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Signature

Name of Supervisor : Dr. Razali Ismail

Date : 14 October 2004
RESOURCE ALLOCATION SCHEMES FOR WIRELESS ASYNCHRONOUS TRANSFER MODE (ATM) NETWORK

MOHD FADEL BIN ALWI

A thesis submitted in fulfillment of the requirements for the award of the degree of Master of Electrical, Electronic and Telecommunication

Faculty of Electrical Engineering
Universiti Teknologi Malaysia

OCTOBER 2004
"I declare that this thesis entitled ‘Resource Allocation Schemes for Wireless Asynchronous Transfer Mode (ATM) Network’ is the result of my own project except as cited in the references. This thesis has not been accepted for any degree and is not concurrently in candidature of any other degree."

Signature : .................................

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Date :  14 OCTOBER 2004
In the name of ALLAH, The Most Gracious, The Most Merciful…

For their love, support and advice, all the encouragement they gave and the understanding, and also patience they showed all this while, I would like to record my deepest appreciation to my beloved family and loves one…

They are always with me and my highest inspiration…
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Sincere gratitude and appreciation is also express to my supervisor, Dr Razali Ismail and for the most valued guidance, advice, encouragement and helpful discussions throughout the study.

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ABSTRACT

With the ever increasing demand for easy portability and mobility of devices supporting diverse mobile multimedia applications, the need for the adaptation of broadband infrastructure to wireless scenario. However, the cost to set up broadband multimedia is very high but ATM network is cheaper. The mobile multimedia networks like the wireless ATM are faced with challenges relating to user mobility management and channel access. In WATM, traffic control is essential in order to protect the network from congestion and to achieve realistic network efficiency. The main objective is designed the resource allocation scheme that can give high (QoS) performance in terms of probability of call lost and handover call. It means performance in terms of probability of new call, handover call dropping for real time connections and the probability of overload for non-real time. All of these will be done in mathematical analysis using Matlab and simulations using COMNET III. The design started with scheme call With Reserved Channel (WRC), which has reserved channels for handover calls. The reserved channel will be getting the good performance in term of QoS. However, this scheme cannot be support for the new incoming calls. Hence, to improve it, the With Queuing and Reserved Channels (WQRC) will be design to handle that problem. This scheme allows queuing of new incoming calls and reserved channels for handover calls. Using WQRC, low probability of handover dropping was maintained and the new call blocking probability was eliminated. The network will be getting the higher link utilization and increase the performance and reliability of the network.
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LIST OF SYMBOLS

\( P_{HO} \) - Probability Of Handover

\( B \) - The Number Of Base Station In A Cell Cluster

\( P_v \) - The Probability That The Mobile User Visit \( K \) Cell Sites

\( \lambda \) - Arrival Rate

\( \mu \) - Departure Rate

\( P_{nr} \) - The Probability Of S Connection At Base Station

\( P_{ov} \) - The Overload Probability

\( \mathcal{I} \) - State Space In Reversible Markov Process

\( P_{NCH} \) - The Probability Of New Call Block

\( P_{HDH} \) - The Probability Of Handover Dropping

\( \gamma_{rt} \) - The Departure Rate Of Handover Calls From The Source Cell To The Target Cell Cite

\( \gamma_{a} \) - The Arrival Of Handover Calls In Source Cell Site

\( \lambda_{w} \) - The New Call Arrival Rate For WRC

\( \mu_{w} \) - The New Call Departure Rate For WRC

\( \mu_{c} \) - The Completion Of Calls In The Source Cell Site

\( \lambda_{cr} \) - The New Incoming Calls In The Source Cell Site

\( P_{HDH_u} \) - The Probability Of Handover Blocking

\( P_D \) - The Probability Of Delay
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CHAPTER 1

INTRODUCTION

1.1 Overview

The service area of wireless network is divided into cell as shown in Figure 1.1a and 1.1b. Each cell has an assigned group of discrete channels from the available bandwidth. The base station (BS) is established in each cell. Subscribers in each cell use channels assigned to each cell. Every mobile unit (MU) in the cell communicates through the BS via a channel. A region initially served by a BS in a conventional mobile system is split into several cells, each of which has its own BS; it serves by a certain number of channels.

The primary rule of a BS is to terminate the radio link in the network side of the user to network interface. In a small micro cell system, a distributed channel assignment function can be implemented in BS. BSs are connected to the switching network through the wired lines. The major requirement for the future mobile communication systems is high network capacity and flexibility to accommodate time varying communication network. The limited availability of the radio frequency spectrum or bandwidth is the main reason that the future cellular mobile systems must be efficiently employed channel assignment methods to increase network capacity and adapt to time varying traffic.
Figure 1.1a: Wireless Network Architecture

Figure 1.1b: Network Architecture of Mobile ATM
The wireless cellular has designed to maximize the number of channel in the space and spectrum under the constraint due to interference is lower than the prescribed limit. The capacity of wireless cellular system is directly related to the number of channels available. The area in which channel can be reused, and the minimum cell size that can be implement in a given by environment. For the rural area, micro cell can be used, and for urban and suburban areas micro cell is introduced. The radius of macro cell about 10 KM and micro cell between 500 meter and 1000 meter.

The large subscriber capability requirement demands on the system are capable of serving many thousands of mobile unit within a local requirement area with fixed number of hundred channels. The same frequency channel is reused in several cells simultaneously if they are separated by a distance that is long enough to avoid co channel interference. When we use the frequency reuses, big problem become like a channel assignment. To find a channel assignment within each cell that avoids co channel interference. When a call is requested, the wireless network goes through the channel assignment process. If a channel is available, the process allocates the channel to the user.

When the MU moves from source cell to destination cell, handover will occurs. Based on that, the destination cell should provide sufficient resources to the call have handed in. This is because if the resources are not sufficient to meet the pertinent QoS needs, premature termination of the calls can occur. In resource allocation at BS should be priority to the handover request over new connection request as premature termination of a call is more detrimental to the network performance than the rejection of new call.

At this time, the non-real time connection such as Short Message Service (SMS) arrival at the BS, but the system put it in the buffer if the systems have a buffer. The strategy to solve this problem is allocated an effective bandwidth for a channel assignment and reservation channel. It is the objective of this project. Future mobile connections will include the requiring Quality of Service (QoS) guarantee with fairly high bandwidth requirement.
These connections will also require being of different bandwidths according to the specific application. In wireless Asynchronous Transfer Mode (WATM), traffic control is an important issue to protect the network from congestion and to achieve network efficiency compliance with the QoS [1].

1.2 Standard ATM

Broadband telecommunication networks, according to the I-300 series of the International Telecommunication Union – Telecommunication Sector (ITU-T) recommendations, are based on packet switching technique, established in 1990/91 the so called asynchronous transfer mode (ATM). ATM is a member of the fast packet switching family called cell delay. As part of its heritage, it is an evolution from many other sets of protocols. In fact, ATM is a statistical time division multiplexed (TDM) from the traffic that is designed to carry any form of traffic and enables the traffic to be delivered asynchronously to the network. In another word, ATM is a data transport technology that supports a single high speed infrastructure that integrates broadband communications which involve voice, video and data. It achieves bandwidth efficiency through statistical multiplexing of transmission bandwidth.

The data are usually transmitters over a coaxial or optic fiber cable of high transfer rate (155,600,1200Mb/s), low bit error rate ($10^{-9}$). In the end of the equipment, an ATM adaptation layer (AAL) maps the service offered by the ATM network to the services required by the application. This enables ATM to handle a wide range of information bit rate together with various types of real time and non real time service classes with different traffic attributes and Quality of Service (QoS) guarantees at cell and call levels.

ATM technology is an integration of the existing circuit mode digital communication technique with the packet mode communication technique. In the
sense that ATM uses ATM cells as its basic unit of transmissions, it has a close connection with packet mode communication. However, there is significant different that ATM is designed from beginning to manage real time CBR/VBR traffic, whereas traditional packet networks were designed primary for non real time data traffic ABR/UBR/nrt-VBR. This topic will be discussed in chapter 2.1.2.

ATM is based on sort of packet technology, it yields an efficient use of network resource through dynamic allocation of resource and statistical multiplexing. In addition, the concept of virtual circuits (VC) results in clear and simple basis for QoS guarantees. ATM technology has four main characteristic that differentiate it from other network technology. First; ATM networks transport traffic in the form of cells; secondly ATM network form virtual circuits using these cells; third, ATM technology uses a multiplexing technique call asynchronous time division multiplexing (ATDM); and fourth, it is designed to support the concept of QoS [2].

The first characteristic of ATM technology is that cells are the basis for multiplexing, switching, and transport. Cells are small fixed sized packets, so ATM can be regarded as a sort of packet based network technology. Fixed sized packets are chosen, as they would enable simple and cheap equipment design, including large and fast switches. The reason for choosing a small packet cell size of only 53 byte was to accommodate voice traffic. If the packet size is large, due to the small size of compressed voice sample, unacceptably large packet utilization delay would occur while filling its payload with voice. This topic will be discussed in chapter II.

The second main characteristic is the use of virtual circuit (VC). ATM is a connection oriented method that transfer service information through the establishment of VCs. A connection identifier is assigned whenever a VC is established and removed when the connection released. This identifier, located at the header of each cell, contains information on the virtual connection a utilized for the multiplexing and switching of the cell. One additional characteristic is that the two tier virtual circuit architecture is used, with the lower layer comprised of VCs, and the upper layer being comprised of virtual paths (VPs). These two tier architecture are seen as a way of aggregation. By aggregating many VCs and VP connection, the
switches in the core network could be built in a simpler manner, yet enable faster operation.

The main reason for adopting virtual circuits is that it is the easiest way to enable easy implementation of QoS guarantee. Since the virtual circuit architecture would set up connections along the path before data transport, it would be easy negotiate and allocate resource along the network path. This contrast to the connectionless packet network architecture that has no obvious means of guaranteeing such resource allocation.

The third characteristic of ATM is that it makes use of ATDM. ATDM is a TDM technique that is based on fixed sized time slots, but the time slots are allocated to users not on a periodical basis but on an occasional basis. The ATM cells of various users are chosen, one at one time, to be transported over the time slots, but no user is guaranteed a fixed time slots on irregular basis. The main advantage of ATDM is that it enables an efficient use of transport and switching resources by allowing intelligent multiplexing of user traffic. A fourth characteristic, ATM has been designed to support QoS. This is basically enabled by the combination of therefore mentioned characteristics; virtual circuits result in fixed routes through the network, allowing easy network resource allocation. The use of fixed sized cells enables easy network resource allocation. Additionally, designing hardware with sophisticated resource management schemes is simpler due to the fixed size. The use of ATDM enables flexible allocation of network resource, allowing the support of a much more varied and flexible QoS infrastructure.

1.3 ATM Architecture

This approach permits each layer to evolve while maintaining consistent interface layer. Several configurations are possible. The use of the ATM technologies improves routing and quality aspects. If such technologies are adopted, the layers are
configured as shown in figure 1.2 Peer-to-peer communications refer to layer in one entity exchanging information with a corresponding layer in another entity.

The AAL user is the higher layer signaling for broadband. The AAL user can be the Digital Subscriber Signaling System No.2 (DSS2) for access signaling or the Broadband ISDN User Part (B-ISUP) for inter-nodal signaling.

The ATM Adaptation Layer (AAL) bridges the function of the ATM layer and the higher layers. The functions performed by the AAL depend upon the type of user and application. The functions are grouped into services that are offered to the higher layers.

The ATM layer provides a service independent transport function. Any requirement that are specific to a particular application is covered by the higher layer functions. The ATM layer transports information supplied by the higher layer in packets of data called cells. ATM standard define a fixed size cell with a length of 53 byte comprised of 5-byte header and a 48-byte payload as shown in figure 1.3. This information is transported transparently by the ATM layer, without any processing of the information field.

The Physical Layer defined the physical characteristics of the transmission links and corresponds to the Physical Layer in the Open System Interconnection (OSI) model. The architecture for the Network Node Interface (NNI ) and User Network Interface (UNI) are based on the same model, but they differ to take account of the varying nature of access and inter-nodal signalling.

Each cell consists of a Header and Information Field. The Header identifies cells belonging to the same Virtual Channel. The information Field contains the data to be transferred. Cell sequence integrity is maintained over the Virtual Channel.

The signaling for ATM is carried on the ATM layer connection that is separate from the user information.
1.3.1 Cell Structure

Follow the figure 1.3, each ATM cell comprises header of five octets and information field (payload) of 48 octets. The format of the header is different for the UNI and NNI. The combination of Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI) forms the Routing Field and is used to route cells through the network. There are 24 bits available at the UNI and 28 bits available at the NNI for Routing Field.

The Generic Flow Control (GFC) Field consists of four bits and is used to assist in the control of information flow and ATM connection queues. The Payload Type (PT) Field consists of three bits and indicates the form of data being carried.

The Cell Loss Priority (CLP) Field is a bit that indicates the priority of the cell. In abnormal network conditions, lower priority cells are discarded before higher priority cells. The Header Error Control (HEC) Field is a means of checking the validity of the header. The receiving node or user can perform several modes of error detection and correction under fault conditions.

Figure 1.2: Architecture for broadband signaling [3]
1.4 Wireless ATM (WATM)

A significant development emerged in the 1900’s, which is the area of wireless personal communication. Wireless communication networks have been growing rapidly in recent years and will extend the service from conventional voice and data to a wide range of multimedia services including voice, data, text, video, graphics, and so on. The number of mobile phone subscribers reaches one million by the year 2010 and surpass fixed phone lines. No bounded offered and freedom, who is using mobile devices such as phones, television and wireless computer.

The capability of ATM network to provide a large bandwidth and to handle multiple QoS guarantees can be realized by preparing effective traffic and effective bandwidth management mechanisms. Traffic management includes congestion...
control, call admission control, and virtual circuit routing. In this project, we concentrate at the call level at the call admission control.

An important issue in the selection of congestion control scheme is the traffic pattern. In Constant Bit Rate (CBR) and Variable Bit Rate (VBR) services the traffic parameters are described in term of peak cell rate, cell delay variation and burst length. The congestion control for CBR and VBR services is administrated through admission control and bandwidth allocation.

If the WATM cannot deliver the resource demand by the connection request, the request will be rejected at call setup time. Voice and video or real time service are examples of source that requires guaranteed traffic service. The remaining bandwidth not used by guaranteed bandwidth services must be shared fairly among all active users by using non real time service as Available Bit Rate (ABR) service such as data, or best effort service.

Designing a high speed wireless network architecture requires careful consideration of the type of services to be supported, the mobility profiles of users, the communication and computation ability of mobile host, the needs for internetworking, the availability and limitations of wires technologies.

1.4.1 Why WATM

The growth of wireless communications paired with the rapid developments in asynchronous transfer mode (ATM) networking technology signals the start of a new era in telecommunications. The growth of cellular radio communications in the past decade has been remarkable. The number of cellular users has exceeded all predictions on cellular use. Demand for cellular communications has placed a heavy demand on the capacity of wireless/air interfaces and the network resources available.
The success of cellular mobile communications has spurned the telecommunications industry to push the implementation Personal Communications Services (PCS). PCS will provide voice, text, video and data. As a result, the demand for higher transmission speed and mobility is even greater. Since the beginning the concept of ATM is for end-to-end communications (i.e. in a WAN environment).

The communication protocol will be the same (i.e. ATM), and companies will no longer have to buy extra equipment (like routers or gateways) to interconnect their networks. Also, ATM is considered to reduce the complexity of the network and improve the flexibility while providing end-to-end consideration of traffic performance. That is why researchers have been pushing for an ATM cell-relay paradigm to be adopted as the basis for next generation wireless transport architectures.

There are several factors that tend to favor the use of ATM cell transport for a personal communication network. These are:

- Flexible bandwidth allocation and service type selection for a range of applications.
- Efficient multiplexing of traffic from bursty data/multimedia sources
- End-to-end provisioning of broadband services over wireless and wired networks.
- Suitability of available ATM switching equipment for inter-cell switching.
- Improved service reliability with packet switching techniques
- Ease of interfacing with wired B-ISDN systems that will form the telecommunications backbone.

In general, internetworking may always be seen as a solution to achieve wireless access to any popular backbone network but the consequence, in this case, is a loss of the ATM quality of service characteristics and original bearer connections. The more internetworking there is in a network, the less harmonized the services
provided will be. Therefore, it is important to be able to offer appropriate wireless extension to the ATM network infrastructure.

One of the fundamental ideas of ATM is to provide bandwidth on demand. Bandwidth has traditionally been an expensive and scarce resource. This has affected the application development and even the user expectations. So far, application development has been constrained because data transmission pipes cannot support various qualities of service parameters, and the maximum data transmission bandwidth that the applications have to interface with is relatively small. Finally, ATM has removed these constraints. Bandwidth has become truly cheap and there is good support for various traffic classes. A new way of thinking may evolve in application development.

The progress towards ATM transport in fixed networks has already started and the market push is strong. It can be expected that new applications will evolve that fully exploit all the capabilities of the ATM transport technology. The users will get used to this new service level and require that the same applications be able to run over wireless links. To make this possible the wireless access interface has to be developed to support ATM quality of service parameters.

The benefits of a wireless ATM access technology should be observed by a user as improved service and improved accessibility. By preserving the essential characteristics of ATM transmission, wireless ATM offers the promise of improved performance and quality of service, not attainable by other wireless communications systems like cellular systems, cordless networks or wireless LANs. In addition, wireless ATM access provides location independence that removes a major limiting factor in the use of computers and powerful telecom equipment over wired networks.

1.4.2 Challenges in WATM

A typical reaction to the concept of wireless ATM is to question the compatibility of several aspects of the ATM protocol and the wireless channel. First,
considering the fact that ATM was designed for the media whose bit error rates are very low (about $10^{-10}$), it is questioned whether ATM will work at all in the highly noisy wireless environment. The environment in question is a multi-access channel that may also be time varying. Second, the wireless channel is an expensive resource in terms of bandwidth, whereas ATM was designed for bandwidth-rich environments. Every ATM cell carries an overhead of about 10%. This is considered too high in a wireless environment where bandwidth is precious. In addition, the potential need to transmit single ATM cell means the system should be capable of transmitting individual cells.

However, the physical layer overhead associated with the transmission of individual cells, due to channel equalization and timing can exceed leading to inefficiency which may outweigh the advantages of the wireless access. Supporting wireless users in an ATM network presents two sets of challenges to the existing ATM protocols. The First set includes problems that arise due to the mobility of the wireless of the users. The second set is related to providing access to the wireless ATM network.

### 1.4.2.1 Challenges Related to the Mobility of Wireless Users

The ATM standards proposed by the International Telecommunication Union (ITU) are designed to support the wire line users at fixed locations; on the other hand, wireless users are mobile. Current ATM standards do not provide for any provisions of support of location lookup and registration transactions that are required by the mobile users. They do not support hand-offs and rerouting functions that are required to maintain connectivity to the backbone ATM network during a move. If a wireless user moves while he is communicating with another user or a server in a network, the network may need to transfer the radio link of the user between radio access points in order to provide seamless connectivity to the user.

The transfer of a user’s radio link is referred to as hand-off. During a hand-off event, the user’s existing connection may need to be rerouted in order to meet delay,
quality of service or cost criteria or simply to maintain the connectivity between two
users or between a server and a user. Since the existing ATM protocols are designed
for wire line networks with fixed users, support for rerouting of existing user is not
included in ATM standards. Rerouting is critical to wireless networks which need to
maintain connectivity to a wireless user through multiple, geographically dispersed
radio access points. In mobile networks, as end points move, segments of
connections have to be turndown and re-establishes. Maintaining cell sequence and
connection QoS while performing hand-offs are important requirements in Wireless
ATM networks.

1.4.2.2 Challenges Related to Providing Access

ATM key benefit of a wireless network is providing tether less access to the
subscribers. The most common method for providing tether less access to a network
is through the use of radio frequencies. There are two problems that need to be
addressed while providing access to an ATM network by means of radio frequencies:

A. Error Performance of the Radio Link:

ATM networks are designed to utilize highly reliable fiber optical or very
reliable copper based physical media. These physical media have very low
probability of bit error and hence ATM does not include error-correcting mechanism.
In order to support ATM traffic in a wireless ATM network, the quality of radio links
needs to be improved by error correction and detection.

B. Multiple Access for Wireless ATM Networks:

A wireless ATM network needs to support multiple traffic types with different
priorities and quality of service guarantees. A Medium Access control protocol that
supports multiple users, multiple connections per user and service priorities with
quality of service requirements must be developed in order to maintain full
compatibility with existing ATM protocols. This medium access protocol needs to
make the maximum use of shared radio resource and needs to achieve full utilization of radio frequencies in a variety of environments.

1.5 Objective

The main objective is designed the resource allocation scheme that can give high (QoS) performance in terms of probability of call lost and handover call. It means performance in term of probability of new call, handover call dropping for real time connections and the probability of overload for non-real time.

1.6 Scope

1.1 Concentrate on the performance of WATM at call level probability of new call, probability of handover call, call delay and non real time service.
1.2 Improves the schemes allowing without reserve channel and with reserved channels for handover call and allowing queuing for new calls include non real time request.
CHAPTER II

LITERATURE REVIEW

2.1 Description of WATM Concepts

WATM network same as mobile communication network and illustrated figure 2.1. The WATM network used fully ATM protocol at the data link layer. Interface between networks defines as:

- Between WATM terminal and WATM terminal adapter call WUNI
- Between WATM terminal adapter and radio port call RAL
- Between radio port and mobility enable ATM switching is MNNI
- Between mobility enable ATM switching and ATM network is NNI
- Between ATM network and ATM host is UNI

Figure 2.1: WATM Reference Model
2.1.1 WATM Protocol Architecture

The wireless ATM was introduced in a lot of recent paper in [15, 13, 12]. Sometimes, wireless ATM is simply considered as a wireless access to ATM system [15], whereas other paper consider it will provide end to end ATM in making the continuity of ATM services everywhere, from the laptop computer to the office [14]. In this latter case, a wireless ATM network is described as a communication system where the base stations will be interconnected within a hierarchical ATM switching network as an illustrated in figure 2.2a, 2.2b and figure 2.3. Two kinds of switches are working in the ATM backbone: the classical switches for routing data in the fixed part and the switches connected to BS for routing data in the radio channel. The BS covers a specific area which represents a cell in the wireless network.

Figure 2.2a: ATM Protocol Architecture Implementation
Figure 2.2b: WATM Protocol Architecture

Figure 2.3: Architecture of WATM network
The radio channel is composed of two frequencies as a pair channel illustrated figure 2.4:

- The uplink (the link between the MT and the BS)
- The downlink (the link between the BS and the MT)

To provide ATM services continuity, the ATM cell will be based on the ATM protocol stack with specific extensions for the wireless environment. A specific error control field is then added in the radio channel to provide error recovery. In contrast to wired ATM links, wireless channels are today’s characterized by very low bit rate and high and time varying error rate. As a consequence, wireless ATM will have to improve the transmission quality and to take care to the bandwidth management in order to support the various service classes specified by the ATM.

Figure 2.4: Representation of downlink and uplink, a paired channel.
2.1.2 Bandwidth Sharing for Multimedia

Nowadays, spread spectrum techniques are widely used in PCS because it enables higher capacity. As wireless data applications will need high bit rate in future WATM network. TDMA is however preferable to CDMA since CDMA, strong synchronization make prohibitive the chip cost for rate higher than 25 Mbps. In this project, consider the bandwidth unit as the time slot of TDMA frame, as pictured in figure 2.5 and the time slot is able to transmit or receive exactly one data cell.

Wireless access channel is multi access channel where multiple users, ignoring among each other will compete for the same resource. As for LAN technology, a lot of MAC protocols has been proposed and analyzed in the literature [14]. Such a MAC protocol has to be designed in such way that mobiles share efficiently the limited communication bandwidth while maximizing the delay experienced by mobiles.
According to a usual classification, five protocol classes are identified: fixed assignment, random access, demand-assignment, and adaptive strategies and mixed modes. It is demonstrated that their performance depend both on the environment and on the traffic characteristics: fixed assignment techniques (TDMA, FDMA) are better for continuous traffic like voice and random access (Aloha, CSMA) is more suitable to bursty traffic like data transfer. Since both of these techniques have their disadvantage like poor channel utilization versus non-determinism, a lot o variants have been proposed to improve performance in various environments. As wireless ATM will combine a lot of types of traffic, thus, it is not possible to serve bursty data traffic in the same manner as stream voice traffic and mixed access control is required.

To provide acceptable end to end ATM performance, bandwidth allocation has to satisfy various QoS parameters such as cell delay variation and cell loss rate, and to support various services such as constant, variable, available and unspecified bit rate. In multi service’s environment, these techniques are interesting because of their adaptively as they allow to process separately each kind of service, to allocate bandwidth depending on the state of the network, to give priority to some service over the others. Furthermore, bandwidth allocation to mobile users request is processed in the same manner as to the request coming from the fixed network.

Multiservices Dynamic Reservation TDMA (MDR-TDMA) in order to provide bandwidth sharing to various multimedia services with high channel utilization, while maintaining a reasonable QoS level on each service. The dynamic allocation is based on a statistical algorithm in which available capacity is prorated among demand based on Usage Parameter Control will discusses at chapter 2.4.2. It allows one also to take into account real time data traffic by using a time of expiry based queue service scheme.

The advantage of this mechanism is that a guaranteed amount of bandwidth is always provided to voice traffic while a minimal amount of resource is also guaranteed to data traffic with the possibility of using all slots in the absence of voice calls. Furthermore, resource allocation to data traffic is really adapted to the available
resource. The major drawback is the difficulty to determine the boundary in order to have a good compromise between data packet delay and voice call blocking.

2.2 ATM Service Classes

The ATM Forum committee tries to integrate all the major traffic classes of the fixed network in WATM network while. In the same time, guaranteeing their quality of service requirement. The following service classes have to be integrated.

2.2.1 Real Time Service

The most distinction among applications concerns the amount of delay and the variability of delay, referred to as jitter that the application can tolerate. Real time application typically involves a flow of information to a user that is intended to produce that flow at a source. For example, a user expected a flow of audio or video information to be presented in continuous, smooth fashion. A lack of continuity or excessive loss results in significance loss quality.

Applications that involve interaction between people have tight constrains on delay. Typically, any delay above a few hundred milliseconds becomes noticeable an annoying. Accordingly, the demands in the ATM network for switching and delivery of real time are high. The types of real time and non real time connection are defined.
2.2.1.1 Constant Bit Rate

The CBR service is perhaps the simplest service to define. It is used by applications that require a fixed data rate that is continuously available during the connection lifetime and a relatively tight upper bound on transfer delay. CBR is commonly used for uncompressed audio and video information. Examples of CBR application include video conferencing, interactive audio (telephony), audio/video distribution (television) audio/video (video on demand, audio library).

2.2.1.2 Real Time Variable Bit Rate (rt-VBR)

The rt-VBR category is intended for time sensitive applications; that is, those requiring tightly constrained delay and delay variation. The principle different between application appropriate for rt-VBR and those appropriate for CBR is that rt-VBR application transmits at an arte that varies with time. Rt-VBR source can be characterized as somewhat bursty. For example, the standard approach to video compression results in a sequence of image frame of varying sizes, because real time video allows the network more flexibility than CBR. The network is able to statistically multiplex a number of connections over the same dedicated capacity and still provide the required service to each connection.

2.2.2 Non Real Time Services

Non real time service are intended for applications that have bursty traffic characteristic and do not have tight constrains on delay and delay variation. Accordingly, the network has greater flexibility in handling such traffic flows and can make greater us of statistically multiplexing to increase network efficiency. The non real time service a further categorized as non real time variable bit rate (nrt-VBR), unspecified bit rate (UBR), and available bit rate (ABR).
2.2.2.1 Non Real Time Variable Bit Rate (nrt-VBR)

For some non real time applications, it is possible to characterize the expected traffic flow so that the network can provide substantially improved quality of service (QoS) in the areas of loss and delay at cell level. Such applications can use the nrt-VBR service. With this service, the end system specifies a peak cell rate, a sustainable or average cell rate, and measure of how bursty or clumped the cells. With this information, the network can allocate resource to provide relatively low delay and minimal cell loss. The nrt-VBR service can be used for data transfer that have critical response time requirement. Example includes airline reservations, banking transactions and process monitoring.

2.2.2.2 Unspecified Bit Rate (UBR)

At any given time, a certain amount of the capacity of an ATM networks is consumed in carrying CBR and the two types of VBR traffic. Additional capacity is available for one or both of the following reasons:

a. Not all of the total resources have been committed to CBR and VBR traffic.

b. The bursty nature of VBR traffic means that at some times less than the committed capacity is being used.

This service suitable for applications that can tolerate variable delays and some cell losses, which is typically true of TCP based traffic with UBR, cell are forwarded on first in first out (FIFO) basis using the capacity not consumed by other services: both delays and variable losses are possible. No initial commitment is made to UBR source and no feedback concerning congestion is provided; this is referred to as best effort service. Example of UBR applications include, text/data/image transfer, messaging, distribution, retrieval, remote terminal.
2.2.2.3 Available Bit Rate (ABR)

Bursty applications that use a reliable end to end protocol such as TCP can detect congestion in a network by means of increased round trip delay and packet discarding. However TCP has no mechanism for causing the resources within the network to be shared fairly among many TCP connections. Further, TCP does not minimize congestion as efficiently as possible using explicit information from congestion nodes within the network.

To improve the service provided to bursty sources that would otherwise use UBR, the ABR service has been defined. An application using ABR specifies a peak cell rate (PCR) that it will use and minimum cell rate (MCR) that it requires. The network allocates resources so that all ABR applications receive at least their MCR capacity.

Any unused capacity is then shared in a fair and controlled fashion among all ABR sources. The ABR mechanism uses explicit feedback to sources to assure that capacity is fairly allocated. Any capacity not used by ABR sources remains available for UBR traffic. An example of an application using ABR is LAN interconnection.

2.3 ATM Traffic Characteristics

ATM network is designed to support many types of traffic simultaneously, including real time application such as video, voice and non real time such as data. Even though both applications are transmitted in one network, each of the different traffic is handling depending on the characteristics and requirement of the application.

For example, real time video traffic must be delivered within minimum variation in delay. Hence, the ATM forum has defined the following characteristic:
2.4 WATM Traffic Control

The purpose of traffic control is to optimize network resources, provide QoS at call (i.e. size of bandwidth for non real time connection) and cell level (i.e. cell delay and cell loss) for existing connections and avoid or limit congestion. To meet this objective, ITU-T and the ATM forum have defined a collection of traffic control functions that operates across a spectrum of timing intervals. Table 2.1 list these functions with respect to the response times within which they operate.

<table>
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<tr>
<th>Response Time</th>
<th>Traffic Control Functions</th>
<th>Congestion Control Functions</th>
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<tbody>
<tr>
<td>Long Term</td>
<td>• Resource management using virtual paths</td>
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<td>Connection Duration</td>
<td>• Connection admission control (CAC)</td>
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<tr>
<td>Round trip Propagation</td>
<td>• Fast resource management</td>
<td>• Explicit forward congestion indication (EFCI) and ABR flow control</td>
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<tr>
<td>Cell Insertion Time</td>
<td>• Usage parameter control (UPC)</td>
<td>• Selective cell discard</td>
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<tr>
<td></td>
<td>• Priority control</td>
<td></td>
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<tr>
<td></td>
<td>• Traffic shaping</td>
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</tbody>
</table>
- Cell insertion time: Functions at this level react immediately to cells as they are transmitted.
- Round trip propagation time: At this level, the network responds within the life time of a cell in the network and may provide feedback indications to the source.
- Connection duration: At this level, the network determines whether a new connection at given QoS (at cell level) can be accommodated and what performance level will be agreed too.
- Long term: These are controls that affect more than one ATM connection and share established for long term use.

The essence of the traffic control strategy is based on determining whether a given new ATM connection can be accommodated and agreeing with the subscriber on the performance parameters that will be supported. In effect, the subscriber and the network enter into a traffic contract where the network agrees to support traffic at a certain level of performance on this connection, and the subscriber agrees not to exceed traffic parameters limits. Traffic control functions are concerned with establishing this traffic parameter and enforcing them. Thus, they are concerned with congestion avoidance, if traffic control fails in certain instances, then congestion may occur. At this point, congestion control functions are invoked to respond to and recover from the congestion.

2.4.1 **Connection Admission Control (CAC)**

The Connection Admission Control performed by the network to determine whether a new arrival call request should be admitted to the network or not. One of the main considerations here is that the QoS at cell level for the existing connections are no under circumstances jeopardized. A connection request for a given call is accepted only when sufficient resources are available to carry the new connection.
through the whole network as its requested QoS at cell level while maintaining the agreed QoS at cell level of already established connections in the network.

During the connection establishment procedure at call set-up phase, the following information, embedded in a traffic contract specification, has to be negotiated and agreed between user and network to enable CAC to make reliable connection acceptance/denial decisions [14].

- Specific limits on the traffic volume the network are expected to carry in terms of well-chosen traffic descriptors.
- A requested QoS class at cell level expressed in terms of cell transfer delay, delay jitter and cell loss ratio
- A tolerance to accommodate cell delay variation introduced by for example the Terminal Equipment (TE) or the Customer Premises Equipment (CPE), which may alter the negotiated limits of the expected traffic volume.

This information may be negotiated during the lifetime of the connection at the request of the user. The network itself may limit the frequency of these renegotiations. CAC schemes are currently not standardized and are at the discretion of the network operators. For WATM an implementation of the CAC function might incorporate Radio Signal Strength (RSS) or Bit Error Rate (BER) measurements and this is illustrates in figure 2.6.

Figure 2.6: RSS measurement location
2.4.2 Usage Parameter Control

When a connection has been accepted by the connection admission control function, the usage parameter control (UPC) function of the network monitors the connection to determine whether the traffic conforms to the traffic contracts. This associates at cell level. The main purpose of UPC is to protect network resources from an overload on one connection that would adversely affect the QoS (at cell level in terms of cell rate and cell burst size) on the other connections by detecting violations of assigned parameters and taking appropriate actions.

In the fixed ATM network, the UPC functions are located at the entry point of a connection into a network, which is in the most cases the port of an ATM switch (See figure 2.7).

In wireless ATM the UPC functions have to be integrated with the radio resource management [15]. User has to be prevented from using more radio bandwidth than specified in the traffic contract and the WATM medium access control (MAC) layer provides this functionality. For a centralized radio resource management in the base station, the UPC functions can be integrated with MAC layer scheduling of the bandwidth requested by the wireless ATM terminals. If the users request to transmit more cells than allowed, the exceeding requests will not be
granted by the WATM base station controller. Another solution is to reject the request in the wireless terminal. In this case the location of the UPC functions has to be changed to the wireless terminal.

2.5 The Traffic Contract and Optimization of Bandwidth Allocation

When an ATM end user requests a connection by signaling or subscription, along with this connection request, comes source traffic descriptor and QoS parameters at cell level. This will characterize quantity and quality of the traffic to be transported over the connection and police the traffic as it enters the network.

Hence, this agreement between the ATM end user and the network is called traffic contract. The ATM network however, will support the connection and associated QoS at cell as well as call level as long as the ATM end user stay within the agreed upon parameters.

For example, if a non real time call need a 64kb/s connection, it will request for 64 kb/s bandwidth from the network. If the amount of bandwidth in the network is sufficient enough to provide 64kb/s for the connection, the network will accept the call. Otherwise, the network will negotiate for a lower amount of bandwidth, for example 40kb/s for VBR connection. However, with less bandwidth provided, it will jeopardize the QoS in term of cell lost, hence call lost.

Amount of bandwidth allocated for any call of performance at each transmission call setup in all multiplexes located on the selected virtual path. The algorithm in charge of admitting or rejecting the user requests to establish a transmission path through ATM network is calling the Call Admission Control (CAC). This allocation is carried out after each transaction of call admittance or rejection, which include bandwidth renegotiation.
The purpose of admission control based on allocating bandwidth is to establish a fair blocking against any new coming call and to ensure that the resource or bandwidth available to each admitted connection remains sufficient to meet the performance objectives. Indeed, any new connection can only be admitted to the network if sufficient resources are available. The bandwidth to be allocated to a connection in each multiplexes will be expressed in [bits per second] and defined as a fraction of the transmission bit rate which empties the multiplexing queue with a exponential service time.

2.6 Mobility Management In WATM

The three basic issues are:

1. Location Management
2. Connection Management
3. Handoff Management

2.6.1 Location Management

Location management has two functions, tracking the position of a mobile and handling queries regarding the location of a mobile which in Public Land and Mobile Network (PLMN). The location information is usually maintained and used by the network to locate the user for call routing purpose. (See figure 2.8)

In location management, each base station keeps broadcasting on a periodic basis, the cell sites identities on the broadcast control channels of the cell site under its coverage. The mobile terminal within each cell site keeps monitoring such information. As change in location are detected, they each report the new location to
the base station, which route it to the Visitor Location Register (VLR) of the Mobile Switch Control (MSC) to which it is connected. The VLR in turn, send the location information to the users Home Location Register (HLM) directs the old VLR to delete the old visiting location of the mobile from its database and also sends a copy of the user’s service profile to the new VLR.

![Location Management Routing](image)

Figure 2.8: Location Management Routing

### 2.6.2 Connection Management

Connection management deals with connection routing and QoS maintenances. ATM is a connection oriented technology with a connection establishment phase prior to data exchange and connection release phase after data
exchange. In wireless networks, as end points move, segments of connection need to be turned down and reestablished. Meanwhile, maintaining call sequence and connection QoS while performing handoff are important requirements in wireless ATM.

In this chapter, different types of connection rerouting methods for handover will be discussed. Some schemes require segment tear down and set up during handover. Some other schemes minimize handover time and packet loss during handover by using pre-establishing virtual connection tree approach.

The latter schemes avoid segment tear down and setup during handover by setting up extra segments of a connection tree during the connection setup phase. In WATM network, it completely avoids this problem of having to tear down and setup segment to reduce delay in data transmission. Hence, handover rerouting methods such as pre-establishing and increment extensions are widely proposed for WATM network.

2.6.3 Handover Management

Handover management quickly and timely handover has crucial effect on how users perceive QoS. However, handover strategies should not be complicated. Wireless ATM provides mobility and freedom or borderless area to its users. Figure 2.9 shows how handover becomes, as a user moves from one base station to another base station. The connection of that particular mobile terminal (MT) needs to handover to a new base station in the current cell or another adjacent cell site and the process continues as the user moves against.

WATM environment uses small cell sites (Pico cell: ~ 1 km, and micro cell: ~ 0.5 km), compared to the amount of handover experienced by the MT in macro cell site (>1.5 km) to accommodate its growing population of mobile users. This phenomenon affects the performance of the systems especially during handover event. In no-protection call handling schemes, handover request are treated the same
manner as the new call incoming, so the probability of handover dropping equal the probability of blocking of the new call incoming calls.

However from the MT point of view, the dropping of handover calls is less than the new call attempt. Hence some measurement must be taken to avoid too many terminations of these handover calls. In this case handover management of the network has to determine a way to reduce unnecessary termination.

There are various technical approaches that have been used to improve the probability of handover dropping in WATM mobile system. The most common used technique is the prioritizing the handover calls. There are two types of prioritizing scheme: queuing of the handover request and reserve channel for handover.

![Figure 2.9](image)

Figure 2.9: In progress cellular calls are transferred from one cell base station to another cell base station while maintaining the call’s connection to the cellular system. All call handovers are coordinated by the mobile switching center (MSC).

### 2.7 QoS in ATM Traffic

A three level hierarchical model with different time scales can describe ATM traffic. In this work the QoS to be upgrade will be at call level as well as cell level. The call level has a typical time seconds up to hours (for example a long lasting
video conference). Their burst level is in millisecond ringed up to seconds and the cell level is microseconds range. These levels have different impact on ATM network.

**Call level:** The level that controls by the connection admission control (CAC) algorithms, which decide whether a connection can be accepted or not. If too many connections are rejected, the utilization of the network will degrade. Additional network resource should be deployed. The number of end user who wants to transmit information concurrently will have to determine the time scale at this level.

**Burst level:** This level can be controlled by fast resource reservation mechanism and adaptive flow control protocol such as ABR. It determines the large buffers needed for non real time connection. Furthermore, specific CAC algorithm may allow connection to be statistically multiplied at burst level. Burst periods are generated by end application, which transmit more or less information during the connection lifetime.

**Cell level:** This level can be controlled by mechanism such as policing, priority controller or traffic shaping. At multiplexing points, the traffic pattern at cell level determines the buffer size required for real time connections. Cell arrival variations are caused by the principles of asynchronous multiplexing of cells.

### 2.7.1 Guaranteed QoS in WATM Network

Qos for mobile are different from that of fixed stations in an ATM Network. More precisely, mobile QoS is an augmented version of fixed station QoS. M-QoS is comprised of the traditional QoS found in wired ATM networks augmented with wireless and handover call QoS. For each major traffic type such ABR, VBR, CBR and UBR, QoS parameters such as end-to-end delay and desired specified by the application.
The specification must then be mapped into wired, wireless and handover QoS requirements. The definitions of wired, wireless and handover QoS are as follow:

a) Wired QoS refers to the traditional quality of service supported by the wired ATM networks, such a link delay variation and bandwidth.
b) Wireless QoS refers to the QoS parameters associated with the wireless link, such as link delay, error rate, and channel reservation.
c) Hanover QoS refers to the performance parameters associated with the handover, such as handover blocking probability and cell loss.

2.7.2 Mobile Quality of Service (M-QoS)

Mobile communication in wireless ATM-LAN environment can take place into two different forms: mobile to fixed and mobile to mobile. For either of these communications, a mobile connection is a concentrate of wireless and wired link. The undoubtedly defines M-QoS parameters to attributes of wireless and wired links. The term wired QoS, includes traditional QoS parameters used in ATM network while wireless QoS is related to bandwidth variation owing to mobile host roaming, and link delay.

While the switching, buffering, and traffic handling policies built into ATM switches have a direct influence on the wired QoS, the channel access, buffering, and traffic handling policies implement at the base station have a similar influence on the wireless QoS.

An essential aspect of M-QoS is handover QoS. As a mobile host migrates from one wireless cell to another, a connection can be forced to terminate due to the failure of a mobile handover. Such failure are characterized by the handover blocking probability and they can result in an inability to provide the desired wireless QoS in the new wireless cell, a failure of a base station, an inability to handover in time, a failure of radio links and inability to fulfill the desired wired QoS in the wired links.
Hence, handover blocking can regarded as another M-QoS metric. In additional, the need to support seamless handover has led to a new M-QoS metric, which is known as handover urgency or deadline. The ability to control handover urgency can be impacted by a number of factors, for example: speed of mobile host moving, size of overlapping wireless cells, direction migration, and types of ongoing mobile traffic.

Handover urgency is interpreted in term of handover request priority. A mobile handover request consists of connection identifier, an intra cell or inters cell flag and a handover urgency flag. The handover QoS parameters related to per virtual circuit are handover urgency, probability of handover blocking, a traffic disruption period during handover, cell loss during handover, ATM cell sequencing during handoffs, and the speed of handover operations. Multiple virtual circuits related parameters include the percentage of calls degrade, QoS consistency during handovers and the virtual circuit degradation priority.

2.8 Handover Mechanism in WATM

After a connection has been established to a mobile endpoint, handover protocols are necessary to reroute existing active connections when the endpoint moves to a different radio port. We try explaining with two ways differentiated method from another researcher. First part discusses at topic 2.8.1, 2.8.2 and 2.8.3 and the second part at the 2.8.4. However, the both topics are same, but represented with two different ways. First part is; a handover procedure consists of three basic steps:

- Selecting a crossover switch and setting up a new connection segment from the crossover switch to the new radio port.
• Shifting the data path at the crossover switch to the new connection segment, and optionally achieving a zero cell loss shift of the data path.
• Extending the connection data path from the new radio port to the mobile terminal after it moves under the radio port’s coverage area. Below, we consider different methods of executing handover.

2.8.1 Partial Path Rerouting Scheme

This approach is based on removing a part of the existing connection and adding a new sub path from the point of detachment (the “crossover” point). In Figure 2.10, let the mobile terminal move from radio port BS1 to BS2. Then the portion of the existing connection from the “handover switch” S1 to BS1 is deleted, and the remainder of the original connection path is extended to BS2.

A key issue in this scheme is the selection/discovery of the crossover point. One method is to iteratively probe each switch on the existing connection path such that the rerouted path through the switch (to the new radio port BS2) satisfies the QoS of the original connection path. As discussed later, if “zero-cell-loss” handover is desired, the path rerouting scheme requires cells to be buffered at BS1 and transferred to BS2 before the mobile terminal can begin to receive cells at BS2.

2.8.2 Path Splicing

In the “path splicing” method for handover, crossover switch discovery is initiated from the destination radio port rather than the mobile terminal’s current radio port. In Figure 2.10 the mobile endpoint sends a handover request to its current radio port, BS1, asking for a handoff to BS2. BS1 forwards this message to BS2 using a pre-established handover control VC between adjacent radio ports. BS2 first executes a local call admission control to check if the wireless link can accommodate the connection to be handed off. It then initiates a connection setup to the remote
endpoint of the original connection, using the QoS parameters supplied within the handover request message. This setup is eventually guaranteed to encounter a switch (S2) on the existing connection path, which is considered the crossover switch. This switch will then splice the new path segment (S2–BS2) onto the existing connection path (between the caller and the crossover switch).

2.8.3 Path Extension Scheme

Another simple approach to handover consists of extending an active connection from the terminal’s current radio port to the next one. The idea here is that after handover, the new connection consists of the existing connection from the source to BS1 followed by an additional sub path (the “extension”) from BS1 to BS2.

This approach offers the advantage that no crossover switch discovery phase is required, and the existing path is maximally reused. However, the extended path also increases the end-to-end delay and reduces network utilization since the extended path may traverse the same link more than once (i.e., loops may be created). A possible solution is to optimize the path lazily after completion of handoff:

- “Looping” points (switches) are detected during the path extension process.
- Loops are removed by sending a specially defined operations, administration, and management (OAM) cell from a looping point.
- While the OAM cell is traversing the loop, incoming cells on the connection are buffered locally at the looping point and then forwarded on the optimized path after the loop has been removed.
Figure 2.10: Scheme of handover

2.8.4 Connection Rerouting Methods

Handover in a Wireless ATM network requires changes in virtual connections. Several handover schemes have been proposed, such as connection extension, full establishment, partial re-establishment and multicast establishment.

The full establishment approach requires the setup of a completely new connection between the end users. Figure 2.11 represents the path evolution for a
mobile terminal that moves within three cells in a network adopting the full establishment approach.

When the handover occurs, the connection extension scheme extends the path between the users by adding one hop that provides the connection from the old base station to the new base station through the fixed network. Figure 2.12 shows the path modifications needed to follow the roaming terminal.

The multicast establishment approach pre-allocates resources in the network portion surrounding the cell where the user is located. When a new mobile connection is established, a set of virtual connections, named a virtual connection tree, is created. The latter reaches all BSs managing the cells toward which the mobile terminal might move in the future. Thus, the mobile user can freely roam in the area covered by the tree without invoking the network call admission control during handover.
The allocation of the virtual connection tree may be static or dynamic during the connection lifetime. Figure 2.13 shows a multicast establishment, assuming that the mobile terminal roams through three cells. Acampora et al propose building a virtual connection tree covering base stations in a local area, with virtual circuits pre-established from the root of the tree to each base station. In this case, a handover only involves the switching to an already established virtual circuit.

Figure 2.12: Path evolution example adopting connection extension approach
In the partial re-establishment handover scheme (see Figure 2.14), a new path is established from the new base station to a node in the original connection path. Hence this method requires the discovery of the crossover switch (COS), the setting up of the new partial path and the tearing down of the old partial path. The purpose of crossover switch discovery is to locate a suitable COS so that a new partial path can be established from the new base station (BS) to the COS.
2.9 The Virtual Connection Tree (VCT)

A wireless ATM network system consists of the wireless and the fixed part. A virtual connection tree (VCT) will have the root, the branch and the leaves. The root is the root switch (RS) of the network; the branch is the connection in the fixed network and the leaves are the base stations. To support the frequent amount of handovers in a small cell site, the network control processors (NCP) have to be invoked every time handover happens.

The NCP is a processor that decides on the route and address number for each route, it also decides on the acceptance or rejection of a call coming into the network. When a mobile terminal (MT) enters a connection network, the NCP will allocate a
multicast route in the fixed and wireless environment. This is to ensure the mobile terminal is free to handover to any base stations in the connection network. Multimedia devices, mobile phones, wireless notebooks are some of the examples of wireless terminals.

The fixed portion includes the base station, sub ATM switch (sub SW), ATM switch (SW) and the root switch (RS). The adjacent group cell sites are included in a cell cluster, the root switch and the NCP will control each cell cluster (See figure 2.15). This architecture reduces call set up and routing load on the NCP. As a mobile enters a VCT, it is free to handover to any base station in the VCT without involving the NCP, thus it will decrease the process of need to set up and tear down the connection during handover event. The fixed portion of the connection will be maintained as the user moves within the virtual connection network. Thus QoS at cell level as well as call level are guaranteed to the connection in the network throughout the connection lifetime. A virtual connection tree has a control mechanism to monitor the network performance.

2.9.1 Implementation of VCT

For new incoming call, when a MT is accepted to a VCT, a collection of virtual circuit numbers (VCN) is assigned to the call. Each of these VCN defines a path between the root of a connection tree and a distinct base station within the neighboring mobile access region. A given MT selects its base station from among those in its connection tree by transmitting ATM cells with the appropriate VCN. Hence, an ATM cells associated with a given tether less connection will eventually flow through the root of the connection tree and from there, to its destination by means of a fixed wired path if the connection is to a hard wired network port or to a root of another mobile user.
For handover call, when a mobile user already admitted to a virtual connection tree wishes to handover to another base station in the same virtual connection tree, it simply begins to transmit ATM cells with the connection number assigned for use between itself and the new base station. Using the pre-established path between the new base stations at the root of the tree that mobile's ATM cells will flow to the root, across the fixed portion of the network, and to their ultimate destination. In this way; the call processor is not involved in the handover.

In ATM different type of service classes will be identify by the cell header where it carries sufficient information for the network to route and distinguish each cell to the destination port and end users.
2.9.2 Why Virtual Connection Tree

As cell site size diminishes, the rate of cell handover escalates. Hence, a strategy is needed to quickly reroute each handed-off virtual connection to its new cell site and maintain QoS guarantees without burdening the admission controller. The VCT totally decentralizes handover operations, with each mobile responsible for managing its own handover events without requiring intervention of the NCP. The admission controller is involved only in establishing new virtual connections and in handling connections to adjacent trees, but not in handing off among cell sites within the same tree.

At the time a mobile connection is admitted to a tree, a set of virtual connection networks (VCNs) are assigned to that connection. Each VCN defines a path between a specified base station and the root of the tree. A given mobile selects its base station from among those in its connection tree merely by transmitting ATM cells with the appropriate VCN. In this way, an ATM cell associated with a given virtual connection will eventually flow through the root of the connection tree and from there, to its destination by means of fixed wired path if the connection is to a hard wired network port, or to the root of another connection tree, the VCN used to route the cell along the appropriate wired path.

Thus, creation of VCT and assignment of a set of VCNs for each connection wastes neither channel capacity nor wire link capacity. Also, the call processor is involved only at connection setup, but is totally uninvolved in processing handoffs within a connection tree, which are handled entirely by handed-off mobile itself in a totally distributed fashion. The call processor becomes involved only in handovers to different VCT; since the geographical area served by a given connection tree, it may include many base stations and becomes quite large, the frequency of call processor involvement remains low and any potential handover problems that might have resulted from the use of small cell sites is avoided.
2.10 Queuing in WATM Network

In ATM wireless network, when very small cell site are used, the handover producers affect the performance of the system. In no protection call handling schemes, handover request are treated the same manner are the new incoming calls, so the probability of handover dropping equals the probability of blocking of the new call incoming calls.

However, from the MT point of view, the dropping of handover call is less desirable than the new call attempt. In order to decrease the probability of blocking of the on going calls, handover protection schemes had been proposed. Two commonly used handover prioritization a scheme:

- Guard channel that reserved a percentage of channels exclusively for handover calls.
- Queuing handover request with priority or without priority.

Handover protection scheme, in general decrease the probability of handover at the cost of increasing of new incoming call and reducing the total admitted traffic. The use of guard channels is a means of improving the probability of successful handover by simply reserving a fixed or dynamically adjustable number of channels exclusively for handover requests.

For wireless ATM network that support real time and non real time connection, both of traffic have to be taken in account to ensure the smooth flow of integrated transmission.

The handover area is the where the ratio of the received power level from the current and target base station is between the handover and the receiver threshold. The MT, whose power level is closest to the receiver threshold, has the higher priority to handover first. However, if the entire channel in the target cell is occupied after the timeout expired; the call will then be terminated. Instead of using priority queuing, used first in first out (FIFO) queuing for handover call with directly retry.
This proved to upgrade the probability of handover dropping. This scheme might be proper for data transmission but for CBR application, delay will cause termination of the going calls.

Since only non real time connection can tolerate delay, we conclude that queuing for handover calls is possible ATM wireless network. However for real time connection queuing scheme can be applied to the new incoming call with considerate amount of queue. A certain number of channels are reserve for handover and the new call incoming calls are allowed to queue with infinite queue. This thesis investigated how efficient the prioritizing scheme with guard channels, queuing and bandwidth allocation for new incoming calls, handover request and data request using integrated WATM wireless with VCT.
CHAPTER III

RESOURCE ALLOCATION SCHEMES

3.1 Multiservice Resource Allocation Policies (MRAP)

This policy is very important when researcher simulate the network and to know how the network serve the request. It is discussed [13]. The MRAP policy aims at allocating time slots between multi services request either on the uplink channels or on the downlink channel. To achieve each service specific QoS requirement, a higher priority level is given to CBR and VBR traffic while only best effort service is offered to ABR and UBR services. The difference between ABR and UBR is that the flow control is provided to ABR traffic in order to meet the desired cell lose rate.

When the CBR and VBR connection request arrives, it takes the first time slot available in the TDMA frame. If no time slot is available the connection is dropped out. If one CBR and one VBR connection request compete for a single time slot, then the CBR connection giants it. The CBR traffic is composed of new connection and handover request. In this thesis, researcher give priority is given to the handover call over the new call, because it is more important to keep a connection in the cell.

During the connection, CBR traffic is continuous whereas VBR traffic is variable alternating idle periods with active periods. MRAP takes benefit from this
burstyness as data traffic can use the idle time slots reserved to the VBR connections. As soon as a VBR connection becomes active again, it retrieves its resource.

3.2 Model of Allocation Bandwidth Policy

The model of this policy is represented in figure 3.1. The resource is represented by a set of multiple server; each server acting as a time slot of the TDMA frame. The time slot duration gives the time unit of the discrete time model. Then consider the arrival rate, departure rate and serving rate during the particular time slot.

Among the five traffic classes described above, researcher divided to two main part classes in the model, each one being subdivided into sub classes (real time and non-real time).

A) The connection request (real time)
   • The new call connection request (CBR)
   • The handover call request (CBR)
   • The VBR connection request

B) The data rate request (non-real time)
   • The ABR data cell request
   • The UBR data cell request

The new call or handover call (CBR and VBR) connections keep their slots during all the communication duration; connection duration defines the services time of these service classes according to a Bernoulli process.
3.3 Rerouting Handover Signaling

When setting up a point-to-point call the ITU-T Recommendations Q.2931 and Q.2961 shall apply. When a VCN value is assigned the allocation is in both directions even when the connection is unidirectional (has zero backward bandwidth). The VPN value cannot be used for another connection until the call is cleared [17]. Selected extensions to ATM signaling syntax (e.g. Q.2931) together with a new wireless control (meta-signaling) sub layer is needed to support mobile ATM users at the connection control level [18]. Specific mobile network functions requiring signaling/control support include address registration for mobile user, wireless network QoS parameter specification/renegotiation and handover.

A basic issue to be addressed is that of delaying wireless control and ATM signaling to support relatively transparent handover operations. In general, this will require rerouting of the ATM connection from the base station to another base station, while also moving wireless network state information to smoothly resume communication with a minimum of cell loss. An experimentally validation example of possible signaling and meta-signaling exchanges during an ATM handover shown
in figure 3.2a. In this example, new signal (handover request) is used to initiates transfer of group of VC’s from one base station to another.

**Figure 3.2a: Example of handover control procedure in wireless ATM**

Whilst in VCT architecture as a mobile terminal (MT) admitted to a connection tree, the call setup procedure is executed in two steps:

1. The fixed portion of the virtual connection is established between the root of the tree and the appropriate fixed point of the wired network (i.e.: the fixed user terminal or the root of a destination tree).

2. Within the connection tree, two sets of connection numbers are assigned to that mobile connection (one in each direction) with one member of each set used to define a path from the root to one of the leaves, and the
routing table of the switches within the connection tree is appropriately updated to include the new connection numbers.

Illustrated in figure 3.2b, base station X is used by MT to make connection to the fixed VC and VC _X had been assigned for it as the connection number. When the MT wishes to handoff to base station Y (which is in the same connection tree), it simply begins to transmit the packet with the connection numbers assigned for use between itself and the new base station.

Figure 3.2b: Call set up for handover event

Using the pre-established path between the new base stations. The mobile packets will flow to the root and across the fixed portion to the destination. In this case the NCP will not be involved. Routing is essential to accommodate the new arrival and handoff calls. The function of NCP is to setup the route and ensure the new established route maintain acceptable QoS at cell level as well as call level to both the wireless connection and pre-existing connection sharing links of the new
route. To execute handoff (to another tree), the NCP must first ensure that the new wireless connection does not overload the base station and then it will create a link between the MT and the new base station. Figure 3.2b, it shows that after setting up has been acknowledge the NCP will not be invoke until the call needs to handoff to another VCT.

3.4 Description of the Markov Chain

The Markov Chain used to determine the stationary distribution and to compute the reward of this system defined by:

- The blocking probability of the new call request when it arrive a new connection when no free slots are available.

- The blocking probability for handover request when no reservation channel and with reservation channel.

- The blocking probability for new call and handover request with queuing.

- The cell loss probability for non-real time (ABR and VBR) traffics when it arrive a new cell and the buffer is full.

The Markov Chain theory will discuss in the next topic.
3.5 Call Admission Control (CAC)

Call admission control (CAC) is the first part of defense for a network in protecting itself from excessive loads. In wireless ATM (WATM), the CAC function is located in the base station. Usually, the CAC function has to be called before a handover procedure can be completed. In order to guarantee a certain degree of mobility for the wireless ATM terminals, the CAC function needs to reserve some of the available bandwidth for handover calls. A handover call drop result an interruption of on going call and occurs if he bandwidth required by a connection cannot be supplied in the new cell site location.

In virtual connection tree (VCT), the CAC function is a mechanism that limits the incoming calls into the network in order to protect the already connected call from being interrupted. An accepted connection request by a mobile terminal (MT) is by the network controller will be given pre established virtual connections to all the base stations in the tree so that the MT can freely move within the tree. The VCT network also monitors the existing connections in the network in order not to exceed the pre determine threshold limit. Moreover, it uses a scheme that gives priority to the handover calls where a percentage of channels are reserved for handover calls whenever the channels in the base station a busy.

3.5.1 CAC for Real Time Connection Without Reserve Channels

In VCT, adjacent cell sites are grouped into a cell cluster and B is the number of base stations in a cell cluster. This model is illustrated in figure 3.3. It is assume that all cell sites have a similar size and shape, which is considered hexagonal. The probability of a MT from a source cell site to handover to another cell site within a cluster is:
\[ P_{HO} = \frac{1}{(B - 1)} \quad (3.1) \]

Figure 3.3  The model of WATM network

Here it is assumed that a target cell site is the cell site the MT intended to move to and a source cell site is the cell site the MT is currently located. Hence, the probability that user will handover to the entire cell site is equal to 1:

\[ \sum_{i=1}^{B-1} P_{HO} = 1 \quad (3.2) \]

To ensure high efficiency of the limited bandwidth for a large number of mobile users, smaller cell site such as pico and micro cell sites are used. They are in the range of 0 – 500m and less than 1 km radius respectively. As the user moves from a source cell site to other cell sites in the cell cluster, it will visits a number of cell sites before coming back to the source cell site. The probability that the mobile user visit \( k \) cell sites is given by:

\[ P_V(k) = P_{HO}(1 - P_{HO})^{(k-1)} \quad \text{for} \quad k = 1, 2, 3, \ldots, \quad (3.3) \]
The real time connection for CAC without reserve channels is said to using the M/M/1 queuing system. The birth death process of M/M/1 queuing system is illustrated as state transition rate diagram of Markov Chain in figure 3.3.

![State transition rate diagram for M/M/1 queuing system.](image)

From the state transition rate diagram in figure 3.4, the equilibrium different equation for state \( k \) can be obtained:

\[
\lambda_{k-1} P_{k-1} + \mu_{rt} P_{k+1} = \left( \lambda_{nt} + \mu_{nt} \right) p_k
\]  

(3.4)

Where \( \lambda_{rt} \) is the call arrival rate, \( \mu_{rt} \) is the call departure rate and \( p_k \) is the probability that the system has \( k \) members.

Hence, the probability of finding \( k \) mobile terminals in a system given

\[
P_k = P_0 \frac{\lambda_{nt}}{\mu_{nt}} \prod_{i=0}^{k-1} \frac{\lambda_{nt}}{\mu_{nt}}
\]  

(3.5)

where

\[
P_0 = \frac{1}{1 + \sum_{k=1}^{\infty} \prod_{i=0}^{k-1} \frac{\lambda_{nt}}{\mu_{nt}}}
\]  

(3.5)
Applying equation 3.5 to M/M/1 queuing system, we have:

\[ p_k = p_o \prod_{i=0}^{k-1} \frac{\lambda_{st}}{(i+1)\mu_{st}} \quad k \leq m \]  

(3.6)

where;

\[ p_o = \left( \sum_{k=0}^{m} \left( \frac{\lambda_{st}}{\mu_{st}} \right)^k \frac{1}{k!} \right)^{-1} \]

Hence the probability of new call blocking and the handover call dropping for CAC without reserve channel is given by:

\[ P_{NC+H} = \frac{\left( \frac{\lambda_n}{\mu_n} \right)^m}{m!} \left/ \sum_{k=0}^{m} \frac{\left( \frac{\lambda_n}{\mu_n} \right)^k}{k!} \right. \]  

(3.7)

In mechanism without reserve channel (WORC), where all the channels in the network are for new call and handover calls, probability of blocking for new call and handover calls will depend on the arrival rate into the network. This is because both types of calls are treated the same manner, as illustrated in figure 3.5.
One of the QoS parameters required in a network is a low probability of handover drop in order to provide better service to the network customers. Hence CAC with reserved channels for handover calls or WRC is presented.

### 3.5.2 CAC for Real Time Connection With Reserve Channels

To solve the problem of handover call droppings when all channels are busy in CAC, we introduce the reserve channel for handover calls. When the new call incoming enter the network, it can be blocked if the number of connections in the network exceeded the pre-determined threshold, $N_{rt}$ and cannot support any additional connections. Here, $N_{rt}$ is given by:

$$N_{rt} = mgB$$  \hspace{1cm} (3.8)

where $B$ is the number of base station in a cell cluster, $m$ is the number of call a base station can handle and $g$ is the percentage of reserve channels.

The range of $g$ between 0 to 100 percent, however this figure is not reasonable in real network because the block incoming call too much. The percentage of reserve channels must be not be less then 20% of the total channel capacity. When a communicating MT moves out of source cell site, the handover attempt is generated. If the network not enough channels in the target cell site to accommodate the handover call, the call will be terminated. This is called handover dropping of the on going call, which is the interruption of successfully connected call. Since a termination due to failure of handover call attempt is more obtrusive than the new call blocking, in this system the handover call may be given priority access to the channels in the following way.
Each base station reserves a certain percentage of the total channel allocated to it for handover call. A new incoming call that originate can use any idle channels, provided that they are less than \( N_{rt}/B \) channels, which are used in the base station at that particular time. If \( N_{rt}/B \) or more channels are already being used, the new call will be blocked. A handover call however, will be able to use any idle channels of the \( m \) channels in the base station. In this way the handover calls have the priority of accessing channels in any base station in the network.

Applying equation 3.7, the probability of having new call blocked for CAC WRC is given by

\[
P_{\text{new}}(f) = \left( \frac{\lambda_{rt}/\mu_{rt}}{f!} \right)^f \left[ \sum_{k=0}^{N_{rt}/B} \left( \frac{\lambda_{rt}/\mu_{rt}}{k!} \right)^k \right]
\]

(3.9)

where \( \lambda_{rt} \) is the new call arrival rate and \( \mu_{rt} \) is the new call departure rate for WRC.

However, considering the real phenomena, call departures from a cell site consist of calls that desire to be terminated and calls that desire to handover to another base station. Thus, the actual or effective call departure rate \( \mu_{rt} \) is higher that the natural call departure rate \( \mu_{rt} \). If observation is made in a call site, there are changes of call coming in and going out of the cell site as follows figure 3.6.
Figure 3.6: The changes of calls going in and coming out of one cell site to another.

- $\mu$ - The completion of calls in the source cell site
- $\lambda$ - The new incoming calls in the source cell site
- $\gamma_{rt}$ - The departure rate of handover calls from the source cell to the target cell site
- $\gamma_a$ - The arrival of handover calls in source cell site

where

\[ \gamma_a = \gamma_{rt} \left(1 - P_{HDH}\right) \]  

(3.10)

The effective departure rate of wireless calls from a source cell is given by:

\[ \mu_{ert} = \mu_{rt} + \gamma_{rt} P_{HDH} \]  

(3.11)

The effective arrival rate is given by:

\[ \lambda_{ert} = \lambda_{rt} \left(1 - P_{NCH}\right) \]  

(3.12)

where $P_{HDH}$ and $P_{NCH}$ are the probability of handover dropping and the probability of new call block, respectively for CAC with reserve channel.

\[ P_{NCH} = B \frac{\left(\frac{\lambda_{rt}}{\mu_{rt}}\right)^{Nrt}}{Nrt!} \sum_{k=0}^{Nrt} \frac{\left(\lambda_{rt}/\mu_{rt}\right)^k}{k!} \]  

(3.13)

Assumed that the mobile movement patterns of handover were independent and similar to the new incoming calls to any cell site. The model of aggregate calls arrival rate to any cell site in the cell cluster is assumed as Poisson Process. When a call departs from a cell site while still communicating, not all of these calls will be successfully handed off to the target cell sites.
Assumed that the handover call going out of the network was statistically identical to the process of handovers going into the network. Again, by using figure 3.4 and equilibrium equation can be found. Note that to find the probability of handover calls that will probably be dropped; actual load that had been accepted to a base station will only be considered.

Hence, with \( m \) number of channels at a base station the probability of handover calls dropping is given by:

\[
P_{\text{HDH}} = B \frac{(\lambda_{\text{m}}/\mu_{\text{m}})^{m}}{m!} / \sum_{k=0}^{m} \frac{(\lambda_{\text{m}}/\mu_{\text{m}})^{k}}{k!}
\]

However, during heavy traffic, \( \gamma_{rt} \) tends to infinity, the cell cluster will always have \( N_{rt} \) number of calls. Hence, the Erlang load at each base station in the network is \( N_{rt}/B \), the probability of handover in heavy traffic:

\[
P_{\text{HDH}} = \lim_{\lambda \to \infty} P_{\text{HDH}} = \frac{(\lambda_{\text{m}}/\mu_{\text{m}})^{m}}{m!} / \sum_{k=0}^{m} \frac{(\lambda_{\text{m}}/\mu_{\text{m}})^{k}}{k!}
\]

\[
= \frac{(N_{rt}/B)^{m}}{m!} / \sum_{k=0}^{m} \frac{(N_{rt}/B)^{k}}{k!}
\]

The block diagram in figure 3.7 illustrated the calls accepted and rejected in a base station using CAC with reserve channel.
3.5.3 CAC for Non Real Time Connection

The ATM network support multiclass applications. Since it accepts different classes of service, each of them is characterized by a specific transmission quality of service. For non real time connection, data are transmitted despite of real time communication. Hence, they can queue at any base station whenever there are no available channels. Generally a queuing network consists in principle of finite number of nodes or base stations, where the queue capacities are supposed to be finite or infinite.

In an open queuing systems, the arrival process entering the network are assumed to be independent of each other and independent of the queuing service processes. All entering calls are supposed to leave the network eventually. This is opposite to the closed system where a fixed number of calls flow in the system from node to node without either entering or departing from the network. A VCT network that support non real time connection and do not limit incoming calls will have problems regarding overload in the network. To solve this problem a system using VCT architecture model with control mechanism of incoming calls is introduced.

Assume that the system is homogeneous for non real time connection in this project to be an open queuing system of M/M/\infty with the probability of \( j \) connections in a system is [19]:

\[
p(j) = \left( \frac{\lambda_{\text{rt}}}{\mu_{\text{rt}}} \right)^j \frac{e^{-\lambda_{\text{rt}}/\mu_{\text{rt}}}}{j!}
\]  

(3.17)

In VCT network, cell sites are grouped into a cell cluster. In order to limit the number of calls coming into the network a threshold is set to control the maximum number of connections, \( N_{\text{rt}} \), in a network with \( B \) number of base stations in a cell
cluster. Hence, the VCT system is said to be the truncation of the open queuing system (see figure 3.8).

![Figure 3.8: The truncated open queuing system where B is the number of base station in a cluster.](image)

Here, $Z \subseteq \mathcal{Z}$ having the equilibrium distribution of:

$$P_A(j) = \frac{p(j)}{\sum_{k \in A} p(k)} \quad j \in Z$$

(3.18)

$\mathcal{Z}$ is a state space in reversible Markov process. The call arrivals are in Poisson process with arrival rate $\lambda_{nrt}$ per base station and call duration is exponential distribution with mean $1/\mu_{nrt}$. In this homogeneous system, $N_{rt}$ is the maximum number of non real time connection allowed to a network and $s$ id the number of connections to a base station.

The process is illustrated as state transition rate diagram of Markov Chain in figure 3.9. The equilibrium distribution of the truncated process is a conditional probability distribution that the original process is in state $j$ given that it is somewhere in $Z$. 

\[\text{ATM Network}\]
State space $Z$ containing all $\{j_1, j_2, \ldots, j_b\}$ with:

$$\sum_{i=1}^{B} j_i \leq N_{nrt} \quad (3.19)$$

Where $i = 1, 2, 3 \ldots B$.

Assumed that $j_i$ is the number of connection at base station $i$. Also, assumed that $Z_k$ is the subspace of $Z$ with:

$$\sum_{i=1}^{B} j_i = k \quad (3.20)$$

Where $i = 1, 2, 3 \ldots B$. 
Figure 3.10: Illustrated of the truncated open queuing system

The truncated open queuing open system is shown in figure 3.10. Expand equation (3.18) using equation (3.17) with:

\[ e^{-\rho} = \frac{1}{1 + \sum_{k=1}^{N_{eq}} \rho_k / k!} \]  \hspace{1cm} (3.21)

Where \( \rho = \frac{\lambda_{nrt}}{\mu_{nrt}} \)

Hence, the probability of \( s \) connection at base station is:

\[
P_{nrt}(s) = \left. \left[ \left( \frac{\lambda_{nrt}}{\mu_{nrt}} \right)^{s / s!} \sum_{k=0}^{N_{eq}} \left( \frac{\lambda_{nrt}}{\mu_{nrt}} \right)^k / k! \right] \right|_{s=0,1,2,\ldots} (3.22)
\]

Since \( \{j_1, j_2, \ldots, j_b\} \) for all base station in the cell cluster is equal to less then \( N_{nrt} \), equation (3.22) can be simplified to:

\[
P_{nrt}(s) = \frac{\left( \frac{\lambda_{nrt}}{\mu_{nrt}} \right)^{s / s!} \sum_{k=0}^{N_{eq}} \left( \frac{(B - 1) \lambda_{nrt}}{\mu_{nrt}} \right)^k / k!}{\sum_{k=0}^{N_{eq}} \left( \frac{B \lambda_{nrt}}{\mu_{nrt}} \right)^k / k!} \hspace{1cm} s = 0,1,2,\ldots (3.23)
\]

However, in heavy traffic condition the equilibrium distribution of number of connection in any base station approaches a Binomial distribution:
The QoS performance for non real time connection is the overload probability at a base station. This is because a call that enters a congested area will queue until there are available channels for it to continue transmitting and the situation will continue until the base station is overloaded.

The overload probability is the summation of the probability that there are more than $m$ connections in a network with limitation of $N_{\text{nrt}}$ and is given by:

$$P_{ov} = \sum_{s=m+1}^{N_{\text{nrt}}} P_{\text{nrt}}(s)$$  \hspace{1cm} (3.25)

### 3.6 The VCT Model

The study will concentrate on the architecture of wireless ATM environment using virtual connection tree (VCT). The VCT architecture and its advantage has already discussed and mentioned before this. This architecture grouped adjacent base stations into a cell cluster. In reality, the cell site shape is affected by the geographical structure of an area but here it is assumed that all cell site shape is hexagon.

Every cell sites has at least one base station and each cell site is given a set of channel frequencies. Adjacent cell site have different set of frequencies, however the same set of frequency can only be use after a certain distance. This is call frequency reuse.
In the VCT network, there are two types of calls coming in and going out, the new incoming calls, which are the newly initiated call and the handover calls, which are the already connected calls. It is assumed that the call arrivals are Poisson process and call duration is exponentially distributed. Also, it is assumed that the handover rate of each mobile terminal, from any cell site to another, experiences the same rate of arrival of handover calls. Furthermore, all wireless real time connections are homogenous (use 64kbps for voice connection) and any base station can support up to \( m \) number of calls.

A predetermined threshold, \( N_{nrt} \) is set to limit the number of connection within the VCT network. This is the CAC function in VCT where priority of access channels is given to the handover calls in the network and the new incoming call will be blocked whenever all of the channels in the base station are busy.

Each base station in a cell site reserved a certain percentage of the total channels allocated to it for handover calls. These reserved channels vary with the capacity of population of an area where \( N_{nrt} \) is the allowed number of connection to a network. The percentage amount of the reserved channels however is fixed for a particular area and will be upgraded or otherwise, yearly depending on request. The speed mobility of a mobile terminal will range in between 20 to 40 kmph.

3.7 Simulation of VCT Network Model

COMNET III a performance analysis tool for computer and communication network. Based on description of a network, its control algorithm and workload, COMNET III simulates the operation of the network and provides measure of the network performance. No programming is required. Also, using COMNET III, network descriptions are created graphically through a highly intuitive interface that speeds up model formulation and experimentation. COMNET III is integrated into a single windows package, which performs all functions of model design, model
execution and presentation of result. A model is built and executed in straight forward steps. COMNET III can be used to model Wide Area Network (WANs) and Local Area Network (LANs).

A simulation of the VCT model for real time connection was analyzed using discrete simulation COMNET III release 2.1 from ASCII software. The system is assumed have 32 numbers of channels at the base station where each channel is 64kbps and the number of base station is one. However the inter arrival time for this simulation is 0.0025 to 0.008 seconds, call duration is 0.007 minute. 64kbps for one channel when the call request, the system can be support to 32 channels, the line is used is E1 line because 2Mbps request for 32 line at a base station.

However, the some parameter can be change for the simulation like a inter arrival rate, departure rate, call rate and so on.

Figure 3.11: Assuming 7 Base station, 20 connection per base station, Arrival rate 40-120 call per seconds, departure rate 2.5 call per seconds and handover departure rate 10 call per seconds
The graph showing the comparison between simulation and theoretical method is plotted in figure 3.11. The call arrival rate was varied from 40-120 calls per seconds. The call departure rate is fixed to 2.5 calls per seconds. The simulation shows an increment of probability of call block when increase traffic. The result of the probability of call block is comparable to the theory (the difference in the last data is due to random process in discrete event simulation).

Figure 3.12: Assuming 7 Base station, 20 connection per base station, Arrival rate 40-120 call per seconds, departure rate 2.5 call per seconds and handover departure rate 10 call per seconds
3.8 Mathematical Analysis of VCT Network Model

The mathematical analysis used Matlab as the analysis tool. Matlab is a combination of powerful technical computing environment centered on real math notation; flexible and full feature technical word processor. With Matlab, the task of performing computations and documenting them are integrated into one seamless process, resulting in substantial increases in productivity.

Matlab on screen interface is a blank worksheet on which equations, graph data and functions are entered with annotate with text, anywhere on the page. The mathematical expression in Matlab looks the way it would seem in text or a reference.

Matlab can handle complex number as seamlessly as its handles units, and standard engineering and math functions are also built into the environment. The analysis for this chapter was done using Matlab simulation tool where the parameter or input to the equation used are the assumptions for that particular situation. The Matlab programs are as attached in appendix.
3.8.1 CAC Without Reserved Channels for Real Time Connection

Assuming that the network has a group of 7 base stations with 20 connections per base station, the new incoming calls arrival rate varies from 15 to 100 calls per seconds. The handover call rate is 2 calls per seconds with call departure rate at 1 call per seconds.

![Graph showing the probability of new call block and handover call drop against Erlang Load per cell site for CAC WORC. Assuming 7 Base station, 20 connection per base station, Arrival rate 15-100 call per seconds, departure rate 1 call per seconds and handover departure rate 10 call per seconds.](image)

Figure 3.13: The probability of new call block and handover call drop against Erlang Load per cell site for CAC WORC. Assuming 7 Base station, 20 connection per base station, Arrival rate 15-100 call per seconds, departure rate 1 call per seconds and handover departure rate 10 call per seconds.

Using the assumptions in equation (3.7), figure 3.13 shows the probability of new incoming call and handover calls increase with Erlang load for WORC.
However when channels are busy both types of calls will be block. It means that handover calls will be block when it enters congested area in WORC. This is because the total channels are for handover calls and new incoming calls.

### 3.8.2 CAC With Reserved Channels for Real Time Connection

A comparison is done between WRC and WORC. Assuming that the network has group of 7 base stations with 20 connections per base station and $q$ equal to 0.085. The new incoming calls arrival rate varies from 15 to 100 calls per seconds.

Figure 3.14: Probability of new call block and handover call dropping for WORC and WRC against Erlang load per cell sites. Assuming 7 Base station, 20 connection per base station, Arrival rate 15-100 call per seconds, departure rate 1 call per seconds and handover departure rate 10 call per seconds.
Using the above assumptions in equation (3.7), (3.13), and (3.15), figure 3.14 shows the performance of WRC and WORC. At normal load, which is less than 3.57 Erlang load per cell sites, both mechanisms gives acceptable rate probability of new call blocking and handover drop but as the load increase higher than 3.57 Erlang load per cell site, WORC cannot seem to maintain the handover calls.

This is shown when the probability to lose call and drop of handover calls in WORC is as same as the probability of new call block in WRC. This is because in WORC, when all of the channels in base station are occupied, the handover calls will be blocked as well as the new arrival calls. However, in WRC priority are given to handover calls during heavy load and this is shown in figure 3.14 where the probability of handover call drop are maintain at a constant value heavy load.

Figure 3.15: Assuming 7 Base station, m=10:20 connection per base station, Arrival rate 15-100 call per seconds, departure rate 2 call per seconds and handover departure rate 10 call per seconds.
The performing of real time connections was analyzed in WRC system. Assuming that call arrival rates vary from 15 to 100 calls per seconds. The handover rate is fixed at 10 calls per seconds. The call departure rate is fixed at 2 calls per seconds. The network has 7 base stations and q equal 0.85. The probability of handover block is assumed 0.01. Using the assumptions in equation (3.13), figure 3.15 shows a graph of probability of call block against Erlang load per cell site at base stations that can supported up to 10,15, and 20 connections (m=10,15,20). A reduction of new call blocking probability was obtained when the number of connections supported by a base station, m, is increased.

For instance 2.142 Erlang load per cell site, the new call blocking probability for a base station with m = 10 connection is 0.02315292726616 but with m = 20 connections per base station the new call blocking is 0.00000171900282. This because when a base station can support more number of connections, more new incoming call will be accepted by the CAC.

Figure 3.16: The probability of handover call dropping against the Erlang load per cell site. Assuming 7 Base station, m=10:20 connection per base station, Arrival rate 15-100 call per seconds, departure rate 2 call per seconds and handover rate 10 call per seconds.
Figure 3.16 shows a graph of probability of handover dropping against Erlang load per cell site at base station that can be support $m$ up to 10, 15, 20 connection per base station. Using the assumptions in equation 3.15 shows that a reduction of handover dropping probability was also obtain when more number of calls can be support by a base station. This is because when a base station can support more number of connections, more reserved channels will be provided to handover calls.

For instance at 2.142 Erlang load per cell site, the new call blocking probability for base station with $m = 10$ connections is $0.25558890540686$, for $m = 15$ connections is $0.05655836586691$ but for $m = 20$ is $0.00430274411494$. This is because when a base station can support more number of connections, more new incoming call will be accepted by the CAC, but it is small if compare without reserved channel.

![Figure 3.17a: Probability of Handover call dropping against the Erlang load per cell site. Assuming 7 Base station, $m = 20$ connection per base station, Arrival rate 15-100 call per seconds, departure rate 5 call per seconds and handover rate 10 call per seconds and $q = 0.85$.](image-url)
Figure 3.17b: Probability of Handover call dropping against the Erlang load per cell site. Assuming 5 Base station, $m = 20$ connection per base station, Arrival rate $15-100$ call per seconds, departure rate $5$ call per seconds and handover rate $10$ call per seconds and $q = 0.85$.

In figure 3.17a and 3.17b a graph of probability of handover call block against Erlang load per cell site is plotted using the assumptions used in the previous graph, in equation (3.15). It is found that the size of cell cluster does not affect the probability of handover dropping. This is due to the constant value of handover call rate through both size of cell clusters and the smaller difference of the reserved channels in both cell clusters.
3.8.3 CAC for Non Real Time Connection

For non real time application, the overload state is considered as a serious issue whenever many calls arrive simultaneously at base station. Certain amounts of bandwidth are assigned to a base station and depending on these amounts the probability of overload state can be found. First, from figure 3.18, using system without control mechanism, the overload probability grow very rapid as the number of non real time connection increase. However, the overload probability will be saturated at the 40 number of connection. This is because in system with control mechanism a threshold is set to limit the incoming calls and this is done to avoid overflow.

Figure: 3.18a: Assuming 7 Base station, m=30 connection per base station, Arrival rate 30 call per seconds, departure rate 1 call per seconds. Equation 3.25

\[ N_{\text{nrt}} = \text{the maximum number of non real time connection allowed to a network} = 80 \]
Figure: 3.18b: Assuming 7 Base station, \( m = 30 \) connection per base station, Arrival rate 30 call per seconds, departure rate 1 call per seconds. Equation 3.25

\[ N_{nrt} = \text{the maximum number of non real time connection allowed to a network} = 70 \]

Now, using system with control mechanism assuming 7 base stations are connected to a VCT, the average call departure is 1 call per seconds. Consider an ideal case where 1 call will have 1 unit of bandwidth (UB), which approximately equals to 64kbps. Here, two situations are analysis where 20 BU and 30 BU are assigned to two base stations. Using the assumptions in equation (3.25), figure 3.18 and 3.19 was obtained and as the results, when the number of connections allowed to base stations is 80, higher probability of overload is obtained and this is because more connection can be connected to the network.
Figure: 3.19a: Assuming 7 Base station, m=20 connection per base station, Arrival rate 30 call per seconds, departure rate 1 call per seconds. Equation 3.25

\[ N_{nrt} = 70 \]

Figure: 3.19b: Assuming 7 Base station, m=20 connection per base station, Arrival rate 30 call per seconds, departure rate 1 call per seconds. Equation 3.25

\[ N_{nrt} = 80 \]
3.9 Summary

In this chapter, the performance of real time and non real time connection had been analyzed theoretically. First, the real time connection was analyzed using call admission control without reserved channels, WORC. It was found that using WORC, the probability of handover calls is treated the same manner as the new incoming calls. Hence when all of the channels in the base station are busy, the new incoming calls as well as the handover calls will be blocked. Rejection of newly initiated call is not as annoying as the rejection of the already connected calls. These chancels for handover callas are introduce in mechanism with reserved channels WRC.

Using threshold Nrt, the calls coming into the network are limited. The effect of Erlang load and reserved channels to the probability of new incoming call block and handover call analyzed for WRC. As Erlang load increased, the probability of new incoming call block and handover call drop increase because the more traffic coming into the network, the more the chances of calls being rejected due to limited channels. However, by expanding the capacity of base station, the probability of handover call drop is lowered.
4.1 Call Admission Control With Queuing

The reserved channels for handover calls were introduced as the previous chapter. Unfortunately this WRC scheme increase the probability of blocking of new incoming calls by blocking all new incoming calls when the channels are busy. In this chapter queuing for the new incoming call is present at the same time the probability of dropping will be maintained. The call admission control (CAC) with queue allows queuing of the new incoming calls when there are no available channels at the targeted base station. It also has reserved channels for handover calls to decrease the probability of handover drop when channels are busy. Hence, it is called the CAC with queue and reserve channels, WQRC.

In the public switching telephone network (PSTN), the signaling needed for dialing is done on the communication channel. Queuing new incoming calls would therefore result in multiple dialing that would unnecessarily occupy some communication channel. In the cellular system, the dialing (the setup of a call) is done on a separate control channel, which can provide the system with a way of queuing new incoming call without affecting the transmission channels.
In this chapter a method of queuing incoming calls will be analyze using the same virtual connection tree (VCT) as the architecture model. In this scheme, new incoming call requests for connection to a congested network will not be block, but instead they will be queued, waiting for the connection to be accepted to the network. WQRC can be applied to any network despite of their sensitiveness to delay. The queue only involves the request for connection service and not during the call already connected. Such application is propose for wireless telephony and teleconferences systems.

4.2 The Queuing Model

Prioritizing scheme is one of many ways to upgrade quality-of-service (QoS) of network to customers. Though not all real-time connection call can tolerate delay, some (e.g. rt-VBR) may consider a redial scheme to ensure the request will be accepted despite of the waiting time. Assumed that all wireless connections are homogenous real time connections for the all base stations. Figure 4.1 describes the arrival of calls to a network. There are two types of calls; the handover calls with handover call rate of $\gamma$, and the new incoming calls with call rate of $\lambda_{nrt}$.

Figure 4.1: The CAC with queue of the new incoming calls during heavy loads.
Both types of calls have access to all channels. When all of the channels are busy, the new incoming calls will be blocked but the handover calls have priority access to the network. In other words, the new incoming calls will have free access to the system as long as $N_{II}$ channels are available. $N_{II}$ is the pre determined threshold for WQRC, where:

$$N_{II} = g(B_{II}, m_{II})$$ (4.1)

where $B_{II}$ is the number of base station in WQRC and $m_{II}$ is the number of connection a base station can handle. As mentioned the figure of $g$ can be in between 0 to 100% but for a reasonable value in order not to block new incoming call too much, the percentage of reserved channels must not be less than 20% of the total channel capacity. However, if all $N_{II}$ channels are fully occupied the new incoming calls will be put in the queue and will receive service on first in first out (FIFO) basis as soon as less than $N_{II}$ channels are idle.

In a wireless ATM network, setting up of a call is done on a separate control channel. This function however will require a slight modification of the existing connection setup protocol. A MT that wants to initiate a call sends a call request to the ATM network on the control channel, and must wait for an answer from the system. The connection setup protocol will have an expire-time procedure that makes the MT automatically resend the call request after the expire-time period. This will then notify the MT that it has been put in a queue waiting for service. This is illustrated in figure 4.2.

![Figure 4.2: The call set-up procedure prior to connection acceptance.](image-url)
4.3 The Representation of Queuing of Incoming Call

A two-dimensional state diagram of figure 4.3 describes the flow of two different types of calls in a cell cluster. The state diagram is divided into two parts. Part 1 consists of less than $N_{II}$ channels are busy and it is the lower part of the column in the state diagram. This is as same as the pure blocking system. Part 2 is when more than $N_{II}$ server is busy and no more new incoming calls will be allow to be connected. However, priority is given to the handover calls. Assuming the pair $(i_1, i_2)$ denotes the following notation: $i_1$ will be the number of customers in the queue, and $i_2$ will be the number of customers in the service.

![Figure 4.3: Two dimensional state diagram of with queuing reserve channel](image-url)

Figure 4.3: Two dimensional state diagram of with queuing reserve channel
In figure 4.3 $B_{II}m_{II}$ is the maximum capacity the network can handle and $\mu_{II}$ is the call departure rate of WQRC. The state diagram can be derived as the balance equations of $E_1$, $E_2$ and $E_3$:

\[E_1 : (i_1 = 0, 0 \leq i_2 \leq N_{II} - 1) \rightarrow P(0, N_{II} - 1) : \]
\[(i_2 + 1)P(0, i_2 + 1) + (b + c)P(0, i_2 - 1) = (b + c + i_2)P(0, i_2) \] (4.2)

\[E_2 : (i_1 = 0, i_2 = N_{II}) \rightarrow P(0, N_{II}) : \]
\[(i_2 + 1)P(0, i_2 + 1) + Pi_2(1, i_2) + (b + c)P(0, i_2 - 1) = (b + c + i_2)P(0, i_2) \] (4.3)

\[(i_1 = 0, N_{II} \leq i_2 \leq B_{II}m_{II}) \rightarrow P(0, N_{II} + 1) : \]
\[(N_{II} + 2)P(0, N_{II} + 2) + cP(0, N_{II}) = [(b + c) + (N_{II} + 2)]P(0, N_{II} + 1) \] (4.4)

\[(i_1 = 0, i_2 = B_{II}m_{II}) \rightarrow P(0, B_{II}m_{II}) : \]
\[cP(0, i_2 - 1) = (B_{II}m_{II} + b)P(0, i_2) \] (4.5)

\[E_3 : (i_1 \geq 1, i_2 = N_{II}) \rightarrow P(1, N_{II}) : \]
\[(i_2 + 1)P(1, i_2 + 1) + Pi_2(2, i_2) + (b)P(0, i_2) = (b + c + i_2)P(1, i_2) \] (4.7)

\[(i_1 \geq 1, N_{II} \leq i_2 \leq B_{II}m_{II}) \rightarrow P(1, N_{II} + 1) : \]
\[(N_{II} + 2)P(1, N_{II} + 2) + bP(0, N_{II} + 1) + cP(N_{II}) = [(b + c) + (N_{II} + 1)]P(1, N_{II} + 1) \] (4.7)

\[(i_1 \geq 1, i_2 = B_{II}m_{II}) \rightarrow P(1, B_{II}m_{II}) : \]
\[bP(0, i_2) + cP(1, i_2 - 1) = (b + i_2)P(1, i_2) \] (4.8)

where $b = \frac{\lambda_{II}}{\mu_{II}}$ and $c = \frac{\gamma_{II}}{\mu_{II}}$

$E_1$ is as same as the pure blocking system in [20] such that:

\[P(0, i_2) = \frac{a^{i_2}}{i_2!}P(0, 0) \] (4.9)
where $0 \leq i_2 \leq N_I$ and $a = \frac{\lambda_I + \gamma_I}{\mu_I}$

To find the general solution for the state diagram, we need to solve the balance equation. Referring the first column starting from the top:

$$P(0, B_{II} m_{II} - 1) = q_k \quad 0 \leq k \leq B_{II} m_{II} - N_{II}$$

$$P(0, N_{II}) = q_g = \frac{a^{N_{II}}}{N_{II}!} P(0,0) \quad (4.10)$$

To express $P(0, i_2), (0 \leq i_2 \leq B_{II} m_{II})$ in terms of $P(0,0)$, from the balance equation, we have the following relations for $q_k$'s:

$$q_o = q_o \Rightarrow P(0, B_{II} m_{II}) = P(0,0) \quad (4.11)$$

$$q_1 = r + B_{II} m_{II} c^{-1} P(0,0) = \left(B_{II} m_{II} c^{-1} + r\right) q_o \quad (4.12)$$

$$q_2 = \left[(B_{II} m_{II} + 1)c^{-1} + (r + 1) - 2c^{-1}\right] q_{k-1} - \left[(B_{II} m_{II} + 2)c^{-1} - 2c^{-1}\right] q_{k-2} \quad (4.13)$$

$$q_k = \left(\alpha - k \delta\right) q_{k-1} - \left(\beta - k \delta\right) q_{k-2} \quad (4.14)$$

where

$$r = \frac{\lambda_{II}}{\gamma_{II}} = \frac{b}{c}, \quad \beta = \left(B_{II} m_{II} + 2\right) c^{-1}$$

$$\alpha = (B_{II} m_{II} + 1)c^{-1} + (r + 1), \quad \delta = c^{-1}$$

Equation (4.13) however is only valid for $2 \leq k \leq B_{II} m_{II} - N_{II}$ but for all $k \geq 2$ we need to define the sequence into a differential equation and depend transform. So we get $k=1 \ldots B_{II} m_{II}$ [20]:
\[ q_{m^{n-m}-k} = q_o \sum_{j-k}^{\frac{m-n}{2}} C_j \left( \frac{j}{j-k} \right)^{-1-k} \left[ \sum_{i=j}^{\frac{m-n}{2}} \left( B_{n,n} \right)_i \right] c_i \]  \hspace{1cm} (4.15) \\

where \( (x)_k = \frac{\Gamma(x+k)}{\Gamma(x)} \)

Since \( q_o = P(0,B_{n,n}) \) and \( q_g = P(0,N_{n,n}) = \frac{a^{N_v}}{N_{n,n}!} P(0,0) \) as in equation (4.10), equation (4.15) can be written as:

\[ P(0,N_{n,n}) = P(0,B_{n,n}) \sum_{j=N_{n,n}}^{\frac{m-n}{2}} C_j \left( \frac{j}{j-N_{n,n}} \right)^{-1-j} \left[ \sum_{i=j}^{\frac{m-n}{2}} \left( B_{n,n} \right)_i \right] c_i \]  \hspace{1cm} (4.16) \\

Hence, \( P(0,B_{n,n}) \) can be expressed in terms of \( P(0,0) \) and \( P(0,k) \) is used in terms of \( P(0,0) \) in equation (4.15), giving:

\[ P(0,i) = P(0,B_{n,n}) \sum_{j=0}^{\frac{m-n}{2}} C_j \left( \frac{j}{j-i} \right)^{-1-j} \left[ \sum_{i=j}^{\frac{m-n}{2}} \left( B_{n,n} \right)_i \right] c_i \]  \hspace{1cm} (4.17) \\

To simplify the equation, substitute the indexes \( i \) for \( k \) and \( \lambda \) for \( j-i \) in equation (4.17), giving:

\[ P(0,i) = P(0,B_{n,n}) \sum_{j=0}^{\frac{m-n}{2}} C_j \left( \frac{j}{j-i} \right)^{-1-j} \left[ \sum_{i=j}^{\frac{m-n}{2}} \left( B_{n,n} \right)_i \right] c_i \]  \hspace{1cm} (4.18) \\

Let,

\[ \lambda_{n,n}(B_{n,n},i) = \sum_{j=0}^{\frac{m-n}{2}} \left[ \frac{(\lambda+i)\lambda}{2} \right]^{-1-j} \left[ \sum_{i=j}^{\frac{m-n}{2}} \left( B_{n,n} \right)_i \right] c_i \]  \hspace{1cm} (4.19) \\

Hence,

\[ P(0,i) = \frac{c_i}{i^2} \lambda_{n,n}(B_{n,n},i) P(0,B_{n,n}) \]  \hspace{1cm} (4.20)
This implies that $P(0,i_2)$ can be written in terms of $\lambda_m(B_m,i_2)$ and $\lambda_m(B_m,N)$ as

$$P(0,i_2) = \frac{e^{\lambda_m(B_m,i_2)}}{i_2!} \lambda_m(B_m,N) P(0,N) \left[ \frac{N!}{C^{N-i_2}} \frac{P(0,N)}{\lambda_m(B_m,N)} \right]$$

$$= \frac{N!}{C^{N-i_2}} \lambda_m(B_m,i_2) \frac{P(0,N)}{\lambda_m(B_m,N)}$$

(4.21)

The state probabilities of the first column are obtained as function of $P(0,0)$ and are given by $0 \leq B_m - N \leq B_m$:

$$P(0,i_2) = \begin{cases} 
\frac{\alpha^{i_2}}{i_2!} P(0,0) & \text{for } 0 \leq i_2 \leq N_m \\
\frac{\alpha^{N_m}}{i_2!} e^{\lambda_m(B_m,N)} \frac{\lambda_m(B_m,i_2)}{\lambda_m(B_m,N)} P(0,0) & \text{for } N_m \leq i_2 \leq B_m
\end{cases}$$

(4.22)

To find the blocking probability of handover calls, we have to solve the case of $i_2 \geq N_m$ ($i_2$ servers are busy).

$$P_{i_2} = \sum_{i_1=0}^{\infty} P(i_1,i_2)$$

(4.23)

A state diagram involving these larger states (for $N_m \leq i_2 \leq B_m$) is shown in figure 4.4, which is a compressed version of figure 4.3. The diagram is typical classic purely blocking system and the state probabilities are easily expressed in terms of $P_{N_m}$:

$$P_{i_2} = e^{\lambda_m(N)} \frac{N!}{i_2!} P_{N_m}$$

(4.24)
which gives

\[
\sum_{i_1=0}^{\infty} P(i_1, i_2) = e^{(\lambda - \mu) N_\mu} \frac{N_\mu!}{i_2!} \sum_{i_1=0}^{\infty} P(i_1, N_\mu) 
\]

(4.25)

where \( N_\mu \leq i_2 \leq B_\mu m_\mu \). From the column of the system (in the two-dimensional state diagram of figure 4.3) it is found that the outgoing flux (to the right) is balance by the incoming flux from state \( i_1 + 1, N_\mu \). Hence, it can be written as:

\[
P(i_1 + 1, N_\mu) = \frac{b}{N_\mu} \sum_{i_2=N_\mu}^{B_\mu m_\mu} P(i_1, i_2) 
\]

(4.26)

where \( i_1 \geq 0 \)

![State Diagram of Busy Servers](image)

Figure 4.4: The state diagram of busy servers

To solve \( P_{i_2} \), the sum of equation for \( i_2 \geq 0 \) using equation (4.25) and (4.26) has to be computed. For equation (4.26), for all values of \( i_1 \geq 0 \):
In equation (4.26a-c) \( i_2 \) is substituted by \( \lambda \). Substituting equation (4.26c) in equation (4.25) gives:

\[
\sum_{i_1=0}^{\infty} P(i_1, i_2) = \frac{b}{N_{H_H}} \sum_{i_2=N_{H_H}}^{\infty} \sum_{i_1=0}^{\infty} P(i_1, i_2)
\]

and:

\[
:\sum_{i_1=0}^{\infty} P(i_1, N_{H_H}) = \frac{b}{N_{H_H}} \sum_{i_2=N_{H_H}}^{\infty} \sum_{i_1=0}^{\infty} P(i_1, i_2)
\]

\[
:\sum_{i_1=0}^{\infty} P(i_1, N_{H_H}) = P(0, N_{H_H}) + \frac{b}{N_{H_H}} \sum_{\lambda=0}^{N_{H_H}} \left( \sum_{i_1=0}^{\infty} P(i_1, \lambda) \right)
\]

\[
(4.26d)
\]

The double summation in equation (4.25d) represents the sum of all state probabilities, except for case where less than \( N_{H_H} \) servers are busy. These missing probabilities are known as functions of \( P(0,0) \) (see equation 4.21).

For stability, all states probabilities must add up to 1, so:

\[
\sum_{\lambda=0}^{N_{H_H}} \left( \sum_{i_1=0}^{\infty} P(i_1, \lambda) \right) = \sum_{\lambda=0}^{N_{H_H}} P(0, \lambda)
\]

\[
(4.27)
\]

Thus,

\[
P_{i_2} = \sum_{i_1=0}^{\infty} P(i_1, i_2)
\]

\[
= e^{\lambda - N_{H_H} \frac{N_{H_H}}{i_2}} \left[ P(0, N_{H_H}) + \frac{b}{N_{H_H}} \left( 1 - \sum_{\lambda=0}^{N_{H_H}-1} P(0, \lambda) \right) \right]
\]
\[ P(0,0) \left( a \right)^N \left( c \right)^i \frac{c^i}{i_2!} + \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \frac{c^i}{i_2!} \left( \sum_{x=0}^{N_{II}-1} \frac{a^x}{A!} \right) + \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \frac{c^i}{i_2!} \right) \] (4.28)

Also stable system holds that:

\[ \sum_{i_2=0}^{N_{II}-1} P(0,i_2) + \sum_{i_2=N_{II}} P(0,i_2) = 1 \] (4.29)

From (4.9),

\[ \sum_{i_2=0}^{N_{II}-1} P(0,i_2) = P(0,0) \sum_{i_2=0}^{N_{II}-1} \frac{a^i}{i_2!} \]

Hence,

\[ 1 = P(0,0) \sum_{i_2=0}^{N_{II}-1} \frac{a^i}{i_2!} + P(0,0) \left( \frac{a}{c} \right)^N \sum_{i_2=N_{II}}^{N_{II}-1} \frac{c^i}{i_2!} - P(0,0) \left( \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \frac{c^i}{i_2!} \right) \left( \sum_{x=0}^{N_{II}-1} \frac{a^x}{A!} \right) + \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \sum_{i_2=N_{II}} \frac{c^i}{i_2!} \]

Let

\[ A = \sum_{i_2=0}^{N_{II}-1} \frac{a^i}{i_2!} + \left( \frac{a}{c} \right)^N \sum_{i_2=N_{II}}^{N_{II}-1} \frac{c^i}{i_2!} - \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \sum_{x=0}^{N_{II}-1} \frac{a^x}{A!} \left( \sum_{i_2=0}^{N_{II}-1} \frac{c^i}{i_2!} \right) \] (4.30)

or

\[ 1 = [P(0,0), A] + \frac{b}{N_{II}!} \frac{N_{II}!}{c^N} \sum_{i_2=N_{II}} \frac{c^i}{i_2!} \] (4.31)
From this, $P(0,0)$ can be found as:

$$P(0,0) = 1 - \frac{b}{N_H} \sum_{i=0}^{B_{mH}} \left( \frac{c^i}{i!} \right) 4^i$$  \hspace{1cm} (4.32)

The probability of handover blocking is when all of the channels in the network are busy including the reserved channels are given by:

$$P_{HDH_b} = P(0,0) \frac{c^{B_{mH}}}{B_{mH}!} \left( a \right)^{N_a} \left( \frac{b}{N_H} \sum_{i=0}^{N_H} \frac{a^i}{i!} \right) + \frac{b}{N_H} \frac{N_H!}{N_H!} \left( c^{B_{mH}} \right)$$  \hspace{1cm} (4.33)

The probability of delay of the new incoming calls in when all of the $N_H$ channels are:

$$P_d = 1 - P(0,0) \left( \sum_{i=0}^{N_H-1} \frac{a^i}{i!} \right)$$  \hspace{1cm} (4.34)

4.4 Effect Queuing

Comparisons between WQRC and WRC are analyzed. Both schemes have the reserved channels for handover calls where certain percentage of channels is reserved from the total channels. The percentage will depend on the number of population in the particular area. Also, both schemes use the VCT architecture as the network model. To make the comparison, some of the results in chapter III are used. The mathematical analysis in this chapter done using the Matlab tool where the detail this programmed had been explained previous chapter.
Figure 4.5: Handover dropping probability against reserved channels. Assuming 3 base stations, 20 connections per base station, Arrival rate 75 calls per seconds, departure rate 5 calls per seconds and handover departure rate 10 call per seconds.

Using equation (4.33), it is found that with 3 base stations per cell cluster the probability of handover dropping for CAC with queuing is extremely low (see figure 4.5). Hence, to make reasonable comparison, the number of base stations is increased to 7 instead of 3. The handover rate for this analysis is also changed to 20 call per second. The new incoming call arrival will still vary from 50 to 75 calls per seconds. The call departure is assumed to be 5 calls-per seconds in order to slow down the departing of calls. This is to balance with the calls, which are queued.
4.4.1 Effect of Queuing to Reserved Channels

Figure 4.6: Handover dropping probability against reserved channels. Assuming 7 Base stations, 20 connections per base station, Arrival rate 50 calls per seconds, departure rate 2 calls per seconds and handover departure rate 30 call per seconds.

A graph is plotted in figure 4.6 showing the handover dropping probability for scheme with queuing of new call block and reserved channels (WQRC), using 20 connections per base station (m = 20). Each call is assigned to a fixed bandwidth. Using the assumptions stated below figure 4.6 in equations (4.33), figure 4.6 was obtained. The handover dropping probability for WQRC dropped as the reserved channels for handover is increased.

Refer chapter III, using WRC, the handover dropping probability also dropped with the increasing of reserved channels but with higher probabilities than WQRC. Since the inputs to the network, the new incoming call rate and the handover
Call rate are fixed, expending reserved channels gave more bandwidth for the handover call and this decreases the probability of blocking.

Figure 4.7: Probability Handover dropping against reserved channels. Assuming 4 Base stations, 20 connections per base station, Arrival rate 50 calls per seconds, departure rate 2 calls per seconds and handover departure rate 30 call per seconds.

In figure 4.7 and 4.8, the effect of queuing to reserved channels is analyzed. For both graphs, the new incoming call arrival rate, arrival rate is 50 calls per seconds and handover call rate is 30 calls per seconds with call departure rate of 2 calls per seconds. The number of channels for a base station is 20 with 4 base stations in a cell cluster. Using the above assumptions in equation (4.33), figure 4.7 show that when more reserved channels are assigned to a network the probability of handover dropping declined rapidly. This is due to the bandwidth capacity reserved for handover calls during handover of access point to the backbone network.
Figure 4.8: Delay probability against reserved channels. Assuming 4 Base stations, 20 connections per base station, Arrival rate 50 calls per seconds, departure rate 2 calls per seconds and handover departure rate 30 call per seconds.

Using the assumptions started below figure 4.8 in equation (4.34) figure 4.8 shows that with more reserved channels assigned to a network, the probability of delay of the new incoming calls rises. This is because when the amount of bandwidth reserved for handover calls is expanded, the remained channels will be occupied and queue will build up. This means that as the reserved channels expanded, queue will be more but the probability of new call block will not increase. Hence, handover dropping probability as in figure 4.7 will be less. This also shows that in deciding the capacity of the reserved channel, the maximum allowed delay would have to be taken into consideration.
4.5 Summary

In chapter IV, the effect of queuing to reserved channels and Erlang load per cell site had been analyzed using mathematical approach. However, the result for Erlang load will be discussed at the previous chapter. From the results, it shows that as the percentage of reserved channels increases, the probability of handoff dropping decreases as well. This is due to the expanded capacity of reserved channels.

However, this increases the probability of delay of the new incoming calls. When probability of delay increases, the probability of new incoming call block is eliminated. The Erlang load per cell site also affected the probability of handoff dropping and new call delay. It is found that the probability of delay and the probability of handover dropping increases as Erlang load increase. This is because of the limited channels available when more traffic is injected into a network. However, the probability of delay can be reducing by expanding the capacity of base station.

The analysis done in chapter III found that scheme with reserved channels (WRC) decreases the rapid increment of probability of handoff call dropping in scheme without reserved channels (WORC) by reserving channels for handoff calls. However, by decreasing the probability of handoff call drop, WRC increased the probability of new call block when all of the channels in the network are not available. Queuing of new incoming is introduced to solve the above problem while maintaining the low probability of handoff call dropping. Eliminating call blocking probability however introduces delay at the set protocol which can be tolerated.
CHAPTER V

CONCLUSION AND RECOMMENDATION

5.1 Conclusion

Handover priority scheme (i.e. reserving channels for handover call during busy hours) is essential in a mobile network to ensure connection continuity whenever a mobile device moves within or out of the network. In WATM network, call admission control (CAC) limits the incoming call in order to avoid congestion and unnecessary call termination. By using Virtual Connection Tree (VCT) architecture model, an accepted connection will enjoy freedom of mobility under one Network Call Processor (NCP) where by handover calls are guaranteed a good Quality-of-Service (QoS) in terms of call and cell levels.

When handover calls are considered as a higher priority than the incoming calls, a scheme is needed to protect these handover calls. The new call blocking probability and handover dropping probability determine QoS performance at call level in terms of message loss, provided by the WATM network. New call blocking probability determines QoS performance of the network, that it can provide the service whenever a mobile terminal (MT) request for a connection. Handover dropping probability determines QoS performance of the network, that it can guarantee a smooth connection during the connection lifetime despite of the changing access point in the VCT by MTs. The overload probability is one of the
QoS metrics of the network that the non real-time applications will not get through congested network.

The technique of this project is to find a scheme that can serve better QoS performance in WATM network for real-time and non real-time connections. The QoS performance concerned is to maintain low probability call block, probability of handover call dropping, probability of overload and probability that the bandwidth is less than the minimum rate \( P(r < r_{\text{min}}) \). The method of finding the right scheme was first started with reserving a certain percentage of channels in the network for handover calls in order to increase the continuity of handover calls as users move from one cell site to another. The scheme is called with reserved channels WRC. The efficiency of WRC in handling handover calls and new incoming calls was highlighted when compared to scheme without reserved channels, WQRC. Low probability of handover dropping was able to achieved using WRC. Further enhancement of WRC depends very much on factors like size of the cell cluster, size of the reserved channels and the capacity a base station can handle. Despite of the improvement WRC had obtained, a set back was found where with the decrement of handover dropping probability; the new call block probability had increased. Hence, further improvement was proposed by allowing queue for the new incoming calls at the setup protocol.

The proposed scheme is called with queuing of the new incoming calls and with reserved channels, WQRC. In WQRC when a new incoming call enters a congested area it will be queued. Hence, this results the increment of the call delay. With the increment of probability of call delay, WQRC eliminates the probability of new incoming call block. Factors that provide better QoS performance using WQRC system are the size of the cell cluster, size of the reserved channels and the capacity a base station can handle. A set back obtained from the improvement of WRC system is the increment in probability of delay.

The probability of new call block can be reduced with the increment of base stations in a cell cluster and expanding the capacity a base station can handle. This means that the VCT has the advantage to reduce initial call rejection. Even though there are reserved channel but if is not use in VCT network, the probability of new
call lost would not reach as low as when reserved channels is used in VCT networking. However, the new calls block probability increase with the increment of reserve channels. Hence, to control the rapid growth of new call block probability when reserve channels increase, a certain amount of reserved channels has to be defined in order to maintain an acceptable increment of new call lost.

The reduction of handover dropping probability can be attained if reserved channels and cell cluster is large. VCT networks decreases the new call block probability as well as handover dropping probability by grouping base station which directly enlarge the network capacity. Hence, when network capacity is large, the percentage for reserved channels can be increase. For non real-time homogeneous system the QoS performance is based on the overload probability. Maximum number of non real-time connection allowed into a network, $N_{nrt}$ and the capacity assigned for base stations contribute to low probability of overload. Hence, CAC plays an important role in homogenous system of non real-time connections for WATM. In addition, VCT network decreases the overload probability of non real-time connection.

The WQRC was simulated using COMNET III and it was found comparable with the theory. Thus, the thesis emphasizes on the mathematical analysis to determine the QoS performance in WATM. With WQRC the handover dropping probability is reduce and the new call block probability is eliminated using reserved channel for handover calls and allowing queue for new incoming calls. Hence, WQRC has upgrade the previous mechanism by avoiding rapid increasing of new call block probability when handover dropping probability declining. The only set back obtained from the improved mechanism is the delay. Such as the VCT network had help to reduce the handover dropping probability, new call block probability and overload probability, here it too somehow reduce the rapid growth of delay probability in the queue. However, the delay will only affect the call set up protocol and not the data transmission.
5.2 Recommendation

This thesis had shown that WQRC system could enhance the VCT network model for WATM network. However, there is a lot of room for improvement for this project. Some of these recommendations include:

a) In this project the queue of the new incoming call is considered infinite. Finite queue can be obtain if the column of the two dimensional state diagram in figure 4.3 is limit to a certain figure. Hence the queue of the new incoming call is limited and the probability of new call block when queue is full can be obtained. In addition the average queue and delay time can be obtained.

b) Comparing the QoS performance of the proposed system using VCT model with other models such as incremental extension, BAHAMA or two-tier architecture.

c) The whole system can be verified with further analysis using discrete event simulation. For more advances in WATM network simulations, Network Simulator is suggested to be used in future, which will take into account several parameters.
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MATLAB PROGRAMMING

This programming is not including my entire thesis.

Eq: 3.7

```matlab
% Probability of new call blocking and handover call dropping
% El = Erlang load
%.............................................................
clf;       % clear current figure
clear all; % remove all variables
c1c;       % clear command window
%......................................................................
dr =1;  % Duparture rate
m = 17;  % m=handover(2)+ new (1) call= 20.
%......................................................................
ar = [5,10,15,20,25,30,35,40,45]; % assume arrival rate 5 to 45 per second
%......................................................................
for   y = [1:9]
    Dr(y) = dr;
    Ar(y) = ar(y);
    q1(y) = Ar(y)/Dr(y);
    q2(y) = (q1(y).^m)/factorial(m);
end
%......................................................................
for y = [1:9]
    for k = [1:m]
        Q(k) = (q1(y).^k)/factorial(k);
    end
    Qk(y) = 1 + sum(Q);
    Pab(y) = q2(y)/Qk(y);
end
%......................................................................
for i = [1:9]
    El(i) = ar(i)/7;
end
%......................................................................
semilogy (El,Pab,'b*-');
ylabel ('Probability of New Call Block and Handover Call Drop');
xlabel ('Erlang Load Per Cell Sites');
grid on;
%......................................................................
```
Eq: 3.9

% Probability of having new call blocking for with reserved channel
% El = Erlang load
clf; % clear current figure
clear all; % remove all variables
clc; % clear command window

%....................................................................
dr = 5; % Duparture rate
m = 20; B = 7;B1 = 5;g = 0.85;c=25;
%....................................................................
Nrt= B*m*g;
Nrt1= B1*m*g;
%ar = [5,10,15,20,25,30,35,40,45,50,55,60,65,70,75,80,85,90,95,100]; %
assume arrival rate 5 to 45 per second
ar= [10,11,12,13,14,15];
ar1= [7.15,7.85,8.55,9.3,10,10.7];
%xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
%for B=7
%....................................................................
for y = [1:6]
Dr(y) = dr;
q1(y) = ar(y)/Dr(y);
q2(y) = (q1(y).^c)/factorial(c);
end
%....................................................................
for y = [1:6]
for k = [1:Nrt]
Q(k) = (q1(y).^k)/factorial(k);
end
Qk(y)= 1+sum(Q);
Pab(y) =q2(y)/Qk(y);
end
%xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
% for B=5
%....................................................................
for y = [1:6]
Dr(y) = dr;
q1a(y) = ar1(y)/Dr(y);
q2a(y) = (q1a(y).^c)/factorial(c);
end
%....................................................................
for z = [1:6]
for v = [1:Nrt1]
Q1(v) = (q1a(z).^v)/factorial(v);
end
Qk1(z)= 1+sum(Q1);
Pab1(z) =q2a(z)/Qk1(z);
end
%....................................................................
for i = [1:6]
El(i) = ar(i)/7;
El1(i) = ar1(i)/5;
end
%....................................................................
semilogy (El,Pab,'b-*',El1,Pab1,'kh-');
ylabel ('Probability of having new call blocking for with reserved channel');
xlabel ('Erlang Load Per Cell Sites');
grid;
%....................................................................
Eq: 3.13

% % Probability of new call blocking with reserved channel
% % % clear current figure
clear all; % remove all variables
clc; % clear command window

% % Departure rate
dr = 2.5;  % % departure handover rate
hr = 10; % % number of channel =20= New call + Handover Call
m = 20;  % % number of base station per cluster
B = 7;  % % percentage of reserved channel 85%
g = 0.85; % % Arrival handover=8
Phdi1=0.2; % % %

% % %
Nrt = m*g;  % % 19 for reserve channel, 25 channel for handover call
ar = [40,50,60,70,80,90,100,110,120];  % % assume arrival rate 20 to 45 per
% % second

% % %
for y = [1:9]
    Dr(y) = dre;
    Ar(y) = ar(y);
    q1(y) = Ar(y)/Dr(y);
    q2(y) = (B*(q1(y).^Nrt))/factorial(Nrt);
end

% %
for y = [1:9]
    for k = [1:Nrt]
        Q(k)=q1(y).^k/factorial(k);
    end
    Qk(y)=1+sum(Q);
    PabII(y) = q2(y)/Qk(y);
end

% %
for i = [1:9]
    El(i) = ar(i)/(7*dr);
end

% %
semilogy (El,PabII,'k*-');
ylabel ('Probability of New Call With Reserve Channel');
xlabel ('Erlang Load Per Cell Sites');
grid on;
% %
% Probability of handover call dropping with reserved channel
%............................................................
clf; % clear current figure
clear all; % remove all variables
clc; % clear command window
%............................................................
dr= 1; % Departure rate
hr= 10; % departure handover rate
m = 20; % number of channel, 20 channel per unit cell, m=140 channel
hr = 10;
%.............................................................
ar = [40,50,60,70,80,90,100,110,120]; % assume arrival rate 20 to 45 per second
%.............................................................
for y = [1:9]
    Dr(y) = dr;
    Ar(y) = ar(y);
    q1(y) = Ar(y)/Dr(y);
    q2(y) = q1(y).^m/factorial(m);
end
%.............................................................
for y = [1:9]
    for k = [1:m]
        Q(k) = (q1(y).^k)/factorial(k);
    end
    Qk(y) = 1 + sum (Q);
    PhdII(y) = q2(y)/Qk(y);
end
%.............................................................
for i = [1:9]
    El(i) = ar(i)/7;
end
%.............................................................
semilogy (El,PhdII,'g*-');
ylabel ('Probability of Handover Call Drop');
xlabel ('Erlang Load Per Cell Sites');
grid;
%.............................................................
Eq: 3.7, 3.13 and 3.15

clf;  % clear current figure
clear all;  % remove all variables
clc;  % clear command window

% ...........................................................................
B = 7;  % number of base station per cluster
dr = 1;  % Departure rate
dr1 = 2;  % departure rate
m = 17;  % m=handover(2)+ new (1) call= 20.
ml = 20;  % number of channel =20= New call + Handover Call
g = 0.85;  % percentage of reserved channel 85%
hr = 10;  % departure handover rate
Phd11 = 0.2;  % Arrival handover=8
% ...........................................................................
dre = dr1 + (hr*Phd11);
Nrt = m1*g;  % 119 for reserve channel, 25 channel for handover call
ar = [5, 10, 15, 20, 25, 30, 35, 40, 45, 50, 55, 60, 65, 70, 75, 80, 85, 90, 95, 100];  %
assume arrival rate 5 to 45 per second
% Probability of new call blocking and handover call dropping
% El = Erlang load
% ...........................................................................
for y = [1:20]
   Dr(y) = dr;
   Ar(y) = ar(y);
   q1(y) = Ar(y)/Dr(y);
   q2(y) = (q1(y).^m)/factorial(m);
end
% ...........................................................................
for y = [1:20]
   for k = [1:m]
      Q(k) = (q1(y).^k)/factorial(k);
   end
   Qk(y) = 1 + sum(Q);
   Pab(y) = q2(y)/Qk(y);
end
% Probability of new call blocking with reserved channel
% ..........................................................................
for z = [1:20]
   Dr1(z) = dre;
   Ar1(z) = ar(z);
   p1(z) = Ar1(z)/Dr1(z);
   p2(z) = ((p1(z)^Nrt))/factorial(Nrt);
end
% Probability of handover call dropping with reserved channel
% ..........................................................................
for y = [1:20]
   for k = [1:Nrt]
      P(k) = (p1(z).^k)/factorial(k);
   end
   Pk(z) = 1 + sum(P);
   PabII(z) = p2(z)/Pk(z);
end
% Probability of handover call dropping with reserved channel
% ..........................................................................
for y = [1:20]
   for k = [1:m1]
R(k) = (r1(y).^(k))/factorial(k);
end
Rk(y) = 1 + sum (R);
PhdII(y) = r2(y)/Rk(y);
end
%.............................................................................
for i = [1:20]
  El(i) = ar(i)/7;
end
%.............................................................................
semilogy (El,PhdII,'c*-',El, Pab, 'bo-', El, PabII, 'kh-');
ylabel ('Probability of New Call Block and Handover Call Dropping');
xlabel ('Erlang Load Per Cell Sites');
grid;
%.............................................................................
\[ Eq: 3.25 \]

\[
\text{% probability for non-real time, } P_{\text{nrt}} \\
\text{% \text{nrt} = the number of connections to base station} \\
\text{% \text{Nnrt} = the maximum number of non real time connection allowed to a network} \\
\text{% \text{Arnrt} = the arrival rate per base station} \\
\text{% \text{Drnrt} = the departure rate per base station} \\
\text{clf; \% clear current figure} \\
\text{clear all; \% remove all variables} \\
\text{clc; \% clear command window} \\
\text{ Arnrt= 30; \% allocate 30 arrival at one base station} \\
\text{Nnrt = 80;} \\
\text{B = 7;} \\
\text{m = 20;} \\
\text{s = m+1;} \\
\text{y = Nnrt-s;} \\
\text{for } k = [1:y] \\
\text{\quad q2(k) = \frac{((B-1)*(Arnrt/Drnrt))^k}{\text{factorial}(k)}; \\
\text{end} \\
\text{Q = 1+sum(q2);} \\
\text{for } x = [1:Nnrt] \\
\text{\quad q3=((B*Arnrt/Drnrt)^x)/\text{factorial}(x);} \\
\text{end} \\
\text{Q1 = 1+sum(q3);} \\
\text{for } S = [s:Nnrt] \\
\text{\quad Q2(S) = Q;} \\
\text{\quad Q3(S) = Q1;} \\
\text{\quad q1(S) = \frac{((Arnrt/Drnrt)^S)}{\text{factorial}(S)};} \\
\text{\quad Pnrt(S) = q1(S)*Q2(S)/Q3(S);} \\
\text{\quad Pov(S) = sum(Pnrt);} \\
\text{end} \\
\text{semilogy (Pov,'k-*');} \\
\text{xlabel ('number of non real time connection');} \\
\text{ylabel ('Overload Probability');} \\
\text{grid; \\
\text{}}
Eq: 4.32

% the handover dropping probability call with queue and reserved channel
% Hr=the handover calls rate
% Ar=the new incoming calls with call rates
% Dr= the call departure rate
N = the numbers of channel are available
% g = the percentage of reserved channels
% B = the number of base station
% m = the number of connection a base station can be handle
% i1=the number of customer in the queue
% i2=the number of customer in the service
% declare the variable
clf; % clear current figure
clear all; % remove all variables
clc; % clear command window

Ar = 75;  
Hr = 20;  
Dr = 2;   
g = 0.30;  
m = 20;   
B = 7;    
N = g*m*B;  

% general formula
b = Ar/Dr;  
c = Hr/Dr;  
a = (Ar+Hr)/Dr;  
x = B*m;  

q1 = (a/c)^N;  
q2 = (b*factorial(N))/(N*(c^N));  
q3 = (c^x)/factorial(x);  

for i22=[N:x]  
A21(i22) = (c^i22)/factorial(i22);  
end  
A22= sum(A21);  

for i21=[1:N-1]  
A11(i21) = (a^i21)/factorial(i21);  
end  
A1 = 1+sum(A11);  
A2 = q1*A22;  

for Ar1=[1:N-1]  
A311(Ar1) = (a^Ar1)/factorial(Ar1);  
end  
A31 = 1+sum(A311);  
A3 = q2*A22*A31;  
A = A1+A2-A3;  

% find the P;  
P = (1-q2*A22)/A;  

% find the Phb;
Phb1 = q2*q3;

for r=[1:N]
    Phb2= (a^r)/factorial(r);
end
Phb3 = 1+sum(Phb2);
Phb4 = q1-q2*Phb3;
Phb = (P*q3*Phb4)+Phb1

semilogy (Phb, 'b*');
xlabel ('Reserved Channel');
ylabel ('Handover Dropping Probability');
Eq: 4.33

% the handover dropping probability call with queue and reserved channel
% Hr= the handover calls rate
% Ar= the new incoming calls with call rates
% Dr= the call departure rate
% N = the numbers of channel are available
% g = the percentage of reserved channels
% B = the number of base station
% m = the number of connection a base station can be handle
% i1= the number of customer in the queue
% i2= the number of customer in the service
% declare the variable
clf;       % clear current figure
clear all; % remove all variables
clc;       % clear command window

Ar = 50; Hr = 30; Dr = 2; g = 0.15; g1 = 0.20; g2 = 0.25; g3 = 0.30;
g4 = 0.35; m = 20; B = 4;

% general formula
b = Ar/Dr;
c = Hr/Dr;
a = g*m*B;
x = B*m;
q1 = (a/c)^N;
q2 = (b*factorial(N))/(N*(c^N));
q3 = (c^x)/factorial(x);

for i22=[N:x]
    A21(i22) = (c^i22)/factorial(i22);
end
A22= sum(A21);

A = A1+A2-A3;

P = (1-q2*A22)/A;

Phb1 = q2*q3;

for r=[1:N]
Phb2 = \frac{(a^r)}{\text{factorial}(r)}; 
end 
Phb3 = 1 + \text{sum}(\text{Phb2}); 
Phb4 = q1 - q2^2 \times \text{Phb3}; 
Phb = (P \times q3 \times \text{Phb4}) + \text{Phb1}; 

% general formula 
\% \text{general formula} 
N1 = g1 \times m \times B; 
q1a = \frac{(a/c)^N1}{\text{factorial}(N1)}; 
q2a = \frac{(b \times \text{factorial}(N1))/(N1 \times (c^N1))}{\text{factorial}(N1)}; 
q3a = \frac{(c^x)}{\text{factorial}(x)}; 
% \text{for} \ i22a=[N1:x] 
A21a(i22a) = \frac{(c^{i22a})}{\text{factorial}(i22a)}; 
end 
A22a = \text{sum}(A21a); 
% \text{find the A}; 
% \text{for} \ i2la=[1:N1-1] 
A11a(i2la) = \frac{(a^{i2la})}{\text{factorial}(i2la)}; 
end 
A1a = 1 + \text{sum}(A11a); 
% \text{A2} = q1 a^A22a; 
% \text{for} \ A1a=[1:N1-1] 
A311a(A1a) = \frac{(a(A1a))}{\text{factorial}(A1a)}; 
end 
A31a = 1 + \text{sum}(A311a); 
% \text{A3} = q2a \times A22a \times A31a; 
Aa = A1a + A2a - A3a; 
% \text{find the P}; 
% \text{Pa} = \frac{(1 - q2a \times A22a)}{Aa}; 
% \text{find the Phb}; 
% \text{Phb1a} = q2a \times q3a; 
% \text{for} \ ra=[1:N1] 
\text{Phb2a} = \frac{(a^ra)}{\text{factorial}(ra)}; 
end 
Phb3a = 1 + \text{sum}(\text{Phb2a}); 
Phb4a = q1a - q2a \times \text{Phb3a}; 
Phba = (Pa \times q3a \times \text{Phb4a}) + \text{Phb1a}; 
% \text{general formula} 
\% \text{general formula} 
N2 = g2 \times m \times B; 
q1b = \frac{(a/c)^N2}{\text{factorial}(N2)}; 
q2b = \frac{(b \times \text{factorial}(N2))/(N2 \times (c^N2))}{\text{factorial}(N2)}; 
q3b = \frac{(c^x)}{\text{factorial}(x)}; 
% \text{for} \ i22b=[N2:x] 
A21b(i22b) = \frac{(c^{i22b})}{\text{factorial}(i22b)}; 
end 
A22b = \text{sum}(A21b); 
% \text{find the A}; 
%
for i21b=[1:N2-1]
A11b(i21b) = (a^i21b)/factorial(i21b);
end
A1b = 1+sum(A11b);

% find the A;

A2b = q1b*A22b;

% find the Phb;
Phb1b = q2b*q3b;

for rb=[1:N2]
Phb2b = (a^rb)/factorial(rb);
end
Phb3b = 1+sum(Phb2b);
Phb4b = q1b-q2b*Phb3b;
Phbb = (Pb*q3b*Phb4b)+Phb1b;

% find the P;
Pb = (1-q2b*A22b)/Ab;
Pc = (1-q2c*A22c)/Ac;

% general formula
N3 = g3*m*B;
q1c = (a/c)^N3;
q2c = (b*factorial(N3))/(N3*(c^N3));
q3c = (c^x)/factorial(x);

for i22c=[N3:x]
A21c(i22c) = (c^i22c)/factorial(i22c);
end
A22c= sum(A21c);

% find the A;
A11c(i21c) = (a^i21c)/factorial(i21c);
end
A1c = 1+sum(A11c);

% find the P;
Pc = (1-q2c*A22c)/Ac;
\begin{verbatim}
% find the Phb;
% find the Phb;
Phb1c = q2c*q3c;

for rc=[1:N3]
    Phb2c= (a^rc)/factorial(rc);
end
Phb3c = 1+sum(Phb2c);

Phb4c = q1c-q2c*Phb3c;

Phbc = (Pc*q3c*Phb4c)+Phb1c;

% general formula
N4 = g4*m*B;
q1d = (a/c)^N4;
q2d = (b*factorial(N4))/(N4*(c^N4));
q3d = (c^x)/factorial(x);

for i22d=[N4:x]
    A21d(i22d) = (c^i22d)/factorial(i22d);
end
A22d= sum(A21d);

A2d = q1d*A22d;

for Ar1d=[1:N4-1]
    A311d(Ar1d) = (a^Ar1d)/factorial(Ar1d);
end
A31d = 1+sum(A311d);

A3d = q2d*A22d*A31d;

Ad = A1d+A2d-A3d;

% find the P;
Pd = (1-q2d*A22d)/Ad;

% find the Phb;
Phb1d = q2d*q3d;

for rd=[1:N4]
    Phb2d= (a^rd)/factorial(rd);
end
Phb3d = 1+sum(Phb2d);

Phbd = (Pd*q3d*Phb4d)+Phb1d;

PHB=[Phb,Phba,Phbb,Phbc,Phbd];
Res=[15,20,25,30,35];

semilogy (Res,PHB,'kh-');
xlabel ('Reserved Channel (%)');
ylabel ('Handover Dropping Probability');
grid;
\end{verbatim}
Eq: 4.34

% the probability of blocking call with queue and reserved channel
% Hr=the handover calls rate
% Ar=the new incoming calls with call rates
% Dr= the the call departure rate
% N = the numbers of channel are available
% g = the percentage of reserved channels
% B = the number of base station
% m = the number of connection a base station can be handle
% i1=the number of customer in the queue
% i2=the number of customer in the service
% declare the variable
clf;       % clear current figure
clear all; % remove all variables
clc;       % clear command window

Ar = 50; Hr = 30; Dr = 2; g = 0.05; g1 = 0.10; g2 = 0.15; g3 = 0.20;
g4 = 0.25; m = 20; B = 4;

% general formula
b = Ar/Dr;
c = Hr/Dr;
a = g*m*B;
x = B*m;
q1 = (a/c)^N;
q2 = (b*factorial(N))/(N*(c^N));
q3 = (c^x)/factorial(x);

% find the A;
for i22=[N:x]
    A21(i22) = (c^i22)/factorial(i22);
end
A22= sum(A21);

% find the A;
for i21=[1:N-1]
    A11(i21) = (a^i21)/factorial(i21);
end
A1 = 1+sum(A11);

% find the A2;
A2 = q1*A22;

% find the A31;
for Ar1=[1:N-1]
    A311(Ar1) = (a^Ar1)/factorial(Ar1);
end
A31 = 1+sum(A311);

% find the P;
P = (1-q2*A22)/A;

% find the Phb;
Phb1 = q2*q3;

for r=1:N
Phb2 = (a^r)/factorial(r);
end
Phb3 = 1+sum(Phb2);
Phb4 = q1-q2*Phb3;
Phb = (P*q3*Phb4)+Phb1;

% find the Pd;
Pd = 1-a31*P;

% find the Phb;
Phb2a = (a^ra)/factorial(ra);
end
Phb3a = 1+sum(Phb2a);
Phb4a = q1a-q2*Phb3a;
Phba = (P*a*q3a*Phb4a)+Phb1a;

% find the Pd;
Pda = 1-A31a*Pa;

% general formula
N1 = q1*m*B;
q1a = (a/c)^N1;
q2a = (b*factorial(N1))/(N1*(c^N1));
q3a = (c^x)/factorial(x);

for i22a=[N1:x]
    A21a(i22a) = (c^i22a)/factorial(i22a);
end
A22a= sum(A21a);

% find the A;
for i21a=[1:N1-1]
    A11a(i21a) = (a^i21a)/factorial(i21a);
end
A1a = 1+sum(A11a);

A2a = q1a*A22a;

for Ar1a=[1:N1-1]
    A311a(Ar1a) = (a^Ar1a)/factorial(Ar1a);
end
A31a = 1+sum(A311a);

A3a = q2a*A22a*A31a;

Aa = A1a+A2a-A3a;

% find the P;
Pa = (1-q2a*A22a)/Aa;

% find the Phb;
Phb1a = q2a*q3a;

for ra=[1:N1]
    Phb2a = (a^ra)/factorial(ra);
end
Phb3a = 1+sum(Phb2a);
Phb4a = q1a-q2a*Phb3a;
Phba = (P*a*q3a*Phb4a)+Phb1a;

% find the Pd;
Pda = 1-A31a*Pa;

% general formula
N2 = q2*m*B;
q1b = (a/c)^N2;
q2b = (b*factorial(N2))/(N2*(c^N2));
q3b = (c^x)/factorial(x);
% find the A;
for i22b=[N2:x]
    A21b(i22b) = (c^i22b)/factorial(i22b);
end
A22b= sum(A21b);
% find the P;
Pb = (1-q2b*A22b)/A31b;
% find the Phb;
Phb1b = q2b*q3b;
for rb=[1:N2]
    Phb2b= (a^rb)/factorial(rb);
end
Phb3b = 1+sum(Phb2b);
Phbb = (Pb*q3b*Phb3b)+Phb1b;
% general formula
N3 = g3*m*B;
q1c = (a/c)^N3;
q2c = (b*factorial(N3))/(N3*(c^N3));
q3c = (c^x)/factorial(x);
for i22c=[N3:x]
    A21c(i22c) = (c^i22c)/factorial(i22c);
end
A22c= sum(A21c);
A2c = q1c*A22c;
for Ar1c=[1:N3-1]
    A311c(Ar1c) = (a^Ar1c)/factorial(Ar1c);
end
A31c = 1+sum(A311c);

A3c = q2c*A22c*A31c;

% find the P;
% Pc = (1-q2c*A22c)/Ac;

% find the Phb;
Phb1c = q2c*q3c;
for rc=[1:N3]
    Phb2c= (a^rc)/factorial(rc);
end
Phb3c = 1+sum(Phb2c);
Phb4c = q1c-q2c*Phb3c;
Phbc = (Pc*q3c*Phb4c)+Phb1c;

% find the Pd;
Pdc = 1-A31c*Pc;

% general formula
N4 = g4*m*B;
q1d = (a/c)^N4;
q2d = (b*factorial(N4))/(N4*(c^N4));
q3d = (c^x)/factorial(x);
for i22d=[N4:x]
    A21d(i22d) = (c^i22d)/factorial(i22d);
end
A22d= sum(A21d);

for i21d=[1:N4-1]
    A11d(i21d) = (a^i21d)/factorial(i21d);
end
A1d = 1+sum(A11d);

A2d = q1d*A22d;

A31d(Ar1d) = (a^Ar1d)/factorial(Ar1d);
end
A31d = 1+sum(A311d);
A3d = q2d*A22d*A31d;
Ad = A1d+A2d-A3d;

% find the P;
Pd = (1-q2d*A22d)/Ad;

% find the Phb;
Phb1d = q2d*q3d;
for rd=[1:N4]
    Phb2d = (a^rd)/factorial(rd);
end
Phb3d = 1+sum(Phb2d);
Phb4d = q1d-q2d*Phb3d;
Phbd = (Pd*q3d*Phb4d)+Phb1d;

semilogy (Res,PHD,'k*-');
xlabel ('Reserved Channel (%)');
ylabel ('Probability Delay');
grid;
APPENDIX B: NETWORK DESIGN USING COMNET III
APPENDIX A

NETWORK DESIGN USING COMNET III

Network 1: 3 link for 3 Base Station
Network 1: 3 link for 1 Base Station

Network 1: 3 link for 3 Base Station, first MT connected to the switch and then switch select either one to connect.
Network 1: 3 link for 3 MT and 1 Base Station