A
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On

A Decision Model for Universal Seamless Handoff Architecture.

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CERTIFICATE

This is certified that the work contained in this dissertation entitled **“A Decision Model for Universal Seamless Handoff Architecture.”** by Padam Bhushan Chamoli is the requirement for the partial fulfillment for the award of degree of Master of Engineering in computer Technology & Application at Delhi College of Engineering is a bona-fide record work done by him under my guidance in the academic year 2006-2007.

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ABSTRACT

Roaming between the various network connection technologies like WLAN, CDMA, GSM etc. will be in high demand in the future. The user would want to roam between different networks without any interruption. It is also desirable for the subscribers to use the best network services without paying particular attention to its type. But the desire of communicating through the best connection suffers a setback with the change of location because of disparate or broken connection at that place.

The remedy for this problem is to takeover the connection by another available network i.e., a Vertical Handover. A handover is called seamless if it can be done without having to restart the running applications.

Various proposals for performing vertical handovers are discussed in this thesis and a prototype to deal with this problem of vertical seamless handover between WLAN , CDMA and GSM is presented. The challenging problem was to keep the session alive while changing the physical connection interface. This was done using a 'Decision model' and 'Velocity of mobile station'.

The decision model deals with chosing the best network out of all available networks depending upon the parameters like monetary cost, offered services, network conditions and user preferences. And the latter deals with the motion of the mobile station. The velocity factor decides whether the station is suitable for a handover.

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LIST OF ABBREVIATIONS

AP..............................Access Point
AuC............................Authentication center
BS..............................Base Station
BSS............................Basic Service Set
COA...........................Care-of address
CMR..........................Call-to-Mobility Ratio
CDMA........................Code Division Multiple Access
ESN............................Electronic Serial Number
EIR.............................Equipment Identity Register
ESS.............................Extended Service Set (ESS)
FCC............................Federal Communications Commission
FDMA........................Frequency Division Multiple Access
GSM ....Global System for Mobile
GPS............................Global Positioning System
GPRS..........................Global Packet Radio Service
HO.............................Handoff
HCC...........................Handoff Control Center
IP...............................Internet Protocol
IMT............................International Mobile Communications
LAN...........................Local Area Network
LA..............................Location Area
MH............................Mobile Host
MAC..........................Medium Access Control
MIN...........................Mobile Identification Number
MTSO.......................Mobile Telephone Switching Office
MSC..........................Mobile Switching Center
MA............................Mobile Agent
PDA...........................Personal Digital Assistant
PSTN.........................Public Switched Telephone Network
QoS...........................Quality of Service
RF............................ Radio Frequency
SDOs ....Standards Development Organizations
SDR...........................Software Defined Radio
SCTP.........................Stream Control Transmission Protocol
SID...........................System Identification Code
SA.............................Subnet Agent
SMS ....Short Message Service
TDMA.......................Time Division Multiple Access
TCP MH....................TCP Multi Homing
UMTS.......................Universal Mobile Telecommunications System
USHA.......................Universal Seamless Handoff Architecture
WCDMA...................Wideband Code Division Multiple Access
Wi-Fi....................... Wireless Fidelity

CHAPTER I

INTRODUCTION

1.1 OBJECTIVE

Various proposals for performing vertical handovers are discussed in this thesis and a prototype to deal with this problem of vertical seamless handover between WLAN , CDMA and GSM is presented. The challenging problem was to keep the session alive while changing the physical connection interface. This was done using a 'Decision model' and 'Velocity of mobile station'.The decision model deals with chosing the best network out of all available networks depending upon the parameters like monetary cost, offered services, network conditions and user preferences. And the latter deals with the motion of the mobile station. The velocity factor decides whether the station is suitable for a handover.

1.2 BACKGROUND

The step beyond 3G is probably a multi access solution. Different access network will be used to meet the optimum cost and performance and the requirement to be always connected. In the future Internet users probably want to roam between different network technologies without any interruptions. Already today there exists several wireless access networks like GSM, GPRS, 3G, Bluetooth and WLAN. The main challenge when performing a vertical seamless handover is to keep the communication session alive while changing physical connection to the Internet. The goal of the thesis is to evaluate and prototype a multi access solution where the connection is kept while moving from one access type to another.

1.3 MOTIVATION

The goal of the thesis is to study different vertical handoff techniques and to implement an application prototype. The prototype application should work with existing applications and without any extensions to the applications. The prototype should work with no modifications in the infrastructure. So the application should be application and infrastructure transparent. The prototype should primary manage to handle soft handovers and optional hard handovers. All communication sessions that the mobile node initiates will be done with the home address as source address.

1.4 PROBLEM STATEMENT

As mentioned the step beyond 3G is probably a multi access solution. Users would probably from time to time also like to connect to fixed broadband networks, WLANs and technologies such as Bluetooth associated to e.g. cable TV and DSL access points. Users also probably want to roam between different access networks with minimal user actions. The IETF Mobile IP standard was created for this but has many limitations. The main idea with Mobile IP is that a mobile node always should be reachable through a home IP address regardless of its point of attachment. Unfortunately today's Mobile IP doesn't support that a mobile node is registered to the home network through multiple network interfaces at the same time. Another important limitation is that mobile IP doesn't support session continuity. Session continuity means that a user should be able to switch network interface (access technology) to the home network without having to restart the running applications. Switching access technology from one to another is called a vertical handover. If a handover can be done without having to restart the applications it's called a seamless handover.

1.5 THE APPROACH

The work started with a requirement specification that were quite unspecific so the initial weeks were used to try out what could be done and what couldn't. Since it was hard to specify how this seamless vertical handoff problem would be solved the implementation part didn't get so extensive. The work in the beginning was also of mixed characteristics, literature studying and experimental implementation. However at the end implementation part was finished.

OVERVIEW OF MOBILE COMMUNICATION.

In the past decade, the telecommunications industry has witnessed an ever accelerated growth of the usage of the mobile communications[4]. As a result, the mobile communications technology has evolved from the so-called second-generation (2G) technologies, GSM in Europe, IS-95(CDMA) and IS-136 (TDMA) in USA, to the third generation (3G) technologies. Along with the standards development for providing voice service to mobile users, a group of standards to deliver data to the mobile users have evolved from both SDOs (Standards development organisations) and industry. Systems and applications, such as Short Message Service (SMS) for sending and receiving short text messages for mobile phone users, have been built and continue to be developed.

The genius of the cellular system is the division of a city into small cells. This allows extensive frequency reuse across a city, so that millions of people can use cell phones simultaneously. In a typical analog cell-phone system, the cell-phone carrier receives about 800 frequencies to use across the city. The carrier divides the entire city into cells. Each cell is typically sized at about 10 square miles (26 square kilometers).Cells are normally thought of as hexagons on a larger hexagonal grid, as shown in Figure1.1.

Each cell has a base station that consists of a tower and a small building containing the radio equipment that is used to communicate with Mobile Terminals over preassigned radio frequencies.Cell phones have low-power transmitters in them.

Many cell phones have two signal strengths: 0.6 watts and 3 watts . The base station also transmits at low power.

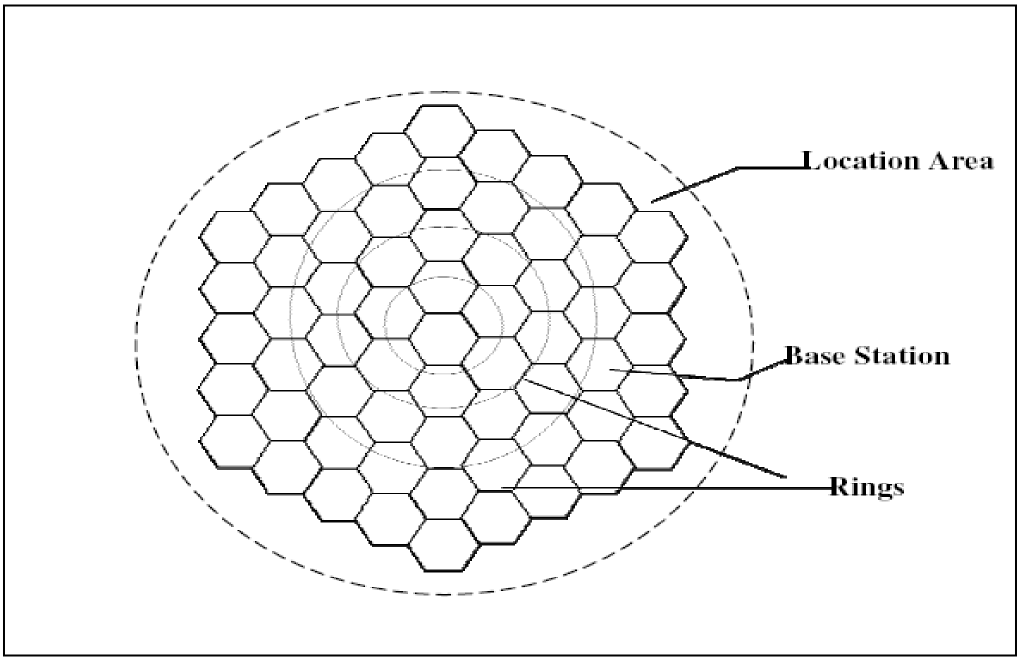

Figure 2.1: *The Cell Topology.*

Low-power transmitters have two advantages:

• The transmissions of a base station and the phones within its cell do not make it very far outside that cell. Therefore, in Figure1.1, both of the cells in alternate rings and non-adjacent cells can reuse the same frequency. The same frequencies can be reused extensively across the city.

• The power consumption of a cell phone, which is normally battery-operated, is relatively low. Low power corresponds to small batteries, and this is what has made handheld cellular phones possible.

The cellular approach requires a large number of base stations in a city of any size. A typical large city can have hundreds of towers. But because so many people are using cell phones, costs remain low per user. Each carrier in each city also runs one central office called the Mobile Telephone Switching Office (MTSO). This office handles all of the phone connections to the normal land-based phone system, and controls all of the base stations in the region. Groups of several cells are connected to a Mobile Switching Center (MSC) through which the calls are then routed to the telephone networks. The area serviced by a MSC is called a Registration Area (RA) or

Location Area (LA). A group of RA’s composes a Service Area (SA). Each SA is serviced by a Home Location Register (HLR). A wireless network may include several SAs and thus several HLRs.

All cell phones have special codes associated with them. These codes are used to identify the phone, the phone's owner and the service provider. Electronic Serial Number (ESN) (a unique 32-bit number programmed into the phone when it is manufactured), Mobile Identification Number (MIN) (a 10-digit number derived from the owners phone's number), and a System Identification Code (SID) (a unique 5-digit number that is assigned to each carrier by the FCC-Federal Communications Commission (A U.S. government agency charged with the task of regulating all forms of interstate and international communication)) are a few of the standard cell phone codes employed. While the ESN is considered a permanent part of the phone, both the MIN and SID codes are programmed into the phone when one purchases a service plan and has the phone activated.

2G systems such as GSM, IS-95, and cdmaOne were designed to carry speech and lowbit rate data. 3G systems were designed to provide higher data rate services. During the evolution from 2G to 3G, a range of wireless systems, including GPRS, Bluetooth, WLAN and HiperLAN have been developed.All these systems were designed independently, targeting different service types , data rates, and users. As these systems all have their own merits and shortcomings, there is no single system that is good enough to replace all the other technologies. In cellular networks such as GSM, a call is seamlessly handed over from one cell to another using hard handover without the loss of voice data. This is managed by networks based handover control mechanisms that detect when a user is in a handover zone between cells and redirect the voice data at the appropriate moment to the mobile node via the cell that the MN has just entered. In 4G networks a handover between different networks is required. A handover between different networks is referred to as a vertical handover.

2.1 EVALUTION FROM 1G TO 4G

From the early analog mobile generation (1G) to the last implemented third generation (3G) the paradigm has changed. The new mobile generations do not pretend to improve the voice communication experience but try to give the user access to a new global communication reality. The aim is to reach communication ubiquity (every time, everywhere) and to provide users with a new set of services.

2.1.1 THE FIRST GENERATION MOBILE (1G TO 2.5G)

The first operational cellular communication system was deployed in the Norway in 1981 and was followed by similar systems in the US and UK. These first generation systems provided voice transmissions by using frequencies around 900 MHz and analog modulation[1].

The second generation (2G) of the wireless mobile network was based on low-band digital data signaling. The most popular 2G wireless technology is known as Global Systems for Mobile Communications (GSM). The first GSM systems used a 25MHz frequency spectrum in the 900MHz band. FDMA (Frequency Division Multiple Access), which is a standard that lets multiple users access a group of radio frequency bands and eliminates interference of message traffic, is used to split the available 25MHz of bandwidth into 124 carrier frequencies of 200 kHz each. Each frequency is then divided using a TDMA (Time Division Multiple Access) scheme into eight timeslots and allows eight simultaneous calls on the same frequency. This protocol allows large numbers of users to access one radio frequency by allocating time slots to multiple voice or data calls. TDMA breaks down data transmission, such as a phone conversation, into fragments and transmits each fragment in a short burst, assigning each fragment a time slot. With a cell phone, the caller does not detect this fragmentation.

While GSM technology was developed in Europe, CDMA (Code Division Multiple Access) technology was developed in North America. CDMA uses spread spectrum technology to break up speech into small, digitized segments and encodes them to identify each call. CDMA distinguishes between multiple transmissions carried simultaneously on a single wireless signal[4]. It carries the transmissions on that signal, freeing network room for the wireless carrier and providing interference-free calls for the user.

While GSM and other TDMA-based systems have become the dominant 2G wirelesses technologies, CDMA technology are recognized as providing clearer voice quality with less background noise, fewer dropped calls, enhanced security, greater reliability and greater network capacity.

The Second Generation (2G) wireless networks mentioned above are also mostly based on circuit switched technology, are digital and expand the range of applications to more advanced voice services[1]. 2G wireless technologies can handle some data capabilities such as fax and short message service at the data rate of up to 9.6 kbps, but it is not suitable for web browsing and multimedia applications.

So-called '2.5G' systems recently introduced enhance the data capacity of GSM and mitigate some of its limitations. These systems add packet data capability to GSM networks, and the most important technologies are GPRS (General Packet Radio Service) and WAP (Wireless Application Protocol). WAP defines how Web pages and similar data can be passed over limited bandwidth wireless channels to small screens being built into new mobile telephones. At the next lower layer, GPRS defines how to add IP support to the existing GSM infrastructure. GPRS provides both a means to aggregate radio channels for higher data bandwidth and the additional servers required to off-load packet traffic from existing GSM circuits.

2.1.2 THIRD GENERATION MOBILE NETWORKS (3G)

All 2G wireless systems are voice-centric. GSM includes short message service (SMS), enabling text messages of up to 160 characters to be sent, received and viewed on the handset. Most 2G systems also support some data over their voice paths, but at painfully slow speeds usually 9.6 Kb/s or 14.4 Kb/s[3]. So in the world of 2G, voice remains king while data is already dominant in wireline communications. And, fixed or wireless, all are affected by the rapid growth of the Internet.

Planning for 3G started in the 1980s. Initial plans focused on multimedia applications such as videoconferencing for mobile phones[1]. When it became clear that the real killer application was the Internet, 3G thinking had to evolve. As personal wireless handsets become more common than fixed telephones, it is clear that personal wireless Internet access will follow and users will want broadband Internet access wherever they go.Today's 3G specifications call for 144 Kb/s while the user is on the move in an automobile or train, 384 Kb/s for pedestrians, and ups to 2 Mb/s for stationary users. That is a big step up from 2G bandwidth using 8 to 13 Kb/s per channel to transport speech signals.

The second key issue for 3G wireless is that users will want to roam worldwide and stay connected. A key goal of 3G is to make this roaming capacity universal.A third issue for 3G systems is capacity. As wireless usage continues to expand, existing systems are reaching limits. Cells can be made smaller, permitting frequency reuse, but only to a point.

The next step is new technology and new bandwidth. International Mobile Telecommunications-2000 (IMT-2000) is the official International Telecommunication Union name for 3G and is an initiative intended to provide wireless access to global telecommunication infrastructure through both satellite and terrestrial systems, serving fixed and mobile phone users via both public and private telephone networks. GSM proponents put forward the universal mobile telecommunications system (UMTS), an evolution of GSM, as the road to IMT-2000. With the advent of mobile Internet access, suddenly the circuit-based backhaul network from the base station and back has to significantly change. 3G systems are IP-centric and will justify an all-IP infrastructure. There will be no flip to 3G, but rather an evolution and, because of the practical need to re-use the existing infrastructure and to take advantage of new frequency bands as they become available, that evolution will look a bit different depending on where you are. The very definition of 3G is now an umbrella, not a single standard, however, the industry is moving in the right direction towards a worldwide, converged, network. Meanwhile, ever-improving DSPs will allow multi-mode, multi-band telephones that solve the problem of diverse radio interfaces and numerous frequency bands. When one handset provides voice and data anywhere in the world, that will be 3G no matter what is running behind the scenes.

2.1.3 FUTURE MOBILE GENERATION NETWORKS (4G)

The objective of the 3G was to develop a new protocol and new technologies to further enhance the mobile experience. In contrast, the new 4G framework to be established will try to accomplish new levels of user experience and multi-service capacity by also integrating all the mobile technologies that exist (e.g. GSM - Global System for Mobile Communications, GPRS - General Packet Radio Service, IMT-2000 - International Mobile Communications, Wi-Fi - Wireless Fidelity, Bluetooth)[2][6].

In spite of different approaches, each resulting from different visions of the future platform currently under investigation, the main objectives of 4G networks can be stated in the following properties:

• Ubiquity;
• Multi-service platform;
• Low bit cost;

Ubiquity means that this new mobile networks must be available to the user, any time, anywhere. To accomplish this objective services and technologies must be standardized in a worldwide scale.

A multi-service platform is an essential property of the new mobile generation, not only because it is the main reason for user transition, but also because it will give telecommunication operators access to new levels of traffic. Voice will loose its weight in the overall user bill with the raise of more and more data services.

Low-bit cost is an essential requirement in a scenario where high volumes of data are being transmitted over the mobile network. With the actual price per bit, the market for the new high demanding applications, which transmit high volumes of data (e.g. video), is not possible to be established.

To achieve the proposed goals, a very flexible network that aggregates various radio access technologies, must be created. This network must provide high bandwidth, from 50-100 Mbps for high mobility users, to 1Gbps for low mobility users, technologies that permit fast handoffs, an efficient delivery system over the different wireless technologies available, a method of choosing the wireless access from the available ones. Also necessary is a QoS framework that enables fair and efficient medium sharing among users with different QoS requirements, supporting the different priorities of the services to be deployed. The core of this network should be based in Internet Protocol version 6 – IPv6, the probable convergence platform of future services (IPv4 does not provide a suitable number of Internet addresses).

The network should also offer sufficient reliability by implementing a fault-tolerant architecture and failure recovering protocols.

2.2 COMPARISON BETWEEN 1G TO 4G.

The History and evolution[4] of mobile services from the 1G (first generation) to fourth generation are discussed in this section. Table1 presents a short history of mobile telephone technologies.

Technology	1G	2G	2.5G	3G	4G
Service	Analog Voice, Synchronous data to 9.6 kbps	Digital Voice, short messages	Higher capacity, packetized data	Higher capacity, broadband data upto 2 Mbps	Higher capacity, completely IP-oriented, multimedia, data to hundreds of megabits
Standards	AMPS, TACS, NMT, etc	TDMA, CDMA, GSM, PDC	GPRS, EDGE, 1xRTT	WCDMA, CDMA2000	Single standard
Data Bandwidth	1.9 kbps	14.4 kbps	384 kbps	2 Mbps	200 Mbps
Multiplexing	FDMA	TDMA, CDMA	TDMA,CDMA	CDMA	CDMA
Core network	PSTN	PSTN	PSTN, packet network	Packet network	Internet

Table 2.1 : *Short History of Mobile Telephone Technologies.*

First Generation	Second Generation	Third Generation	Fourth Generation
Mobile Telephone Service: car phone	• Digital Voice and Messaging –Data • Mobile Phone • Fixed Wireless Loop	• Integrated High Quality Audio, Video and Data. • Narrowband and Broadband Multimedia Services	• Dynamic information access • Telepresence (virtual meetings, education, and training) • Wearable devices
Analog Cellular Technology Macro cellular Systems	• Digital Cellular Technology • Microcellular and Pico cellular: Capacity, quality Enhanced Cordless Technology	• Broader Bandwidth CDMA Radio Transmission • Information Compression Higher Frequency Spectrum Utilization • Network Management integration & IP technology	• Unified IP and seamless combination of Broadband hot spots • WLAN/LAN /PAN • 2G/3G + 802.11 • Knowledge-Based Network Operations

Table 2.2 : *Wireless Network and Service Evolution.*

INTRODUCTION OF 4G MOBILE COMMUNICATION.

A 4G or 4th generation network, a new generation of wireless is intended to complement and replace the 3G systems. Accessing information anywhere, anytime, with a seamless connection to a wide range of information and services, and receiving a large volume of information, data, pictures, video, and so on. The future 4G infrastructure will consist of a set of various networks using IP as a common protocol so that users are in control because they will be able to choose every application and environment[2]. A 4G or 4th generation network is the name given to an IP based mobile system that provides access through a collection of radio interfaces. A 4G network promises seamless roaming/handover and best connected service, combining multiple radio access interfaces (such as WLAN, Bluetooth, GPRS) into a single network that subscribers may use . With this feature, users will have access to different services, increased coverage, the convenience of a single device, one bill with reduced total access cost, and more reliable wireless access even with the failure or loss of one or more networks.

4G was simply an initiative by R & D labs to move beyond the limitations, and address the problems of 3G which was having trouble meeting its promised performance and throughput. In the most general level, 4G architecture includes three basic areas of connectivity: Personal Area Networking (such as Bluetooth), local high-speed access points on the network including wireless LAN technologies, and cellular connectivity. 4G calls for a wide range of mobile devices that support global roaming. Each device will be able to interact with Internet-based information that will be modified on the fly for the network being used by the device at that moment.

The roots of 4G lie in the idea of pervasive computing . The glue for all this is likely to be software defined radio (SDR) . SDR enables devices such as cell phones, PDAs, PCs and a whole range of other devices to scan the airwaves for the best possible method of connectivity, at the best price. In an SDR environment, functions that are formerly carried out solely in hardware – such as the generation of the transmitted radio signal and the tuning of the received radio signal – are performed by software. Thus, the radio is programmable and able to transmit and receive over a wide range of frequencies while emulating virtually any desired transmission format. As the number of wireless subscribers rapidly increases guaranteeing the quality of services anytime, anywhere, and by any-media becomes indispensable. These services require various networks to be integrated into IP-based networks, which further require a seamless vertical handoff to 4th generation wireless networks [6]. And as one of the next generation mobile communications, the 4th generation mobile communications provides various services, such as high-speed data services and IP-based access to Radio Access Network, etc. Various interface techniques such as WLAN, Bluetooth, UTMS, and CDMA2000 are integrated into the IP-based networks as an overlay structure. In this structure, the optimum services are provided to mobile hosts. Mobile hosts in this structure can be connected to the network through various access points. Moreover, a seamless handoff should also be supported between different air interface techniques during internetwork movement.

3.1 FEATURES OF 4G

The goal of 4G is to 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. This will provide uniform video, voice, and data services to the mobile host, based entirely in IP.

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 [4].

An IP wireless network replaces the old Signaling System SS7, telecommunications protocol, which is considered massively redundant[9]. This is because SS7 signal transmission consumes a larger part of network bandwidth even when there is no signaling traffic for the simple reason that it uses a call setup mechanism to reserve bandwidth, rather time/frequency slots in the radio waves. IP networks, on the other hand, are connectionless and use the slots only when they have data to send. Hence there is optimum usage of the available bandwidth.

Today, wireless communications are heavily biased toward voice, even though studies indicate that growth in wireless data traffic is rising exponentially relative to demand for voice traffic. Because an all IP core layer is easily scalable, it is ideally suited to meet this challenge. The goal was a merged data/voice/multimedia network.

Other feature includes:

High Speed - 4G systems should offer a peak speed of more than 100Mbits per second in stationary mode with an average of 20Mbits per second when traveling.

High Network Capacity – Should be at least 10 times that of 3G systems. This will quicken the download time of a 10-Mbyte file to one second on 4G, from 200 seconds on 3G, enabling high-definition video to stream to phones and create a virtual reality experience on high-resolution handset screens.

Fast/Seamless handover across multiple networks – 4G wireless networks should support global roaming across multiple wireless and mobile networks.

Next-generation multimedia support – The underlying network for 4G must be able to support fast speed volume data transmission at a lower cost than today.

3.2 POSSIBLE ARCHITECTURES FOR 4G NETWORKS

Accessing different mobile and wireless networks is one of the most challenging problems to be faced in the deployment of 4G technology . Figure 3.1 shows three possible architectures [6]:

- Using a multimode device
- An overlay network
- A common access protocol

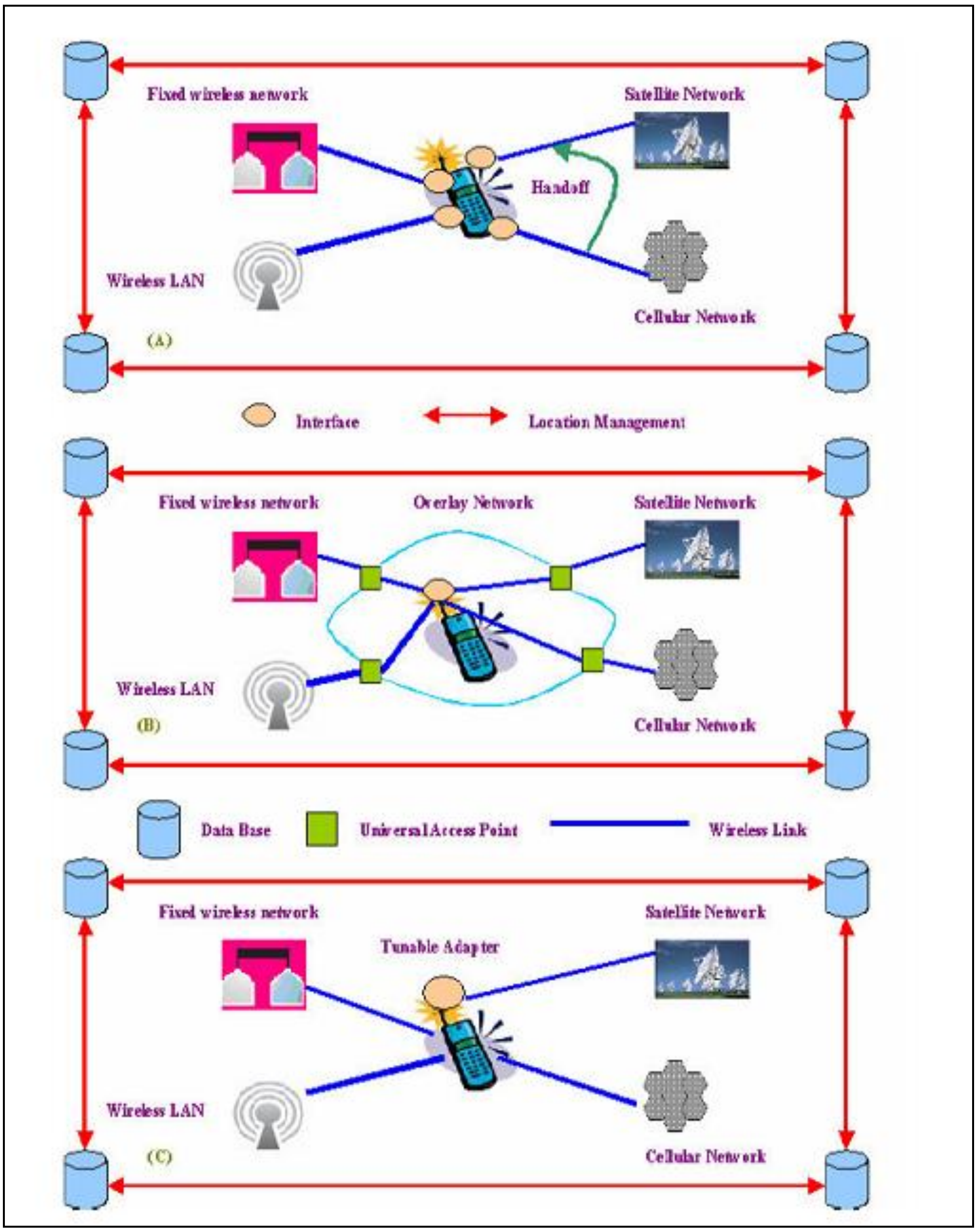

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Figure 3.1: *Architectures for 4G Networks.*

2.2.1 MULTIMODE DEVICES

To access services on different wireless networks, one single physical terminal with multiple interfaces is used. Existing advanced mobile phone system on code division multiple access dual function cell phone, dual function satellite cell phone and global system for mobile telecommunications are examples of Multimode Device architecture. Call completion can be improved and coverage area is expanded effectively using this architecture.

When there is network, link or switch failure, reliable wireless coverage should be provided. The handoff between networks can be initiated by user, device or network. There is no requirement of wireless network modification or employment of interworking devices as the device itself incorporates most of the additional complexity. A database can be deployed by each network which stores the information to keep track of user location, devicecapabilities, network conditions and user preferences.

2.2.2 OVERLAY NETWORK

There are several universal access points in overlay network with which a user can access. A wireless network is selected by each universal access points based on availability, quality of service specifications and user defined choices [8]. Protocol and frequency translation, content adaptation is performed by universal access point on behalf of users.

As the user moves from one universal access point to another, rather than the user or the device, handoffs are performed by overlay networks. User, network, device information, capabilities and preferences are stored by the universal access point. Single billing and subscription is supported as universal access points keep track of the various resources a caller uses.

2.2.3 COMMON ACCESS PROTOCOL

Supporting one or two standard access protocols by wireless networks allows this protocol to become viable. Using wireless asynchronous mode requires interworking between different networks as one possible solution. Transmission of ATM cells with additional headers or wireless ATM cells requiring changes in the wireless networks must be allowed by every wireless network to implement wireless ATM. One protocol might be used by one or more types of satellite based networks while another protocol is used by one or more terrestrial wireless networks.

3.3 CHALLENGES IN 4G NETWORKS

4G Networks are all IP based heterogeneous networks that allow users to use any system at anytime and anywhere. Users carrying any integrated terminal can use a wide range of applications provided by multiple wireless networks. 4G systems provide not only telecommunications services, but also a data-rate service when good system reliability is provided. At the same time, a low per-bit transmission cost is maintained[5].

Users can use multiple services from any provider at the same time. Imagine a 4G mobile user who is looking for information on movies shown in nearby cinemas. The mobile may simultaneously connect to different wireless systems. These wireless systems may include Global Positioning System (GPS) (for tracking users current location), a wireless LAN (for receiving previews of the movies in nearby cinemas), and a code-division multiple access (for making a telephone call to one of the cinemas). In this example, the user is actually using multiple wireless services that differ in quality of service (QoS) levels , security policies, device settings, charging methods, and applications[10]. There are number of challenges faced by 4G networks in integrating all the services.

The challenges mentioned above are grouped into three different aspects:

- Mobile Station
- System
- Service

Mobile Station

To use large variety of services and wireless networks in 4G systems, multimode user terminals are essential as they can adapt to different wireless networks by reconfigure- ring themselves. The need to use multiple terminals is eliminated. Adapting software radio approach is the most promising way of implementing multimode user terminals . The analog part of the receiver consists of an antenna, a bandpass filter, and a low noise amplifier. The received analog signal is digitized by the analog/digital converter immediately after the analog processing. The processing in the next stage is then performed by a reprogrammable baseband digital signal processor. The digital signal processor will process the digitized signal in accordance with the wireless environment.

Unfortunately, the current software radio technology is not completely feasible for all the different wireless networks due to the following technological problems. It is impossible to have one antenna and one low noise amplifier to serve the wide range of frequency bands. Using multiple analog parts to work in different frequency bands is the only solution. The design complexity and physical size of a terminal are increased. And existing analog/digital converters are not fast enough.

System

For 4G infrastructure to provide wireless services at any time and anywhere, terminal mobility is a must. Terminal mobility allows mobile clients to roam across geographic boundaries of wireless networks. The two main issues in terminal mobility are location management and handoff management. The system tracks and locates a mobile terminal for possible connection. Location management involves handling all the information about the roaming terminals, such as original and current located cells, authentication information, and QoS capabilities. Handoff Management maintains ongoing communication when the terminal roams. Mobile IPv6 is a standardized IP-based mobility protocol for IPv6 wireless systems.

Each terminal has an IPv6 home address. Whenever the terminal moves outside the local network, the home address becomes invalid, and the terminal obtains a new

IPv6 address called care-of address in the visited network. A binding between the terminal's home address and care-of address is updated to its home agent in order to support continuous communications. This kind of handoff process causes an increase in system load, high handover latency, and packet losses. It is hard to decide the correct handoff time because measuring handoffs among different wireless systems is very complicated. The uncertain handoff completion time adds to the complexity in designing good handoff mechanisms.

Services

More comprehensive billing and accounting systems are needed, with the increase of service varieties in 4G systems. Customers may subscribe to many services from a number of service providers at the same time rather than only one operator. Dealing with multiple service providers might be inconvenient for customers. Operators need to design new business architecture, accounting processes, and accounting data maintenance. It is challenging to formulate one single billing method that covers all the billing schemes involved. 4G networks support multimedia communications, which consists of different media components with possibly different charging units. This adds difficulty to the task of designing a good charging scheme for all customers.

WIRELESS STANDARDS

4.1 GSM OVERVIEW

Before GSM networks there were public mobile radio networks (cellular).They normally used analog technologies, which varied from country to country and from manufacturer to another. These analog networks did not comply with any uniform standard. There was no way to use a single mobile phone from one country to another. The speech quality in most networks was not satisfactory.GSM became popular very quickly because it provided improved speech quality and, through a uniform international standard, made it possible to use a single telephone number and mobile unit around the world. The European Telecommunications Standardization Institute (ETSI) adopted the GSM standard in 1991, and GSM is now used in 135 countries.The benefits of GSM include:

- Support for international roaming.
- Distinction between user and device identification
- Excellent speech quality
- Wide range of services
- Interworking (e.g. with ISDN, DECT)
- Extensive security features

A GSM network can be divided into three groups (Fig. 4.1): The mobile station (MS), the base station subsystem (BSS) and the network subsystem.They are characterized as follows:

The mobile station (MS):

A mobile station may be referred to as a handset, a mobile, a portable terminal or mobile equipment (ME). It also includes a subscriber identity module (SIM) that is normally removable and comes in two sizes. Each SIM card has a unique identification number called IMSI (international mobile subscriber identity). In addition, each MS is assigned a unique hardware identification called IMEI (international mobile equipment identity).

In some of the newer applications (data communications in particular), an MS can also be a terminal that acts as a GSM interface, e.g. for a laptop computer. In this new application the MS does not look like a normal GSM telephone. Besides providing a transceiver (TRX) for transmission and reception of voice and data, the mobile also performs a number of very demanding tasks such as authentication, handover, encoding and channel encoding.

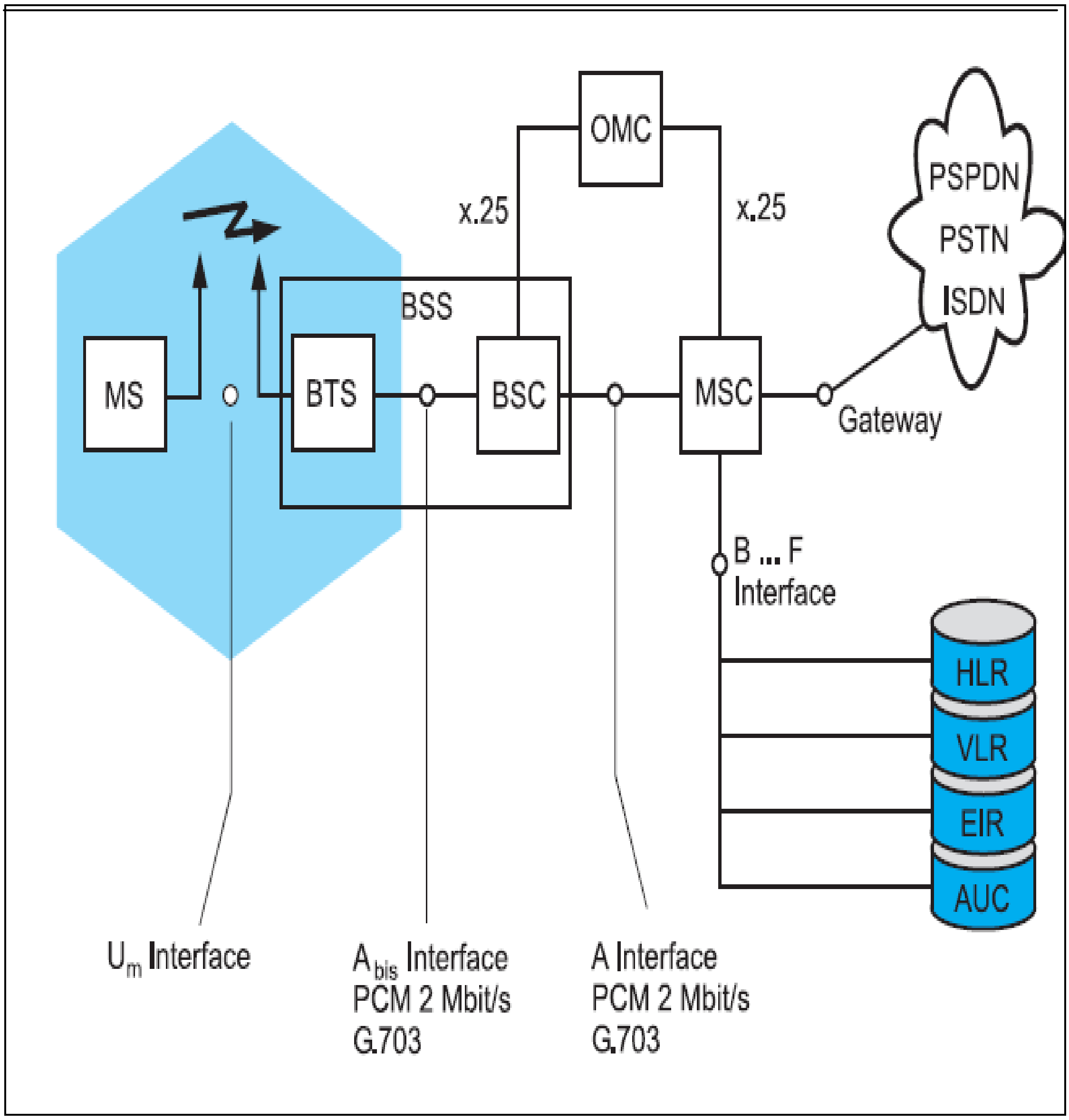

Figure 4.1: *GSM system architecture.*

The base station subsystem (BSS): The base station subsystem (BSS) is made up of the base station controller (BSC) and the base transceiver station (BTS).

The **base transceiver station (BTS)**: GSM uses a series of radio transmitters called BTSs to connect the mobiles to a cellular network. Their tasks include channel coding/decoding and encryption/decryption. A BTS is comprised of radio transmitters and receivers, antennas, the interface to the PCM facility, etc. The BTS may contain one or more transceivers to provide the required call handling capacity. A cell site may be omnidirectional or split into typically three directional cells.

The **base station controller (BSC):** A group of BTSs are connected to a particular BSC which manages the radio resources for them. Today's new and intelligent BTSs have taken over many tasks that were previously handled by the BSCs. The primary function of the BSC is call maintenance. The mobile stations normally send a report of their received signal strength to the BSC every 480 ms. With this information the BSC decides to initiate handovers to other cells, change the BTS transmitter power, etc.

The network subsystem:

The **mobile switching center (MSC):** Acts like a standard exchange in a fixed network and additionally provides all the functionality needed to handle a mobile subscriber. The main functions are registration, authentication, location updating, handovers and call routing to a roaming subscriber. The signaling between functional entities (registers) in the network subsystem uses Signaling System 7 (SS7). If the MSC also has a gateway function for communicating with other networks, it is called Gateway MSC (GMSC).

The **homelocation register (HLR):** A database used for management of mobile subscribers. It stores the international mobile subscriber identity (IMSI), mobile station ISDN number (MSISDN) and current visitor location register (VLR) address. The main information stored there concerns the location of each mobile station in order to be able to route calls to the mobile subscribers managed by each HLR. The HLR also maintains the services associated with eachMS.OneHLRcan serve several MSCs. .

The **visitor location register (VLR):** Contains the current location of the MS and selected administrative information from the HLR, necessary for call control and provision of the subscribed services, for each mobile currently located in the geographical area controlled by the VLR. A VLR is connected to one MSC and is normally integrated into the MSC's hardware.

The **authentication center (AuC):** A protected database that holds a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and encryption over the radio channel. The AuC provides additional security against fraud. It is normally located close to each HLR within a GSM network.

The **equipment identity register (EIR):** The EIR is a database that contains a list of all valid mobile station equipment within the network, where each mobile station is identified by its international mobile equipment identity (IMEI). The EIR has three databases:

- White list: for all known, good IMEIs
- Black list: for bad or stolen handsets
- Grey list: for handsets/IMEIs that are uncertain

Operation and Maintenance Center (OMC):

The OMC is a management system that oversees the GSM functional blocks. The OMC assists the network operator in maintaining satisfactory operation of the GSM network. Hardware redundancy and intelligent error detection mechanisms help prevent network down-time. The OMC is responsible for controlling and maintaining the MSC, BSC and BTS. It can be in charge of an entire public land mobile network (PLMN) or just some parts of the PLMN.

4.2 GPRS OVERVIEW

The general packet radio service (GPRS) provides wireless packet transfer over a wide geographic area. GPRS is a service designed for the existing global system mobile communication (GSM) network. GPRS introduces two new network nodes: The Service GPRS Support Node (SGSN) keeps track of the mobile phones location and performs security functions and access control. The Gateway GPRS Support Node (GGSN) is a gateway in GSM that allow mobile phones to access the public data network (PDN) or private IP networks.

Data is transferred transparently between the mobile phone and the external data networks with encapsulation and tunnelling. The connection between the SGSN and the GGSN is enabled through a protocol named GPRS Tunnelling Protocol (GTP). Data packets are equipped with GPRS-specific protocol information and are transferred between a mobile phone and a GGSN. The connection between the GGSN and the PDN is enabled through the Internet Protocol (IP).

To be able to connect to the Internet the mobile phone has to be assigned an IP address through the GPRS connection. The GGSN uses the Dynamic Host Configuration Protocol (DHCP) to assign an IP address to a mobile phone.

GPRS typically assigns private IP-addresses to mobile phones and uses Network Address Translator (NAT) for mapping a private IP-address to a public at the GGSN. So it is not possible to reach the mobile phone from the Internet through a public IP-address. The mobile phone has to initiate the communication if a host from the Internet will be able to reach the mobile phone. A mobile phone dials up with a point-to-point (PPP) functionality. Figure 4.2 shows an overview over the GPRS entities. Figure 4.2 shows the protocol architecture from a mobile node to the IP network, Internet.

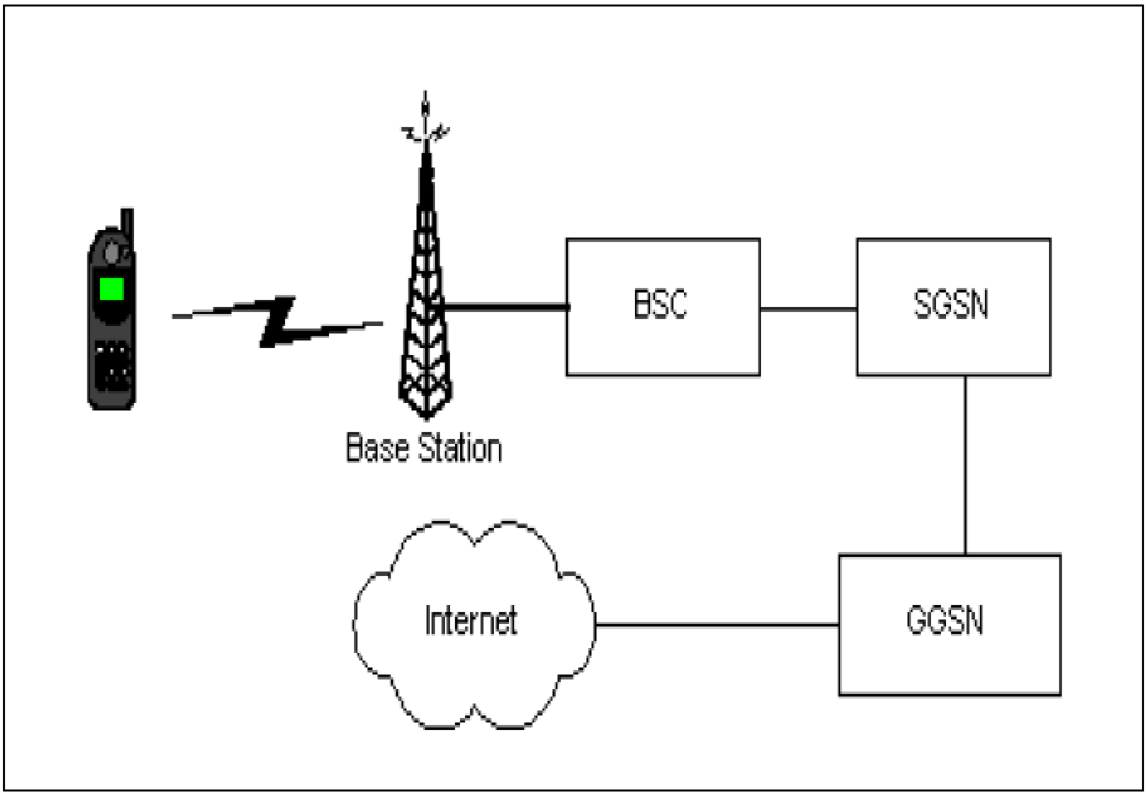

Figure 4.2: *GPRS overview.*

4.3 CDMA OVERVIEW

Heightened growth in the demand for Internet access and services has paralleled the explosion in demand for mobile communications. Users want access to the Internet while they are away from their homes and offices.

CDMA2000 1X supports high speed packet data services and allows data and voice communications to be sent and received over the existing CDMA communications network. CDMA2000 1X provides a maximum over-the-air data rate of 153.6kbps which is up to sixteen times that of a CDMA IS-95A circuit-switched connection, and provides the efficiency of packet data so that more users can be connected simultaneously. CDMA uses unique coding for each call or data session which allows the mobile device to disregard other transmissions on the same frequency. In effect

CDMA lets everybody in the area use the same piece of spectrum, and separates the calls by encoding each one uniquely.

CDMA data transmission is analogous to an international cocktail party—dozens of people are in the room, all talking at once, and all talking in different languages that you don't understand. Suddenly, from across the room, you hear a voice speaking in your own familiar language, and your brain tunes out all the background gibberish and locks onto to that one person. Your brain understands the "code" being used by the other person, and vice versa.

What is CDMA2000?

CDMA2000 is the name used by the Telecommunications Industry Association to refer to Third Generation CDMA (3G). The Third Generation of mobile communications (3G) is a concept outlined in a set of proposals called International Mobile Telecommunications-2000 (IMT2000), to define an "anywhere, any time" standard for the future of universal personal communications. The CDMA path to 3G is through CDMA2000 1X, then onto CDMA2000 1xEV-DO and 1xEV-DV.

What is CDMA2000 1X?

CDMA2000 1X enables operators with existing IS-95A systems to double overall system capacity, and increase data rates to as much as 614kbps. The first implementation of CDMA2000 1X is expected to yield data rates up to 153.6 kbps.

Most applications like email or short messaging only need to send or receive data for short periods of time. By using the airwave resource on an as-packetsare- needed basis rather than a dedicated circuit for one user at a time, many users may share a space with coded packets within the channel.

This results in more efficient usage of the radio frequency, in addition to the fast peak data rates. Actual pricing plans for data usage may vary from operator to operator; however, it is anticipated that the cost for data will be based on the amount of data sent and received, not on the length of the user’s session.

4.4 WLAN (IEEE 802.11) OVERVIEW

A Wireless LAN (WLAN) is a data transmission system designed to provide location independent network access between computing devices by using radio waves rather than a cable infrastructure. In the corporate enterprise, wireless LANs are usually implemented as the final link between the existing wired network and a group of client computers, giving these users wireless access to the full resources and services of the corporate network across a building or campus setting.

Wireless LANs have gained strong popularity in a number of vertical markets, including the health-care, retail,manufacturing, warehousing, and academia. These industries have profited from the productivity gains of using hand-held terminals and notebook computers to transmit real-time information to centralized hosts for processing. Today wireless LANs are becoming more widely recognized as a general-purpose connectivity alternative for a broad range of business customers.

Wireless local area networks (WLAN) provides a high bandwidth service over a narrow geographic area and are typically restricted in their diameter to buildings, a campus or single rooms. The IEEE standard 802.11 specifies the most common family of WLANs. There are two types of WLANs, infrastructure and ad-hoc. Infrastructure networks often provide access to other networks such as Internet.

Communication typically only takes place between the wireless nodes and an access point (AP). The stations and the access point that are within the same radio coverage form a basic service set (BSS). Several BSS:es may form one logical wireless network called extended service set (ESS) and is identified by a name (ESSID). So it is possible to reach Internet through a WLAN with a wireless node where the node is located within the radio coverage for the WLAN.

An overview of an infrastructure WLAN (IEEE 802.11) that is bridged to the Internet is shown in figure 4.3. Ad-hoc networks don't need any infrastructure to work. Each wireless node can communicate directly with other nodes, so no access points are needed. The complexity of each node in an ad-hoc network is much higher than in an infrastructure. The IEEE 802.11 standard defines the physical and medium access control (MAC) layer. The 802.11 link layer is transparent to the IP layer together with upper part of the link layer called logical link control (LLC) . The LLC layer provides an interface to the IP layer and covers the differences of the medium access control layers needed for the different media.

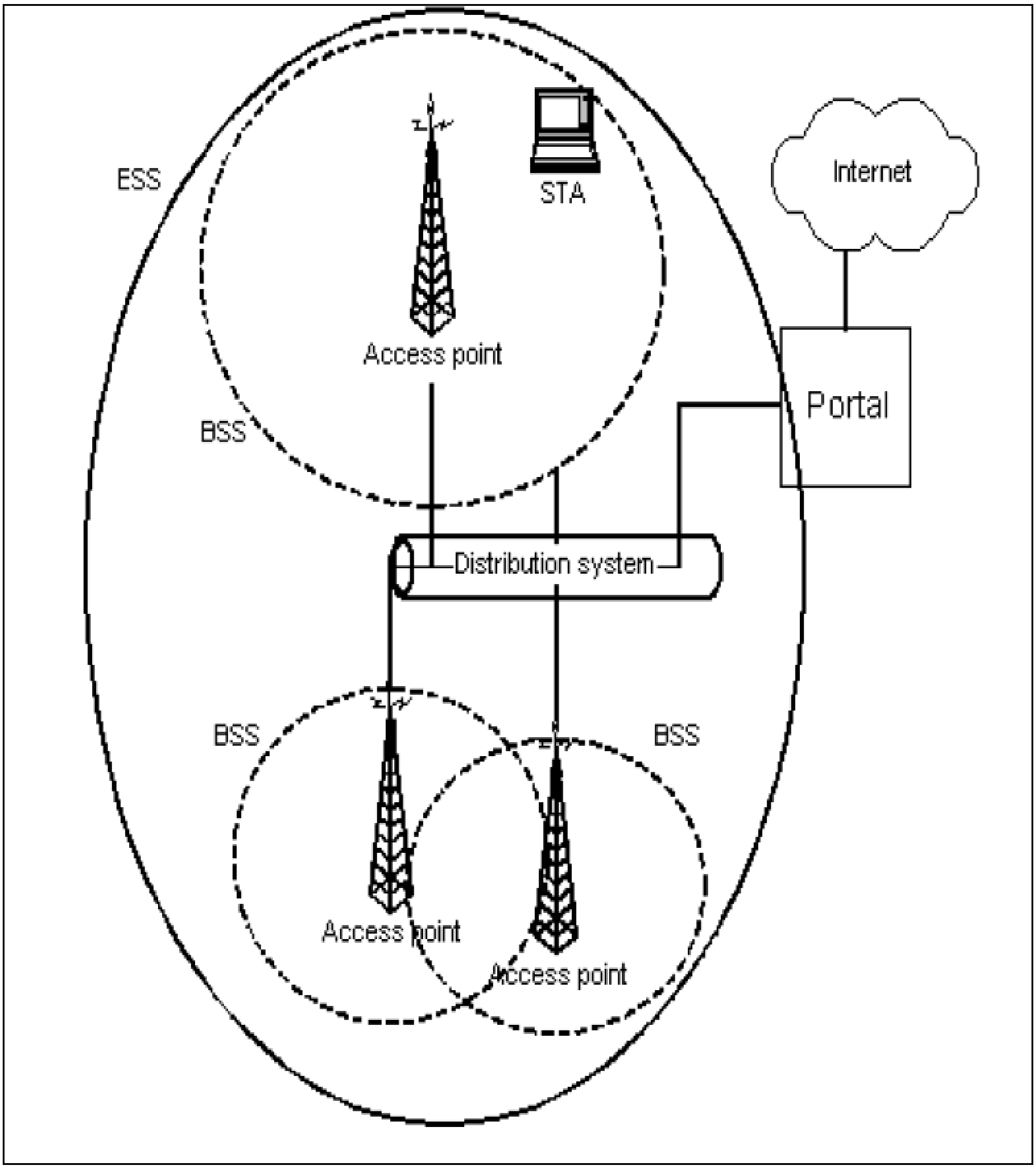

Figure 4.3: *Wireless LAN (IEEE 802.11) overview.*

HANDOFF IN NETWORKS

The services provided by the public switched telephone networks (PSTN) [23] are leveraged by wireless mobile telephone network of public land mobile networks (PLMN). PSTNs ate backbones to PLMNs. Infrastructure for wireless access, mobility management and external network gateways are provided by the network elements of PLMNs.

A simple PLMN [23] consists of the following components:

- Base stations
- Mobile switching service centres (MSC)
- Home Location Register (HLR)
- Visitor Location Registers (VLR)
- Authentication Centre (AUC)
- Equipment Identification Register (EIR).

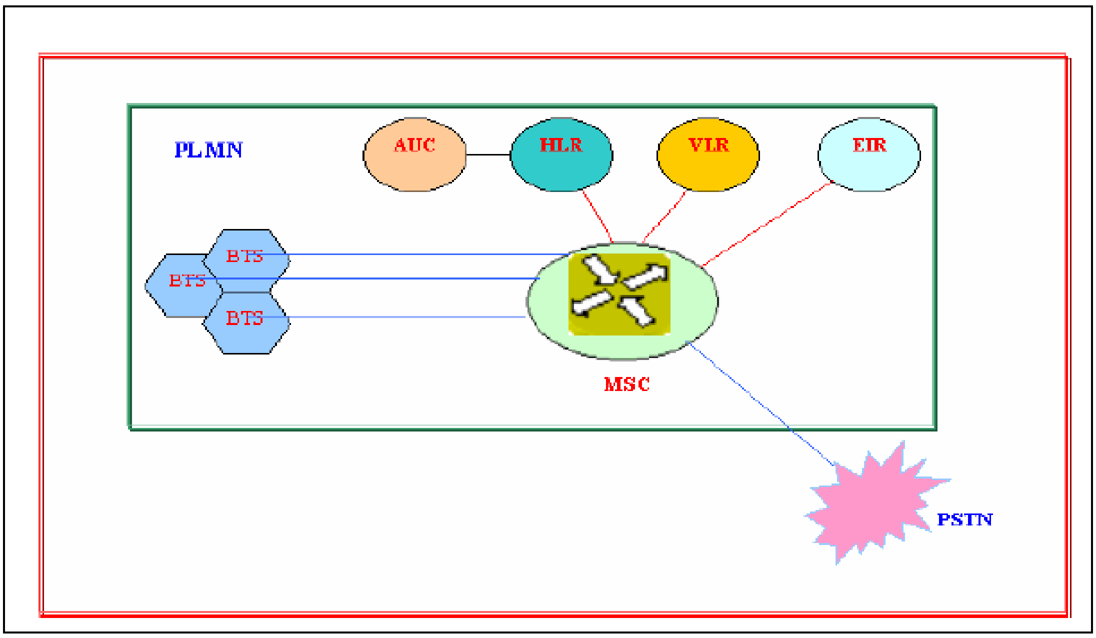

Figure 5.1: *Simple PLMN.*

Radio interface for mobile subscribers are used to provide network access by the base stations. Managing base stations, consulting PLMN database to establish subscriber access rights, routing mobile traffic is managed by MSC. MSC also serves as a gateway to external networks. Subscriber profiles, location encryption codes and equipment data are stored in PLMN databases. HLR, VLR, AVC and EIR are PLMN databases.

All telephone networks require fundamental services like Call establishment and connection maintenance. Call initiation signalling, connection path establishment, alerting called party, call acceptance and preservation of connection until end of session signalling is detected consists of the PSTN call process for two authorized fixed location subscribers. The PSTN uses the fixed location of the subscriber to simplify network functions. Authentication, call establishment and call preservation are simplified by fixed subscriber location.

Subscriber mobility significantly complicates network operations, although a PLMN call process performs the same functions of a PSTN. PLMNs must implement mobility management technologies to provide PSTN services. These technologies enable PLMNs to establish and maintain calls to authorized mobile subscriber. Mobility Management uses the HLR, VLRs, MSCs and Base Stations.

Call quality, reliability and availability are strongly influenced by Mobility Management technologies. Mobility Management is the ability of a PLMN to orchestrate calls for its subscribers and radio management maintains the call regardless of the mobility of the subscribers. PLMNs must track and dynamically route calls to its subscribers in a transparent fashion. The main functions of mobility management are locating, authenticating and tracking mobile subscribers. PLMNs use a registration process to report a mobile station’s right to access network services. Roaming allows authorized mobile subscribers to use networks other than their home PLMN. Signal quality assessments, base station selection and switching constitute Radio Resource Management (RRM).

5.1 HANDOFF CRITERIA

Increasing distance from the base station attenuates the radio signal as a subscriber travels away from its base station[11]. Prior to loss of communication, reliable detection of this condition is crucial. When deteriorating signal strength is detected the PLMN responds by seeking an alternative base station with higher signal strength. The PLMN moves the call to the new base station and releases the previous base station after selecting and reserving a new base station channel.

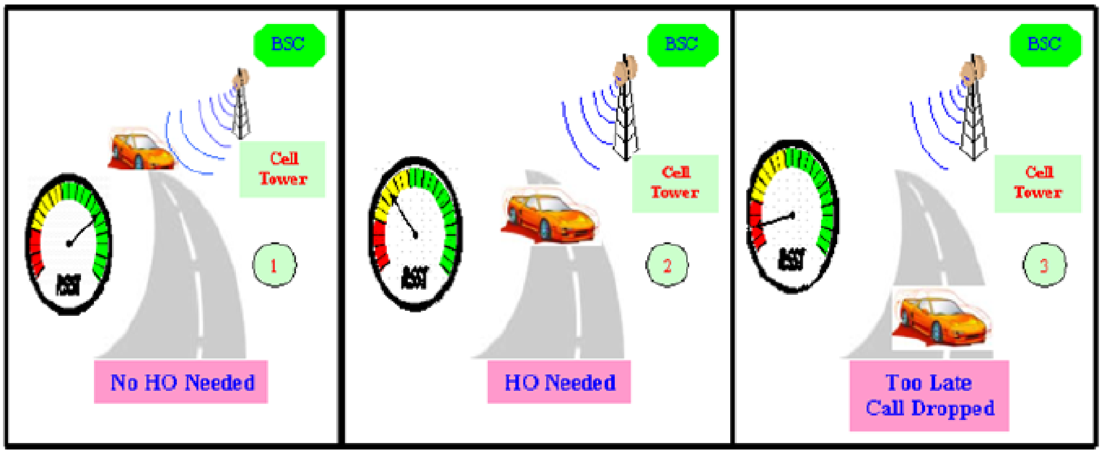

Figure 5.2: *Signal Quality Behaviors.*

Received signal strength is described by Received Signal Strength Indicator (RSSI) and handoff process relies on this signal strength. As the characteristics of base station and mobile handset receivers are well known, it is possible to predict performance ranges based on received signal levels. The crucial factor for PLMN's radio resource management is accurate and reliable signal quality assessments between the mobile station and its serving base station.

Real time measurements effect PLMN's rapid execution of handoffs. Another key parameter for radio resource management is the reference point of signal strength measurements. The PLMN can use measurements made at the base station, handset or both as a reference for resource switching decisions.

5.2 HANDOFF SCENARIOS

An event when a mobile station moves from one wireless cell to another is called Handoff[11]. Handoff can be of two types: horizontal (intra-system) and vertical (inter-system) cases. Handoff within the same wireless access network technology is considered as Horizontal handoff, and handoff among heterogeneous wireless access network technologies is considered vertical handoff. As mobiles traverse cell or sector boundaries, majority of handoffs support calls. The following are scenarios where Handoff processes are required:

- Intra-MSC - Involve crossing cell boundaries within a MSC's service area
- Inter-MSC - Involves crossing cell boundaries between MSCs
- Roaming - Involves crossing cell boundaries between different network operators
- Intra-cell – Involves crossing sector boundaries within a cell
- Switching channels to circumvent persistent interference

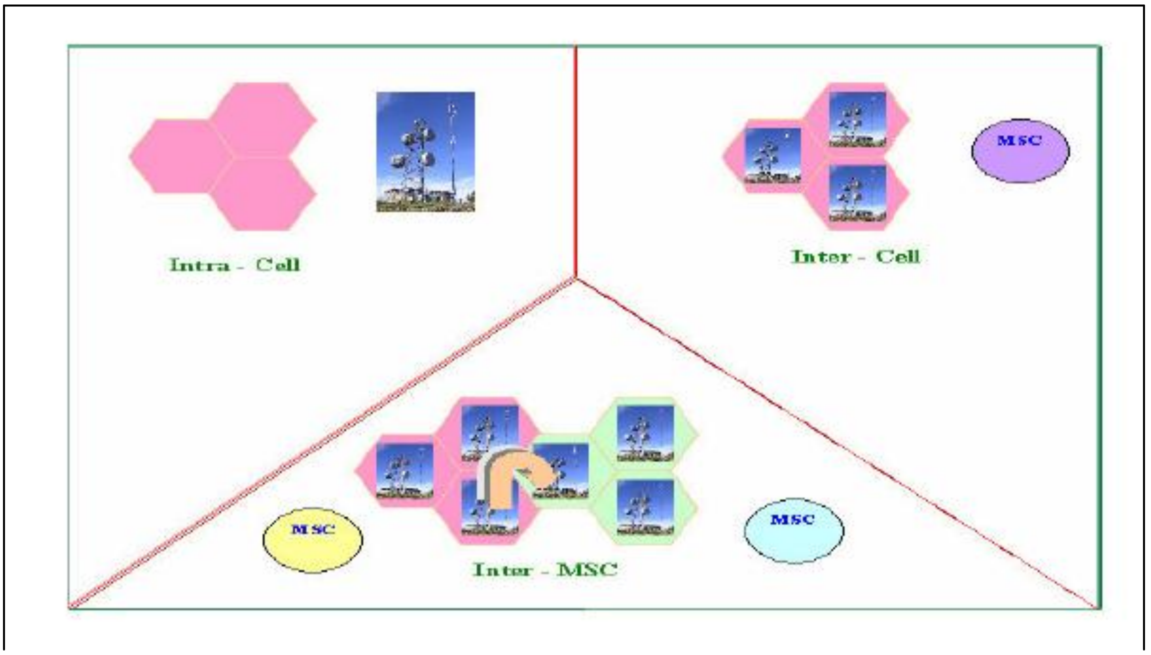

Figure 5.3 : *Basic Handoff Scenarios .*

5.3 HANDOFF METHODS

Handoffs have several methods and they are technology dependent. The two main handoff methods are:

• **Hard Handoff**: It has a brief disruption of service as it has to break before a making a switching action. Hard Handoffs are used by Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) systems.

• **Soft Handoff**: It has no disruption of service action as it makes a switching action before the break. Multiple network resources are used by soft handoffs.Soft handoffs are used by CDMA system.

Figure 5.4 shows these two different handoffs.

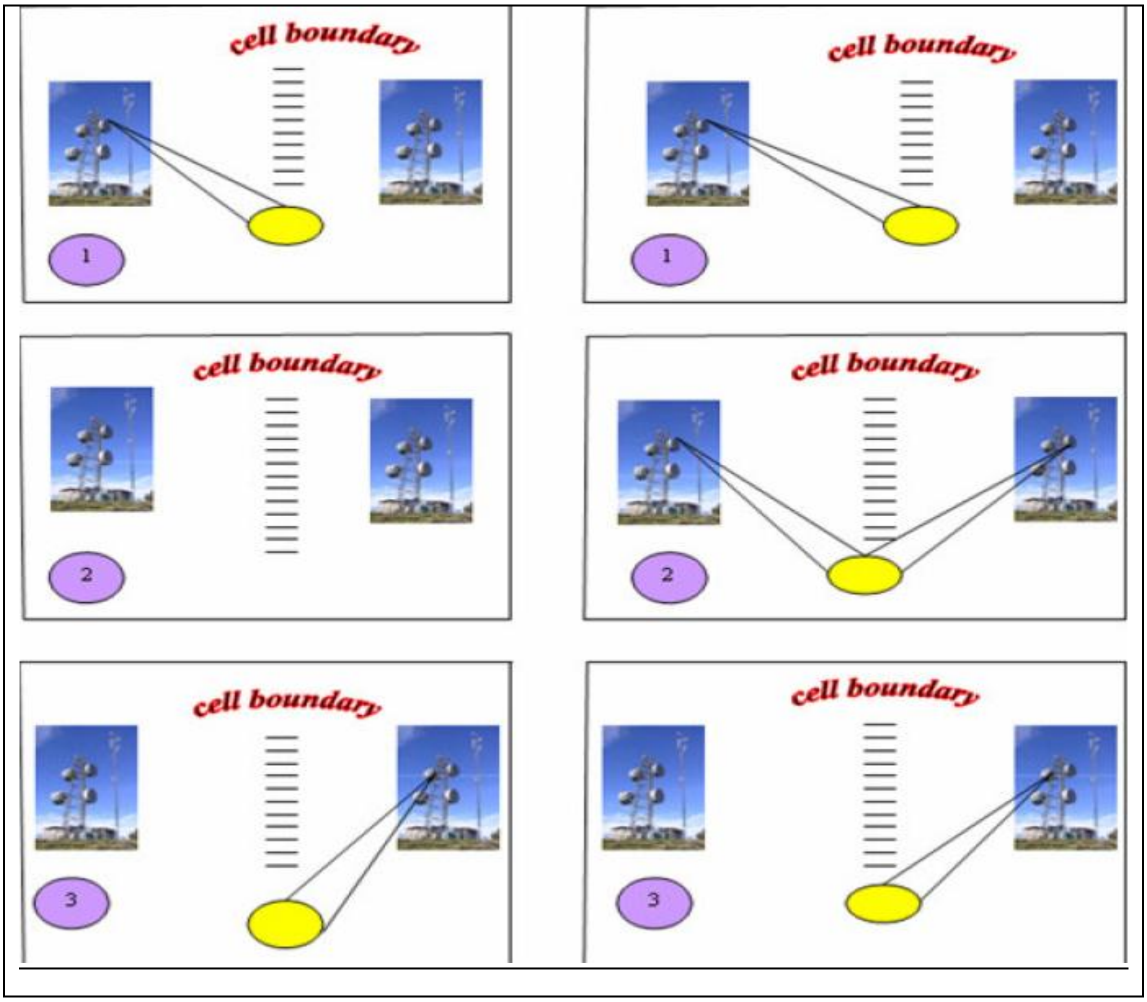

Figure 5.4: *Hard and Soft Handoff Methods .*

Seamless handover:

A seamless handover is a handover scheme that maintains the connectivity of all applications on the mobile device when the handover occurs.

Horizontal handover:

A horizontal handover is a handover between base stations that are using the same kind of wireless network interface.

Vertical handover: Vertical handover is a handover between two network access points using different connection technologies. Verticals handovers can be divided into two categories: upward and downward. An upward handover is a handover to larger cell size and a downward to a smaller cell size[8].

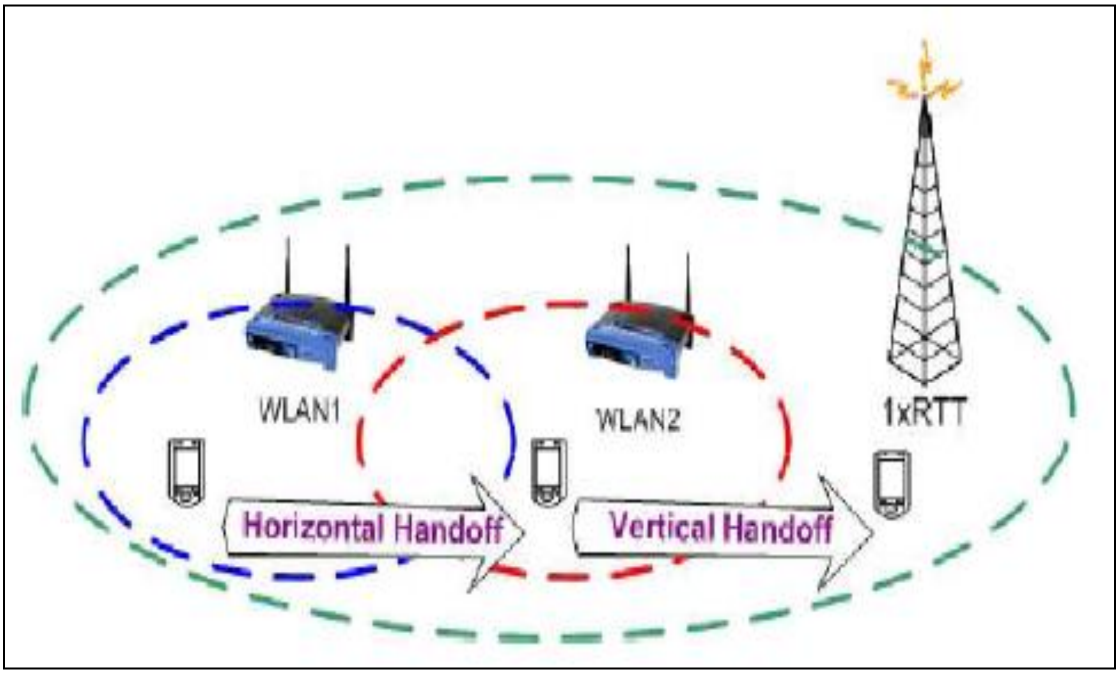

Figure 5.5: *Horizontal and Vertical Handoff Methods.*

DIFFERENT TYPES OF APPROACHES FOR VERTICAL HANDOVER PROBLEM

Future wireless networks must be able to coordinate services within a diverse-network environment. One of the challenging problems for coordination is vertical handoff, which is the decision for a mobile node to handoff between different types of networks.While traditional handoff is based on received signal strength comparisons, vertical handoff must evaluate additional factors, such as monetary cost, offered services, network conditions, and user preferences.

It is essential for modern businesses to have all their employees online including the mobile employees. These employees should be able to access their organisation's communication system with various types of networks and devices. To support connectivity with any device, at any place and with any type of network, handover of disparate networks has emerged as an essential requirement to the industry. As the industry migrates towards an all IP future, it is desirable for subscribers to seamlessly use network services without paying particular attention to its type. What is important is that each subscriber has a multi-mode terminal that supports different bearers such as GPRS, WLAN, 3G.

There are two types of handover, horizontal handover within the same network and vertical handover between networks[8]. The advantages of network handover are many. One of them being the ability of field workers to access the organisation’s Intranet or Internet to receive and update their work schedules, respond to business mails from where ever they are in the field without the restrictions of network and device types.

Seamless vertical handover:

A vertical handover is a handover between different network technologies. Since each network interface on a mobile node has its own IP address a vertical handover has to handle changing of IP addresses. Since a TCP communication session is identified by the tuple (source IP, source Port, destination IP, destination Port) an issue arises. When performing a vertical handover the mobile node has to switch physical IP address. The WLAN network interface and the GPRS network interface cannot use the same IP address. The challenging problem is how to keep the session alive while changing the physical connection interface (IP address). This topic has been studied widely and the solutions can be classified into three categories: Upper-layer approaches, new transport layer approaches and network layer approaches. Network layer approaches are typically based on mobile IP or mobile IPv6. These approaches require implementing agents on Internet for forwarding data. They require infrastructure modifications to work properly. New transport layer approaches require already existing applications to be rewritten. Since most existing applications are using either TCP or UDP as transport protocol they have to be rewritten. Upper-layer approaches implements a new layer above the transport layer. A session layer can make a session exist between two applications instead of between two hosts. Vertical handover solutions have to deal with two main issues: latency and packet loss[15].

6.1 NETWORK LAYER APPROACHES

6.1.1 MOBILE IP :

Mobile IP is an Internet protocol for allowing transparent routing of IP datagram's to mobile nodes in Internet. If one wants to reach a mobile node wherever it is located it has to have a static home IP address. That is achieved with mobile IP. When a mobile node moves and attaches itself to another network, it obtains a new IP address. This is necessary as the IP routing mechanism rely on the topological information embedded in the IP address to deliver the data to the correct end-point. Mobile IP handles this by network agents. No modifications on the routers or end hosts are required. Each mobile node is identified by a static home network address from its home network, regardless of its current point of attachment[12].

Terminology

- **Mobile node (MN):** A mobile node with a static IP address. The mobile node can change its point of attachment to the Internet using mobile IP.

- **Correspondent node (CN):** Mobile nodes communication partner. The correspondent node can be fixed or a mobile node.

- **Home network:** The subnet the mobile nodes home IP address is belonging to.

- **Foreign network:** The current subnet the mobile node is visiting.

- **Home Agent (HA):** Is located in the home network. The home agent can be implemented on the router at the home network or at a regular node in the home network. The home agent can work as manager for the mobile node. With the manager solution the mobile node is always in a foreign network. When a mobile node is outside the home network the home agent receives all packets destined to the mobile node and tunnels them to the current location of the mobile node.

- **Foreign Agent (FA):** The foreign agent provides services to the mobile node during its visit to the foreign network. The foreign agent acts as the tunnel end-point, decapsulates incoming packets and forwards them to the mobile node. The foreign agent is typically implemented on the router at the foreign network. A foreign agent is not necessary needed; if the foreign agent is discarded the mobile node has to decapsulate the incoming packets itself. The mobile node is then co-located.

- **Care-of address (COA):** The care-of address defines the current IP address of the mobile node. All packets sent to the mobile node are sent to the home agent and tunneled to the care-of address. The care-of address is the tunnel endpoint. The care-of address can be located at two different points, at the foreign agent or at the mobile node directly. If the care-of address is located at the mobile node directly, the mobile node is then co-located.

- **Tunnel:** The path followed by a datagram while it is encapsulated. The model is that, while it is encapsulated, a datagram is routed to a knowledgeable decapsulating agent, which decapsulates the datagram and then correctly deliver it to its ultimate destination.

While a mobile node is away from its home network, it updates the home agent with information about its current IP address. The home agent receives all incoming packets destined to the mobile node, encapsulates and tunnels them to the mobile nodes current IP address (COA). When the mobile node wants to send packets to the correspondent node two options is available.

The simpler way is to send the IP packet to the correspondent node with the mobile nodes home address as source address instead of the care-of address. But there are some problems with this option, many intranets only allow packets with topologically correct addresses to pass. Since the source address of the IP packet is changed, the address will not be topologically correct.

The other alternative is to use reverse tunneling, when the mobile node wants to send a packet to the correspondent node it encapsulate and tunnels the packet to the home agent. The home agent then decapsulates the packet and forwards it to the correspondent node as when a packet is sent from the correspondent node to the mobile node. This is called reverse tunneling.

Figure 6.1 shows the scenario when the mobile node is located in a foreign network and communicates via a foreign agent. The home agent and the foreign agent are acting as tunnel endpoints. When a mobile node has found a subnet, either the home or a foreign network, it has to register at the agent. If the mobile node is in a foreign network the registration can be done in two ways, through a foreign agent or directly to the home agent.

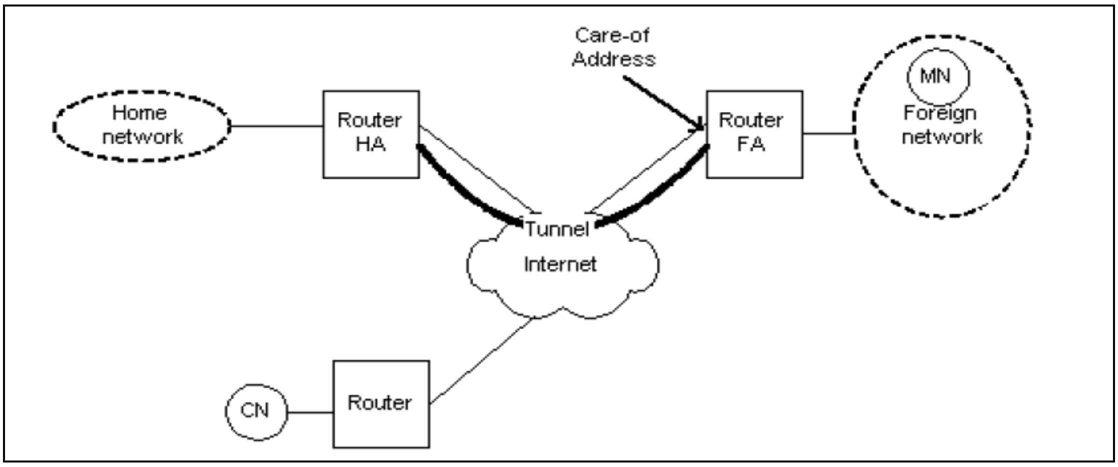

Figure 6.1: *Mobile IP overview.*

If the mobile node is using a foreign agent the registration goes through the foreign agent. The mobile node sends a registration request containing the care-of address to the foreign agent, which forwards the request to the home agent. The home agent sets up tunnel from the home agent to the foreign agent. The registration expires after negotiated lifetime, this is for avoiding mobility bindings which are no longer used.

The home agent sends a reply message to the mobile node through the foreign agent after setting up the tunnel. If the mobile node not is using a foreign agent the registration message is sent directly to the home agent. All registration packets are sent using UDP as transport protocol. Figure 6.2 shows both register cases.

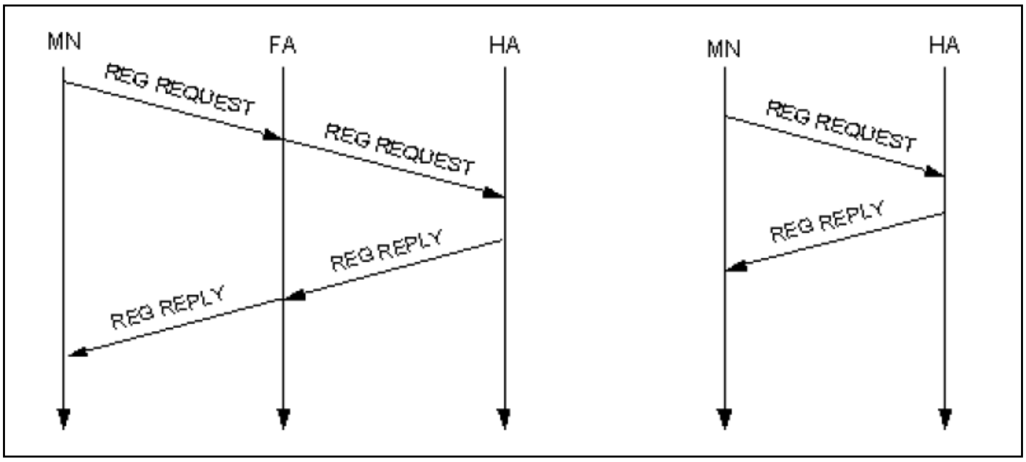

Figure 6.2: *Registration of a mobile node via the FA or directly with the HA.*

6.1.2 MOBILITY SUPPORT IN IPV6

IPv6 is a new version of IP and is intended to replace the current version IPv4 as Internet protocol. The length of the address has been increased from 32 bits to 128 bits. In mobile IPv6] as in mobile IPv4 each mobile node is always identified by its home address, regardless of its current point of attachment to the Internet. When the mobile node is away from the home network the mobile node is associated with a care-of address (COA)[12][13].

Each time the mobile nodes move from one subnet to another the node will configure its COA with another COA belonging to the new subnet. The configuration can be done with DHCPv6 or PPPv6. Mobile IPv6 enables any IPv6 node to learn and cache the COA for a mobile node. This is for avoiding triangular routing and the packet is sent using an IPv6 routing header instead of IPv6 encapsulation. A mobile node's association to a COA is known as binding and has a remaining lifetime. This is for other nodes to know how long to store the COA in the binding cache.

When sending an IPv6 packet to any destination, a node checks its binding cache for an entry for the packet's destination address. If an address is found in the cache the packet is sent directly to the COA instead through a home agent where it has to get encapsulated. Mobile IPv6 introduces a set of new messages to achieve this, *Binding Update* and *Binding Acknowledgement*. After a mobile node has configured its COA, it must send a *Binding Update* to the HA and all corresponding nodes. The Binding Message contains the current COA for the mobile node. The recipients updates their binding cache and sends a *Binding Acknowledgement* if so was requested in the *Binding Update* message. The *Binding Update* message can be sent separate or together with any payload such TCP or UDP.

Although messages can be sent directly to the care-of address from a correspondent node to avoid triangular routing the mobile node can always be reached through its home address. So the movement of the mobile node is thus transparent to the transport and higher layers protocols.

6.2 TRANSPORT LAYER APPROACHES

Today most applications are communicating with either TCP or UDP as transport protocol. Introducing a new transport protocol to solve the seamless vertical handover problem has its disadvantages. Unfortunately almost all today's existing applications have to be rewritten to support a new transport protocol. Two proposals for a new transport layer protocol are considered, Stream Control Transmission Protocol (SCTP) and TCP Multi Homing (TCP MH).

6.2.1 STREAM CONTROL TRANSMISSION PROTOCOL (SCTP)

Stream Control Transmission Protocol offers a reliable delivery service for application over an IP network and is session-oriented[19]. The most interesting feature of SCTP is multi-homing. An SCTP session can be established over multiple IP addresses. SCTP sends packets to a primary IP address, but can reroute packets to an alternative, secondary IP address if the primary IP address becomes unreachable. A SCTP session has a primary path between two SCTP hosts, but can also have multiple paths between the hosts. This type of session is defined as an association in SCTP. An SCTP association between two hosts A and B, is defined as:

{[IP addresses of A] + [port A]} + {[IP addresses of B] + [port B]}

In the base version of SCTP the endpoints exchange their IP addresses before the SCTP association is established and these addresses cannot be changed during the session. However mobile SCTP (mSCTP) supports adding, deleting or changing IP addresses during an active session using Address Configuration (ASCONF) messages. SCTP supports the end-to-end principle; anything that can be done in the end system should be done there. Since the transport layer is the lowest end-to-end layer in the Internet protocol stack, the vertical handover should be done there . The end-to-end principle says that anything that can be done in the end system should be done there. Figure 6.3 shows the protocol stack with mSCTP as transport protocol. The mSCTP is transparent to the IP layer.

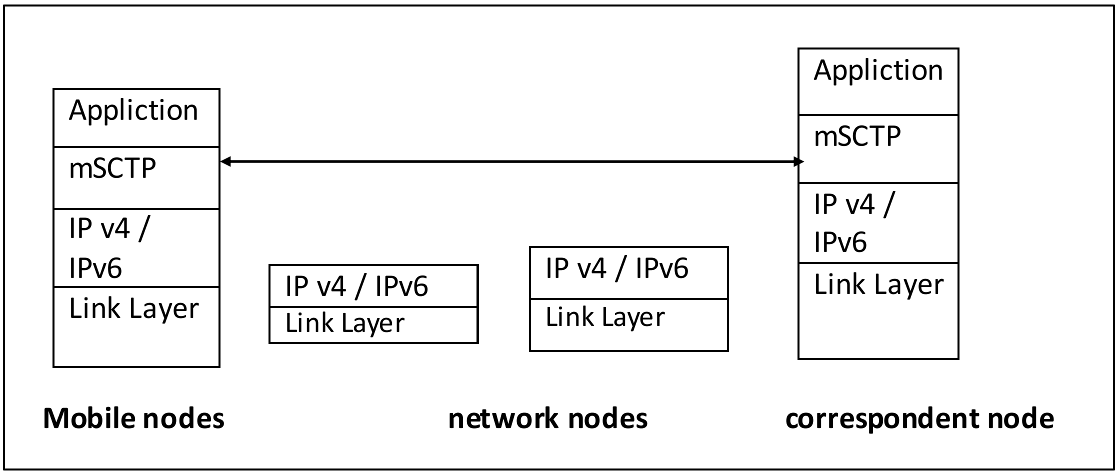

Figure 6.3: *Protocol architecture.*

In figure 6.4 a message procedure is shown. The Mobile node communicates to the correspondent node through the WLAN interface. The mobile node sends an ASCONF message and adds a GPRS IP address. Later the mobile node performs a handover and sends a ASCONF message which switches the primary IP address. Now all data to and from the mobile node are sent through the GPRS interface.

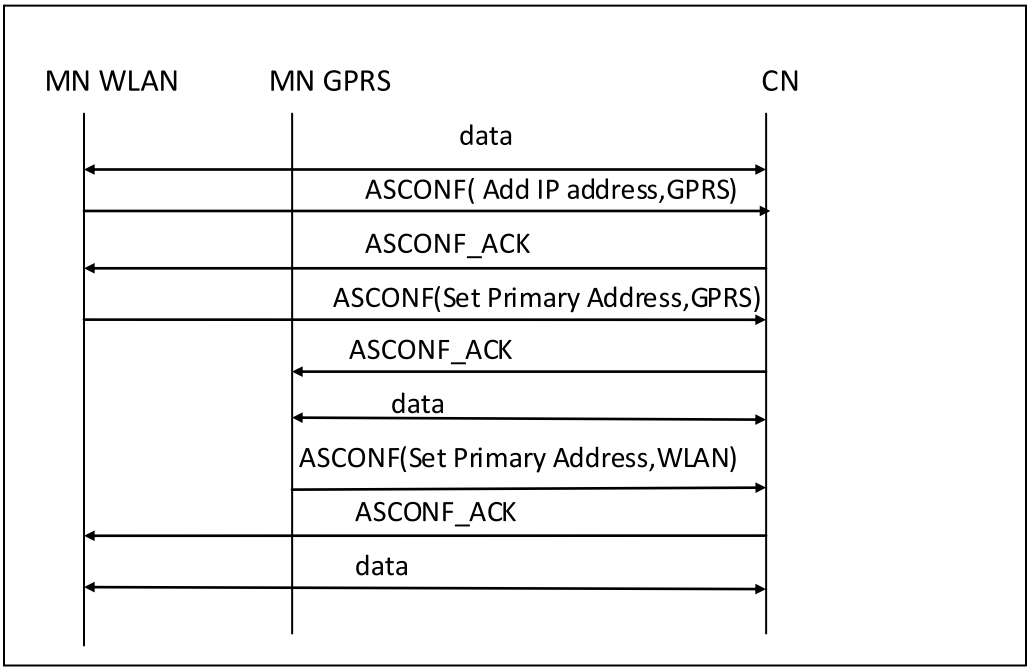

Figure 6.4: Vertical handover procedure with mSCTP

6.2.2 TCP MULTI HOMING (TCP MH)

TCP Multi Homing resembles SCTP but TCP MH is just an extension of the existing TCP, not a complete new transport protocol[18]. The existing TCP is only designed for communication between one local and one remote IP address. The TCP MH option makes it possible to handle multiple local and remote addresses during a TCP session. TCP MH provides multi-home feature to TCP without modification and dependence on any other elements in the Internet.

Features as flow control, slow start, collision avoidance and fast retransmission in TCP are kept in TCP MH. A TCP MH session starts with a MH-Permit option in a SYN packet. If host accepts MH-permit a SYN-ACK with MH-Permit packet is returned and MH options can be used. A TCP session can be kept even though the source and/or the destination address changes. A session can also switch from IPv4 address to IPv6 address and vice versa. Hosts exchange their IP addresses with MH-Add-IPv4 or MH-Add-IPv6 options. After an endpoint has received a MH-Add option the endpoint register the new transmission path.

There are also MH-Delete options for deleting addresses. Figure 6.5 shows a mobile node trying to establish a connection from the WLAN interface, but no response returns from the correspondent node. Later the mobile node switches interface to the GPRS connection and tries to connect with that. The connection is established after an ACK from the correspondent node to the mobile node.

The three- way handshake is finished after the correspondent node receives an ACK from the mobile node. When data is sent the mobile node uses the “MH-Add-IPv4” option, which tells the correspondent node that another IP address for the mobile node (IP for the WLAN interface) can be used. The correspondent node accepts that option and sends back an ACK. Later as seen in the figure the correspondent node sends data that doesn’t reach the mobile node and no ACK was received. The correspondent node then tries to send the data to the other IP address (WLAN) that is available for the session.

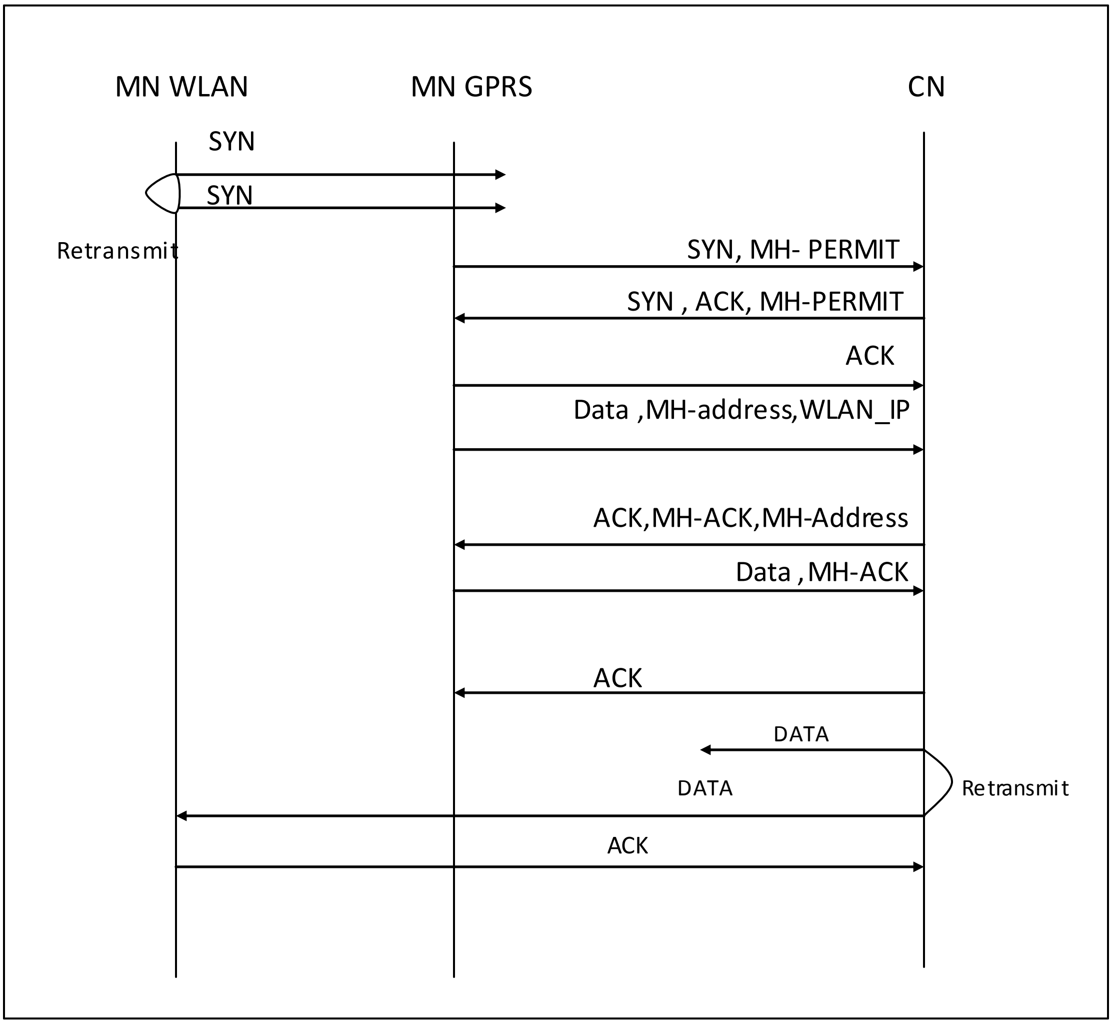

Figure 6.5: *TCP-MH message overview.*

6.3 UPPER-LAYER APPROACHES

These approaches implement a new session layer above the transport layer. The application sessions will be transparent to the connection changes in the underlying layers. A session layer can make a session exist between two applications instead of between two hosts. The session then has to be identified in another way than the IP numbers and port numbers. Some session ID has to identify a session between two applications.

6.3.1 MIGRATE APPROACH

The migrate approach is a session-oriented architecture end-to-end host mobility approach. The migrate approach propose a session layer. Sessions exists between application end points, and should survive changes in the transport and network layer protocol states. Once a session is established a locally unique token or Session ID identifies it. There are five important fundamental issues that has to be handled in Internet mobility.

1. Locate the mobile host: The desired end point must be located and mapped to an addressable destination.
2. Preserving communication: Once a session is established, communication should be able to handle changes in the network location of the end points.
3. Disconnecting gracefully: Disconnection should be rapidly detected.
4. Hibernating efficiently: If a host is unavailable for a period of time, the connection should be suspended and resources should be reallocated.
5. Reconnecting quickly: Communication peers should detect resumption of connectivity in timely manner. The system should be able to re-establish the connection without any extra effort.

Mobility support should be provided at the end hosts. Many previous approaches like mobile IP rely on proxies. Proxy-based solutions have to deal with some performance issues. The proxies have to be well engineered and well located in the network to perform acceptably. application's control. Naming can abstract location details.

The migrate approach provides a naming service, a mobile host isn't bound to a home IP address like in mobile IP. Instead the host is identified by a hostname. To locate mobile node hosts the widely deployed Domain Name System (DNS) is used.

Many applications resolve hostnames to an IP address at the beginning of a connection. No home agent is necessary as in mobile IP. When a mobile node changes its location and IP address, it sends a DNS update to one of the name servers. Since the session is identified with a session ID the session can remain from the new location. The session layer has to re-synchronize the session between the hosts. The session layer has to handle and save the state of the connection to be able to continue the session after a reconnection from another IP address. It is possible to hijack the connection with this solution, a secure key exchange part has to be implemented in the session layer. The same problem arises for the DNS update sent by the mobile node.

The migrate approach uses the security of dynamic DNS updates in RFC 2137 . Figure 6.6 shows the components of the Migrate architecture. The session layer has four interfaces: session establishment, connectivity status, policy decisions and application up-calls.

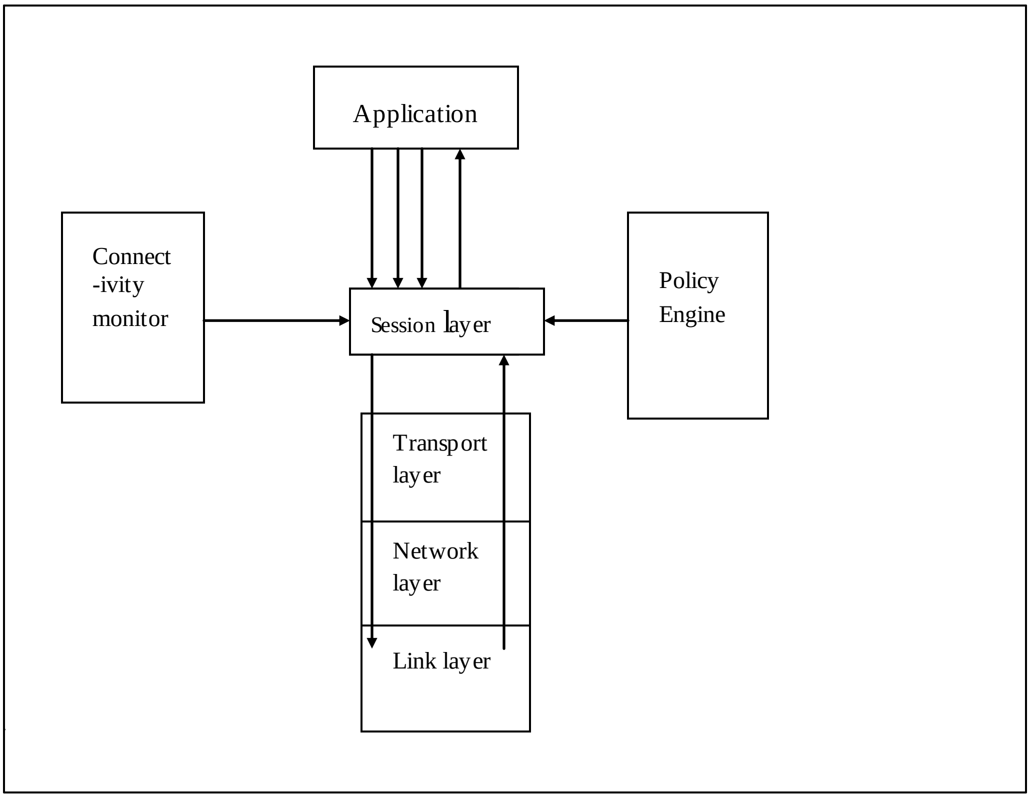

Figure 6.6 : *The Migrate session layer framework.*

6.3.2 MSOCKS

MSOCKS is a proxy-based solution with a proxy inserted into the communication path between a mobile node and its correspondent node[17]. MSOCKS is using a technique called TCP Splice. TCP Splice preserves the end-to-end semantics as normal TCP connection. Normally in proxy-based solutions each session between a mobile node and a correspondent node is split into two separate TCP connections. TCP Splice allows a machine where two independent TCP connections terminate to splice the two connections together. The connection will form a single end-to-end TCP connection between the endpoints of the two original connections with the proxy in the middle. The mobile node is communicating via a MSOCKS library that runs under the application.

At the proxy a MSOCKS proxy process is running; an in-kernel modification on the proxy machine to provide the TCP Splice service. The correspondent host doesn't need any modifications. The MSOCKS protocol is built on top of the SOCKS protocol . The MSOCKS library has similar functions like *bind, accept* and *connect* as SOCKS does. An identifier identifies the session between mobile host and proxy. MSOCKS also has a *reconnect* function, when a mobile host wants to change network interface (IP-address) it opens a new connection to the proxy and sends a reconnect message with the session identifier. The proxy unsplices the old mobile-to-proxy connection and splices in the new mobile-to-proxy connection. The end-to-end semantics of TCP are maintained together with TCP Splice. TCP Splice makes it appear to the endpoints of two separate TCP connections that those two connections are, in fact, one. Data can be lost with this solution; ACK'd data to the correspondent host but lost in the transmission to the mobile host is lost forever.

MSOCKS library is a layer between the application and the transport layer, it provides an interface to the application while internally using the normal TCP stack. To get this to work the applications has to use this library instead of the existing SOCKS or the existing application has to be recompiled. Figure 6.7 shows a message exchange diagram when a MSOCKS client tries to connect to a server on a correspondent host. The MSOCKS library function *Mconnect*() is used for making this split connection. *Mconnect*() first makes a connection to the proxy then it sends the server's address and port number to the proxy in a Connect message. The proxy connects to the desired server and splices the mobile client-proxy and proxy-server connections together. When the splice is set up the proxy finally sends an OK message back to the mobile client.

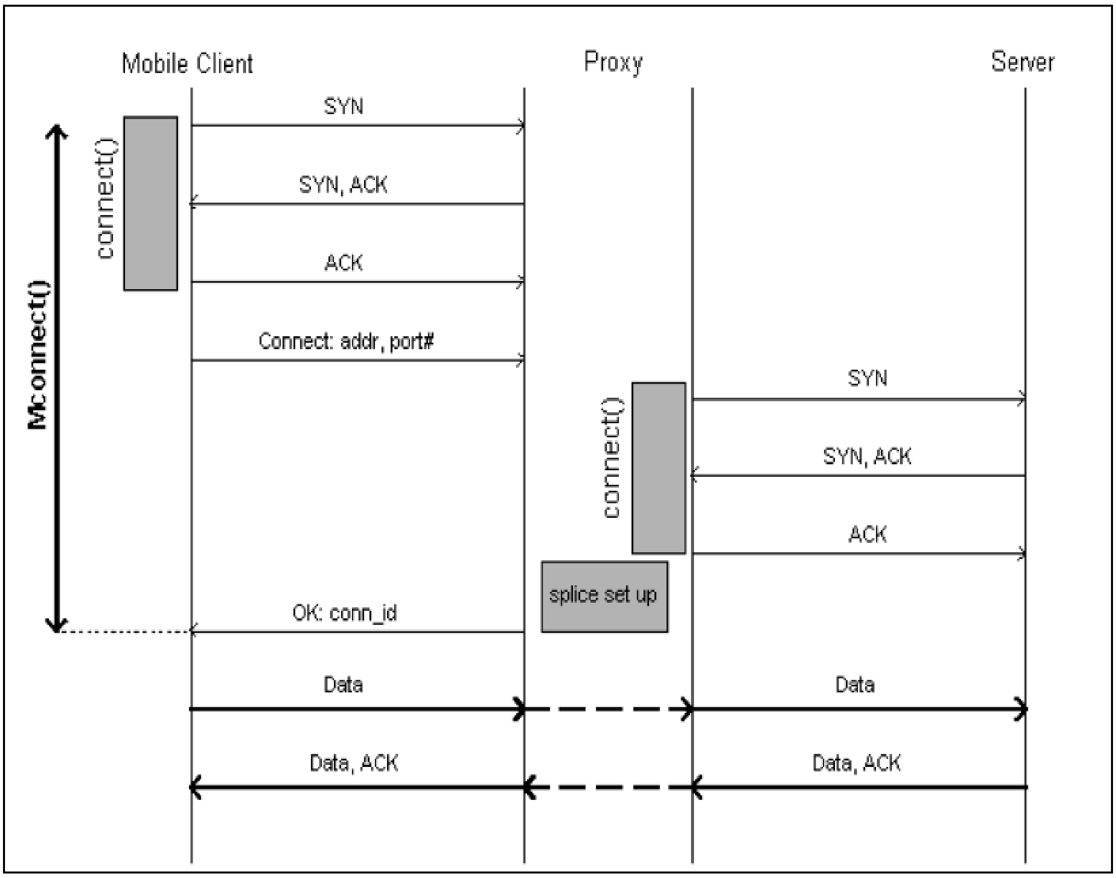

Figure 6.7: *Message exchange diagram for connection establishment between a MSOCKS client and correspondent host via a MSOCKS proxy*.

6.3.3 UNIVERSAL SEAMLESS HANDOFF ARCHITECTURE (USHA)

Universal seamless handoff architecture achieves seamless handoff by following the middleware design[16]. The USHA doesn't require any infrastructure modification, but the solution assumes that handoff only occurs on overlaid networks. The USHA network is composed of a handoff server (HS) and several mobile hosts (MH). The HS and the MH are communicating using an IP tunnel. Each MH maintains a tunnel to the HS. All applications are communication using the tunnel interface instead of any physical IP addresses available. All packets communicating via the tunnel are encapsulated and transmitted to the HS using the UDP protocol. The tunnel has two virtual and two physical IP addresses.

The applications are communicating using the virtual addresses and the tunnel is using the physical addresses to communicate. When a handoff occurs the physical IP address is switched on the mobile host. A handoff client is responsible for switching the physical address of the virtual tunnel to a new interface.

The end-to-end TCP semantics are kept since the session is bound to the virtual addresses and to the tunnel is using UDP.

CHAPTER VII

UNIVERSAL SEAMLESS HANDOFF ARCHITECTURE(USHA)

A *seamless* handoff is defined as a handoff scheme that maintains the connectivity of all applications on the mobile device when the handoff occurs. Seamless handoffs aim to provide continuous end-to-end data service in the face of any link outages or handoff events[16]. To achieve low latency path switching should be completed almost instantaneously and service interruptions should be minimized. In case of an actual connection failure, the architecture should attempt to re-connect as soon as the service becomes available; packet losses during the switching should also be minimized.

Various seamless handoff techniques have been proposed[12][14]. These proposals can be classified into two categories: network layer approaches and upper layer approaches. Network layer approaches are typically based on IPv6 or Mobile IPv4 standards, requiring the deployment of several agents on the Internet for relaying and/or redirecting the data to the moving host (MH). Most upper layer approaches implement a session layer above the transport layer to make connection changes at underlying layers transparent to the application layer . Other upper layer approaches suggest new transport layer protocols such as SCTP and TCP-MH to provide the necessary handoff support.

Previous seamless handoff solutions, whether network layer or upper layer approaches, are often complex to implement and operate. For the network layer solutions, deployment means upgrading every existing router without mobile IP capabilities. The cost imposed by these solutions hinders their chances of deployment. For the upper layer solutions, a new session layer or transport protocol requires an update to all existing applications and servers not supporting it, the potential cost is also discouraging. Consequently, even though many handoff solutions have managed to minimize both latency and packet loss, they are often deemed impractical by the majority of service providers and are still rarely deployed in reality. With the proliferation of mobile applications and mobile users, a “simple” and “practical” seamless handoff solution with minimal changes to the current Internet infrastructure remains necessary.

7.1 INTRODUCTION TO USHA

A Universal Seamless Handoff Architecture (USHA) deals with both horizontal and vertical handoff scenarios with minimal changes in infrastructure. (i.e., USHA only requires deployment of handoff servers on the Internet.) USHA is an upper layer solution; however, instead of introducing a new session layer or a new transport protocol, it achieves seamless handoff by following the middleware design philosophy,integrating the middleware with existing Internet services and applications[16].

USHA is based on the fundamental assumption that handoff, either vertical or horizontal, only occurs on overlaid networks with multiple Internet access methods (i.e. soft handoff), which translates to zero waiting time in bringing up the target network interface when the handoff event occurs. If coverage from different access methods fails to overlap (i.e. hard handoff), it is possible for USHA to lose connectivity to the upper layer applications.

In Figure 7.1, a handoff server (HS) and several mobile hosts (MHs) are shown. USHA is implemented using IP tunneling techniques (IP encapsulation), with the handoff server functioning as one end of the tunnel and the mobile host as the other. An IP tunnel is maintained between every MH and the HS such that all application layer communications are “bound” to the tunnel interface instead of any actual physical interfaces.

All data packets communicated through this IP tunnel are encapsulated and transmitted using the connectionless UDP protocol. The IP tunnel above utilizes two pairs of virtual/fixed IP addresses, one on HS and one on MH. The fixed IP addresses are necessary for an MH to establish a physical connection to the HS. When the handoff event occurs and the physical connection from MH to HS changes, the MH is responsible for automatically switching the underlying physical connection of the virtual tunnel to the new interface, as well as notifying the HS of its change in physical connection.

Upon handoff notification, the HS immediately updates its IP tunnel settings so that any subsequent data packets will be delivered to MH's new physical link. Since all data packets are encapsulated and transmitted using UDP, there is no need to reset the tunnel after the handoff. Therefore, end-to-end application sessions (e.g. TCP) that are bound to the IP tunnel are kept intact. This provides handoff transparency to upper layer applications.

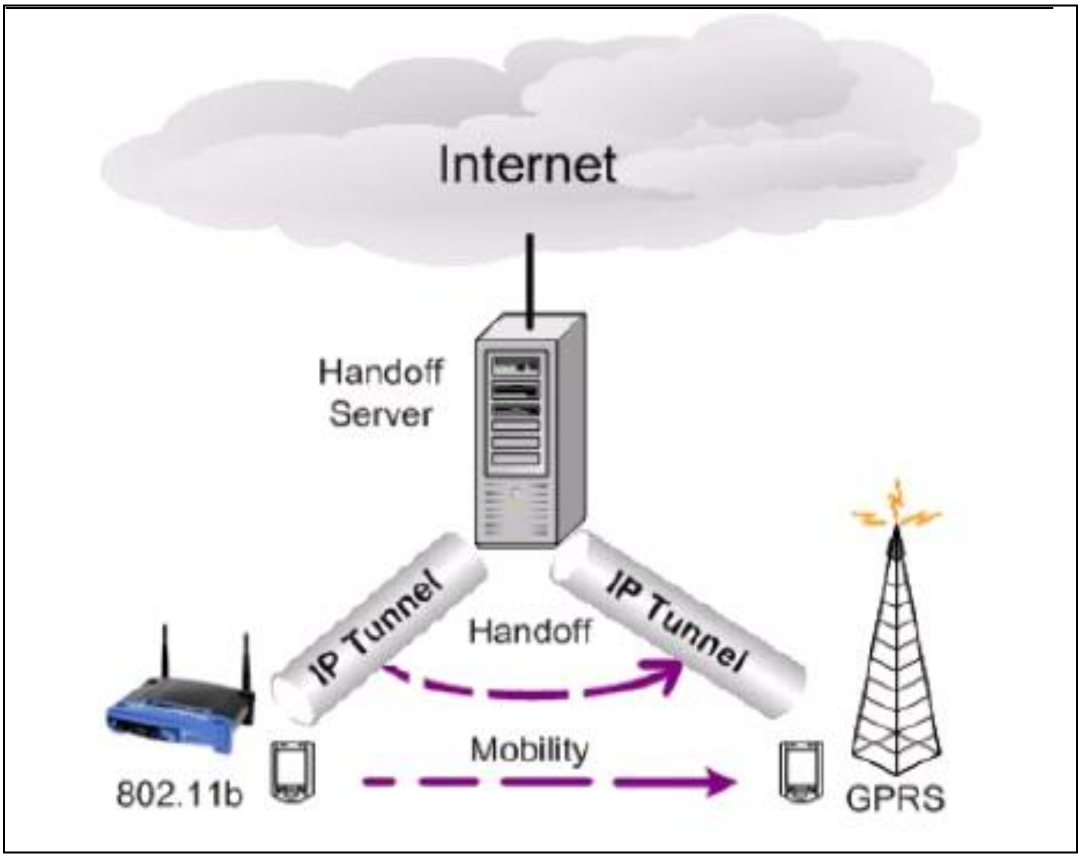

Figure 7.1: *Universal Seamless Handoff Architecture.*

7.2 USHA ENHANCEMENTS

Though USHA provides a simple and practical seamless handoff solution, it still has difficulty in performing handoff smartly, i.e. the handoff cannot be triggered automatically so far. It turns out that a manual handoff solution is still far from becoming practical, since an appreciable solution should keep mobility transparent to the mobile users, i.e. the seamless handoff solution should be "smart" enough so that it is able to perform a handoff properly at the proper moment[16].

In addition its support for continuous connectivity, USHA could be easily extended by inserting additional '*plug-ins*'. For instance, on the MH side, one can implement *policy-enabled handoff system* to determine the "best" target interface at any given moment through a well-designed cost function. Therefore, by continuously updating the cost function, MH can automatically trigger a vertical handoff if necessary.

7.3 DECISION MODEL FOR USHA

A *Decision Model* exists for USHA namely "*Smart Decision Model*"[22]. In the model, a handoff decision is made based on the network properties, system information, and user preferences. Additionally, it is able to perform a handoff automatically to the "most appropriate" network interface at the "most appropriate" moment.

Figure 7.1 depicts the Smart Decision Model. In this figure, a Handoff Control Center (HCC) provides the connection between the network interfaces and the upper layer applications. HCC is composed of four components: Device Monitor (DM), System Monitor (SM), Smart Decision (SD), and Handoff Executor (HE). DM is responsible for monitoring and reporting the status of each network interface (i.e. the signal strength, link capacity and power consumption of each interface).

SM monitors and reports system information (e.g. current remaining battery). SD integrates user preferences (obtained from user set default values) and all other available information provided by DM, SM to achieve a "Smart Decision", to identify the "best" network interface to use at that moment. HE then performs the device handoff if the current network interface is different from the “best” network interface.

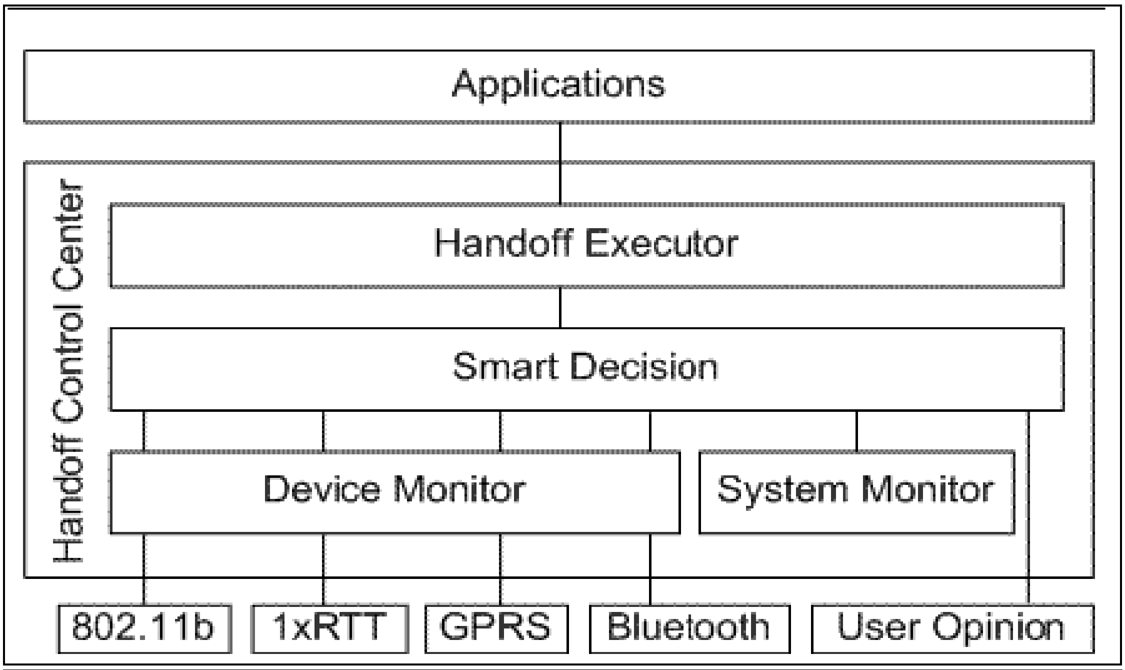

Figure 7.2: *Smart Decision Model.*

The SD algorithm is described as follow:

Priority Phase:

1. Add all available interfaces into candidate list.

2. Remove user specified devices from the candidate list.

3. If candidate list is empty, add back removed devices from step 1

4. Continue with Normal Phase.

Normal Phase:

1. Collect information on every wireless interface in the candidate list from the DM component.

2. Collect current system status from SM component.

3. Use the score function to obtain the score of every wireless interface in the candidate list.

4. Handoff all current transmissions to the interface with the highest score if different from current device.

Priority and normal phases are necessary in SD to accommodate user-specific preferences regarding the usage of network interfaces. For instance a user may decide not use a device when the device may cause undesirable interferences to other devices (e.g. 802.11b and 2.4GHz cordless phones). With priority and normal phases in place, the SD module provides flexibility in controlling the desired network interface to the user[22].

Additionally, SD deploys a *score function* to calculate a score for every wireless interface; the handoff target device is the network interface with the highest score. More specifically, suppose there are *k* factors to consider in calculating the score, the final score of interface *i* will be a sum of *k* weighted functions. The *score function* used is the following:

$$Si = \sum_{j=1}^{k} w_j f_{j,i} \qquad 0 < Si < 1, \ \sum_{j=1}^{k} w_j = 1 \qquad \textbf{(1)}$$

In the equation, wj stands for the weight of factor *k*, and *fj,i* represents the normalized score of interface *i* of factor *j*. The “best” target connection interface at any given moment is then derived as the one which achieves the highest score among all candidate interfaces.The score function can be further break down into three components where each accounts for usage expense (*E*), link capacity (*C)*, and power consumption (*P*), respectively.

Therefore Eq. 1 becomes

$$\mathbf{S_i = W_e f_{e,i} + W_c f_{c,i} + W_p f_{p,i}} \qquad \textbf{(2)}$$

Additionally, there is a corresponding function for each term *fe,i*, *fc,i*, and *fp,i*, and the ranges of the functions are bounded from 0 to 1. The functions are illustrated below:

$\mathbf{f_{e,i} = 1/(e^{\alpha i})}$

$\mathbf{f_{c,i} = (e^{\beta i,i}) / (e^{M})}$

$\mathbf{f_{p,i} = 1 / (e^{\gamma i})}$

where αi≥0, M≥βi≥0 and γi≥0 **(3)**

The coefficients α*i*, β*i*, γ*i* can be obtained via a lookup table (fig 7.3) or a well-tuned function. In Eq. 3, inversed exponential equation for *fe,i* and *fp,I* can be used to bound the result to between zero and one (i.e. these functions are normalized), and properly model users preferences. For *fc,i*, a new term *M* is introduced as the denominator to normalize the function, where *M* is the maximum bandwidth requirement demanded by the user. Without specified by the user, the default value of *M* is defined as the maximum link capacity among all available interfaces.Note that, the properties of bandwidth and usage cost/power consumption are opposite (i.e. the more bandwidth the better, whereas lower cost/power consumption is preferred).

$\alpha = X_i/24$;	**; X_i : Rs/min**
$\beta = Min(Y_i,M)/M$	**; M=2Mbps**
$\gamma = 2 / Z_i$	**; Z_i : hours**

Figure 7.3: *An coefficient function example*

VERTICAL HANDOVER BETWEEN WLAN,CDMA AND GSM

As mentioned earlier In the future Internet users probably want to roam between networks that are using different connection technologies for example, WLAN,CDMA etc. Users probably want to roam between the networks without any interruptions. A user may want to communicate with the best connection currently available, but when a user changes location the best connection may be broken. Another available connection should then takeover the connection, a vertical handover. A handover is called seamless if it can be done without having to restart the running applications.

A application for seamless vertical handover between WLAN,CDMA and GSM is presented here. For simplicity it is assumed that the CDMA network and GSM network is spread almost everywhere, while WLAN network is present in some specified places (fig 8.1).

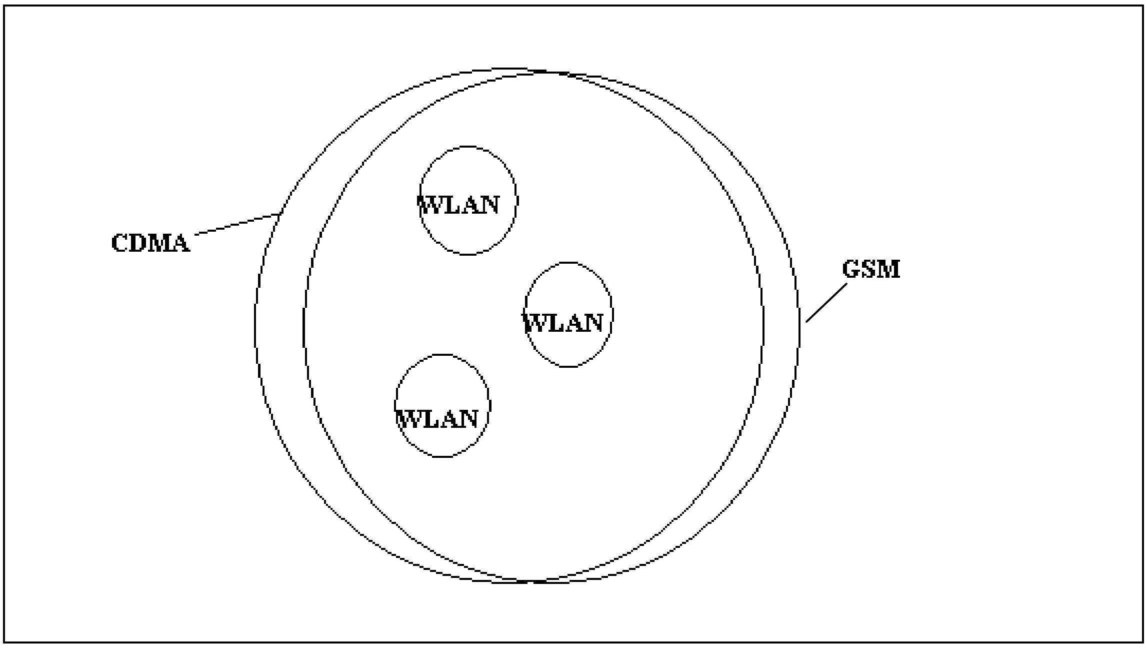

Figure 8.1: *WLAN,CDMA and GSM networks.*

No technology makes it possible to always be connected to a high-bandwidth and a wide-area connection [8]. It would be interesting for a user with multiple wireless network interfaces to always use the best current wireless technology available. The user should be able to decide what is best for the moment and may prioritize different things like, bandwidth, battery consumption, security requirements or cost. The idea is that a mobile user should be able to switch from one network access to another depending on what's available and what needs the user currently has. The mobile user should always be able to choose the best fitting connection without having to restart all running applications.

Further as we know that for a user ,working in WLAN network as compared to that of CDMA and GSM networks, is much more convinient and cheaper. So it desirable here that a user should be handover to WLAN network as much as possible.

Suppose a user is downloading a file from Internet to his laptop (mobile node) using a WLAN connection. The WLAN interface has received an IP address from a DHCP server in the hotspot area he is currently located in. he is moving outside the WLAN coverage physically and lose the connection to the WLAN interface. Now he want his GSM telephone to continue downloading the file without any interruption and active user actions. The GSM connection has received a different IP address from a GGSN. After a while he return to the WLAN hotspot and his WLAN interface receives a new IP address. The download should now switch interface to the WLAN since it has better bandwidth than the GSM connection. These handovers should be done automatically without any user actions.

8.1 METHODOLOGY

Before going into the details of how the application for handover between these three networks is implemented some general terms are needed to be known like what are the parameters on which a user wants to work. As already mentioned in the decision model for USHA there exists three parameters namely, usage expense (*E*), link capacity (*C)*, and power consumption (*P*), respectively.

i,e. the score function can be written as:

$$S_i = W_e f_{e,i} + W_c f_{c,i} + W_p f_{p,i} \quad (4)$$

In this equation, W_e stands for the weight factor for usage expense, W_p stands for the weight factor for power consumption, W_c stands for the weight factor for link capacity and W_r stands for the weight factor for received signal strength respectively.

8.1.1 CALCULATION OF WEIGHT FACTORS

The function for usage expense, $f_{e,i}$ can be calculated as :

$$f_{e,i} = 1/(e^{\alpha}); \quad (5)$$

where from figure 7.3 , α can be written as:

$$\alpha = X_i/24; \quad ; X_i : Rs/min \quad (6)$$

For CDMA, X_i i,e. charges per min is assumed to be equal to 20.

For GSM, X_i is assumed to be equal to 5.

Thus from eq. 5 and 6.

$f_{e,i}$ for CDMA and GSM is equal to **0.4** and **0.8**, respectively.

Now the function for link capacity, $f_{c,i}$ can be written as:

$$f_{c,i} = (e^{\beta,i}) / (e^{M}); \quad (7)$$

where from figure 7.3, β can be calculated as :

$$\beta = Min(Y_i,M)/M \quad ; M=2Mbps \quad (8)$$

For CDMA, Y_i i,e. capacity is equal to 100 Kbps.

For GSM, Y_i is equal to 150 Kbps.

Thus from eq. 7 and 8.

$F_{c,i}$ for CDMA and GSM is equal to **0.95** and **0.92**, respectively.

Similarly , the function for power consumption, $f_{p,i}$ can be calculated as :

$$f_{p,i} = 1 / (e^{\gamma i}) \qquad (9)$$

where from figure 7.3 , γ can be written as:

$$\gamma = 2 / Z_i \qquad ; Z_i : \text{hours} \qquad (10)$$

For CDMA, Z_i i,e. battery capacity is equal to 2.5.

For GSM, Z_i is equal to 2.

Thus from eq. 9 and 10.

$f_{p,i}$ for CDMA and GSM is equal to **0.4** and **0.36**, respectively.

Thus eq. 4 can be written as:

$$S_{CDMA} = 0.4W_e + 0.95W_c + 0.4W_p$$

$$S_{GSM} = 0.8W_e + 0.92W_c + 0.36W_p$$

Here the values of W_e ,W_c and W_p is received from the user at the start of simulation.

8.1.2 GRADE OF SERVICE(GOS)

In loss systems, the traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers. The overall traffic is rejected and hence is not carried by the network. The amount of traffic rejected by the network is an index of the service offered by the network.This is termed as grade of service (GOS) and is defined as the ratio of lost traffic to offered traffic. Offered traffic is the product of the average number of calls generated by the users and the average holding time per call. On the other hand ,the actual traffic carried by the network is the network is called the carried traffic and is the average occupancy of the servers in the network as given by the following equation.Accordingly , GOS is given by:

$$GOS = (A - A_0)/A$$

Where

A= offered traffic

A_0=carried traffic

$A-A_0$=lost traffic

The smaller the value of grade of service, the better is the service. The recommended value for GOS in India is 0.002 which means that two calls in every 1000 calls or one call in every 500 calls may be lost. Usually, every common subsystem in a network has an associated GOS value. The GOS of the full network is determined by the highest GOS value of the subsystems in a simplistic sense.

The blocking probability P_B is defined as the probability that all the servers in a system are busy. When all the servers are busy, no further traffic can be carried by the system and the arriving subscriber traffic is blocked. At the first instance, it may appear that the blocking probability is the same measure as the GOS. The probability that all the servers are busy may well represent the fraction of the calls lost,which is what GOS is all about. For example , in a system with equal number of servers and subscriber,the GOS is zero as there is always a server available to a subscriber. On the other hand , there is a definite probability that all the servers are busy at a given instant and hence the blocking probability is nonzero.

The fundamental difference is that the GOS is a measure from the subscriber point of view whereas the blocking probability is a measure form the network or switching system point of view. GOS is arrived at by observing the number of rejected subscriber calls, whereas the blocking probability is arrived at by observing the busy servers in the switching system.

Among many system performance measures, GOS is mostly used. In this application GOS is thus used for measuring the system performance. Grade of service can be written as:

$$\mathbf{GOS = (1 - \alpha)\, PB + \alpha\, PD} \qquad \textbf{(11)}$$

where the P_B and P_D represent the overlaid system's blocking probability and the dropping probability, respectively. The weight $\boldsymbol{\alpha}$ emphasizes the dropping effect with the value of larger than the half. As a decision criterion, velocity threshold ***VT*** is to be determined to optimize system perfomance. The optimal velocity threshold controls to assign the available channels to the mobile stations with various mobility.

8.1.3 VELOCITY THRESHOLD, VT

Overlaid network architecture takes advantage of different cell sizes: Micro- or pico-cells can transmit a large data rate and support more users while the macro-cells accommodate faster moving terminals without much signal traffic load induced.The size of the cells in cellular mobile communications system become smaller as users require more data rate and better quality of the voice transmission. Upcoming mobile cellular systems, such as IMT-2000, require the system to support users in different speed and of different data rates. A multitier system typically with macrocells overlaying microcells offers the system providers with new opportunities for the purpose. However the resulting high operation cost is to be managed. The service providers want to minimize the signals trafiic of the multitier cell structured systems.

The two-tier cellular systems typically become to existence by additionally deploying microcells onto the existing macrocell structures. The microcells provides the higher data rate since the channel bandwidth is large. Generally, in microcellular systems, handoff occurs frequently. It means also that the microcell systems must handle a large number of handoff requests. On the other hand, the macrocell systems are large enough to accommodate the fast moving users, or lower handoff request rate, hence, the smaller system burden. It is safe to assume that slow or stationary users transmit more data and that fast moving stations communicate with less data rates. Appropriate handoff control becomes important issue in system management for the sake of the above benefits by the overlaid cell structures.

A system parameter is to be defined to assign the channels to the mobile stations in order to appropriately assign macro and micro channels to the fast moving and the slow moving mobiles, respectively. For this purpose, the velocity threshold VT could be used to divide the users into two groups, slow moving and fast moving users. For example when the signal from the WLAN access point (AP) is strong, the MS is connected to the WLAN when the speed of the MS is smaller than velocity threshold (VT). This is because a fast moving MS having speed greater then VT, may cross the WLAN acess point, much earlier then it is given controlled to WLAN. As the MS moves away from the coverage of the access point, the signal strength falls and the distance between AP and MS far away. These two criteria (RSS and distance) reduce the *unnecessary handoff probability*. The MS then scans the air for other access points. If no other access point is available, or if the signal strengths from available access points are not strong enough, the handoff algorithm uses this information along with other possible information to make a decision on handing off to the CDMA network or to the GSM network[22].Thus VT can improve system performance by reducing the unnecessary handoff probability which may occur in case hanoff is entirely dependent on RSS value.

Estimation Of Velocity Threshold, VT

The relationship between Grade fo service(GOS) and velocity threshold(VT) is given in following graph[21].

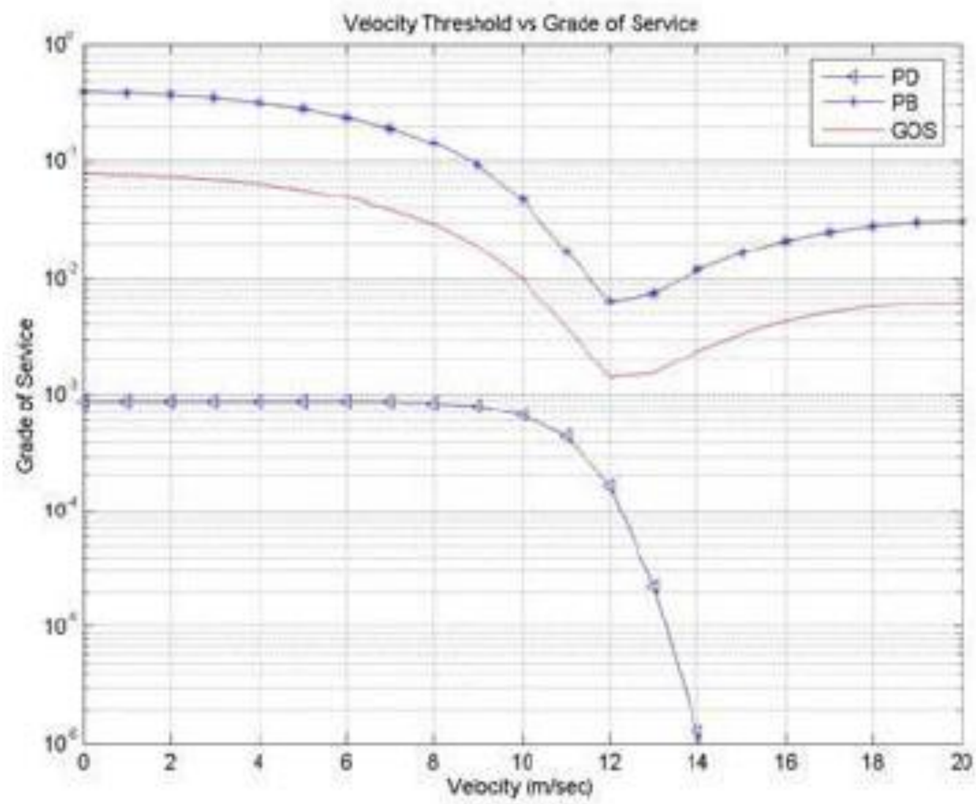

Figure 8.2 : *Relationship between GOS and VT.*

The problem then is to find such VT that minimize the GOS given traffic and terminal's mobility .First, given the fixed size of the microcell in the overlaid structure the upper limit of the VT needs to be considered If the VT is set larger than the system's ability with respect to the new call set up time and the handoff delay time, the system will experience unpredictable P_B and P_D. Secondly, the capacity of microcell's handoff handling is prespecified. Therefore the number of handoff in a microcell must be less than the maximum number of handoff request rate. The minimum of the two upper limits should be the new upper bound for the optimal velocity threshold. The estimation of the velocity threshold (VT) procedure at carried out the system is shown in Figure 8.3[22].

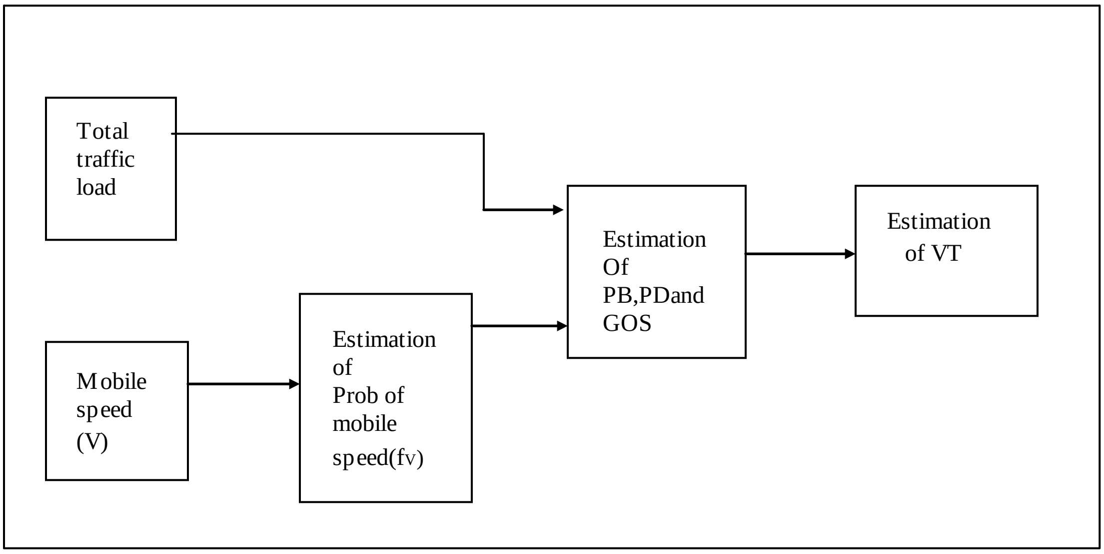

Figure 8.3 : *Estimation of Velocity Threshold.*

For the estimation of the mobile speed, Global Positioning System (GPS) can provide adequate location information. The problem here is to find *VT* improving the GOS with the given traffic parameters. The terminals' mobility distribution fv is assumed not to change as the total traffic changes.

It has been experimentally proved that for PB=PD=0.2, VT lies in the range of 11 to 13 m/sec, such that the grade of service have its minimum value.

In the application PB and PD are assumed to have value each equal to 0.2 and the value of velocity threshold is assumed to be **12 m/sec**.

8.2 ALGORITHM USED

The algorithm that is used to perform a handoff automatically to the “most appropriate” network interface at the “most appropriate” moment is shown below:

Step 1: *Collect the values of the weight factor of the three parameters namely usage expence (E),power consumption (P) and capacity (C).*

Step 2: *If working in WLAN, goto Step 3. if working in CDMA ,goto Step 9 else goto Step 13.*

Step 3: *Working in WLAN. Measure the received signal strength (RSS) value.*

Step 4: *If RSS_wlan>RSS_threshold, goto Step 3.*

Step 5: *Check whether another WLAN network exists nearby. If no goto Step 8.*

Step 6: *Measure its RSS value.if RSS_nwlan<RSS_threshold, goto Step 8.*

Step 7: *Measure the vilocity(V). If V<V_threshold, goto Step 3.*

Step 8: *Compare the score function of CDMA and GSM. If* $S_C > S_G$*, goto Step 10,else goto Step 13.*

Step 9: *Working in CDMA. Check whether WLAN network exists nearby. If no goto Step12.*

Step 10: *Measure the received signal strength (RSS) value.If RSS_wlan < RSS_threshold, goto Step 12.*

Step 11: *Measure the vilocity(V). If V<V_threshold, goto Step 3.*

Step 12: *Measure the received signal strength (RSS) value.If RSS_cdma > RSS_threshold, goto Step 9.*

Step 13: *Working in GSM. Check whether WLAN network exists nearby. If no goto Step 16.*

Step 14: *Measure the received signal strength (RSS) value.If RSS_wlan < RSS_threshold, goto Step 16.*

Step 15: *Measure the vilocity(V). If V<V_threshold, goto Step 3.*

Step 16: *Measure the received signal strength (RSS) value.If RSS_gsm > RSS_threshold, goto Step 13. Else goto Step 9.*

The procedure that is carried during handoff from WLAN is shown below:

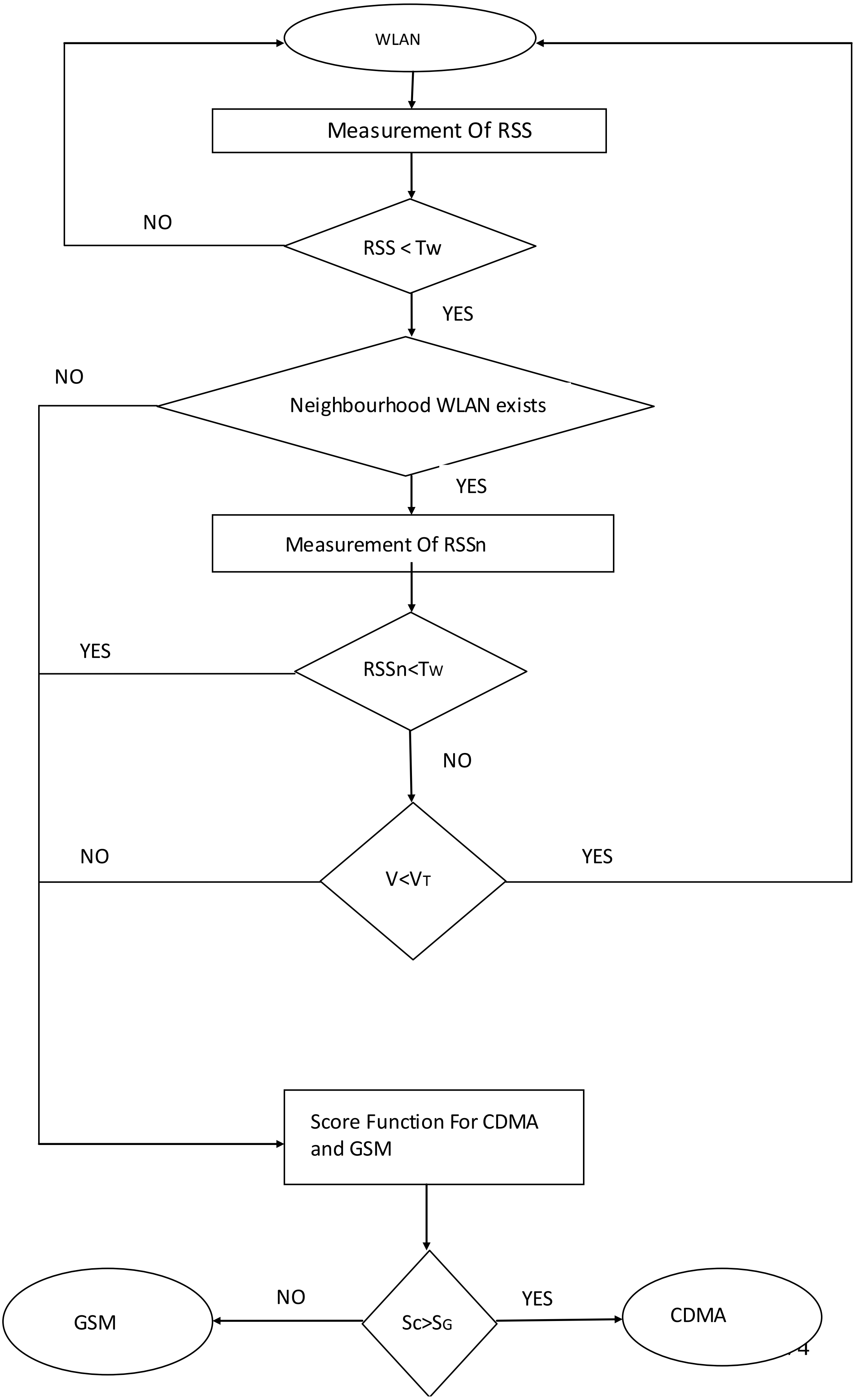

Figure 8.4: *Handover From WLAN.*

The procedure that is carried during handoff from CDMA is shown below:

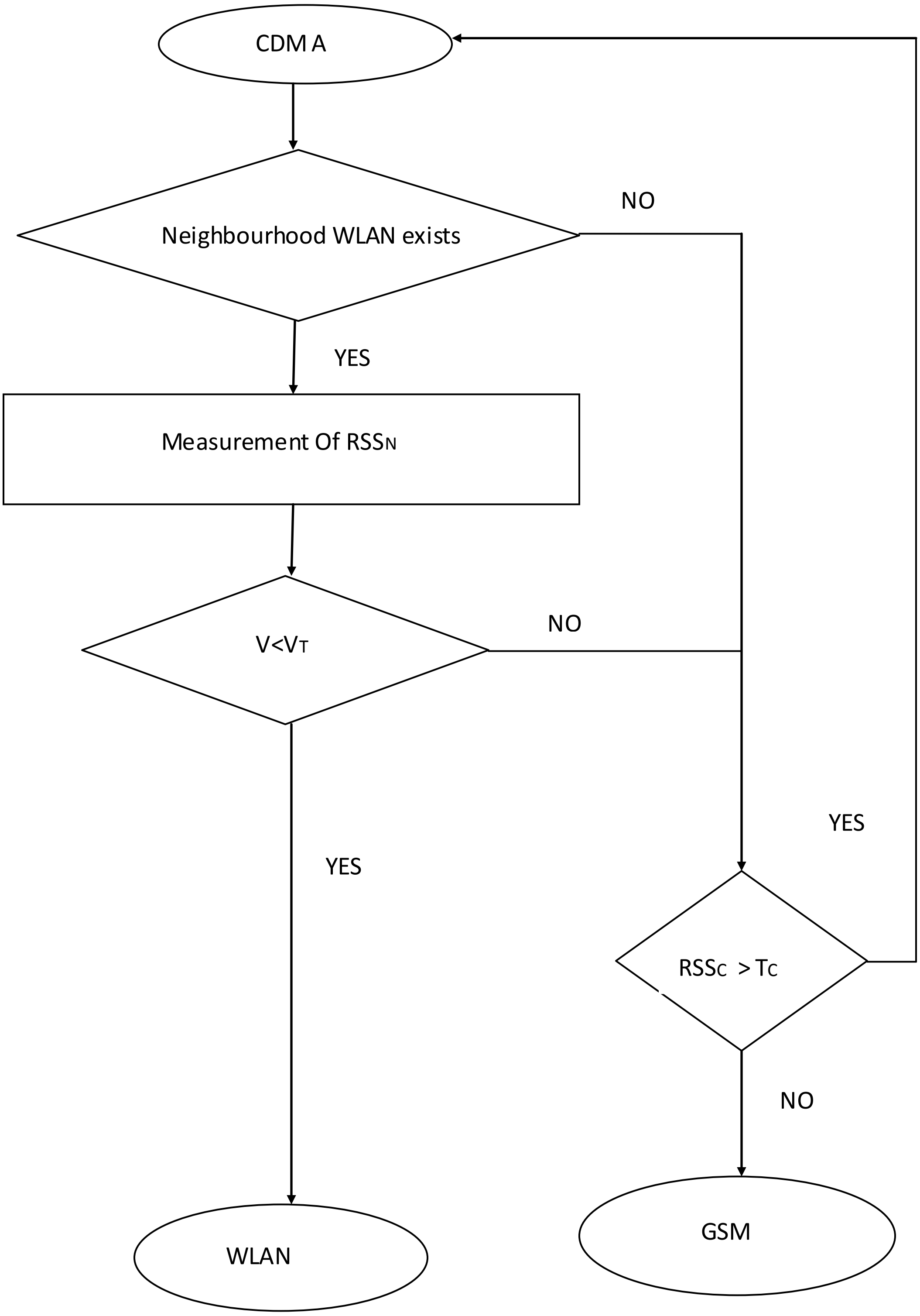

Figure 8.5: *Handover From CDMA.*

The procedure that is carried during handoff from GSM is shown below:

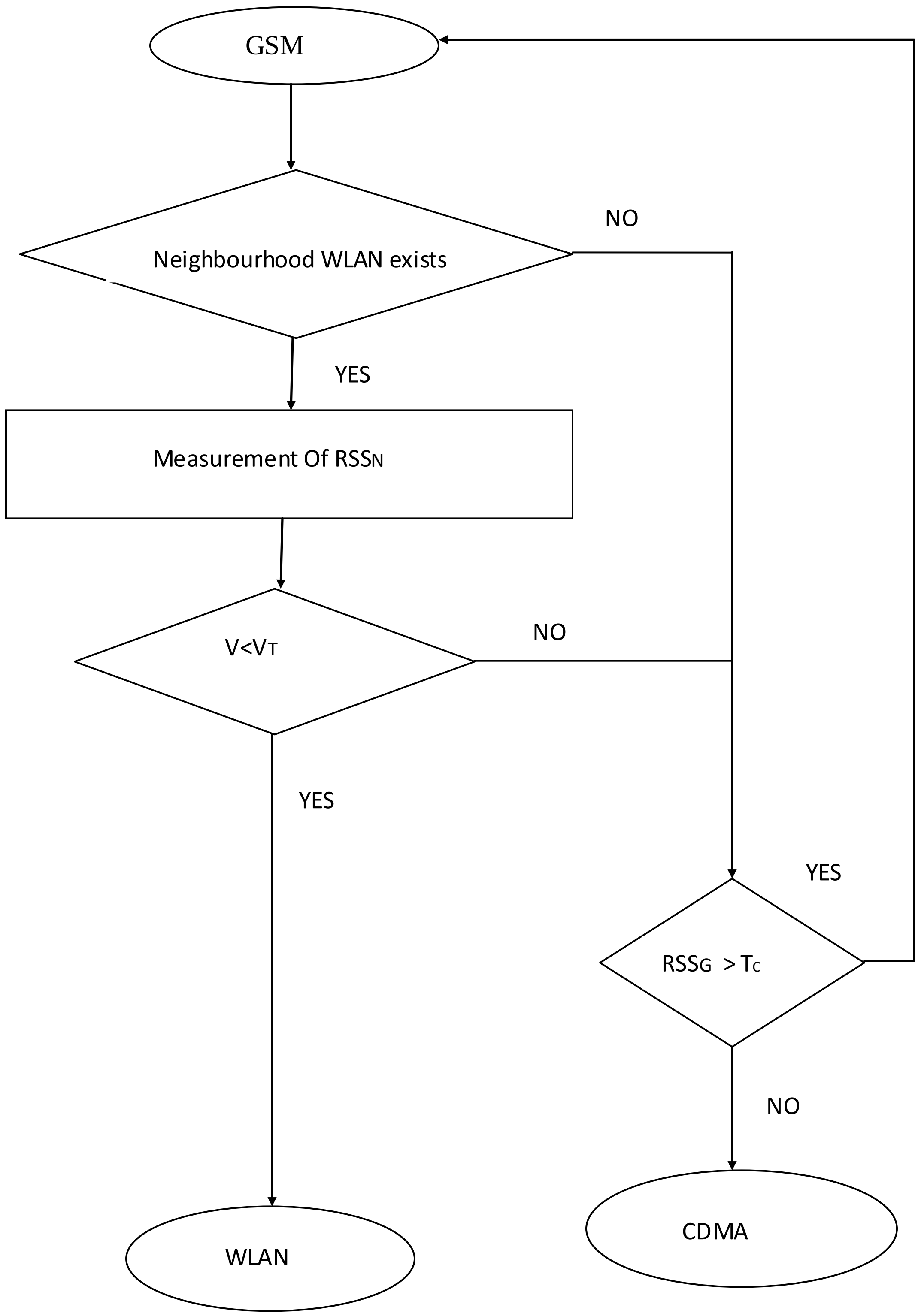

Figure 8.6: *Handover From GSM.*

8.3 RESULTS

The code for the simulator is implemented in C programming language.

Following are some of the results :

Sample Result 1:

```
INPUT VALUES OF THE SIMULATION
--------------------------------
......SIMULATION OF VERTICAL HANDOVER......
TOTAL SIMULATION TIME IN SECONDS:200
TIME PERIOD IN SECONDS:10
WEIGHT FACTOR FOR COST:0.2
WEIGHT FACTOR FOR CAPACITY:0.3
WEIGHT FACTOR FOR POWER CONSUMPSION:0.2
WORKING IN WIRELESS LAN?:ENTER 1 FOR  YES OR 0 FOR NO:1
--------------------------------
```

WORKING IN WLAN AT START OF SIMULATION

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

RECIEVED SIGNAL STRENGTH IS WEAK..

GENERATING SCORE FUNCTION FOR GSM AND CDMA

ENTERING CDMA NETWORKS AS ITS SCORE EXCEEDS THAT OF GSM

WORKING IN CDMA

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

USER IS MOVING VERY FAST...THUS IT CAN NOT BE HAND OVER TO NEIGHBOURHOOD WLAN.

USER STILL *WORK IN CDMA* NETWORK

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

TIME OF SIMULATION HAS BEEN EXPIRED

-----SIMULATION STATISTICS---

time spend in wireless lan is equal to W= 50

time spend in cdma network is equal to C= 150

time spend in gsm network is equal to G= 0

TOTAL NO OF SWITCHES

from current WLAN to neighbour WLAN = 0

from WLAN to CDMA= 1

from WLAN to GSM = 0

from CDMA to WLAN = 0

from CDMA to GSM = 0

from GSM to WLAN = 0

from GSM to CDMA = 0

Sample Result 2:

```
INPUT VALUES OF THE SIMULATION
--------------------------------
......SIMULATION OF VERTICAL HANDOVER......
TOTAL SIMULATION TIME IN SECONDS:200
TIME PERIOD IN SECONDS:5
WEIGHT FACTOR FOR COST:0.5
WEIGHT FACTOR FOR CAPACITY:0.3
WEIGHT FACTOR FOR POWER CONSUMPSION:0.2
WORKING IN WIRELESS LAN?:ENTER 1 FOR  YES OR 0 FOR NO:1
--------------------------------
```

WORKING IN WLAN AT START OF SIMULATION

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

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USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
RECIEVED SIGNAL STRENGTH IS WEAK....
GENERATING SCORE FUNCTION FOR GSM AND CDMA

ENTERING CDMA NETWORKS AS ITS SCORE EXCEEDS THAT OF GSM

WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
ENTERING IN WIRELESS LAN NETWORK AS USER'S SPEED IS LESS THAN VT

USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER SWITCHES TO NEIGHBOURHOOD WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
TIME OF SIMULATION HAS BEEN EXPIRED

-----SIMULATION STATICS----
time spend in wireless lan is equal to W= 190
time spend in cdma network is equal to C= 10
time spend in gsm network is equal to G= 0
TOTAL NO OF SWITCHES
from current WLAN to neighbour WLAN = 1
from WLAN to CDMA= 1
from WLAN to GSM= 0

from CDMA to WLAN= 1

Sample Result 3:

......SIMULATION OF VERTICAL HANDOVER......

………..INPUT VALUES FOR SIMULATION…………..

TOTAL SIMULATION TIME IN SECONDS :300

TIME PERIOD IN SECONDS :5

WEIGHT FACTOR FOR COST :0.3

WEIGHT FACTOR FOR CAPACITY :0.3

WEIGHT FACTOR FOR POWER CONSUMPTION :0.4

WORKING IN GSM AT START OF SIMULATION

GSM

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

USER STILL WORK IN GSM NETWORK

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

USER IS MOVING VERY FAST...THUS IT CAN NOT BE HAND OVER TO NEIGHBOURHOOD WLAN..

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

USER IS MOVING VERY FAST...THUS IT CAN NOT BE HAND OVER TO NEIGHBOURHOOD WLAN..

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

ENTERING IN WIRELESS LAN NETWORK AS USER'S SPEED IS LESS THAN VT

USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER STILL WORKS IN WIRELESS LAN
USER IS MOVING VERY FAST THUS IT CAN NOT BE HANDED TO NEIGHBOURHOOD WLAN……BUT BECIEVED SIGNAL DEGRADES..
...GENERATING SCORE FUNCTION FOR GSM AND CDMA
ENTERING CDMA NETWORKS AS ITS SCORE EXCEEDS THAT OF GSM

WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
USER STILL WORK IN CDMA NETWORK
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
USER STILL WORK IN CDMA NETWORK
WIRELESS LAN DOES NOT EXIST NEAR BY………
AND CDMA SIGNAL DEGRADES:
ENTERING INTO GSM NETWORK:

GSM
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
USER IS MOVING VERY FAST...THUS IT CAN NOT BE HAND OVER TO NEIGHBOURHOOD WLAN..
WIRELESS LAN EXISTS NEAR BY
WIRELESS LAN EXISTS NEAR BY
USER IS MOVING VERY FAST...THUS IT CAN NOT BE HAND OVER TO NEIGHBOURHOOD WLAN..
WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

ENTERING IN WIRELESS LAN NETWORK AS USER'S SPEED IS LESS THAN VT

USER STILL WORKS IN WIRELESS LAN

USER STILL WORKS IN WIRELESS LAN

RECIEVED SIGNAL STRENGTH IS WEAK....

GENERATING SCORE FUNCTION FOR GSM AND CDMA

ENTERING CDMA NETWORKS AS ITS SCORE EXCEEDS THAT OF GSM

USER STILL WORK IN CDMA NETWORK

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

WIRELESS LAN EXISTS NEAR BY

TIME OF SIMULATION HAS BEEN EXPIRED

-----SIMULATION STATICS----

time spend in wireless lan is equal to W= 45

time spend in cdma network is equal to C= 140

time spend in gsm network is equal to G= 115

TOTAL NO OF SWITCHES

from current WLAN to neighbour WLAN = 0

from WLAN to CDMA= 2

from WLAN to GSM= 0

from CDMA to WLAN= 0

from CDMA to GSM= 1

from GSM to WLAN= 2

from GSM to CDMA= 0

The relationship between the weight factors (Wc,Wp and We) and Score functions (Sc and SG) is shown in the following graphs. The graphs are ploted using MATLAB.

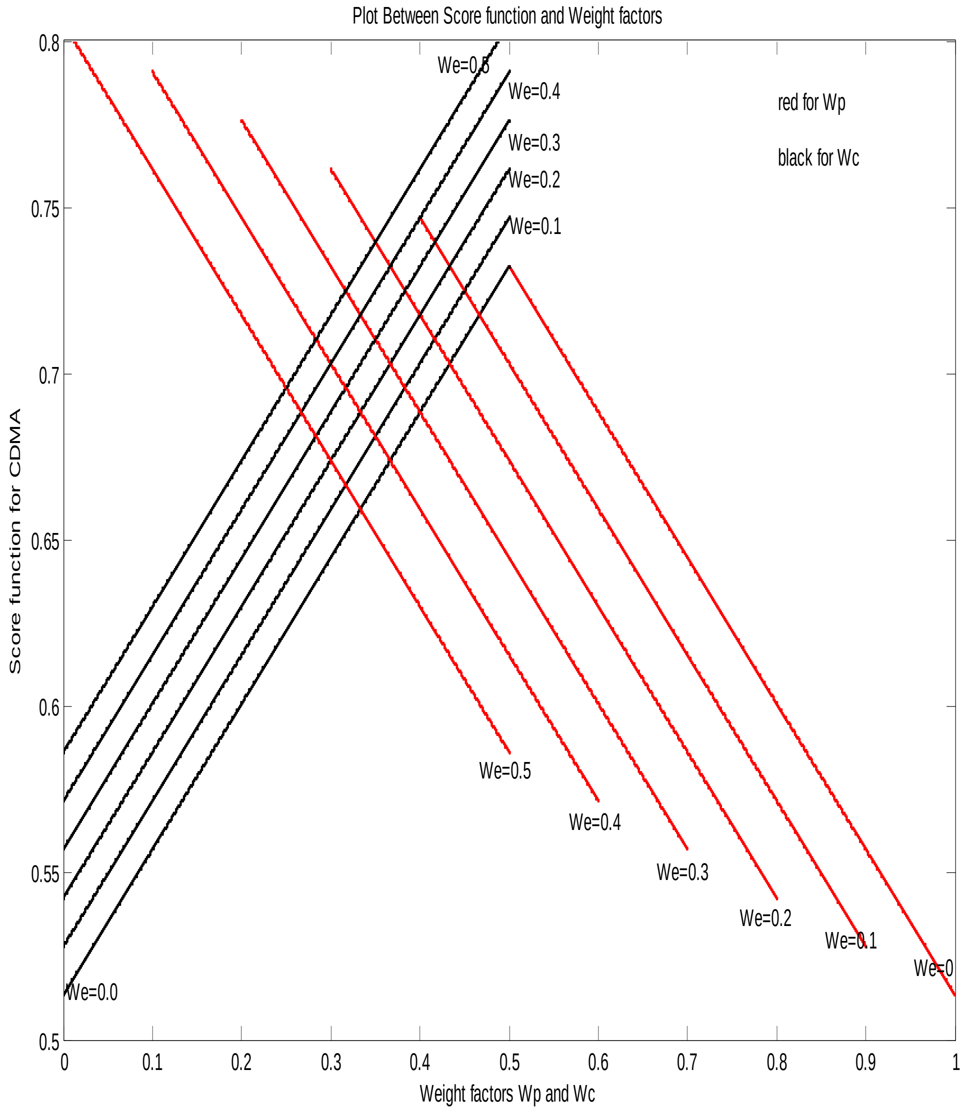

Figure 8.7 : *Graph between Score Function for CDMA and weight factors .*

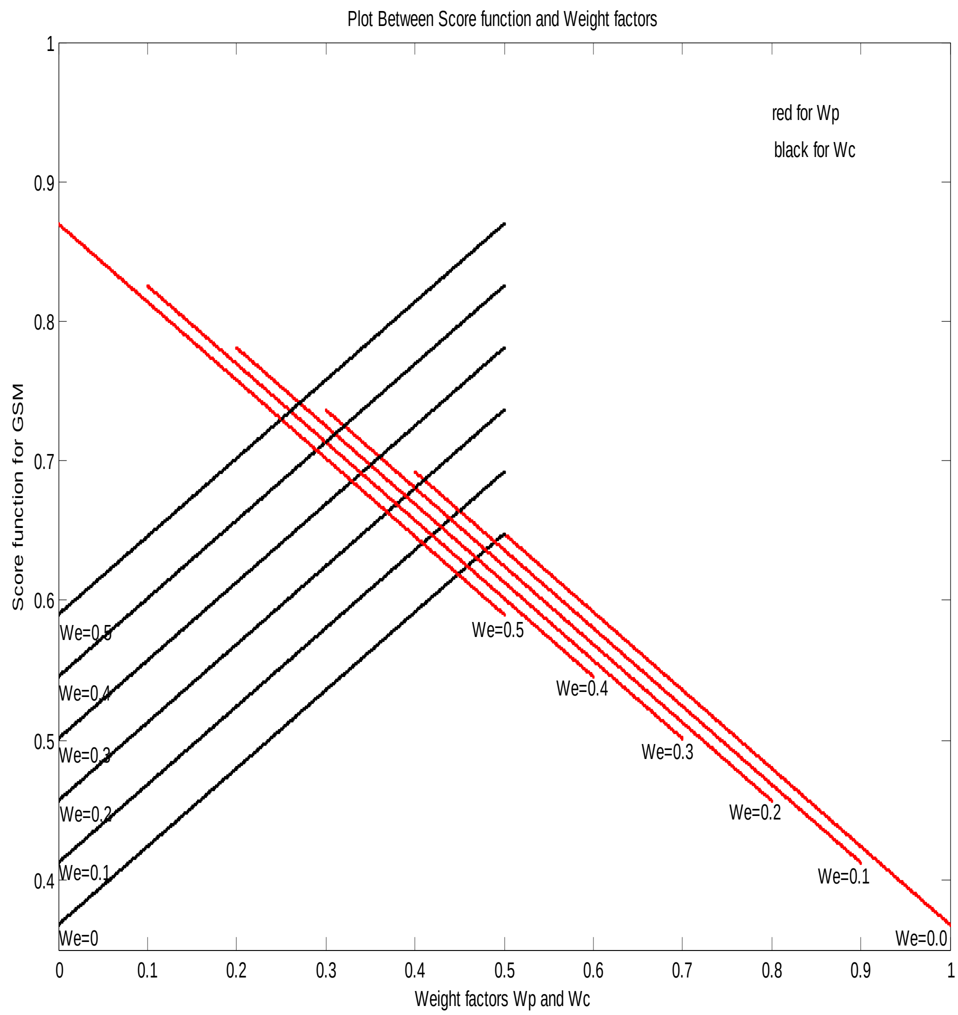

Figure 8.8 : *Graph between Score Function for GSM and weight factors.*

CHAPTER IX

CONCLUSION

Various seamless vertical handoff techniques have been studied in this thesis. Previous seamless handoff solutions, whether network layer or transport layer approaches, are often complex to implement and operate. For the network layer solutions, deployment means upgrading every existing router without mobile IP capabilities. The cost imposed by these solutions hinders their chances of deployment. For the transport layer solutions, a new session layer or transport protocol requires an update to all existing applications and servers not supporting it, the potential cost is also discouraging. Universal Seamless Handoff Architecture an upper layer approach deals with both horizontal and vertical handoff scenarios with minimal changes in infrastructure (i.e., USHA only requires deployment of handoff servers on the Internet.)

An implementation of a seamless vertical handoff procedure and the handoff algorithm for the handoff transition region between the WLAN ,GSM and CDMA cellular network is presented. The challenging problem was to keep the session alive while changing the physical connection interface. This was done using a 'Decision model' and 'Velocity of mobile station'.The decision model deals with chosing the best network out of all available networks depending upon the parameters like monetary cost, offered services, network conditions and user preferences. And the latter deals with the motion of the mobile station. The velocity factor decides whether the station is suitable for a handover.

In chapter 8, we have gone extensively about the various parameters that govern the smooth functioning of the algorithm and how they affect the time spent in these different networks.

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