to maintain the connectivity of all of the applications while hand-off occurs as well as power saving, low handoff latency and low bandwidth overhead (Narisetti, 2006). A seamless handoff is also considered a vertical handoff because the mobile node is switching from one network in the heterogeneous network to the other network with different technology. Since the IEEE 802.11 WLAN has high bandwidth, it is chosen to cover limited hot spot areas such as offices or campus areas. In the 4G networks, the WLAN coverage is overlaid by cellular networks cells. This means if a mobile node can get out of WLAN coverage it still is in the UMTS cell coverage and in this case, a seamless handoff is executed. Because of minimum size of cells in the cellular networks, the handoff is executed frequently (Narisetti, 2006).

In the handoff execution, several factors are taken into the account. The main question in this regard is: What are the characteristics of a good handoff? Recently, several studies have been done to find the best situation for the handoff execution. The first and the most important factor in taking handoff decision is the best moment. The best moment could be defined as when the best network is available and when the handoff should occur in order to provide user's application with high quality services. Ignoring this factor could result in several handoff drop calls in which users could not connect to another network. In the series of the available networks for the handoff, the best of them should be chosen. These networks should be evaluated to ensure which of them are the best networks in order to satisfy the user's application. All the handoff issues are centered on how the best network interface is selected. The best moment along with the best network are the most essential characteristics of a good handoff which helps to reduce the number of unnecessary handoff and reduces call drop probability.

In scoring networks to find the best one, several factors are taken into the account, the most important include load, delay, through put, cost, and security. Many research studies have been done to find the best algorithms for handoff decision and each of them has considered some of the factors mentioned above. It is obvious if any of the selected networks could satisfy users with high bandwidth and high throughput, minimum delay and cost and high secure data transmission, this network can be selected as the best target network for handoff operations. The question here is how to decide which the target network to use since not all of the networks could satisfy users with the mentioned factors. Here there should be a grouping to give a preference to the parameters which plays important role in handoff. In this thesis the Received Signal Strength (RSS), one of the fundamental factors in the handoff execution, as well as the application type (data session or voice session) were two parameters considered for the proposed algorithm.

Statement of the Problem

Handoff execution should be considered as an important issue in 4G networks. Since handoff is happening often in the heterogeneous wireless network, most of the time the number of dropped calls is high. The user's expectation from a network is having seamless connection as it should switch between adjunct cells in the networks with same or different technologies. Several problems could happen in handoff execution. There may be several call drops, lack of signal strength, and insufficient bandwidth. As a result, the probability of failure in handover is high and the user may lose the connection. Several studies have been done in this regard to provide users with high quality services through handover mechanism and many algorithms have been designed to improve the handoff issues in which the user experiences less dropped calls and remained satisfied with the best quality of services in the network.

The first purpose of this study is comparison different vertical handoff algorithms and suggesting our proposed algorithm and our proposed model for calculating the probability of occurring handoff between an IEEE 802.11 (WLAN) and a cellular network in heterogonous wireless networks.

The second purpose of this study is comparing two different scenarios (one simple and one crowded network) using OPNET in the WiMax network in order to examine the effects of several voice applications on the delay, throughput, and handover delay during horizontal handoff.

Definition of Terms

Delay –The delay of a network specifies how long it takes for a bit of data to travel across the network from one node or endpoint to another. It is typically measured in multiples or fractions of seconds. Delay may differ slightly, depending on the location of the specific pair of communicating nodes.

Throughput – Network throughput is the average rate of successful message delivery over a communication channel. The throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per time slot.

Load – Network load is the average rate of message which is transmitted over a communication channel. The load is usually measured in bps.

QoS – Quality of services (QoS) is to provide guarantees on the ability of a network to deliver predictable results.

Cellular – A cellular network is a radio network made up of a number of radio cells (or just cells) each served by at least one fixed-location transceiver known as a cell site or base station.

WLAN – A wireless local area network (WLAN) links devices via a wireless distribution method (typically spread-spectrum or OFDM radio), and usually provides a connection through an access point to the wider internet.

Bandwidth – A data transmission rate; the maximum amount of information (bits/second) that can be transmitted along a channel

CDMA – Code division multiple access is a channel access method used by various radio communication technologies.

GSM – Global System for Mobile Communications, or GSM (originally from Groupe Spécial Mobile), is the world's most popular standard for mobile systems.

TDMA – Time division multiple access (TDMA) is a channel access method for shared medium networks. It allows several users to share the same frequency channel by dividing the signal into different time slots. The users transmit in rapid succession, one after the other, using their own time slot.

GPS – The Global Positioning System (GPS) is a space-based global navigation satellite system (GNSS) that provides reliable location and time information in all weather and at all times and anywhere on or near the Earth when and where there is an unobstructed line of sight to four or more GPS satellites.

PCM – Pulse Code Modulation (PCM) is a method used to digitally represent sampled analog signals.

GSM – Global System for Mobile Communications (GSM) is the world's most popular standard for mobile telephone systems.

IP telephony – Internet Protocol telephony (IP telephony) is a general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, video, and other forms of information

Significance of the Research

Quality of services in handoff decisions have been considered as an important factor for communications in heterogeneous wireless networks. For handoff decisions at the best moment many factors should be considered in order to provide the user with the best quality of services. A good handoff should occur in the best time to the best network while it maintains the connectivity of all of the applications. Moreover, there must be a minimum number of call drops, a minimum number of unnecessary handovers and a minimum number of packet losses.

The significance of our research in the first part of the thesis is useful to show what the most important factors are in the vertical handoff as well as demonstrating the effect of each factor during the handoff execution. Our proposed algorithm based on the RSS and application type is also helpful since it provides mathematical models for calculating the probability of occurring handoff for service providers. The significance of our research in the second part of the thesis is providing a seamless handoff in the networks. Nowadays, service providers face many problems during the horizontal handoff in the cellular networks. Since WiMax network is a good potential network for voice communications, it is good to know what kind of voice application will provide an acceptable performance in the networks. Therefore, this research study helps the service providers to provide better services and to help them to know what services and what types of applications are the best for different kinds of networks.

Chapter II. Review of Literature

A Brief Historical Review

Because of the popularity of integrated networks and vertical handover issues, many studies have been conducted in order to optimize vertical handover execution. Various strategies have been developed and researchers have considered many factors and have offered several algorithms. Several vertical handoff algorithms were classified as are shown in Figure 2. As it is clear in Figure 2, these algorithms have been classified based on different parameters. The vertical handoff algorithms which have been considered in this study are: Smart decision algorithm, the algorithm considering history of information, algorithm considering loosely integration model, and a novel scheme algorithm. Moreover, three kinds of classifications have been made based on RSS (Received Signal Strength), bandwidth and cost of network. For the RSS based algorithms and the cost based algorithm, three different algorithms have been introduced for each. For the bandwidth algorithm, two different algorithms have been mentioned. In the next section, all of these algorithms and the advantages and disadvantages of each will be described.

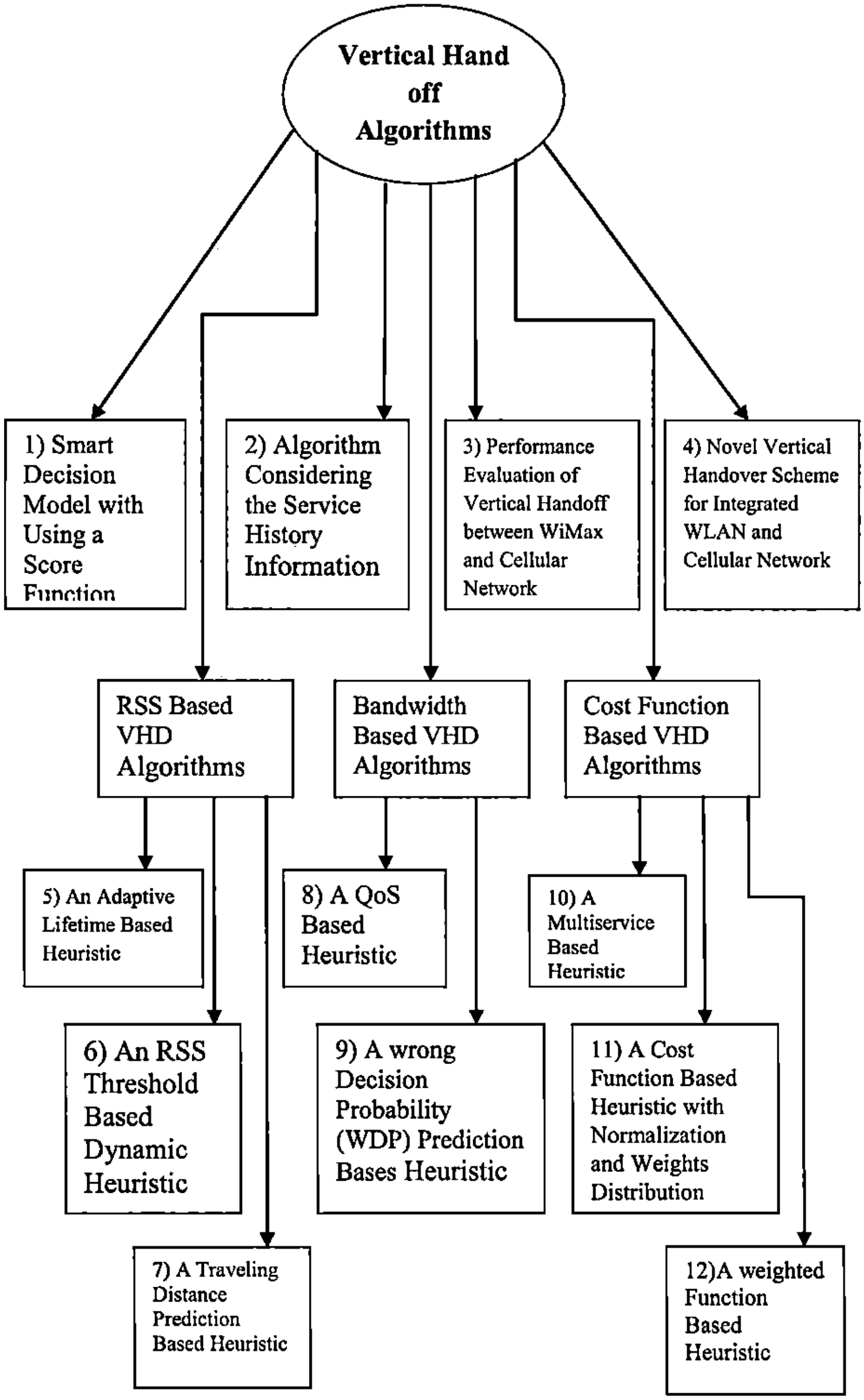

Figure2. Vertical handoff algorithms

A Smart Decision Model for Vertical Handoff

Chen, Sun, Cheung, Nguyen, and Gerla (2004) proposed a Universal Seamless Handoff Architecture (USHA) which L.Chen, Sun, B.Chen, Rajendran, and Gerla (2007) further developed it to solve the smart decision problems. The design of USHA was based on the fact that the handoff occurs only in specific circumstances. It happens on overlaid networks with multiple internet access methods. In this design, the best network would be chosen with zero waiting time. The main important factor in this design was based on overlapping the coverage for different kinds of access methods. In a case that the coverage fails to overlap, the USHA may lose the connection with the upper layer (Chen et al., 2007).

In this architecture, there is a Handoff Server (HS) which is connected to several mobile hosts via an IP tunnel. All the applications in communications layers are connected to the tunnel interface. All the packets are encapsulated for transmission through this channel and a UDP protocol is used for transmissions. In order to maintain the connectivity between the Mobile Host (MH) and HS, there should be two set of addresses at both ends of the IP channel, one for HS and the other for MH. After the handoff occurs, since the location has changed, the MH should inform the HS about the new address in order to continue the connection. The UDP protocol prevents the IP channel from resetting after handoff occurring (Chen et al., 2007).

This algorithm was further developed by Chen et al. (2007) in order to add the smart decision model in which a handoff will occur in the appropriate moment and to the most appropriate network. The proposed design will consist of four parts: A Handoff Executor (HE), Smart Decision (SD), Device Monitor (DM) and System Monitor(SM). DM is responsible for monitoring the status of each network, the SM reports the system information, the SD provides a list of all user interface along with the information provided by DM and applies a score function for calculating the score for each wireless interface and finally, SM will identify the best network for handoff. The HE performs the handoff to the target network.

The score function which is used in this model includes three components: Usage expense (e), link capacity (c) and power consumption (p). The function is illustrated as follows:

$$S_i = W_e\, f_{e,i} + W_c\, f_{c,i} + W_p\, f_{p,i}$$

Where

$f_{e,i} = \frac{1}{e^{\alpha i}}$ $f_{c,i} = \frac{e^{\beta i}}{e^{M}}$ $f_{e,i} = \frac{1}{e^{\sigma i}}$ where $\alpha i \geq 0$, $M \geq \beta i \geq 0$ and $\sigma i \geq 0$

The coefficients αi, βi, and σi are obtained from a specific table or a well-tuned function. The M is the maximum bandwidth requirement which is demanded by a user and is used to normalize the function. This model is simple and it is able to perform handover to the best network at the best time since it is able to make smart decision based on different parameters such as link capacity, power consumption, and link cost (Chen et al., 2007).

A QoS-Aware Vertical Handoff Algorithm Based on Service History Information

In distributed VHO decisions algorithm all the users choose the target network simultaneously, ignoring each other. Several problems will arise in this design one of which might be experiencing high congestion by blindly choosing the network which could not provide the quality of services for the users and may cause handoff call drops and handoff to other networks as well (Kim, S.Han, & Y.Han, 2010).

For optimizing the mentioned algorithms, Kim et al. (2010) has introduced a remedy in which the service history of user traffic was considered and was added to the VHO algorithms. Through usage of this new architecture, the instable handoff decisions were alleviated and the qualities of services were improved. Kim et al. (2010) considered two parameters from service history information for designing this algorithm:

t_0^s , is the service time for network0. This time is taken to the account from the last handoff. It is clear that the bigger this number, is the more efficient the system.

t_i^F , is the time which is calculated from when the last handoff was dropped. If this time is small, it means that the system experienced more call drops, therefore, for improving system functionality, this number should be large. From all the information stated above, it is obvious that if the T is small, the user should be kept in the current serving network to prevent the frequent call drops (Lee, Sriram, K.Kim, Y.Kim, & Golmie, 2009). The proposed evaluation function for this architecture can be expressed as follows:

$$E_i^h(t_i^s) = \begin{cases} \exp(-t_i^s) & if\ i = 0, 0 < t_i^s < T\bar{c} \\ 0 & otherwise \end{cases}$$

$$E_i^h(t_i^F) = \begin{cases} -\exp(-t_i^F) & if\ i \neq 0, 0 < t_i^F < Tc \\ 0 & otherwise \end{cases}$$

The Tc is the maximum effective time of history information and it can be set differently for t_i^s and t_i^F.

This algorithm improves the performance by reducing the number of handoffs, decreasing the probability of handoff occurring as well as reducing the cost (Kim et al., 2010).

Performance Evaluation of Vertical Handoff Scheme between Mobile WiMax and cellular Networks

Park, Yu, and Ihm (2007) proposed an algorithm for vertical handoff between Mobile WiMax and cellular network which is based on the Loosely Integration Model. In this model, WLAN and 3G network exist independently as well as providing autonomous services. For authentication and accounting for roaming services, a gate way has been added to this incorporative model and for the mobility between WLAN and 3G network, this model uses a mobile IP. One of the advantages of this model is that it can easily be adapted to the existing communications and it reduces the effort to make new standards (Park et al., 2007).

The algorithm called smoothly integration scheme has an architecture similar to the loosely integration model but only an IWG (Interworking Gateway) has been added for interworking between Mobile WiMax and CDMA. The IWG helps by using an extended fast handoff scheme in CDMA packet which provides gateway function for protocol adaption. In the fast handoff scheme, serving PDSN (Packet Data Serving Node) sends the traffic to a target PDSN by setting up a tunnel. This traffic will be forwarded to other mobile nodes by target PDSN. In this method the packet loss is minimized since the service anchor point is not changed (Park et al., 2007).

A Novel Vertical Handover Scheme for Integrated WLAN and Cellular Wireless Networks

Wang and Kong (2004) have proposed a novel vertical handover scheme for integrated WLAN and cellular wireless networks handover. In this algorithm, WLAN is overlaid within the coverage area of the cellular network. There is one access point for the WLAN as well as one base station for the cellular network. A Crossover switch connects the access point and the base station. If the user starts communication with the access points it is considered to be connected to the WLAN. However, if the packet exchange is through base stations, a user is considered to be attached to the cellular network.

The Crossover switch decides to handover a user from one network to another as well as transmits subsequent downlink packet to the new access point or base station. This algorithm aims to optimize system utilization without considering packet delay requirements. Two strategies have been defined in order to achieve the objective: The first one is performing unconditional handover when a mobile node is moving out of WLAN coverage and the second one when the mobile node is entering the WLAN coverage. These two conditions are called imperative and alternative respectively (Wang & Kong, 2004).

An unconditional imperative handover will be executed if a user's RSS is lower than a threshold while an alternative handover occurs when a certain number of consecutive handover requests are received by the access point of WLAN from the user. The number of these requests depends on the user's traveling speed and current load of cellular networks. For this algorithm three different classes have been considered: Class A, B, and C. If the current speed is less the $Speed_{low}$, between $speed_{low}$ and $speed_{high}$, or higher than $speed_{high}$, it is class A, B or C respectively. L_{preset} is the threshold of the cellular network (Wang & Kong, 2004). If the current cellular load is equal or greater than L_{preset}, handoff will occur regardless of the class of the user. In a case the L is greater than L_{preset}, the classes are considered. This algorithm can support a larger user arrival rate without dealing with packet delay violation ratio as well as reducing the number of handovers by 10% (Wang & Kong, 2004).

RSS Based VHD Algorithms

An Adaptive Lifetime based Handover Heuristic

For handover between 3G networks and WLAN, an algorithm was proposed by Zahran, Liang, and Saleh (2006). The algorithm evolves two different scenarios which will be described as follows:

First Scenario:

In this scenario, a handover from WLAN to 3G network will happen if the RSS average of WLAN connection is less than the predefined threshold and if the lifetime is less than or equal to the handover delay as well. The RSS average should be calculated continuously from following equation (Zahran et al., 2006; Yan, Sekercioglu, & Narayanan, 2010).

$$\overline{RSS}[k]=\frac{1}{Wav}\sum_{i=0}^{wav-1}\overline{RSS}[k-i]$$

Here Wav is a variable that changes with the velocity of mobile terminal and is called window size. By using RSS[k], the life time metric or EL[k] is calculated through the following formula (Zahran et al., 2006; Yan et al., 2010).

$$EL[k]=\frac{\overline{RSS}[k]-ASST}{S[k]}$$

The Application Signal Strength Threshold (ASST) is an application which represent a composite of the channel bit error rate, application error resilience and application QoS requirements. The S[k] varies with the window size of the slope estimator and the RSS sampling interval (Zahran et al., 2006; Yan et al., 2010).

Second Scenario:

In this scenario, a handover is initiated if a mobile terminal moves from a 3G network to WLAN network. The handover will be triggered if sufficient bandwidth is available on the WLAN network and if the threshold of 3G network falls below the average RSS measurement of WLAN signal (Zahran et al., 2006; Yan et al., 2010).

Zahran et al. (2006) could achieve many benefits in handover between mentioned networks. By using the lifetime metric, the number of extra handoffs will be decreased and throughput of the network will dramatically increase. On the other hand, increasing the lifetime causes an increase in the packet delay which is taken into account as a disadvantage of this algorithm. For solving this problem, the ASST is adjusted based on different parameters such as delay thresholds, mobile terminal velocities, handover signaling costs and packet delay penalties (Zahran et al., 2006; Yan et al., 2010).

An RSS Threshold Based Dynamic Heuristic

Mohanty and Akyildiz (2006) proposed an RSS Threshold Based Dynamic Heuristic algorithm. In this algorithm a dynamic RSS threshold (S_{dth}) is defined when a mobile terminal is connected to a WLAN access point and is used for handover decision from WLAN to 3G through comparison of the current RSS and S_{dth}. By using S_{dth} in this algorithm, the number of false handovers will be reduced and the handover failure will be kept below a limit while the number of superfluous handovers will remain the same (Mohanty & Akyildiz, 2006; Yan et al., 2010).

S_{dth} is calculated from following formula (Mohanty & Akyildiz, 2006).

$$S_{dth} = \text{RSS min} + 10\ \beta \log 10\ \left(\frac{d}{d - \text{LBA}}\right) + \varepsilon$$

Here RSS_{min} (in dBm) is the minimum RSS needed for the mobile terminal to communicate with an access point, B is the path loss coefficient, d is the side length of WLAN cell in meters, here the assumption is that WLAN cells have a hexagonal shape in this study, L_{BA} is the shortest distance between the point at which handover is initiated and WLAN boundary, and ε (in dB) is a zero-mean Gaussian random variable with a standard deviation. This represents the statistical variation in RSS caused by shadowing (Mohanty & Akyildiz, 2006; Yan et al., 2010).

The distance L_{BA} varies with the desired handover failure probability pf, the velocity of the mobile terminal v, and the handover delay from WLAN to 3G which is shown as τ. L_{BA} calculated as follows:

$$L_{BA} = [\tau^2 v^2 + d^2 (pf - 2 + 2\sqrt{1 - pf}]^{1/2}$$

Mohanty and Akyildiz (2006) assumed the failure probability from 3G to WLAN is zero, so the handover can happen anytime a mobile terminal enters WLAN coverage (Mohanty & Akyildiz, 2006; Yan et al., 2010). This algorithm is shown in Figure 3.

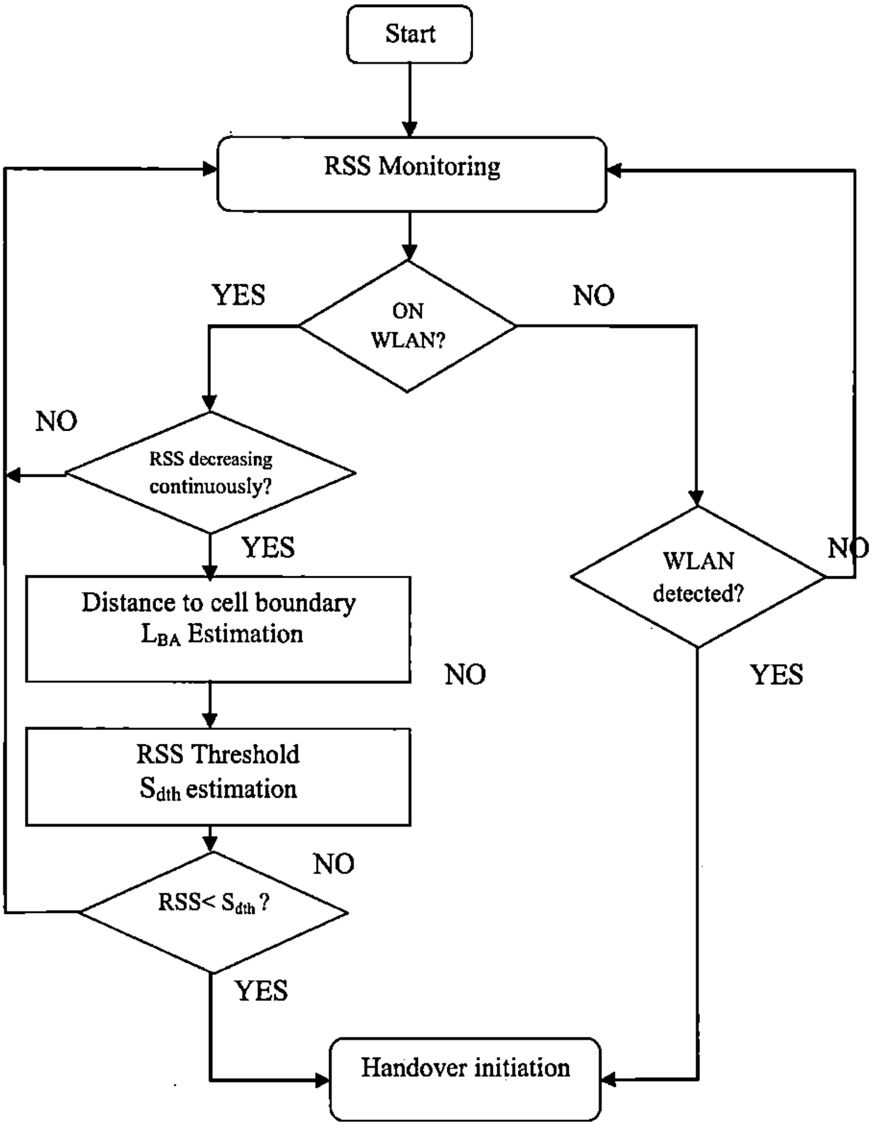

Figure3. An RSS Threshold Based Dynamic Heuristic (Mohanty & Akyildiz, 2006; Yan et al., 2010).

A Traveling Distance Prediction Based Heuristic

Yan, Mani, and Sekercioglu (2008) proposed an algorithm to optimize vertical handoff. They considered the time it takes for a mobile terminal to travel via a WLAN cell (t_{WLAN}) in order to reduce the number of unnecessary handoff. In this design, a handover will occur in a case that the traveling time is greater than the time threshold (T_{WLAN}). The traveling time (t_{WLAN}) is calculated as follows (Yan et al., 2008; Yan et al., 2010).

$$t_{WLAN} = \frac{R^2 - I_{OS}^2 + v^2(ts-tin)^2}{v^2(ts-tin)}$$

Here R is the radius of the WLAN cell, I_{OS} is the distance between where the mobile terminal takes an RSS sample and the access point, v is the velocity of the mobile terminal, and t_s is the time at which the RSS sample is taken, and t_{in} is the time the mobile terminal enters the WLAN cell coverage. I_{OS} can be calculated by using the RSS information and log-distance path loss model (Yan et al., 2008; Yan et al., 2010).

The time threshold (T_{WLAN}) is calculated based on various parameters as

$$T_{WLAN} = \frac{2R}{V} \sin\left(\sin^{-1}\left(\frac{v\tau}{2R}\right) - \frac{\pi}{2}P\right)$$

Here P is the maximum tolerable handover failure, unnecessary handover or connection breakdown probability and τ is the handover delay. For the handover to be initiated the WLAN RSS should fade continuously and the mobile terminal should reach a handover commencement boundary area which size is dynamic to the mobile terminal's speed (Yan et al., 2008; Yan et al., 2010). This algorithm is shown in Figure 4.

While this algorithm reduces the number of extra handoff and minimizes handoff failures, mobile terminal's traveling time is still less than the handover delay which causes loss of network resources (Yan et al., 2008; Yan et al., 2010).

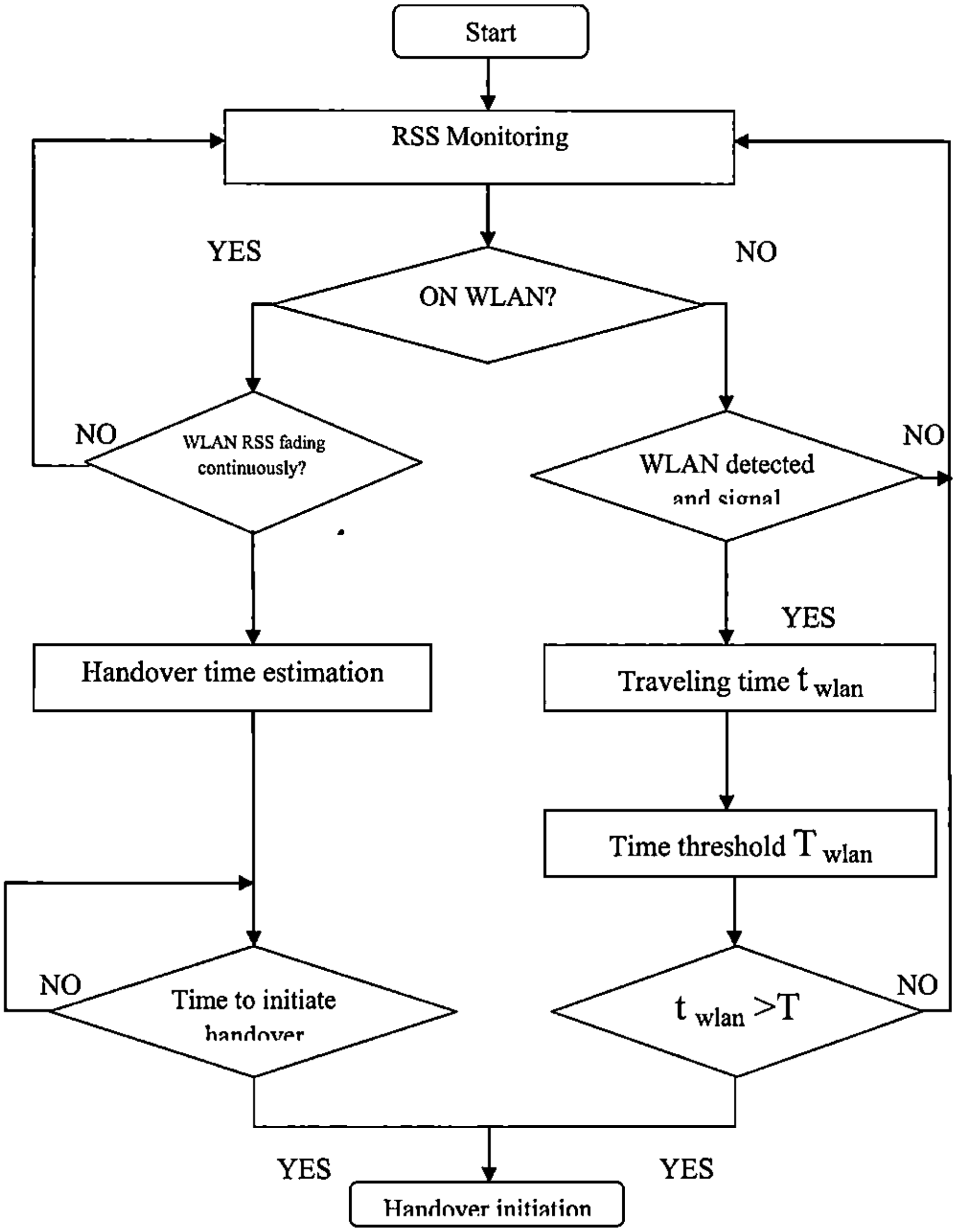

Figure4. A Traveling Distance Prediction Based Heuristic (Yan et al., 2008; Yan et al., 2010).

Bandwidth Based VHD Algorithms

A QoS Based Heuristic

Lee, M.Chen, Y.Chen, and Sun (2005) proposed another algorithm for handover between WLAN to WWAN (Wireless Wide Area Network). In this algorithm the remained bandwidth, the state of the mobile terminal, and the user service requirements are taken into account. In this algorithm two different scenarios are described: Handover from WLAN to WWAN and vice versa (Lee et al., 2005; Yan et al., 2010).

In the first scenario, for handover decision, while the mobile terminal is connected to WLAN, the measured RSS should fall below a threshold (RSS_{T1}). Handover will be performed to the best network if the mobile terminal is in the idle state, otherwise the handoff decision is based on user application type (Lee et al., 2005; Yan et al., 2010).

Here two types of applications take into consideration.

1. Delay sensitive application

   For this type of application a handover occurs if there is no sufficient bandwidth is provided on WLAN to serve the user, while WWAN provides available bandwidth for the user's application.

2. Delay tolerant application

   WWAN provides higher bandwidth for the user than the WLAN.

   Here, the remained bandwidth should be calculated for the WLAN as follows in order to take the handoff decision.

Remained bandwidth=Throughput×(1-α×Channel Utilization)×(1-Packet loss rate)

Here Throughput is the throughput that can be shared among mobile terminals in the WLAN. Channel utilization is the percentage of time that the access point, by using a carrier sense mechanism, senses the medium is busy. α is a factor that reacts IEEE 802.11 MAC overhead and here is set to 1.25. Finally packet loss rate is the part of transmitted medium access control (MAC) protocol data units (MPDUs) that require retransmission, or are discarded as the packets that are not delivered. The values of Channel Utilization and Packet loss rate are obtained from the information in the beacon frame carrying the QoS basic service set (QBSS) load which is sent by an access point (Lee et al., 2005; Yan et al., 2010).

In the second scenario, a handover occurs from WWAN to WLAN if the RSS in the WWAN is less than the threshold (RSS_{T2}) (Lee et al., 2005; Yan et al., 2010). Figure 5 shows this algorithm.

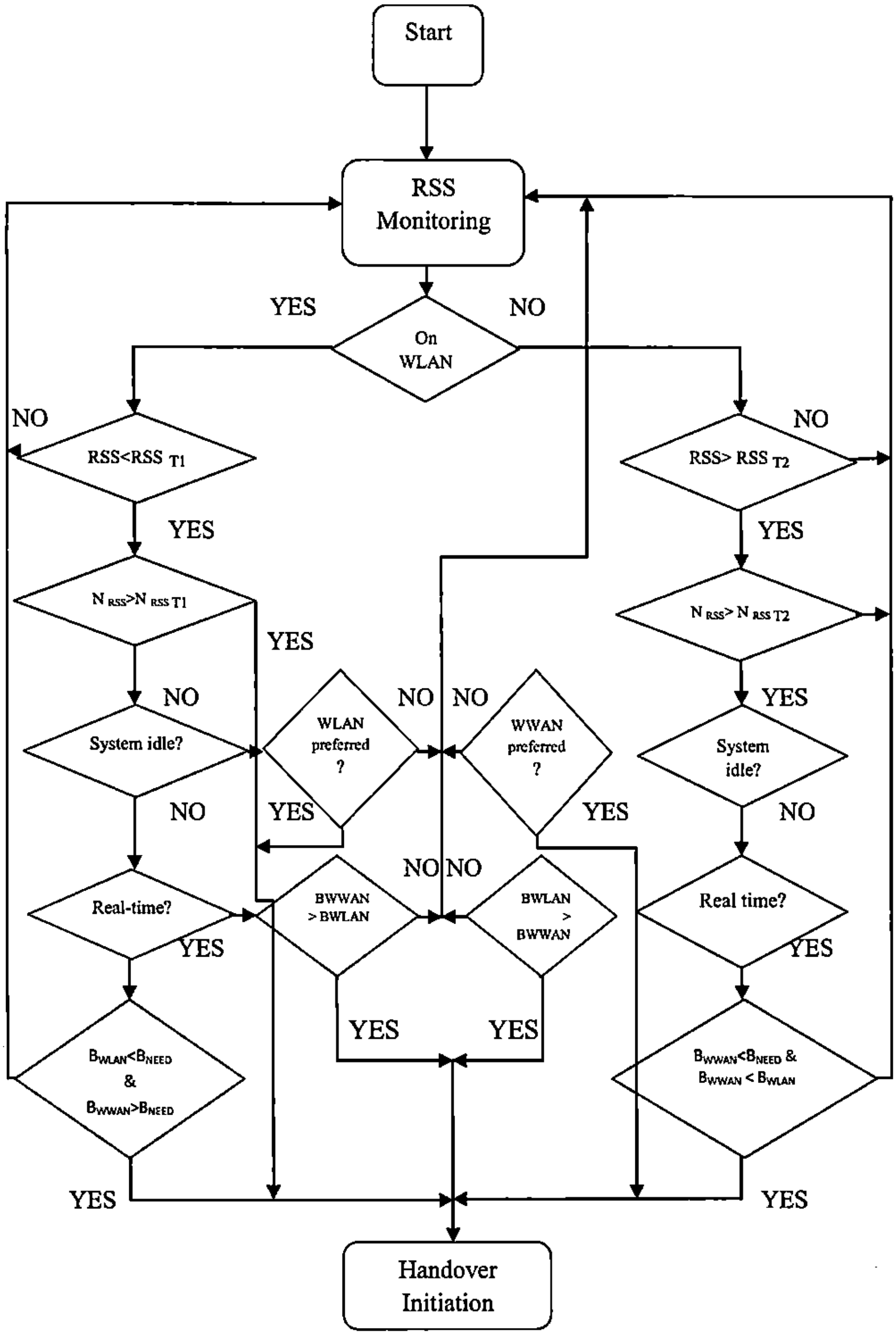

Figure5. A QoS Based Heuristic (Lee et al., 2005; Yan et al., 2010).

A Wrong Decision Probability Algorithm (WDP) Prediction Based Heuristic

Chi, Cai, Hao, and Liu (2007) proposed a WDP algorithm which is based on the probability of unnecessary handovers and missing handovers. If you consider two kinds of networks as x and y, and you also consider the bandwidth associated with these networks as B_x and B_y, an unnecessary handover occurs when a handoff is performed from network x to network y, while the available bandwidth in network x (B_x) is less than the available bandwidth in network y (B_y). On the other hand a missing handover occurs when a mobile terminal in network x should perform handover to network y because of lack of available bandwidth in network x but maintains its connectivity to network x (Chi et al., 2007; Yan et al., 2010).

A handover from network x to network y is initiated if $P_r < \rho \times L_0$ or $b_y - b_x \leq L$

Here P_r is the unnecessary handover probability, ρ is the traffic load of network x, $L_0 = 0.001$, and L is shown bandwidth threshold (Yan et al., 2010).

The proposed algorithm by Chi et al. (2007) has several advantages. This algorithm reduces the Wrong Decision Probability (WDP) and balances the traffic load. However, it does not consider RSS which is a main factor in handoff decision. Received signal strength is a main factor in every handover and a handover to a network with high bandwidth but weak signal strength is undesirable (Chi et al., 2007; Yan et al., 2010).

Cost Function Based VHD Algorithms

A Multiservice Based Heuristic

Zhu and McNair (2004) proposed a cost function based algorithm which works based on a cost function. The algorithm gives priority to active applications which need to perform a handover to a target network. Therefore, the service with highest priority is selected On the other hand, the cost of a series of target networks will be calculated and then the handover occurs between the application with higher priority and the network with minimum cost. The cost of target network is calculated as follows (Zhu & McNair, 2004; Zhu & McNair, 2006; Yan et al, 2010).

$$C_s^n = \sum W_{s,j}^n Q_{s,j}^n \quad E_{s,j}^n \neq 0,$$

Here, C_s^n is defined as cost of service for network n, $Q_{s,j}^n$ is the normalized QoS provided by network n for the parameter j and service s, $W_{s,j}^n$ is the weight which shows the impact of the QoS parameter on the user or on the network. Here network elimination factor($E_{s,j}^n$) is defined which indicate whether the minimum requirement of parameter j for service s can be met by network n. Sum of all the cost in the network will be total cost which includes bandwidth, battery power and delay. The target network for handover is the network with the minimum cost (Zhu & McNair, 2004; Zhu & McNair, 2006; Yan et al, 2010).

This algorithm provides user's applications with reduced blocking probability. It also satisfies more user's requests, however, it is not mentioned in what manner the QoS factors are weighted and normalized. Nasser, Hasswa, and

Hassanein (2006) further developed this algorithm, in which normalization and weight distribution methods were provided (Zhu & McNair, 2004; Zhu & McNair, 2006; Yan et al, 2010).

A Cost Function Based heuristic with Normalization and Weights Distribution

In this algorithm proposed by Nasser et al. (2006), by calculating the network quality factor the performance of a target handover will be evaluated. In this algorithm if the handover is necessary, then the network parameters will be collected. Then the weight and the quality factor will be calculated and if the current quality is less than the candidate quality, the handover will be initiated (Nasser et al., 2006; Yan et al. 2010).

Furthermore, for avoiding superfluous handover, a metric which is called handover necessity estimator was introduced. The network quality factor is calculated as follows (Nasser et al., 2006; Yan et al. 2010).

$$Q_i = w_c C_i + w_s S_i + w_p P_i + w_d D_i + w_f F_i$$

Here Q_i is the quality factor of network i, C_i is cost of service, S_i is security, P_i is power consumption, D_i is network conditions and F_i stands for network performance. Here, w_c, w_s, w_p, w_d and w_f are the weights for these network parameters (Nasser et al., 2006). Since each network parameter has a different unit, a normalization procedure is used and the normalized quality factor for network n is calculated as follows:

$$\frac{w_c(1/Ci)}{\max((1/C1),\ldots,(1/Cn))}+\frac{w_sS_i}{\max(S_1,\ldots S_n)}+\frac{w_p(1/pi)}{\max((1/P1),\ldots,(1/Pn))}$$

$$+\frac{wdDi}{\max(D1,\ldots D_n)}+\frac{wfFi}{\max(f_1,\ldots f_n)}$$

Advantages of this algorithm include: Increasing throughput of the system and user's satisfaction. However, this algorithm does not provide information for estimating security and interferences levels (Nasser et al., 2006; Yan et al. 2010).

A Weighted Function Based Heuristic

Tawil, Pujolle, and Salazar (2008) designed a Weighted Function Based Heuristic. Despite other algorithms in which mobile terminal was responsible for the VHD calculation, in this algorithm VHD calculation will be done in the visited network. The quality of network (Qi) will be calculated as follows:

$Qi = W_b B + W_d D + W_c C$

Where B, D, C are bandwidth, dropping probability and cost of services. And W_b, W_d and W_c are their weights where

$W_b + W_d + W_c = 1$

In this algorithm the network with the highest Qi will be selected as target network for handover. As a result, the handover delay will be decreased, the handover blocking rate will be lowered and the throughput will be increased as well. However, since there should be extended communication between mobile terminal and access point of the visited network, there might be additional delay and load when there is large number of mobile terminals (Tawil et al., 2008;Yan et al., 2010). This algorithm is shown in Figure 6.

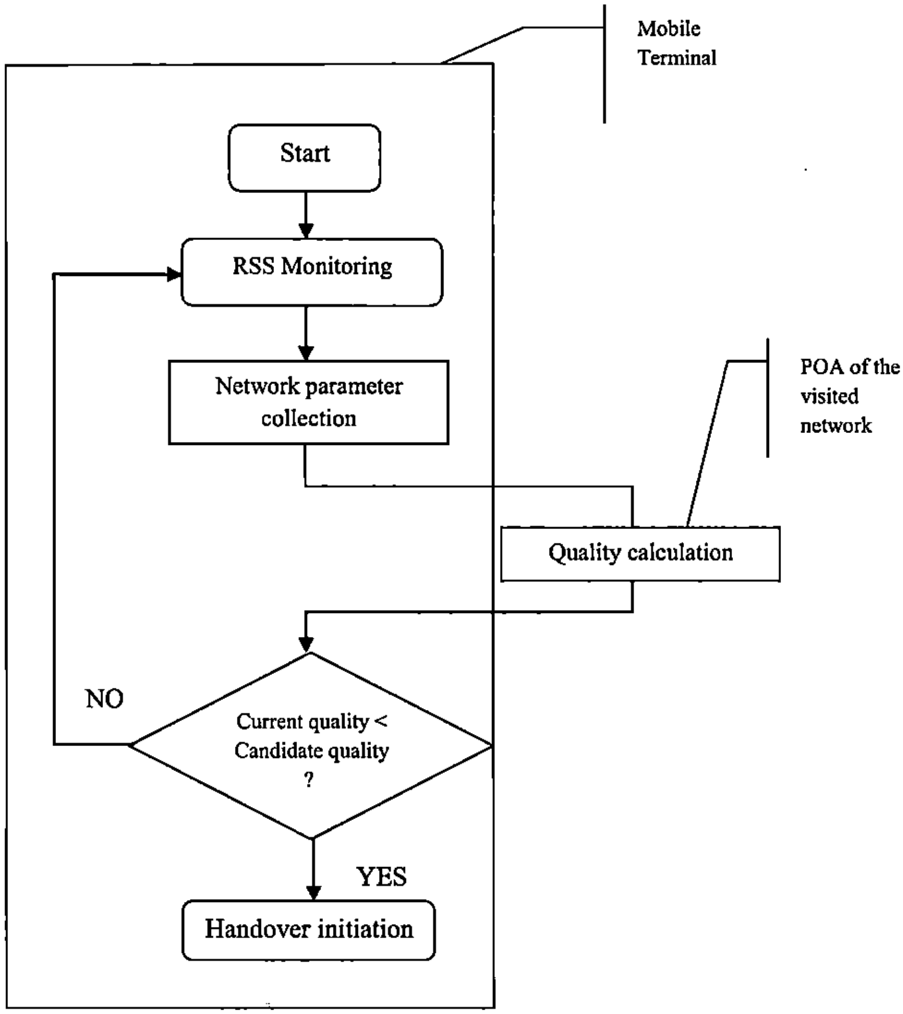

Figure6. A Weighted Function Based Heuristic (Tawil et al., 2008; Yan et al., 2010).

Chapter III – Methodology (Vertical Handoff)

In this part of thesis, the advantages and disadvantages of mentioned algorithms in Chapter II were mainly discussed because the vertical handover between WLAN and cellular network is a critical issue which needs to be handled efficiently in order to provide user's applications with the best quality of service. A table was also prepared in which the result of comparison of these algorithms in vertical handoff were summarized. In the next part, the author's algorithm for vertical handoff between WLAN and cellular network and the mathematical model for the probability of occurring handoff between these two different networks were proposed.

Comparison

In the smart decision algorithm, based on the several network parameters such as link capacity, power consumption and link cost, the author has proposed a model to decide smartly which network to choose in order to execute vertical handoff. By considering many factors this algorithm decides which network is the best for executing handover and it helps to overcome many problems which may arise in the handoff execution

In the algorithm based on system history information, the number of handoff, the numbers of handoff probability and the cost have been decreased. This algorithm also works better in more complicated networks. In the Vertical Handoff Scheme between Mobile WiMax and cellular Networks, the number of packet loss is minimized.

In the Novel Vertical Handover Scheme between WLAN and Cellular Networks, the total number of handover has reduced and by using L_{preset} and monitoring the load in the network, the number of unnecessary handover has been reduced. Moreover, this algorithm can support a larger user arrival rate without dealing with packet delay violation ratio.

An Adaptive Lifetime based Handover Heuristic algorithm could reduce the number of unnecessary handoffs as well as increasing the throughput of the network by considering life time metric. However, if there is an increase in the lifetime, the delay in the network will increase, so this algorithm may not work properly for delay sensitive applications.

The RSS Threshold Based Dynamic Heuristic reduces the number of false handover and keeps the handover failure below a certain limit. However, the disadvantages of this algorithm are as follows: The number of extra handoff will remain the same and if the mobile station's traveling time inside a cell is less than handover delay, there is wastage in the network resources.

The Traveling Distance Prediction Based Heuristic reduces the number of handover failure, superfluous handover and connection breakdown. However, sampling and averaging RSS will increase the handover delay. Furthermore, mobile terminal's traveling time is still less than the handover delay which causes loss of network resources

The QoS Based Heuristic, by considering bandwidth, has increased the throughput of the network. This algorithm works well for delay sensitive applications since it decreases the delay by considering the application type.

In the Wrong Decision Probability Algorithm, the RSS has not been considered but this algorithm reduces the wrong decision probability while balancing the traffic load. Since this algorithm has not considered the RSS, it is not efficient because it may cause several breakdowns in the network. The Multiservice Based Heuristic reduces the blocking probability.

A Cost Function Based heuristic with Normalization and Weights Distribution algorithm provides high throughput for the system. But some of the parameters such as security and interference level are difficult to measure in the network. The Weighted Function Based Heuristic provides short handover decision delay, low handover blocking rate and high throughput. However, it may cause extra delay and load to the network.

In Table1 and Table2, the comparison between these algorithms is shown. It shows the effects of different parameters on the vertical handoff execution. It is clear that considering different factors have different effects on the vertical handoff between WLAN and cellular network.

Table1.

Comparison of different algorithms for vertical handoff.

Algorithms	**Delay**	**Number of handoff**	**Handover failure probability**	**Through put**	**Number of packet loss**
Smart Decision Algorithm	Handover in the BEST time to the BEST network				
A QoS Aware Vertical Handoff Algorithm Based on Service History Information		Reduced number of handoff	Reduced handover failure probability	High through put	
Performance Evaluation of Vertical Handoff Scheme Between Mobile WiMax and Cellular networks.					The number of packet loss is decreased
A Novel Vertical Handover Scheme for Integrated WLAN and Cellular Wireless Networks			Number of handover and Number of unnecessary handover has reduced.		
An Adaptive Lifetime based Handover Heuristic	By increasing lifetime, the delay may increase	Reduced number of handoff		High through put	
An RSS Threshold Based Dynamic Heuristic		Reduces the number of false handover			
A Traveling Distance Prediction Based Heuristic	Increases the handover delay	Reduces the number of superfluous handover	Reduces the number of handover failure		

Table 2.

Comparison of different algorithms for vertical handoff.

Algorithms	Delay	Number of handoff	Handover failure probability	throughput	Number of packet loss
A QoS Based Heuristic	Reduced delay			High throughput	
A Wrong Decision Probability Algorithm (WDP) Prediction Based Heuristic			Reduces the wrong decision probability		
A multiservice based heuristic			Reduces the blocking probability		
A Cost Function Based heuristic with Normalization and Weights Distribution				Increased throughput	
A weighted Function Based Heuristic	may cause extra delay		low handover blocking rate	high throughput	

By looking at the Table1, it is obvious that the RSS based algorithms, reduce the number of handoff. Therefore RSS was chosen as one of the parameters for author's proposed algorithm. The other parameter which was also considered in this algorithm is application type (Data and Voice). In the following sections, the proposed algorithm for the vertical handoff between WLAN and

Cellular network is shown and this algorithm is described step by step. A mathematical model was also found for the probability of handoff occurring between WLAN and cellular network. This algorithm is shown in Figure 7.

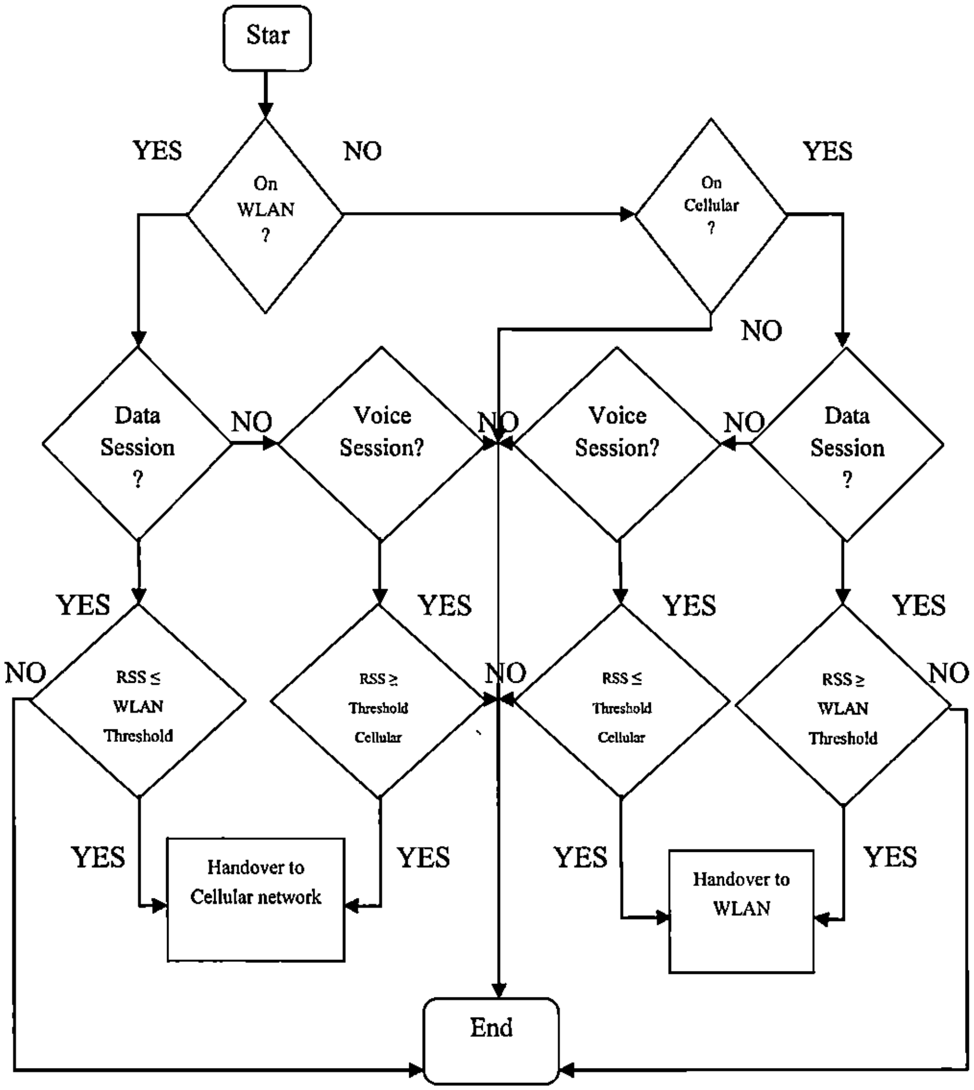

Figure7. Vertical Handoff Algorithm between WLAN and Cellular Network

Algorithm

Assumption: There are two different networks (WLAN and Cellular Network)

1-A user's application wants to start handover

If it is connected WLAN

If it is data session

If RSS $\leq$ WLAN Threshold

Then Handover to Cellular network

If it is voice session

If RSS $\geq$ Threshold Cellular

Then Handover to Cellular Network

If it is connected to cellular network

If it is data session

If RSS $\geq$ WLAN Threshold

Then Handover to WLAN

If it is voice session

If RSS $\leq$ Threshold Cellular for all cells

Then Handover to WLAN

Algorithm description

In the proposed algorithm, two different networks were considered, WLAN and cellular network. The algorithm is divided into two parts:

In the first part, the user is in the WLAN and wants to initiate a handover to cellular network. If the application type is data, the user prefers to stay in the

WLAN since WLAN is a good network for data applications. But if the received signal strength (RSS) is less than the threshold for the WLAN, then this signal is very weak and the user needs to initiate handover to the cellular network. If the application type is voice, the user prefers to handover to the cellular network since the voice applications work better in cellular networks. Therefore, if the received signal strength is greater that than the threshold for the cellular network, the signal is very strong and works better in the cellular network. Therefore, the handover from WLAN to the cellular network will occur.

In the second part, the user is in a cellular network that wants to initiate a handover to WLAN. If the application type is data, the user prefers to handover to the WLAN. Therefore, if the received signal strength is greater that than the threshold for the WLAN, the signal is very strong and handover happens. If the application type is voice, the user prefers to stay in the cellular network. However, if the RSS is less than the threshold for the cellular network, then this signal is very weak and the handover happens.

In this algorithm, the probability functions follow Gaussian distribution which is as follows:

$$P(t) = \frac{1}{\sigma\sqrt{2\pi}} \exp((t-\mu)^2/2\sigma^2) \tag{1}$$

The cumulative distribution function (cdf) was also used which describes probabilities for a random variable. The cdf of the standard normal distribution is denoted by the Q and can be computed as an integral of the probability density function:

$$Q(\beta) = \frac{1}{\sqrt{2\pi}} \int_{\beta}^{\infty} e^{-y^2/2} \quad dy \qquad (2)$$

Handoff from WLAN to Cellular Network (Data Session)

If the mobile station is in the WLAN and wants to initiate a handover to cellular network (WLAN→Cellular network), according to the algorithm, if it is data session, the handover will be executed if (RSS< WLAN Threshold)

The method which was used is based on what Zahran et al. (2006) and Zhang and Holtzman (1996) did for their research. For calculation several terms should be defined as follows:

$P_{c|w}[i]$ = the probability that in the time i the handover occurs from WLAN to Cellular network.

RSS = the received signal strength and is the base of our algorithm.

Thr_{WLAN}= the threshold of WLAN

So according to what was stated above, $P_{c|w}[i]$ is as follows:

$$P_{c|w}[i] = P\{Rss_i < Thr_{WLAN}\} \qquad (3)$$

The equation (3) can be written like the equation (4) and (5), since in the time i-1, the MS should be in the WLAN coverage and in that time, the RSS should be greater than Thr_{WLAN}.

$$P_{c|w}[i] = P\{Rss_i < Thr_{WLAN} \mid WLAN[i-1]\} \qquad (4)$$

$$P_{c|w\,[i]} = P\,\{Rss_i < Thr_{WLAN} \mid Rss_{i-1} > Thr_{WLAN}\} \tag{5}$$

Therefore, according to the probability formula,

$$P_{c|w\,[i]} = \frac{P\,\{Rss_i < Thr_{WLAN}\,,\; Rss_{i-1} > Thr_{WLAN}\}}{P\,\{Rss_{i-1} > Thr_{WLAN}\}} \tag{6}$$

For simplicity, Thr_{WLAN} is shown as tw and equation (6) is written like equation (7).

$$P_{c|w\,[i]} = \frac{P\,\{Rss_i < t_w\,,\; Rss_{i-1} > t_w\}}{P\,\{Rss_{i-1} > t_w\}} \tag{7}$$

The conditional probability can be computed using Gaussian distribution (Zhang, & Holtzman, 1996). The Rss_{i-1} and Rss_i have Gaussian distribution so the mean and the variance should be defined for that. By considering two functions of Z_1 and Z_2 as follows, the mean and the variance for each can be found as shown in (8) and (9).

$Z_1 = Rss_{i-1}$ and $Z_2 = Rss_i$

$E\,(Rss_{i-1}) = \mu_{Rss\,i-1}$ (8) $\qquad VAR\,(Rss_{i-1}) = \sigma_{Rss}$ (9)

According to the definition of Q function,

$$P\,\{Rss_{i-1} > t_w\} = Q\left(\frac{-tw - \mu_{Rss\,(i-1)}}{\sigma_{Rss}}\right) \tag{10}$$

According to what Zhang and Holtzman (1996) found for the probability, the numerator of equation (7) is as follows:

$$P\{ Rss_i < t_w ,\ Rss_{i-1} > t_w\} = \int_{-\infty}^{tw} Q(\frac{-tw - \mu\, Rss\, i-1 \ \ -_{\gamma}(t - \mu\, Rss\, i-1)}{\sigma\, Rss \sqrt{1-_{\gamma}{}^{2}}}) \times$$

$$P_{RSS\, i}(t)dt \tag{11}$$

In equation (11), γ is the correlation coefficient between Rss_{i-1} and Rss_i.

According to (Zhang, & Holtzman, 1996) both numerator and denominator were found therefore, the probability of a mobile station being in WLAN and wanting to execute a handover to cellular network when it receives a data session is as follows:

$$P_{c|w}[i] = \frac{\int_{-\infty}^{tw} Q(\frac{-tw - \mu\, Rss\, i-1 \ \ -_{\gamma}(t - \mu\, Rss\, i-1)}{\sigma\, Rss \sqrt{1-_{\gamma}{}^{2}}}) \times \ P_{RSS\, i}(t)dt}{Q\,(\ \frac{-tw - \mu\, Rss\, (i-1)}{\sigma\, Rss})} \tag{12}$$

Handoff from WLAN to Cellular Network (Voice Session)

If the mobile station is in the WLAN and wants to initiate a handover to cellular network (WLAN→ Cellular network), according to the algorithm, if it is voice session, the handover will be executed if (RSS> Threshold cellular network).

$$P_{c|w}[i] \ = P\ \{Rss_i > Thr_{cellular}\} \tag{13}$$

The equation (13) can be written like the equation (14) and (15), since in the time i-1, the MS should be in the WLAN coverage and in that time, the RSS should be greater than Thr $_{WLAN}$.

$$P_{c|w}[i] = P\{Rss_i > Thr_{cellular} \mid WLAN[i-1]\} \quad (14)$$

$$P_{c|w}[i] = P\{Rss_i > Thr_{cellular} \mid Rss_{i-1} > Thr_{WLAN}\} \quad (15)$$

According to the probability formula,

$$P_{c|w}[i] = \frac{P\{Rss_i > Thr_{cellular},\ Rss_{i-1} > Thr_{WLAN}\}}{P\{Rss_{i-1} > Thr_{WLAN}\}} \quad (16)$$

For simplicity, Thr $_{WLAN}$ is shown as t_w and Thr $_{cellular}$ is shown as t_c therefore, the equation (16) is written like equation (17).

$$P_{c|w}[i] = \frac{P\{Rss_i > t_c,\ Rss_{i-1} > t_w\}}{P\{Rss_{i-1} > t_w\}} \quad (17)$$

Since the (Rssi $>t_c$) and (Rss i-1$> t_w$) are independent from each other, according to the probability function,

$$P_{c|w}[i] = \frac{P\{Rss_i > t_c\} \times P\{Rss_{i-1} > t_w\}}{P\{Rss_{i-1} > t_w\}} \quad (18)$$

And therefore, according to the definition of Q function,

$$P_{c|w}[i] = P\{Rss_i > t_c\} = Q\left(\frac{tc - \mu Rss\,(i)}{\sigma Rss}\right) \tag{19}$$

Therefore, the probability of a mobile station being in WLAN and wanting to execute a handover to cellular network when it receives a voice session is as follows:

$$P_{c|w}[i] = Q\left(\frac{tc - \mu Rss\,(i-1)}{\sigma Rss}\right) \tag{20}$$

Handoff from Cellular Network to WLAN (Data Session)

If the mobile station is in the Cellular network and wants to initiate a handover to WLAN (Cellular network → WLAN), according to the algorithm, if it is data session, the handover will be executed if (RSS> WLAN Threshold).

$$P_{w|c}[i] = P\{Rss_i > Thr_{wlan}\} \tag{21}$$

The equation (21) can be written like the equation (22) and (23), since in the time i-1, the MS should be in the WLAN coverage and in that time, the RSS should be greater than Thr_{WLAN}.

$$P_{w|c}[i] = P\{Rss_i > Thr_{wlan} \mid Cellular\,[i-1]\} \tag{22}$$

$$P_{w|c}[i] = P\{Rss_i > Thr_{wlan} \mid Rss_{i-1} > Thr_{cellular}\} \tag{23}$$

Therefore, according to the probability formula,

$$P_{w|c}[i] = \frac{P\{Rss_i > Thr_{wlan},\ Rss_{i-1} > Thr_{cellular}\}}{P\{Rss_{i-1} > Thr_{cellular}\}} \tag{24}$$

Which can be written as equation (25)

$$P_{w|c}[i] = \frac{P\{Rss_i > t_w\ ,\ Rss_{i-1} > t_c\}}{P\{Rss_{i-1} > t_c\}} \tag{25}$$

Since the (Rssi $>t_w$) and (Rss i-1$>t_c$) are independent from each other, the equation (25) can be written as equation (26).

$$P_{w|c}[i] = \frac{P\{Rss_i > t_w\} \times P\{Rss_{i-1} > t_c\}}{P\{Rss_{i-1} > t_c\}} \tag{26}$$

And therefore, according to the definition of Q function,

$$P_{w|c}[i] = P\{Rss_i > t_w\} = Q\left(\frac{tw - \mu\, Rss\,(i)}{\sigma\, Rss}\right) \tag{27}$$

Therefore, the probability of a mobile station being in cellular network and wanting to execute a handover to WLAN when it receives a data session is as follows:

$$P_{w|c}[i] = Q\left(\frac{tw - \mu\, Rss\,(i)}{\sigma\, Rss}\right) \tag{28}$$

Handoff from Cellular Network to WLAN (Voice Session)

If the mobile station is in the Cellular Network and wants to initiate a handover to WLAN (Cellular network → WLAN), according to the algorithm, if it is voice session, the handover will be executed if (RSS< Threshold cellular network).

The method which was used is based on what Zahran et al., (2006) and Zhang and Holtzman (1996) did for their research. For calculation, several terms should be defined as follows:

$P_{w|c}[i]$ = the probability that in the time i the handover occurs from Cellular network to WLAN.

RSS_i = the received signal strength which is received for the cellular network in time i and is the base of our algorithm.

$Thr_{Cellular}$ = the threshold of Cellular Network

So according to what was stated above,

$$P_{w|c}[i] = P\{Rss_i < Thr_{cellular}\} \quad (29)$$

The equation (29) can be written like the equation (30) and (31). Since in the time i-1, the MS should be in the WLAN coverage and in that time, the RSS should be greater than Thr_{WLAN}.

$$P_{w|c}[i] = P\{Rss_i < Thr_{cellular} | \text{ Cellular } [i-1]\} \quad (30)$$

$$P_{w|c}[i] = P\{Rss_i < Thr_{cellular} | Rss_{i-1} > Thr_{cellular}\} \quad (31)$$

According to the probability formula,

$$P_{w|c}[i] = \frac{P\{Rss_i < Thr_{cellular}, Rss_{i-1} > Thr_{cellular}\}}{P\{Rss_{i-1} > Thr_{cellular}\}} \quad (32)$$

Or,

$$P_{wlc}[i] = \frac{P\{ Rss_i < t_c ,\ Rss_{i-1} > t_c \}}{P\ \{Rss_{i-1} > t_c\}} \tag{33}$$

The conditional probability can be computed using Gaussian distribution (Zhang, & Holtzman, 1996). The Rss_{i-1} and Rss_i have Gaussian distribution so the mean and the variance should be defined for that. By considering two functions of X_1 and X_2 as follows, the mean and the variance for each can be found as shown in (34) and (35).

$X_1 = Rss_{i-1}$ and $X_2 = Rss_i$

$E\ (Rss_{i-1}) = \mu_{Rss\ i-1}$ (34) $VAR\ (Rss_{i-1}) = \sigma_{Rss}$ (35)

According to the definition of Q function,

$$P\ \{Rss_{i-1} > t_c\} = Q\left(\frac{-tc - \mu\, Rss\ (i-1)}{\sigma\, Rss} \right) \tag{36}$$

As Zhang and Holtzman (1996) mentioned, the formula for the numerator of equation (33) is as follows:

$$P\{ Rss_i < t_c ,\ Rss_{i-1} > t_c \} = \int_{-\infty}^{tc} Q\left(\frac{-tc - \mu\, Rss\ i-1 \ - \gamma(t - \mu\, Rss\ i-1)}{\sigma\, Rss \sqrt{1-\gamma^2}} \right)$$

$$\times\ P\ RSS\ i(t)dt \tag{37}$$

According to (Zhang, & Holtzman, 1996) both numerator and denominator were found therefore, the probability of a mobile station being in cellular network and wanting to execute a handover to WLAN when it receives a voice session is as follows:

$$P_{w|c}[i] = \frac{\int_{-\infty}^{tc} Q\left(\frac{-tc-\mu_{Rss\,i-1} - \gamma(t-\mu_{Rss\,i-1})}{\sigma_{Rss}\sqrt{1-\gamma^2}}\right) \times P_{RSS\,i}(t)dt}{Q\left(\frac{-tc-\mu_{Rss\,(i-1)}}{\sigma_{Rss}}\right)} \tag{38}$$

Finding

WLAN→ Cellular Network

a) Data $P1 = P_{c|w}[i] = \frac{\int_{-\infty}^{tw} Q\left(\frac{-tw-\mu_{Rss\,i-1} - \gamma(t-\mu_{Rss\,i-1})}{\sigma_{Rss}\sqrt{1-\gamma^2}}\right) \times P_{RSS\,i}(t)dt}{Q\left(\frac{-tw-\mu_{Rss\,(i-1)}}{\sigma_{Rss}}\right)}$

b) Voice $P2 = P_{c|w}[i] = Q\left(\frac{tc-\mu_{Rss\,(i)}}{\sigma_{Rss}}\right)$

Cellular Network → WLAN

a) Data $P3 = P_{w|c}[i] = Q\left(\frac{tw-\mu_{Rss\,(i)}}{\sigma_{Rss}}\right)$

b) Voice $P4 = P_{w|c}[i] = \frac{\int_{-\infty}^{tc} Q\left(\frac{-tc-\mu_{Rss\,i-1} - \gamma(t-\mu_{Rss\,i-1})}{\sigma_{Rss}\sqrt{1-\gamma^2}}\right) \times P_{RSS\,i}(t)dt}{Q\left(\frac{-tc-\mu_{Rss\,(i-1)}}{\sigma_{Rss}}\right)}$

Two new mathematical models for the handoff from WLAN to cellular network (voice session) and for the handoff from cellular network to WLAN (data session) were suggested. The two other models were similar to what Zhang & Holtzman (1996) found for the handoff probability. Moreover, Zhang & Holtzman

(1996) mentioned the following formula for calculating the probability of occurring handoff. Those probabilities can be computed in a recursive way as follows:

$$P_{ho}[i] = P_w[i-1]\, P_{c|w}[i] + P_c[i-1]\, P_{w|c}[i] \tag{39}$$

Here

$P_{ho}[i]$ = the probability that handoff occurs between WLAN and cellular network in the time i

$P_w[i-1]$ = the probability that the mobile station is in the WLAN in the time i-1

$P_{c|w}[i]$ = the probability that mobile station executes handoff from WLAN to Cellular network in time i

$P_c[i-1]$ = the probability that the mobile station is in the cellular network in the time i-1

$P_{w|c}[i]$ = the probability that mobile station execute handoff from cellular network to WLAN in time i

The following formulas also were introduced for the $P_w[i]$ and $P_c[i]$

$$P_w[i] = p_w[i-1]\,(1 - P_{c|w}[i]) + P_c[i-1]\, P_{c|w}[i] \tag{40}$$

$$P_c[i] = p_c[i-1]\,(1 - P_{w|c}[i]) + P_w[i-1]\, P_{w|c}[i] \tag{41}$$

In the following part, the probability of handoff occurring for i=0 and i=1 for handoff between WLAN and Cellular network are calculated.

(WLAN→ Cellular Network: Data Session):

$P_w[0] = 1$ and $P_c[0] = 0$

If i=1

$$P_w[i] = p_w[i-1](1-P_{c|w}[i]) + P_c[i-1]P_{c|w}[i]$$

$$P_w[1] = p_w[0](1-P_{c|w}[1]) + P_c[0]P_{c|w}[1]$$

$$P_w[1] = 1(1-P1[1]) + 0 = 1-P1[1]$$

$$P_c[i] = p_c[i-1](1-P_{w|c}[i]) + P_w[i-1]P_{w|c}[i]$$

$$P_c[1] = p_c[0](1-P_{w|c}[1]) + P_w[0]P_{w|c}[1]$$

$$P_c[1] = 0$$

Therefore,

$$P_{ho}[i] = P_w[i-1]P_{c|w}[i] + P_c[i-1]P_{w|c}[i]$$

$$P_{ho}[1] = P_w[0]P_{c|w}[1] + P_c[0]P_{w|c}[1]$$

$$P_{ho}[1] = P1[1] + 0 = P1[1]$$

If i=2

$$P_w[i] = p_w[i-1](1-P_{c|w}[i]) + P_c[i-1]P_{c|w}[i]$$

$$P_w[2] = p_w[1](1-P_{c|w}[2]) + P_c[1]P_{c|w}[2]$$

$$P_w[2] = (1-P1[1])(1-P1[2])$$

$$P_c[i] = p_c[i-1](1-P_{w|c}[i]) + P_w[i-1]P_{w|c}[i]$$

$$P_c[2] = p_c[1](1-P_{w|c}[2]) + P_w[1]P_{w|c}[2]$$

$$P_c[2] = 0$$

Therefore,

$$P_{ho}[i] = P_W[i\text{-}1]\, P_{c|w}[i] + P_c[i\text{-}1]\, P_{w|c}[i]$$

$$P_{ho}[2] = P_w[1]\, P_{c/w}[2] + P_c[1]\, P_{w/c}[2]$$

$$P_{ho}[2] = (1\text{-}P1[1])\, P1[2]$$

(WLAN→ Cellular Network: Voice Session):

$P_w[0] = 1$ and $P_c[0] = 0$

If i=1

$$P_W[i] = p_W[i\text{-}1]\,(1\text{-}P_{c|w}[i]) + P_c[i\text{-}1]\, P_{c|w}[i]$$

$$P_w[1] = p_w[0]\,(1\text{-}P_{c/w}[1]) + P_c[0]\, P_{c/w}[1]$$

$$P_w[1] = 1(1\text{-}P2[1]) + 0 = 1\text{-}P2[1]$$

$$P_c[i] = p_c[i\text{-}1]\,(1\text{-}P_{w|c}[i]) + P_W[i\text{-}1]\, P_{w|c}[i]$$

$$P_c[1] = p_c[0]\,(1\text{-}P_{w/c}[1]) + P_w[0]\, P_{w/c}[1]$$

$$P_c[1] = 0$$

Therefore,

$$P_{ho}[i] = P_W[i\text{-}1]\, P_{c|w}[i] + P_c[i\text{-}1]\, P_{w|c}[i]$$

$$P_{ho}[1] = P_w[0]\, P_{c/w}[1] + P_c[0]\, P_{w/c}[1]$$

$$P_{ho}[1] = P2[1] + 0 = P2[1]$$

If i=2

$$P_w[i] = p_w[i-1](1 - P_{c|w}[i]) + P_c[i-1] P_{c|w}[i]$$

$$P_w[2] = p_w[1](1 - P_{c/w}[2]) + P_c[1] P_{c/w}[2]$$

$$P_w[2] = (1-P2[1])(1-P2[2])$$

$$P_c[i] = p_c[i-1](1 - P_{w|c}[i]) + P_w[i-1] P_{w|c}[i]$$

$$P_c[2] = p_c[1](1 - P_{w/c}[2]) + P_w[1] P_{w/c}[2]$$

$$P_c[2] = 0$$

Therefore,

$$P_{ho}[i] = P_w[i-1] P_{c|w}[i] + P_c[i-1] P_{w|c}[i]$$

$$P_{ho}[2] = P_w[1] P_{c/w}[2] + P_c[1] P_{w/c}[2]$$

$$P_{ho}[2] = (1-P2[1]) P1[2]$$

(Cellular Network → WLAN: Data Session):

$P_w[0] = 0$ and $P_c[0] = 1$

If i=1

$$P_w[i] = p_w[i-1](1 - P_{c|w}[i]) + P_c[i-1] P_{c|w}[i]$$

$$P_w[1] = p_w[0](1 - P_{c/w}[1]) + P_c[0] P_{c/w}[1]$$

$$P_w[1] = 0$$

$$P_c[i] = p_c[i-1]\,(1- P_{w|c}[i]) + P_w[i-1]\,P_{w|c}[i]$$

$$P_c[1] = p_c[0]\,(1- P_{w/c}[1]) + P_w[0]\,P_{w/c}[1]$$

$$P_c[1] = (1- P3[1])$$

Therefore,

$$P_{ho}[i] = P_W[i-1]\,P_{c|w}[i] + P_c[i-1]\,P_{w|c}[i]$$

$$P_{ho}[1] = P_w[0]\,P_{c/w}[1] + P_c[0]\,P_{w/c}[1]$$

$$P_{ho}[1] = 0 + P3[1] = P3[1]$$

If i=2

$$P_w[i] = p_w[i-1]\,(1- P_{c|w}[i]) + P_c[i-1]\,P_{c|w}[i]$$

$$P_w[2] = p_w[1]\,(1- P_{c/w}[2]) + P_c[1]\,P_{c/w}[2]$$

$$P_w[2] = 0$$

$$P_c[i] = p_c[i-1]\,(1- P_{w|c}[i]) + P_w[i-1]\,P_{w|c}[i]$$

$$P_c[2] = p_c[1]\,(1- P_{w/c}[2]) + P_w[1]\,P_{w/c}[2]$$

$$P_c[2] = (1- P3[1])\,(1-P3[2])$$

Therefore,

$$P_{ho}[i] = P_W[i-1]\,P_{c|w}[i] + P_c[i-1]\,P_{w|c}[i]$$

$$P_{ho}[2] = P_w[1]\,P_{c/w}[2] + P_c[1]\,P_{w/c}[2]$$

$$P_{ho}[2] = (1-P3[1])\,P3[2]$$

(Cellular Network → WLAN: Voice Session):

$P_w[0] = 0$ and $P_c[0] = 1$

If i=1

$$P_w[i] = p_w[i-1](1 - P_{c|w}[i]) + P_c[i-1] P_{c|w}[i]$$

$$P_w[1] = p_w[0](1 - P_{c/w}[1]) + P_c[0] P_{c/w}[1]$$

$$P_w[1] = 0$$

$$P_c[i] = p_c[i-1](1 - P_{w|c}[i]) + P_w[i-1] P_{w|c}[i]$$

$$P_c[1] = p_c[0](1 - P_{w/c}[1]) + P_w[0] P_{w/c}[1]$$

$$P_c[1] = (1 - P4[1])$$

Therefore,

$$P_{ho}[i] = P_W[i-1] P_{c|w}[i] + P_c[i-1] P_{w|c}[i]$$

$$P_{ho}[1] = P_w[0] P_{c/w}[1] + P_c[0] P_{w/c}[1]$$

$$P_{ho}[1] = 0 + P4[1] = P4[1]$$

If i=2

$$P_w[i] = p_w[i-1](1 - P_{c|w}[i]) + P_c[i-1] P_{c|w}[i]$$

$$P_w[2] = p_w[1](1 - P_{c/w}[2]) + P_c[1] P_{c/w}[2]$$

$$P_w[2] = 0$$

$$P_c[i] = p_c[i-1](1 - P_{w|c}[i]) + P_w[i-1] P_{w|c}[i]$$

$$P_c[2] = p_c[1](1 - P_{w/c}[2]) + P_w[1] P_{w/c}[2]$$

$$P_c[2] = (1 - P4[2])$$

Therefore,

$$P_{ho}[i] = P_W[i-1] P_{c|w}[i] + P_c[i-1] P_{w|c}[i]$$

$$P_{ho}[2] = P_w[1] P_{c/w}[2] + P_c[1] P_{w/c}[2]$$

$$P_{ho}[2] = (1 - P4[1]) P4[2]$$

Chapter IV – Methodology (Horizontal Handoff)

In this part of thesis, two different networks were designed, using OPNET simulator in order to scrutinize the effect of different parameters in the network. OPNET Modeler is a leading commercial network simulator which is used for analyzing and designing communication networks, devices, protocols and applications. It includes a "library of detailed protocol and application models including Voice, HTTP, TCP, IP, Ethernet, ATM, 802.11 Wireless LANs, 802.16, UMTS, IP Multicast, Circuit Switch and many more. The Standard Model Library includes hundreds of vendor specific and generic device models including routers, switches, workstations, and packet generators" (National Science Foundation, 2010).

In this study, two different scenarios were designed; one simple and one crowded. Figure 8 and Figure 9 show these network models.

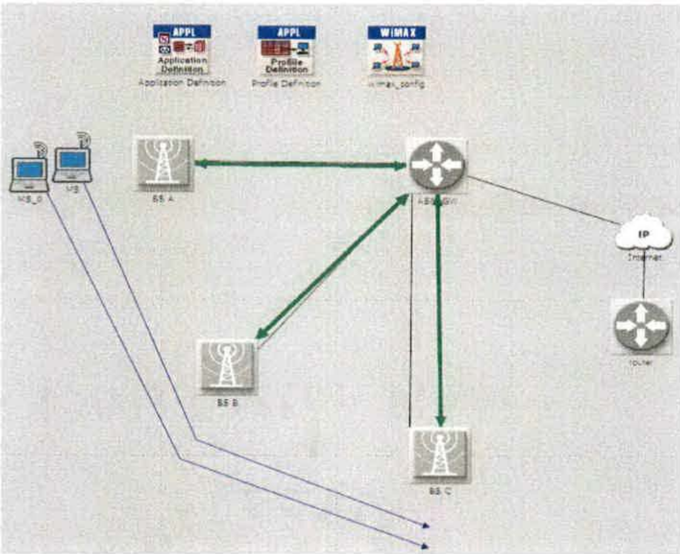

Figure8. Simple Network

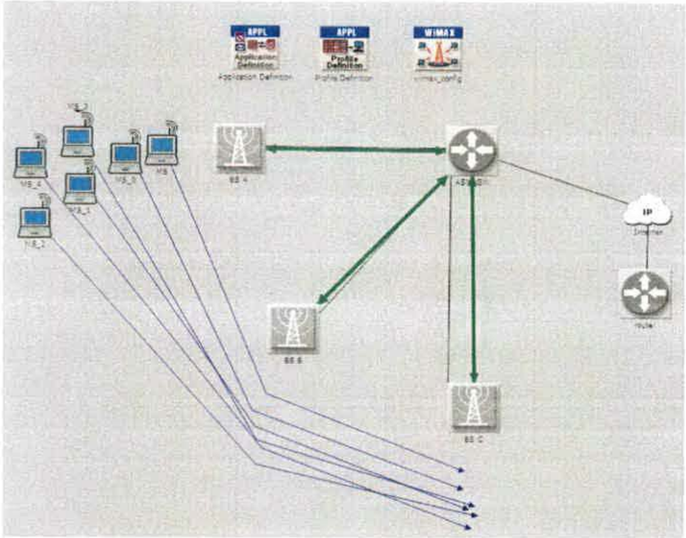

Figure9. Crowded Network

In the simple one, there are 2 mobile stations and 3 base stations and in the crowded one, the number of mobile stations was increased to 6 while the number of base stations remained the same. For each mobile station, a trajectory was set in order to perform horizontal handoff between base stations.

In both scenarios, an ASN gateway (ASN-GW) was used which is a router to handle handover between WiMax base stations. To implement ASN-GW, a separate router configured with GRE tunnels between the ASN-GW router and the base stations was needed. The entire WiMax network under the same ASN- GW must have the same IP subnet. Therefore, all WiMax interfaces for the BS nodes under the same ASN-GW must have an IP address configured under the same IP subnet. There are two steps for ASN-GW configuration

Step 1: Configure IP routing parameters

Step 2: Configure GRE tunnel information

The mobile stations were registered with the base station that was on the same subnet. In order for handoff to occur, the base stations must be in the same subnet. After configuring the IP routing parameters and configuring the GRE tunnel (green lines), the network was ready for simulation. The best simulation time for these models was estimated around 20 minutes (1200 seconds). The time which Kumar and Nagarajan (2011) considered for their handoff simulation in WiMax was 400 seconds. Furthermore, the simulation time which Klein, Pries, and Staehle (2006) considered for their WiMax network was 1000 seconds. Because the longer the simulations run time, the better the result, the simulation run time for this model was chosen as 20 minutes. During the simulation, it was proved this simulation run time was the best since by the end of simulation the needed results were collected from each base station. The applications which were used in these two scenarios were different kind of voice traffics as follows: PCM quality speech, GSM quality speech, and IP telephony. In the following section, these technologies and their advantages are described.

Pulse-code modulation (PCM)

Pulse-code modulation (PCM), which is the standard form for digital audio in computers, compact discs and etc., is a method used to digitally represent sampled analog signals. The first intention of using PCM was in telephone systems. However, it has later been considered as a standard way for digitalizing analog data such as in digital audio, digital video and CD formats (Pulse Code, 2010).

One of the benefits of using PCM service is that when the signal exceeds the noise level by value of 20 dB or more, the noise interference is eliminated. Moreover, the retransmission of the signal can be repeated as many times as desired without any distortion of the signal (Electrical Engineering, 2010).

Global System for Mobile Communications (GSM)

Global System for Mobile Communications (GSM) is considered as second generation mobile phone systems since the signaling is digital as well as speech channel. GSM is used by many people in more than 212 countries and according to the international roaming arrangement between mobile network operators, people can use the phones throughout the world with this service. GSM uses a variation of time division multiple access (TDMA) and is the most widely used of the three digital wireless telephony technologies (TDMA, GSM, and CDMA). GSM, together with other technologies, is part of the evolution of wireless mobile telecommunications (GSM, 2006).

The advantages of the GSM systems for the consumer include higher digital voice quality, low cost alternatives such as text messaging as well as offering roaming services to the subscriber which can use their phones all around the world. The inter-operability provides advantages for the network operators since they can deploy equipment from different vendors (Imran, 2007).

IP telephony

IP telephony is kind of technology which uses the Internet Protocol's Packet-Switched for delivering voice, fax and video packets. It is a trustable communication to users. IP telephony is an important part of the convergence of computers, telephones, and television into a single integrated information environment. VoIP or voice over Internet Protocol is an organized effort to standardize IP telephony (IP telephony, 1998).

IP telephony has many advantages. First of all it is cost effective, high distance calls can be made at a very cheap price. Next, it is flexible in which integration of other services and applications is possible. Furthermore, the voice quality has been improved and the delay of transmission has been reduced (Srivastava, 2011).

Simulation

The both scenarios were simulated based on these technologies in order to see the effect of each one on the delay, load, throughput and handover delay of each network. In both scenarios, the handover delay is collected from each mobile station and at the end the average handover delay for the whole network will be calculated. In the following sections, the results of simulation are shown.

Simulation for PCM Quality Speech

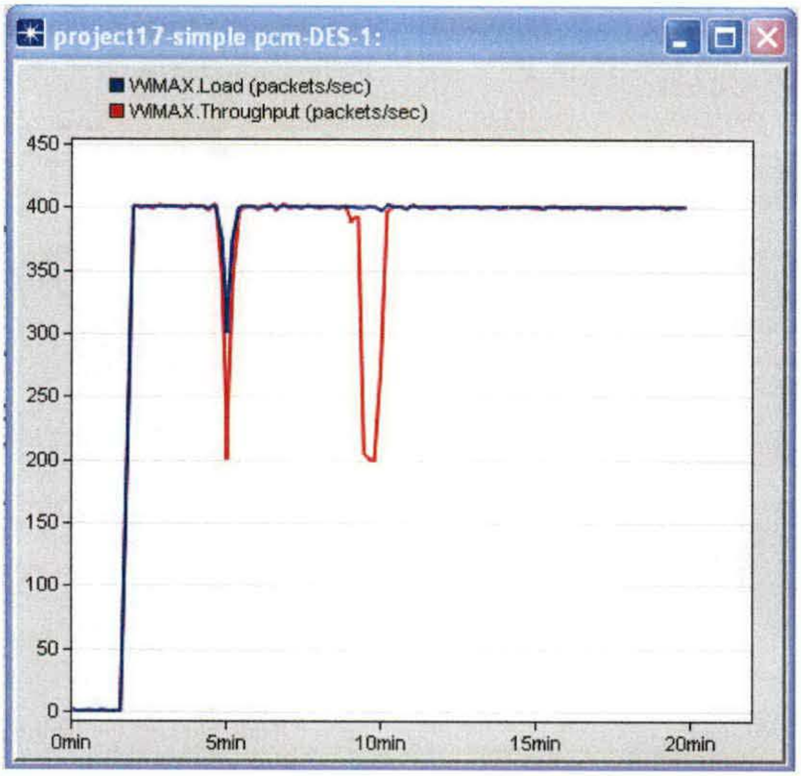

*Figure*10. Load versus Throughput (PCM, simple scenario)

The Figure 10 shows the load versus throughput for the simple scenario. As it is clear, in minutes5 and 10 of the simulation, there are several dropped voice data in the load which has a significant effect on the throughput of the network. In minute5, there are 100 packets/seconds (p/s) dropped voice data in the load which obviously caused about the 200 (p/s) dropped data in the throughput of the network. The condition is worse around minute10 of the simulation since almost no data is dropped in the load while the throughput has been dropped by 200 (p/s).

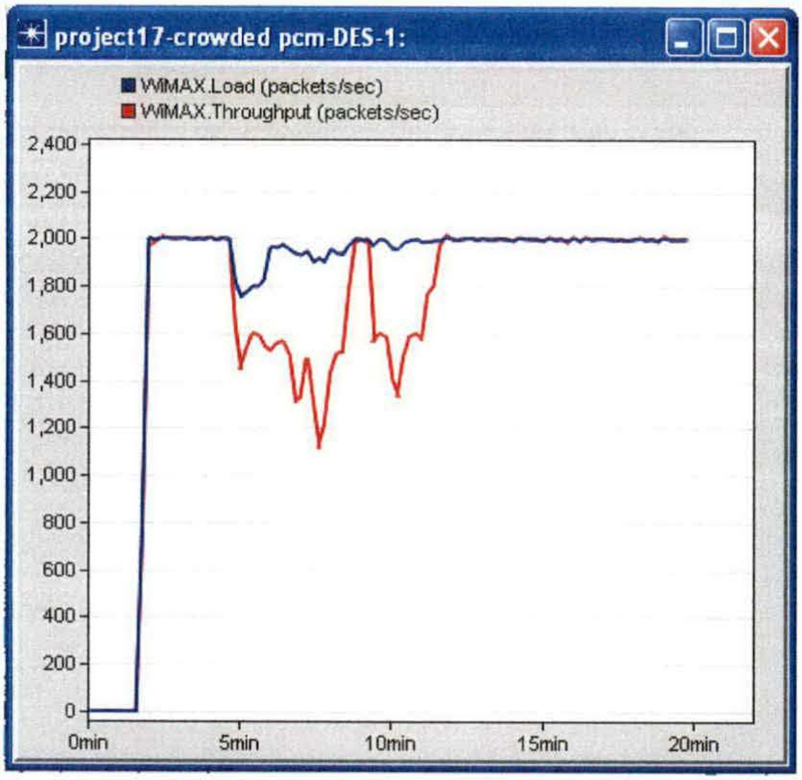

Figure11. Load versus Throughput (PCM, crowded scenario)

The Figure 11 shows the load versus throughput for the crowded scenario. In this scenario, there are continuous dropped voice data in the load from minute5 to 10 of the simulation which has a significant effect on the throughput and has caused continuous dropped data in the throughput of the network. The average dropped data in the load of the network is about 100 (p/s) while the average dropped data in the throughput of the network is about 400 (p/s). This can be explained as if by increasing the number of mobile stations, the distributed load among the mobile stations caused continuous dropped data in the throughput.

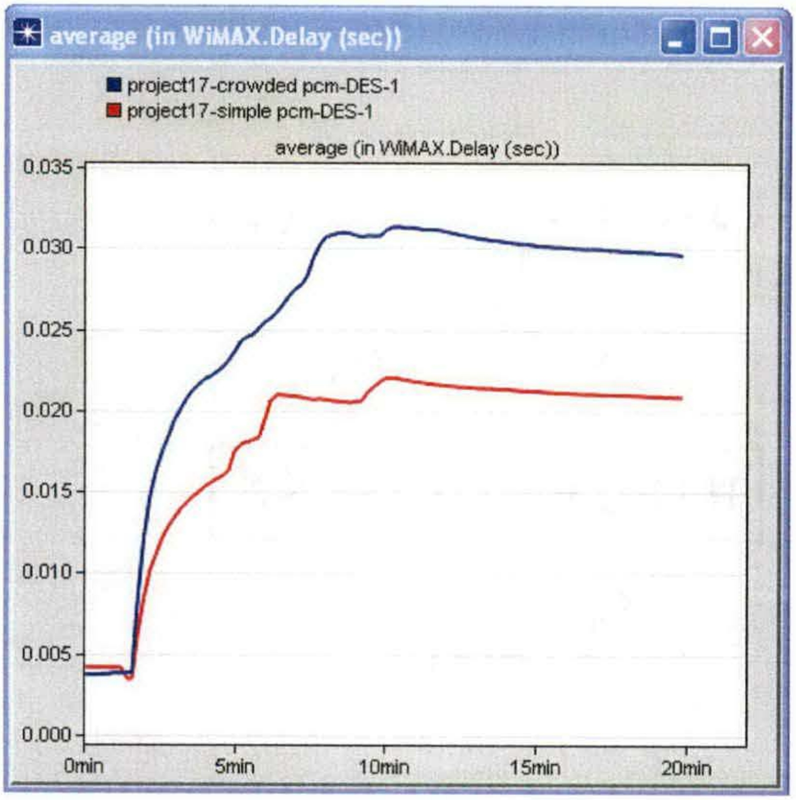

Figure12. Delay (PCM, simple and crowded scenario)

Figure 12 shows the delay of the simple scenarios versus crowded scenario. In the Figure 9, the red line shows the average delay for the simple network while the blue line shows the average delay for the crowded network. It is clear that by increasing the number of mobile stations, the delay of the network has been increased from 0.020 seconds to 0.030 seconds.

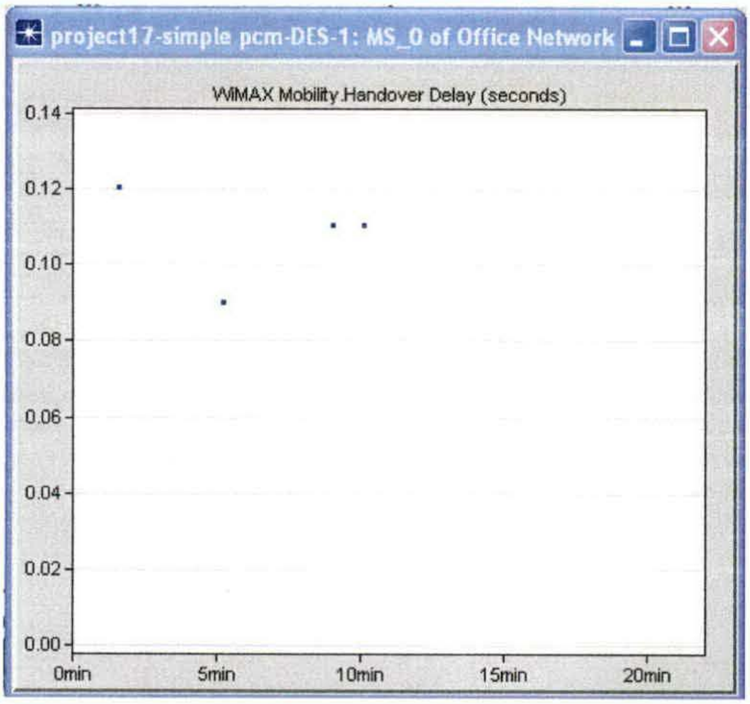

Figure13 .Handover delay for first mobile station (PCM, simple scenario)

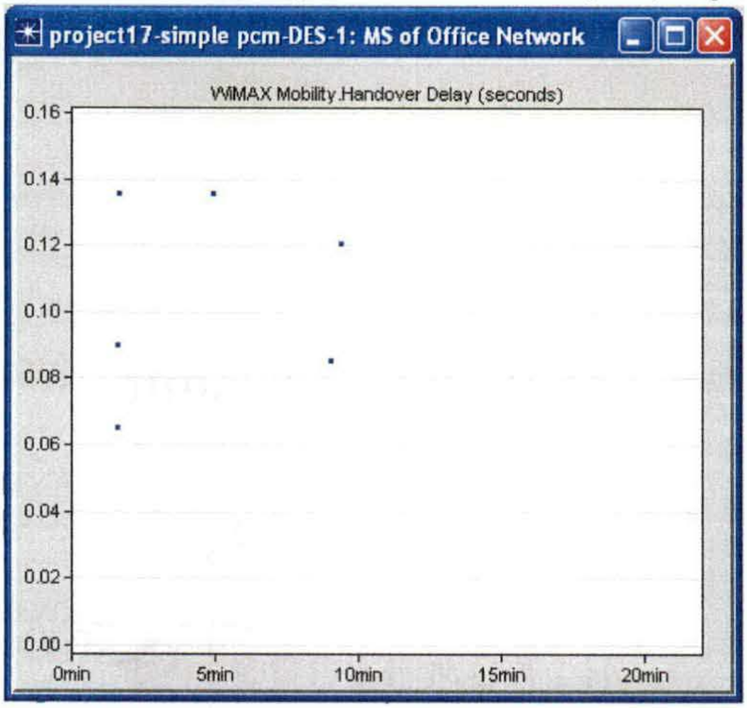

Figure14 .Handover delay for the second mobile station (PCM, simple scenario)

Figure 13 and Figure 14 show the handover delay for the mobile stations. Here, the greatest handover delay was considered. For the first mobile station, the handover delay is 0.12 seconds and for the second mobile station, the delay is about 0.135 seconds. The average of these two delays is around 0.128 seconds which is an acceptable handover delay for the simple network. In the following figures, the handover delay for the crowded scenario is calculated and the average is found.

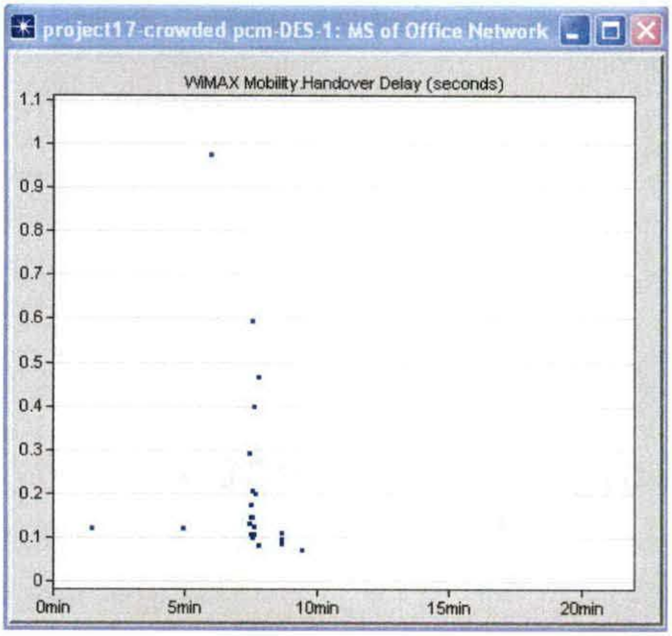

Figure15. Handover delay for the first mobile station (PCM, crowded scenario)

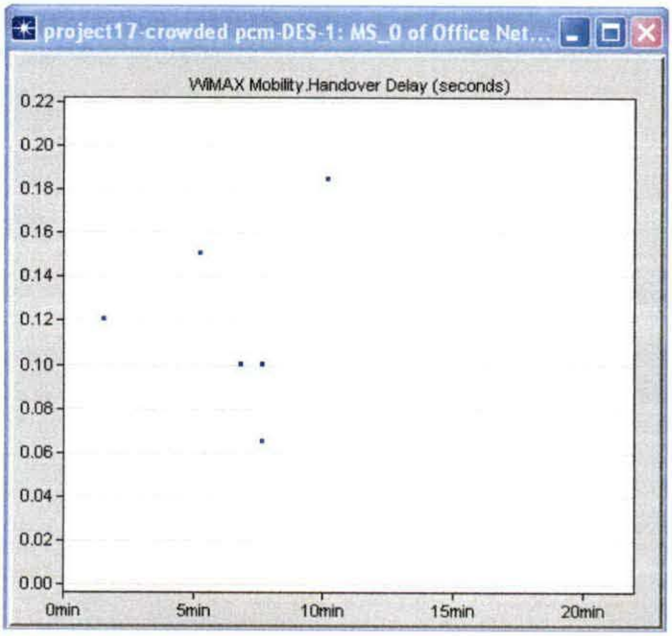

Figure16. Handover delay for the second mobile station (PCM, crowded scenario)

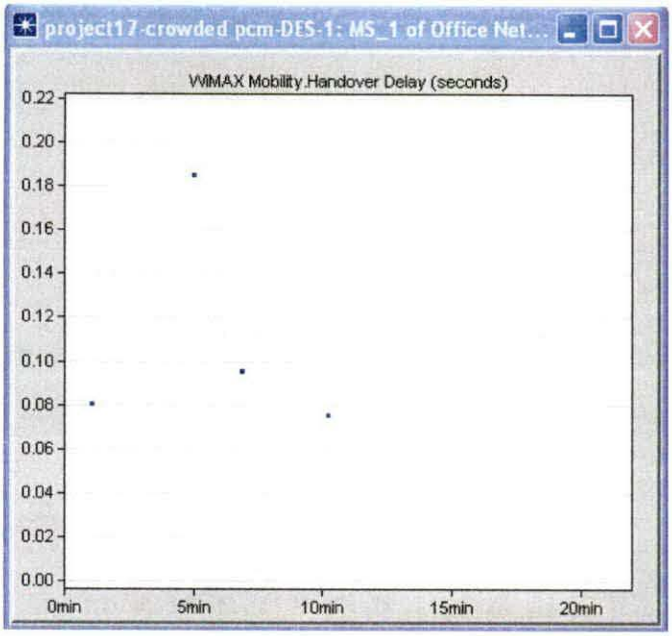

Figure17. Handover delay for the third mobile station (PCM, crowded scenario)

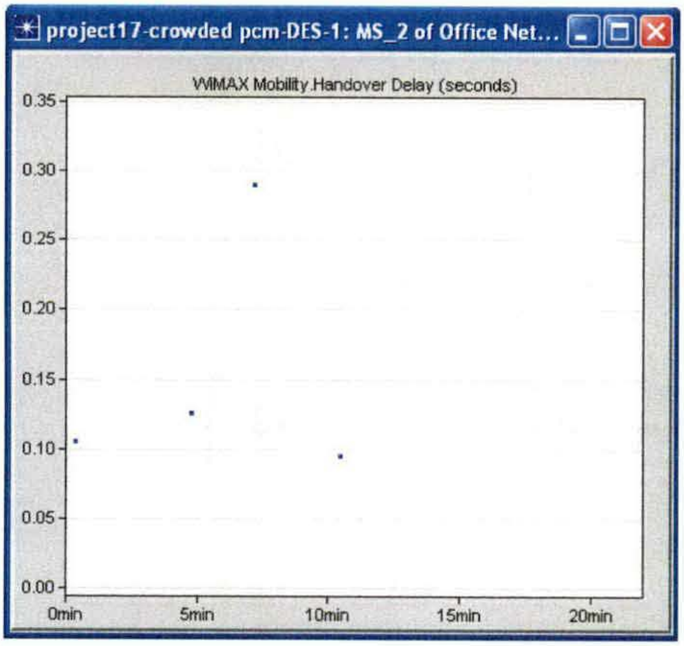

Figure18. Handover delay for the forth mobile station (PCM, crowded scenario)

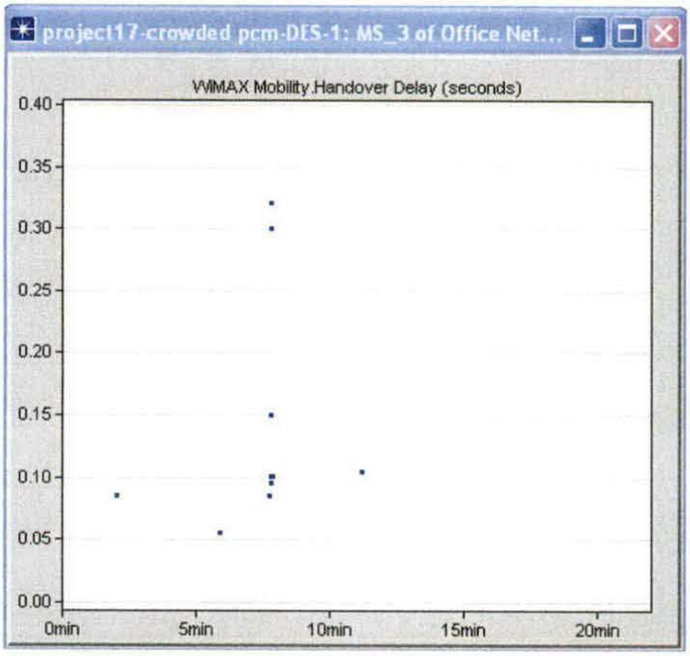

Figure19. Handover delay for the fifth mobile station (PCM, crowded scenario)

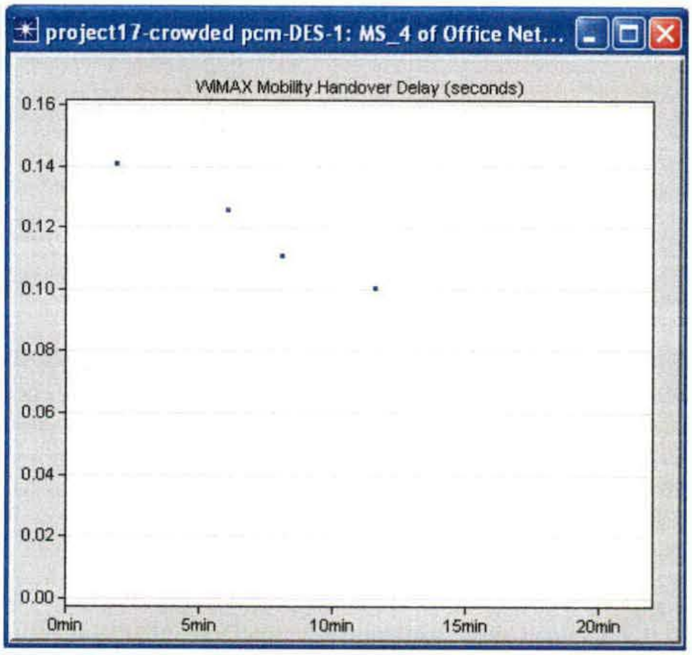

Figure20. Handover delay for the sixth mobile station (PCM, crowded scenario)

In the Figures 15 to 20, the handover delay for the mobile stations are as follows: 0.98 s, 0.185s, 0.185s, 0.29s, 0.33s and 0.14s. The average handover delay for the crowded scenario was calculated as 0.35 s.

Simulation for GSM Quality Speech

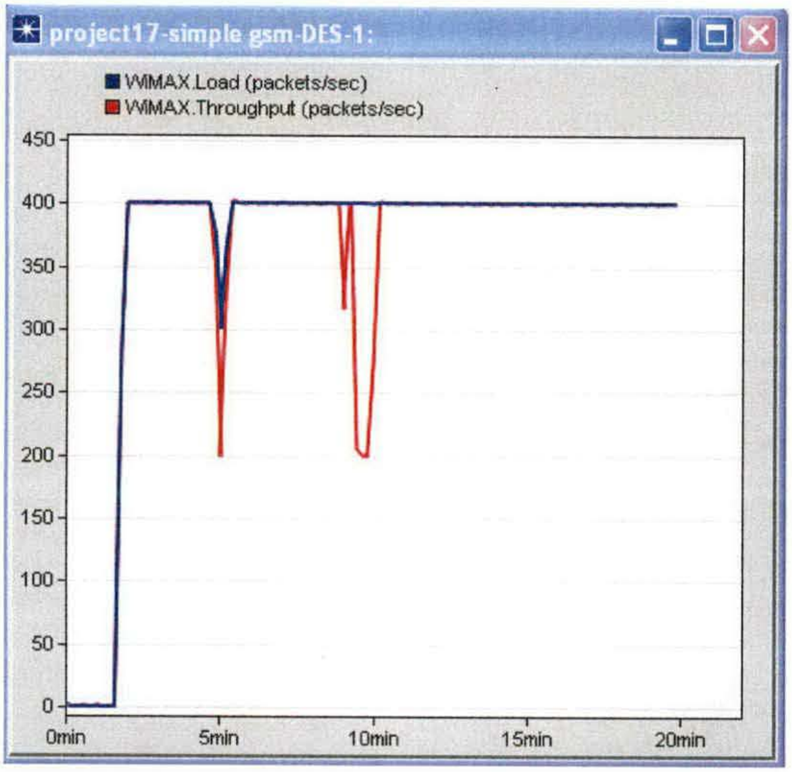

Figure21. Load versus Throughput (GSM, simple scenario)

The Figure 21 shows the load versus throughput for the simple scenario. Here, again there is about 200 (p/s) dropped voice data in minutes 5, 9 and 10 of the simulation. In minute5, 100 (p/s) dropped data caused 200 (p/s) dropped data in the throughput. In minute9 and 10 of the simulation, although there is not any dropped data in the load, there is significant dropped data in the through put of the network (100 (p/s) dropped data in minute9 and 200 (p/s) dropped data in minute10).

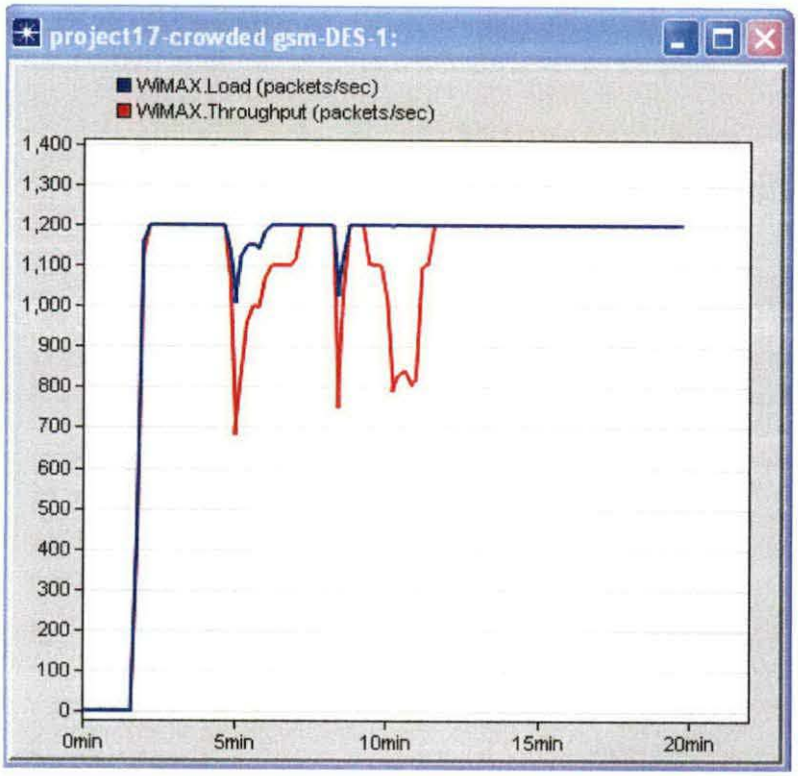

Figure22. Load versus Throughput (GSM, crowded scenario)

The Figure 22 shows the load versus throughput for the crowded scenario. In the crowded scenario, as it is clear, there are plenty of dropped voice data in the throughout compared to the load of the network. Specifically, in minute12 of the simulation, while there is no dropped data in the load, there are around 400 p/s dropped data in the throughput of the network.

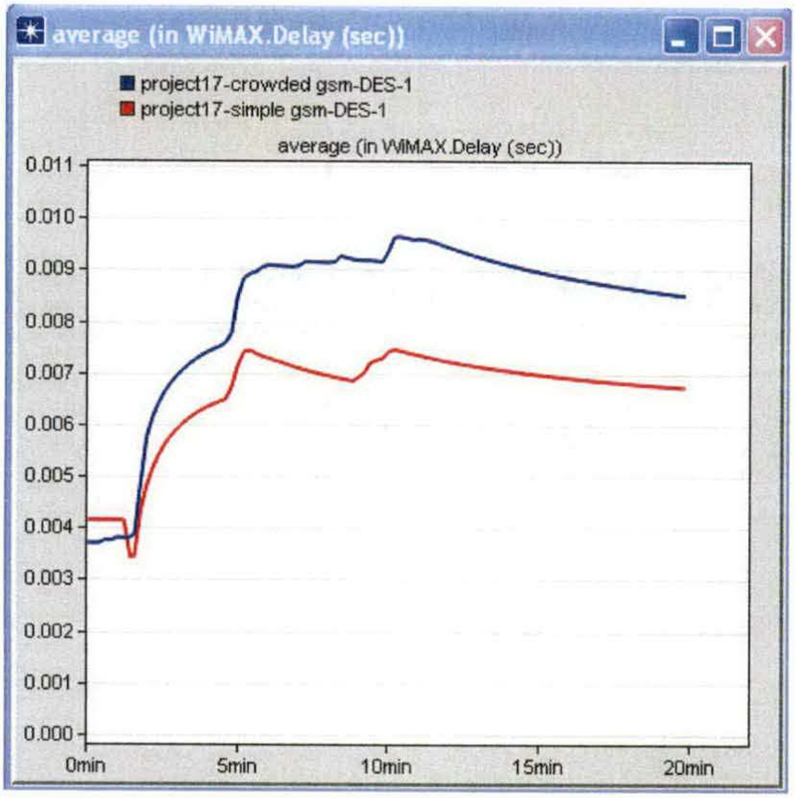

Figure23. Delay (GSM, simple and crowded scenario)

Figure 23 shows the delay of the simple scenario versus crowded scenario. In this figure, the red line again shows the delay in the simple network while the blue line shows the delay in the crowded network. As it is clear in the Figure 23, the delay for the simple network is about 0.007 seconds while for the crowded network is about 0.009 seconds.

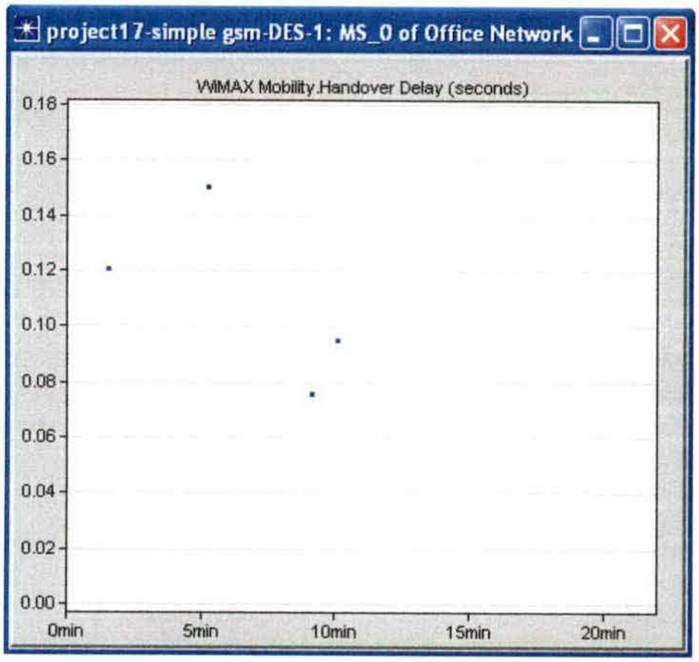

Figure24. Handover delay for the first mobile station (GSM, simple scenario)

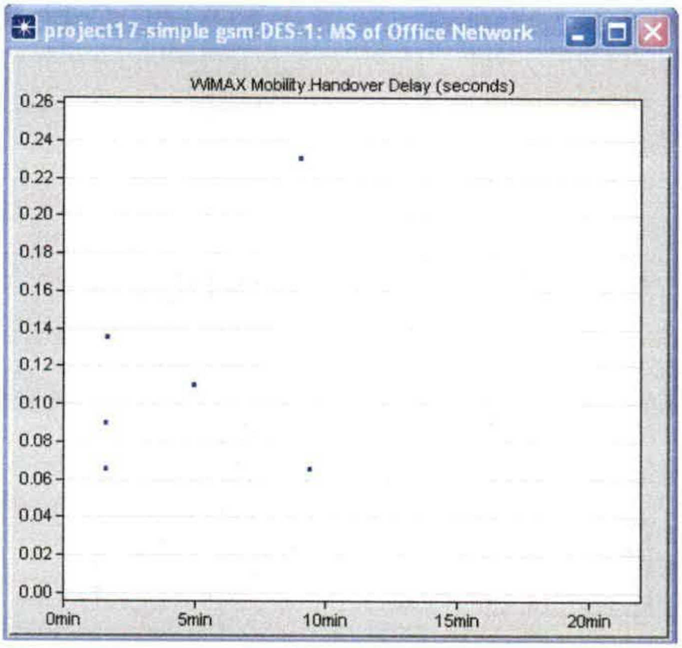

Figure25. Handover delay for the second mobile station (GSM, simple scenario)

Figure 24 and Figure 25 show the handover delay for the two mobile stations in the simple scenario. In this scenario with the GSM quality of speech the handover delay for the first mobile station is 0.15 seconds and for the second one is around 0.23 seconds which will give the average handover delay around 0.19 seconds for the simple scenario. The following figures show the handover delay for each mobile station for crowded scenario.

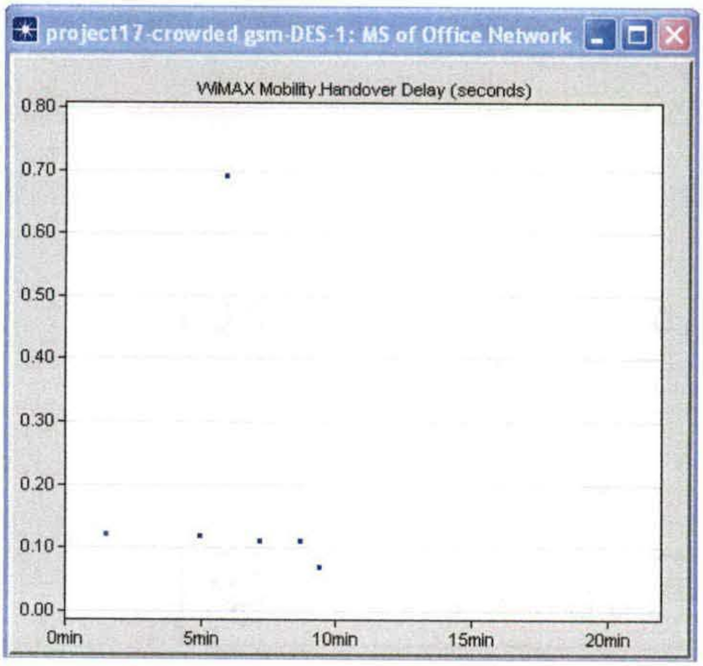

Figure26. Handover delay for the first mobile station (GSM, crowded scenario)

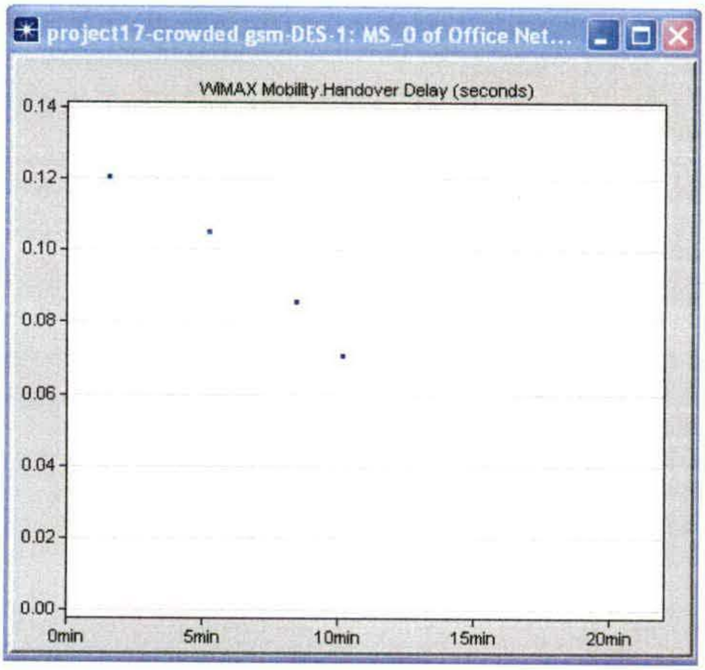

Figure27. Handover delay for the second mobile station (GSM, crowded scenario)

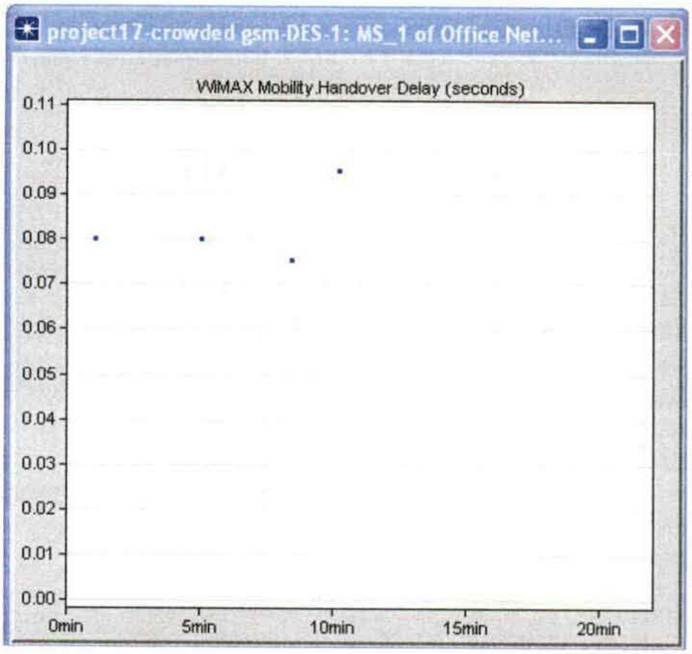

Figure28. Handover delay for the third mobile station (GSM, crowded scenario)

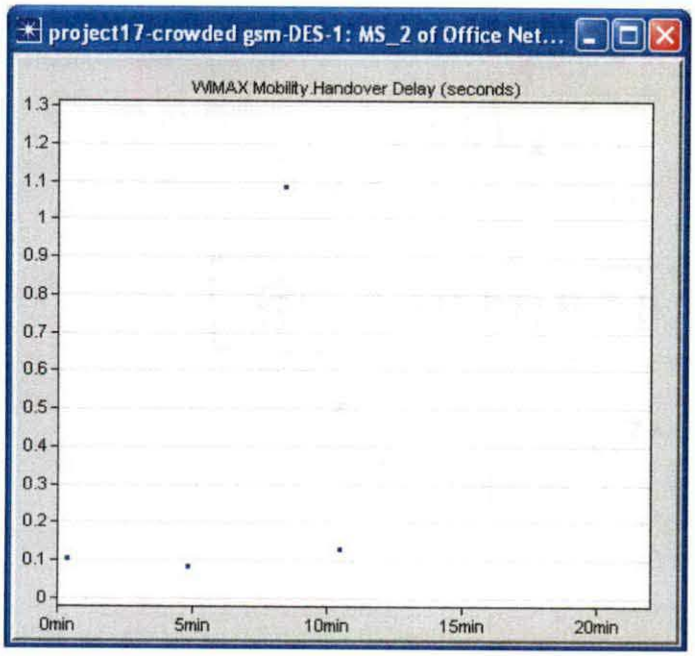

Figure29. Handover delay for the forth mobile station (GSM, crowded scenario)

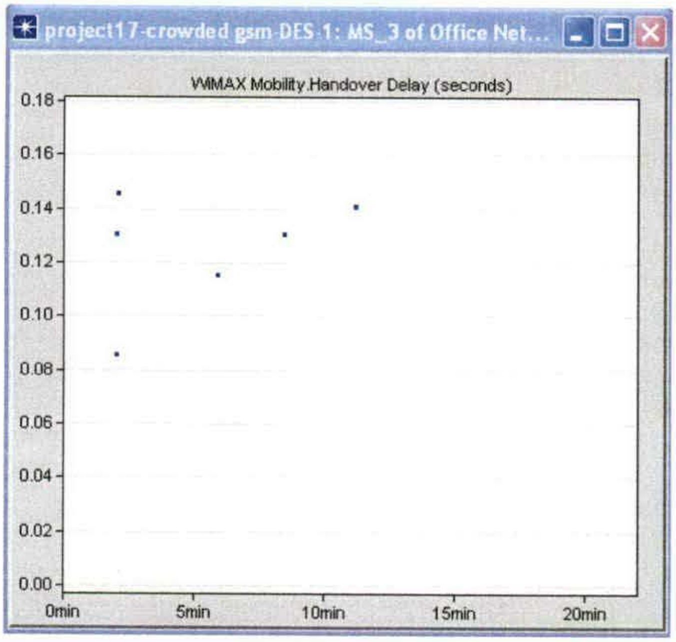

Figure30. Handover delay for the fifth mobile station (GSM, crowded scenario)

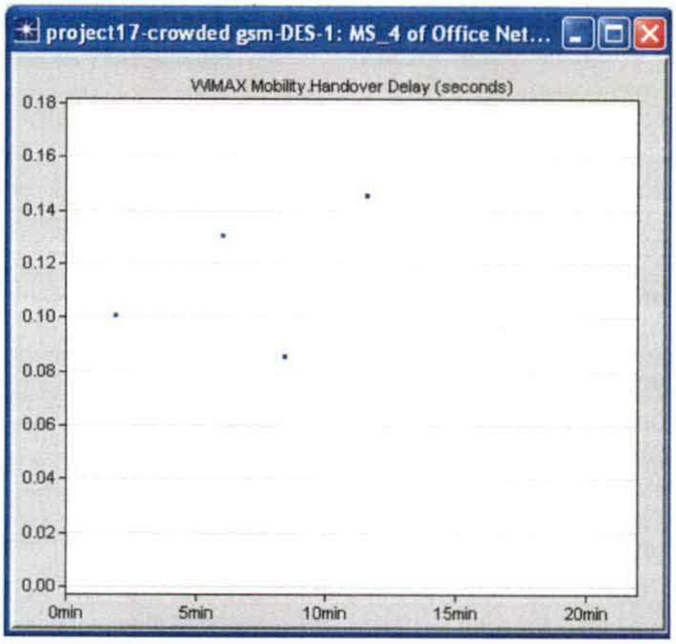

Figure31. Handover delay for the sixth mobile station (GSM, crowded scenario)

In the Figures 26 to 31, the handover delay for each mobile station as follows was gathered as follows: 0.7s, 0.12s, 0.095s, 1.1s, 0.145s, 0.145s. The average handover delay was calculated as 0.38 seconds for the crowded scenario.

Simulation for IP telephony

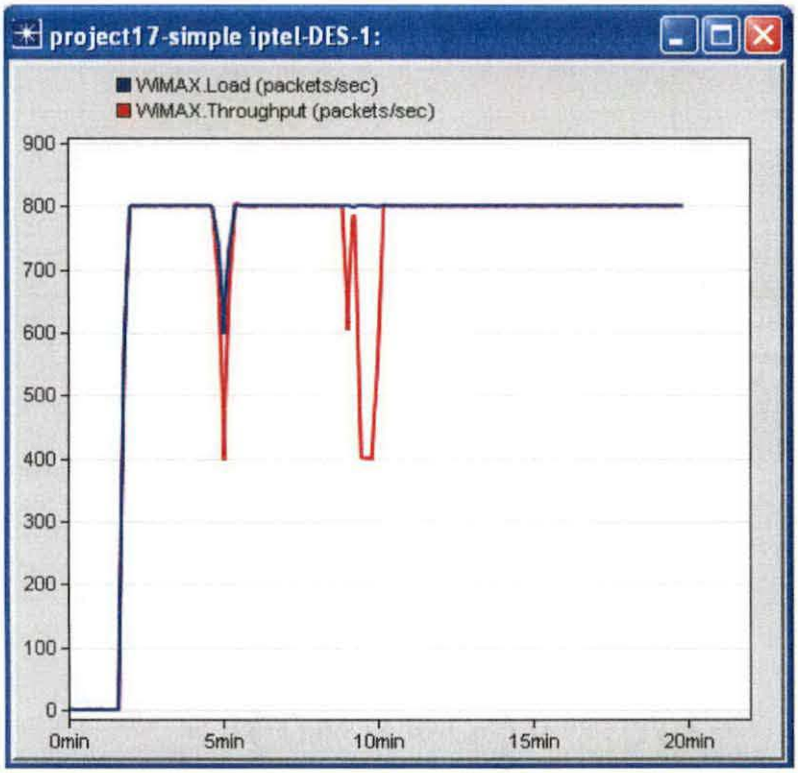

Figure32. Load versus Throughput (IP telephony, simple scenario)

The Figure 32 shows the load versus throughput for the simple scenario. In this scenario, in minute5 and 10 of the simulation, there are 400 (p/s) in the throughput of the network.

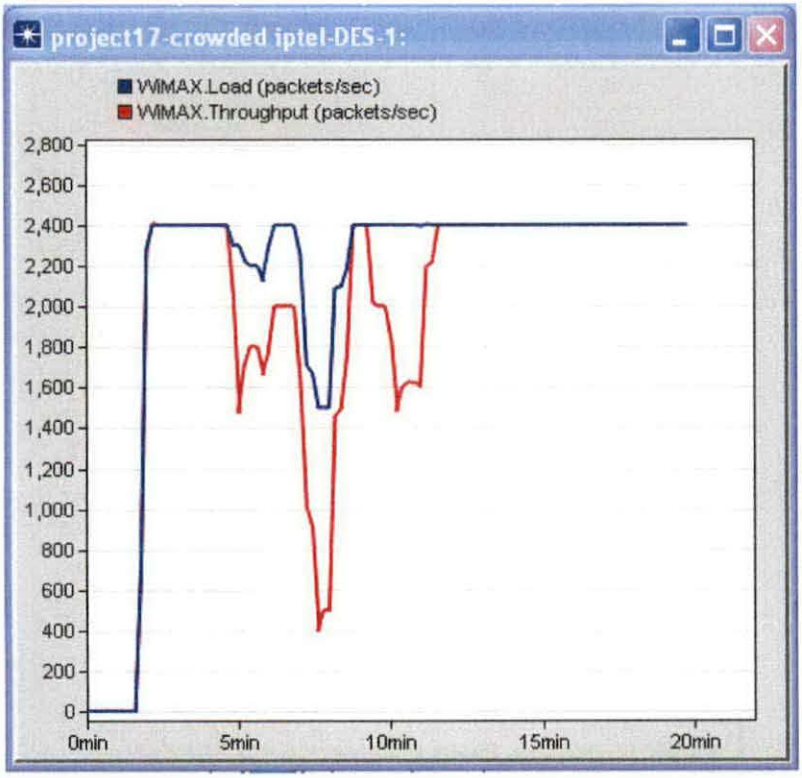

Figure33. Load versus Throughput (IP telephony, crowded scenario)

In the Figure 33 for the IP telephony for the crowded network, from minute5 to 7 and 10 to 12 of the simulation, there are about 800 (p/s) dropped data in the throughput of the network. The condition even is worse in minutes 6 to 8 of the simulation since there are about 2000 (p/s) data dropped which has a very bad effect on the performance of the network.

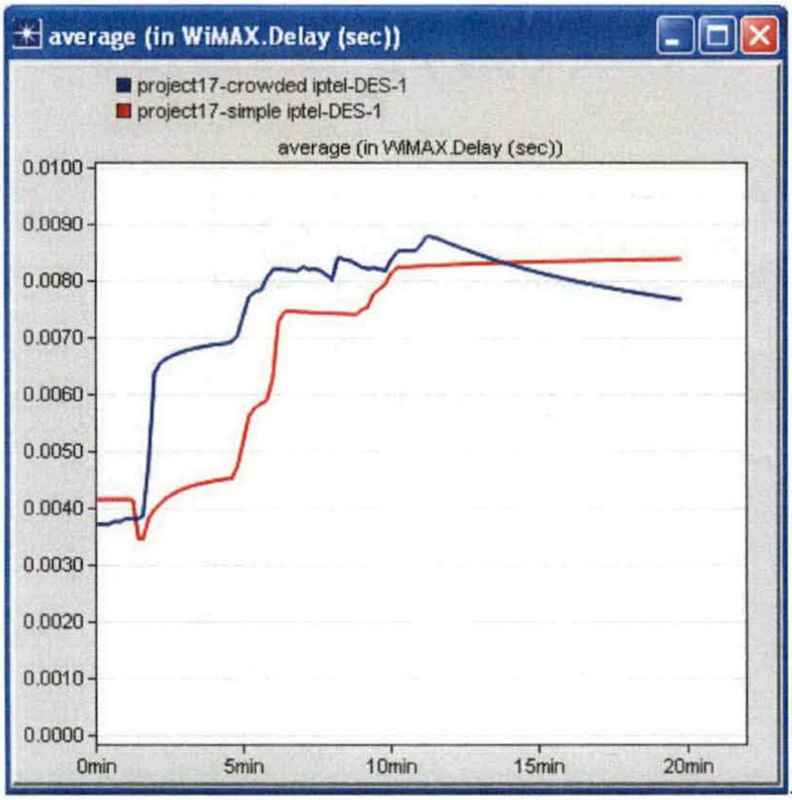

Figure34. Delay (IP telephony, simple and crowded scenario)

Figure 34 shows the delay of the simple scenario versus crowded scenario. In this figure, the average delay for the crowded network (blue line) until minute14 of the simulation is more than the average delay for the simple one (red line). But, from minute14 until the end of the simulation, the delay for the simple network is more than the crowded one.

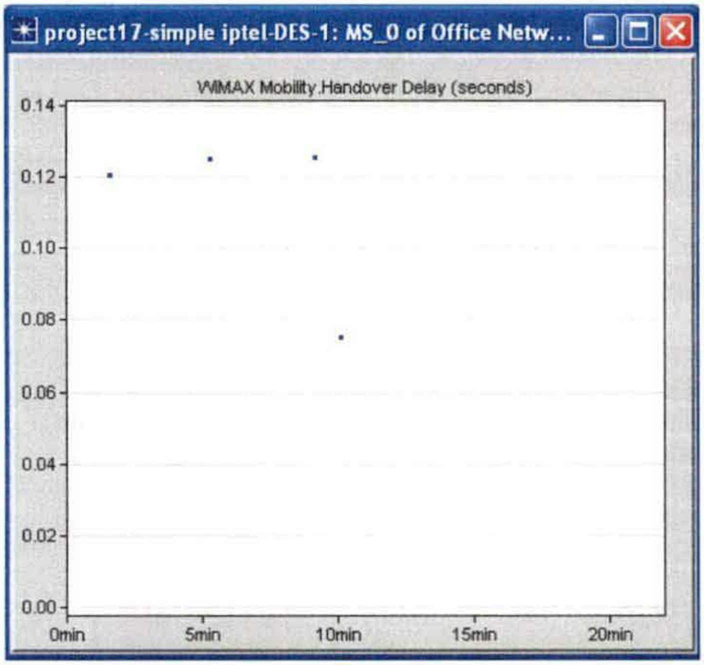

Figure35. Handover delay for the first mobile station (IP telephony, simple scenario)

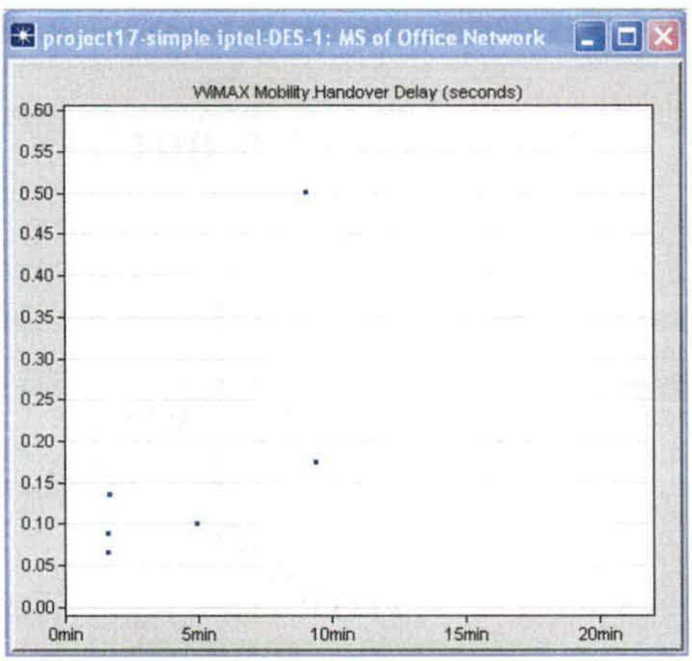

Figure36. Handover delay for the second mobile station (IP telephony, simple scenario)

Figure 35 and Figure 36, show that the handover delay for the first mobile station is around 0.125 and for the second one is o.50 which gave the average handover delay around 0.31s for the network. In the following figures, the handover delay for each mobile station in the crowded scenario is shown.

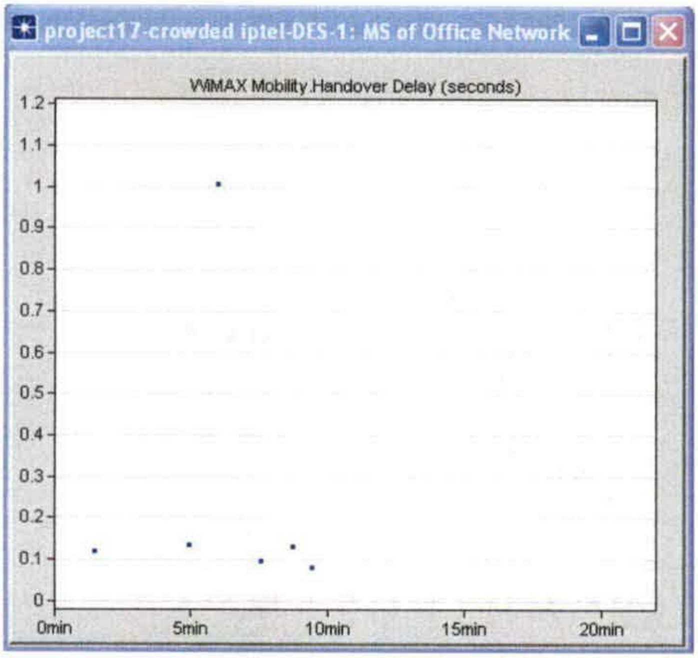

Figure37. Handover delay for the first mobile station (IP telephony, crowded scenario)

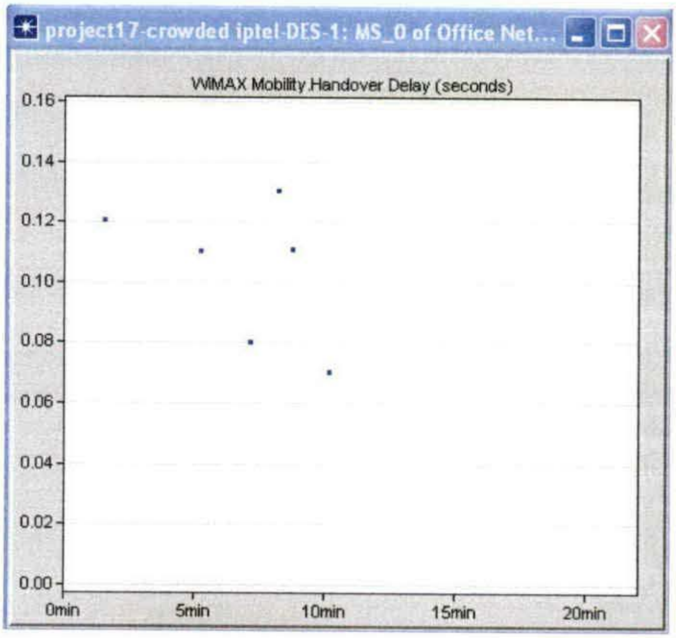

Figure38. Handover delay for the second mobile station (IP telephony, crowded scenario)

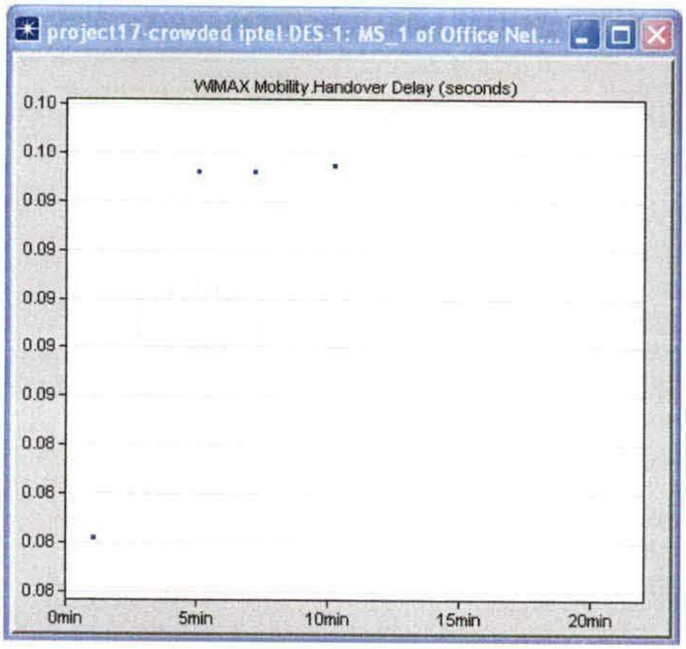

Figure39. Handover delay for the third mobile station (IP telephony, crowded scenario)

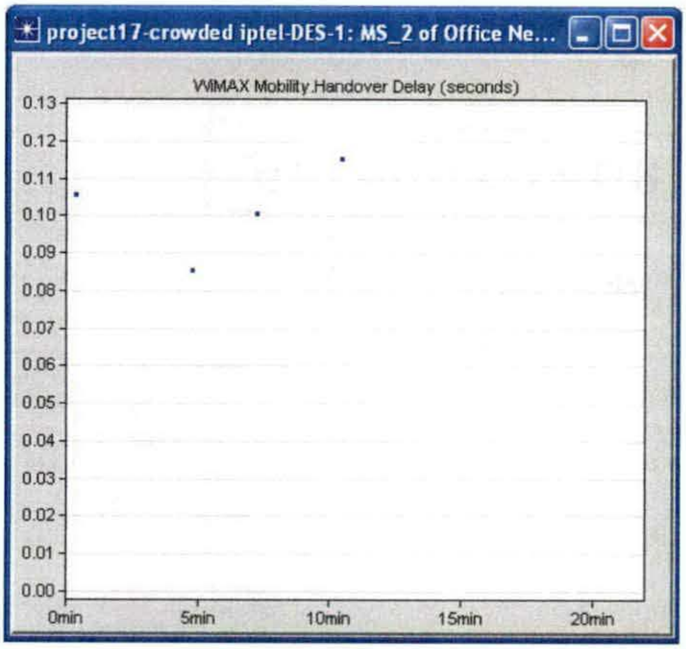

Figure40. Handover delay for the forth mobile station (IP telephony, crowded scenario)

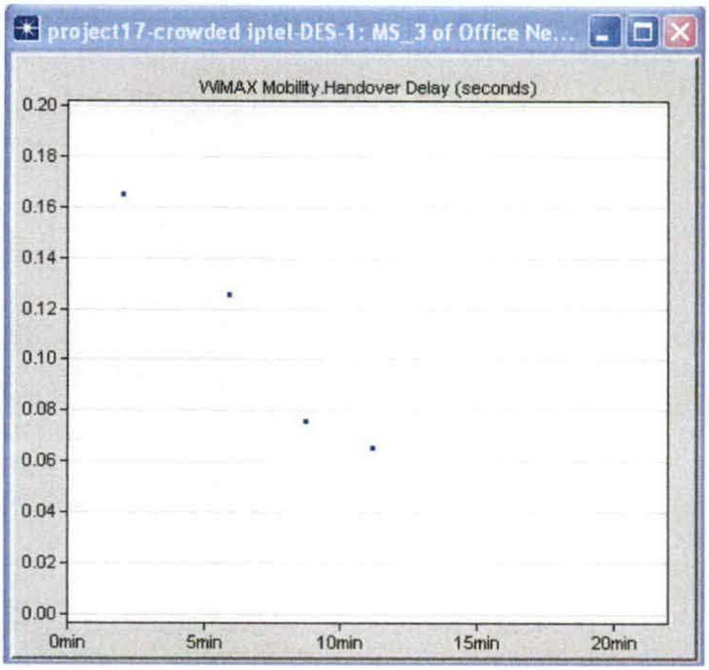

Figure41. Handover delay for the fifth mobile station (IP telephony, crowded scenario)

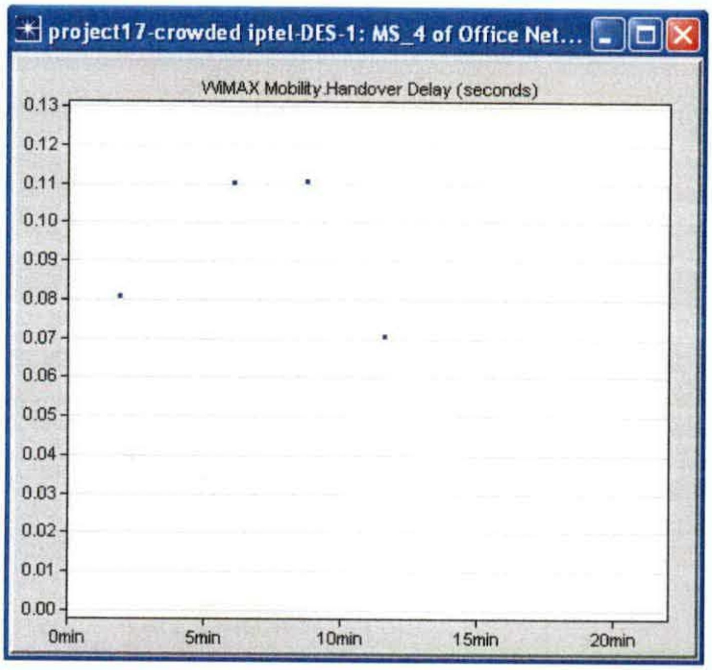

Figure42. Handover delay for the sixth mobile station (IP telephony, crowded scenario)

The Figures 37 to 41 show the handover delay for the each mobile station in the crowded network with IP telephony. The handover delays are as follows: 1s, 0.13s, 0.095s, 0.115s, 0.165s, and 0.11s. The average handover was calculated as 0.26 s for this scenario.

Finding

In the PCM quality speech, in the simple network, the dropped data was obvious in two positions in the simulation and was around 200 (p/s) while in the crowded scenario, there were more dropped data (around 400 to 800 (p/s)). The delay for the simple network was also about 0.02 seconds and for the crowded scenario it was about 0.03 seconds. Furthermore, the handover delay in the simple scenario was approximately 0.128 seconds which was less than the handover delay for the crowded scenario which was 0.35 seconds. The results showed that by increasing the number of mobile stations, both network delay and handover delay were increased. Moreover, the rate of dropped data was increased as well around an average of 400 (p/s). Therefore, in the network with PCM quality of speech, increasing the number of users had a negative effect on the delay, the handover delay and the throughput of the network.

In the GSM quality speech, the situation is again similar to the PCM quality speech. In the simple network, the dropped data was obvious in several positions in the simulation and was around 200 (p/s). In the crowded scenario, however, there were more dropped data (around 400 (p/s)). The delay for the simple network was approximately 0.007 seconds and was 0.009 seconds for the crowded scenario. Moreover, the handover delay in the simple scenario was 0.19 seconds which was less than the handover delay for the crowded one (0.38s). It seems that by increasing the number of mobile stations, the network delay and the handover delay were increased. Furthermore, the number of dropped data was increased from 200 (p/s) to 400 (p/s). The GSM quality of speech worked better in the simple network.

In the IP telephony, the condition was a little different. In the simple network, the dropped data was around the 400 (p/s) while for the crowded scenario, there was significant dropped data in minutes5 to12 of the simulation. Furthermore, the delay for the simple network until minute 14 of the simulation was around 0.0082 and it was 0.0089 seconds for the crowded network. Surprisingly, the delay was decreased in the crowded network compared to the simple one from minute 14 of the simulation to the end of simulation. Moreover, the handover delay was also different in both scenarios since it was around 0.31 seconds in the simple network which was greater than the handover delay for the crowded network (0.26s)

The result of comparing the 3 technologies (PCM, GSM and IP telephony) showed PCM and GSM both worked better in the simple network with fewer numbers of mobile stations. Obviously, the number of dropped data in the throughput of the crowded network was greater than the number for the simple network. Furthermore, the network delay and handover delay was less in the simple network compared to the crowded network. Between GSM and PCM, GSM worked better for delay sensitive applications since the delay of the network for both the simple and the crowded scenario was less than the delay of the network for both scenarios in the PCM quality speech. The results which were collected for IP telephony showed it was a good option for the delay sensitive applications. The handover delay and the network delay for the IP telephony showed the IP telephony can work better in the crowded network than in the simple network but the high amount of dropped data in the throughput of the network was not a favorable result and there should be several enhancements to the throughput of the networks which are working under IP telephony. On the one hand, because the delay was decreased in minute14 of the simulation in the crowded network compared to the simple network, it is good for delay sensitive applications. On the other hand, for the application that the data is important and all the data needs to be received, the IP telephony does not work well.

Chapter V. Conclusions and Future Works

In the first part of this thesis regarding vertical handoff, different algorithms for the vertical handoff between WLAN and cellular network were compared. The summery of comparison of these algorithms was also shown in two tables. Then the author's model for calculating the probability of occurring vertical handoff between WLAN and cellular network for two types of applications (data and voice) was proposed and the probability of handoff occurring between these two networks for time1 and 2 was calculated. It was a recursive model and the probability of handoff occurring can be calculated for different value of i. Since in this algorithm, the RSS was used, it is predicted this algorithm decreases the probability of occurring handoff. However, the proof and the simulation will be for the future work.

In the second part of this thesis regarding horizontal handoff, two different scenarios, one simple and one crowded in the WiMax network were simulated during horizontal handoff using OPNET simulator with three types of voice application (PCM, GSM and IP telephony). It was observed that the PCM and GSM worked better in the simple network and by increasing the user of the network, the delay of the network, handover delay and the throughput of the network were increased. Therefore, for the crowded network, the performance was decreased. In the IP telephony, condition was different because from the first of the simulation until minute 14 of the simulation, the delay was less in the simple network than the crowded network. However, from minute 14 until the end of the simulation, the delay was decreased in the crowded network compared to the simple network. Though the conditions changed, there was plenty of dropped data in the throughput of the network which had a negative effect on the performance of the network. The results which were gathered from the simulation for the delay and for the handover delay for the crowded network showed IP telephony can work better in the crowded networks compared to the other voice applications. Specially, IP telephony is good for delay sensitive applications. It does not work well, however, for the application in which the data is very important and the data needs to be received without any corruption since the number of dropped data was increased in the simulation. Therefore, for the future study, an algorithm should be designed to improve the throughput of the network and by decreasing the number of dropped data, to help improve the performance of the network. In this case, IP telephony could be the best choice for many delay sensitive applications as well as important data and voice applications.

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