AN INVESTIGATION INTO THE ADVANCED TIME DIVISION MULTIPLE ACCESS(ATDMA) PROTOCOL FOR A PERSONAL COMMUNICATION NETWORK

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CONTENTS

CHAPTER III ATDMA AIR INTERFERENCE IN A MICROCELL ENVIRONMENT

ABSTRACT

The performance of the Advanced Time Division Multiple Access (ATDMA) protocol in a microcell environment has been investigated in this thesis. The ATDMA protocol is a new generation protocol which can support both circuit switched and packet switched transmission modes. The protocol can also adapt in a varying propagation environment. This thesis examines the efficiency of the protocol in a microcell environment and also examines different access techniques for voice and data traffic to improve the efficiency of the protocol. To study the performance of the protocol a discrete event based simulation model has been developed which includes a microcell channel model of a city area.

A data block reservation scheme has been developed in this work, which increase the traffic efficiency of the protocol. By combining the data block reservation scheme and capture effect, the ATDMA protocol's performance in transmitting mixed voice and data traffic in an urban microcell environment was investigated by means of computer simulation. The simulation model was used to find out the appropriate parameters for the optimum performance of the protocol and then to investigate the performance of the protocol. With consideration of the capture ratio, the effect of capture has also been evaluated in a more practical manner.

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CHAPTER I

INTRODUCTION

1.0 General introduction

The first-generation mobile communication systems were introduced at national level, for example NMT in Scandinavia, AMPS in the USA, C450 in Germany, RMTS in Italy and TACS in the UK [1][2]. These systems use analogue modulation and were specified essentially as separate dedicated systems for paging, cordless phone, mobile terrestrial and mobile satellite communication. In the early 1990s, second generation systems started to be introduced at regional (continental) level, e.g. GSM in Europe [3]. These systems use digital modulation techniques. The next step, now expected for the first decade of the next century, will be the introduction of a global system called Universal Mobile Telecommunication System (UMTS) in Europe and Future Public Land Mobile Telecommunication System (FPLMTS currently renamed as IMT2000) in the rest of the world [4][5]. This system, which will use digital modulation techniques and support bit-rates of up to 2 Mbit/s, is based upon 2 GHz technology and will possibly merge paging, cordless phone, mobile terrestrial and mobile satellite standards into a single, unified standard. A wide range of services will be supported by the UMTS. This third generation mobile system is expected to integrate audio, video and data information with 'anyone', 'anywhere' and 'anytime' at low cost and using handy devices [6][7][8].

An important question when designing and standarding mobile radio systems is the development of the multiple access protocol. The multiple access protocols are used to allow mobile users to share a finite amount of radio spectrum for transmitting and receiving information. The protocol has a significant influence on the spectral efficiency and capacity, and there is much more research in progress to enhance the different multiple access techniques. The multiple access protocol also influences the design of the fixed or backbone network which is used to connect the mobile network.

Recently several access techniques have been proposed for higher capacity third generation mobile communication systems [6][7][8]. The protocols proposed fall into three different categories. These are TDMA based protocols, CDMA based protocols and random access protocols. The Advanced Time Division Multiple Access (ATDMA) protocol is one of the new generation protocols which is under investigation for the third generation mobile radio system. ATDMA protocol is the further development of the Packet Reservation Multiple Access (PRMA) protocol which combines the advantages of both packet switched and circuit switched systems [6]. The ATDMA protocol is one of the strong contenders for the UMTS. The protocol shows high multiplexing efficiency for the integrated voice, data and video traffic.

The ATDMA protocol use TDMA frame and slot structure. Each UP link frame contains a number of reservation slots and a number of traffic slots. Mobiles which want to transmit information send their request packet via an R slot using the slotted ALOHA protocol and therefore is subject to collisions. Hence, when collision occurs, either the capture effect will allow one mobile to gain access or no mobile is successful. The successful mobile will get a traffic slot for transmitting information packets. The mobiles which do not gain access enter a collision resolution phase. Collision resolution involves the mobile retransmitting on the next available R slot with a specified permission probability.

In this work a data block reservation scheme has been developed, which is used to transmit data traffic in an integrated voice and data environment. By combining the proposed data block reservation scheme and the capture effect the performance of the ATDMA protocol can be improved. A simulation model has been developed to measure the performance of the ATDMA protocol in a microcellular situation. The system performance of the ATDMA protocol in a microcellular environment transmitting mixed voice/data traffic with different transmission rates was investigated by using the developed computer simulation model. Also the performance of the protocol with voice and data traffic was studied by using the capture effect and the data block reservation scheme.

Microcells are a subject of major interest for the furture generation mobile radio systems [9]. The term microcell has now been used for a decade to describe small cells with cell size smaller than 1 km for cellular mobile radio communications. Several advantages can be achieved by using microcells for mobile radio applications. Use of microcell can increase systems capacity considerably over the macrocell based systems and, if mobile telephones become truly ubiquitous, use of microcells may be a suitable way to support the resultant increase in teletraffic. Measurements show that microcells have improved propagation properties, with less severe fading, and a much increased coherence bandwidth [9], allowing low error transmission at higher bit rates than currently available within conventional cells. Microcells require substantially lower transmitter powers than conventional cells, so that battery life is increased and the size of handheld mobile telephones reduced. They only require small base stations (BS) with short antennas which can be sited unobtrusively and their coverage area can be predicted more accurately than that of conventional cells, thus facilitating network design [10].

In this work, a microcell propagation simulation model is developed to asses the performance of the ATDMA protocol. The microcell model was developed to investigate the path loss and carrier to interference ratio characteristics of the protocol in a microcell environment.

1.1 Thesis structure

This thesis is organized into five chapters and one appendix. The thesis presents performance of the ATDMA protocol with mixed voice and data traffic in a microcell environment. The ATDMA protocol structure and microcell propagation characteristics are discussed in the thesis. Also a simulation model was established which includes the ATDMA protocol model and the microcell propagation simulation model.

The work of the thesis is presented in the following sequence: At the beginning, characteristics of the mobile radio environment and different multiple access protocols are discussed in detail. Then the ATDMA frame structure and ATDMA system requirements are discussed. Following is the discussion of the microcell propagation environment. and then, an overall simulation model including the ATDMA protocol model and a microcell propagation model are established to simulate the ATDMA protocol performance in a microcell environment. Finally the performance of the protocol in the microcell environment are investigated using the simulation model.

Chapter II gives an overview of multiple access protocols for cellular wireless network. Some of the typical mobile radio systems and multiple access protocols are discussed here. Comparison of the advantages and disadvantages of these multiple access methods have also been made. The ATDMA protocol is a TDMA based packet access mechanism and is a strong contender for the third generation mobile communication system UMTS.

In Chapter III, ATDMA air interference and the microcell environment are described. ATDMA frame structure and ATDMA system requirements are discussed first. In the discussion of ATDMA protocol structure, a method which combines the data block reservation scheme and the capture effect is proposed to enhance the performance of the ATDMA protocol. Later microcell propagation issues and microcell modeling method are discussed.

Chapter IV describes the simulation of the ATDMA protocol in the microcell environment. SIMSCRIPT II.5, a discrete event simulation language, was used to study the ATDMA protocol and microcell propagation. Performance of the ATDMA protocol with mixed voice and data traffic in the microcell environment is investigated with different transmission bit rates and different system parameters. Due to the simulation, optimum system parameters and optimum system performance is obtained and the effectiveness of the developing method of using a data block reservation scheme and the capture effect is proved.

Chapter VI contains the overall conclusion and some final comments about the future work.

The appendix contains the developed simulation program code.

CHAPTER II

OVERVIEW OF MULTIPLE ACCESS PROTOCOLS FOR CELLULAR WIRELESS NETWORKS

2.0 Introduction

Modern communication systems have developed on the basis of resource sharing. The idea is to maximize the number of users by utilizing resources optimally. This idea gave rise to the development of different techniques of resource sharing [11][12]. For mobile telephone the frequency spectrum is the most crucial resource. Different techniques have been employed to efficiently use the spectrum. One of the technique used to increase the number of users is the use of low bit rate souce coders, by using real time advanced signal processing techniques [13]. On the other hand channel access techniques enable all the mobiles to effectively share the common radio spectrum. Mobile radio is an example of a typical multi-access medium where a channel is shared by many users. Multiple access protocols are used to allow mobile users to share a finite amount of radio spectrum for transmitting & receiving information.

The wireless network differs from other multi-access networks due to its 'hidden users' problem [12]. In a fixed network all the terminals who share the common channel are aware of the other transmitting terminals, but in the case of mobile radio communication in some situations, due to the absence of line-of-sight of communication, all the mobiles can not hear each other. In such cases the protocols require a central coordination to avoid conflict. Multiple access techniques perform a crucial role in system design and system performance.

The work in this chapter will introduce the cellular mobile radio concept and different multiple access protocols for wireless networks. Section 1 will discuss the cellular mobile radio concept and cellular wireless network architecture. Section 2 will discuss the classification of multiple access protocols. Section 3 will describe all the multiple access protocols in detail. The comparison of these protocols are given in section 4. Finally, section 5 presents conclusions.

2.1 Cellular wireless networks

Cellular wireless networks provide their users with opportunities to travel freely within the service area and simultaneously communicate with any telephone, fax, data modem, and electronic mail subscriber anywhere in the world. The distinguishing feature of cellular systems compared to a non-cellular/trucked mobile radio system is the use of multiple base stations with relatively small coverage radii for each base station. Each frequency is used simultaneously by multiple base stations. This "frequency reuse" allows much higher subscriber densities than a non-cellular system. System capacity can be further increased by reducing the cell size (the coverage area of a base station). In addition to supporting much higher subscriber density than a non-cellular system, this approach make available the use of small, battery-powered portable handsets with lower RF (radio frequency) transmit power than the larger vehicular mobile units used in earlier systems. In cellular systems, continuous coverage is achieved by executing a "hand-off" (the seamless transfer of the call from one base station to another) as the mobile unit crosses cell "boundaries". This requires the mobile to change frequencies under the control of the cellular network.

Cellular networks with frequency reuse is employed in cellular mobile telephone networks, personal communication networks, mobile data networks, and some WLANs (Wireless Local Area Networks) .

2.1.1 The cellular concept and frequency reuse

Cellular radio systems rely on an intelligent allocation and reuse of channels throughout a coverage region [14]. Each cellular base station is allocated a group of radio channels to be used within a small geographic area called a cell. Base stations in adjacent cells are assigned channel groups which use different carrier frequencies than neighboring cells. The base station antennas are designed to achieve a desired coverage within the particular cell. By limiting the coverage area to within the boundaries of a cell, the same group of channels may be used to cover different cells that are separated from one another by distances large enough to keep interference levels within tolerable limits. The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called frequency reuse / frequency planning.

Figure 2.1 illustrates the concept of cellular frequency reuse, where cells labeled with the same letter use the same group of channels. The frequency reuse plan is overlaid upon a map to indicate where different frequency channels are used. The hexagonal cell shape shown in Figure 2.1 is conceptual and is a simplistic model of the radio coverage for each base station, but it has been universally adopted since the hexagon permits easy and manageable analysis of a cellular system. The actual radio coverage of a cell is known as the footprint and is determined from field measurements or propagation prediction models. Although the real footprint is amorphous in nature, a regular cell shape is needed for systematic system design and adaptation for future growth. While it might seem natural to choose a circle to represent the coverage area of a base station, adjacent circles can not be overlaid upon a map without leaving gaps or creating overlapping regions. Thus, when considering geometric shapes which cover an entire region without overlap and with equal area, there are two sensible choices: a equilateral triangle, and a hexagon. A cell must be designed to serve the weakest mobiles within the footprint, and these are typically located at the edge of the cell. For a given distance between the center of a polygon and its farthest perimeter points, the fewest number of cells can cover a geographic region, and the hexagon closely approximates a circular radiation pattern which would occur for an omni-directional base station antenna and free space propagation. Of course, the actual cellular footprint is determined by the contour in which a given transmitter serves the mobiles successfully.

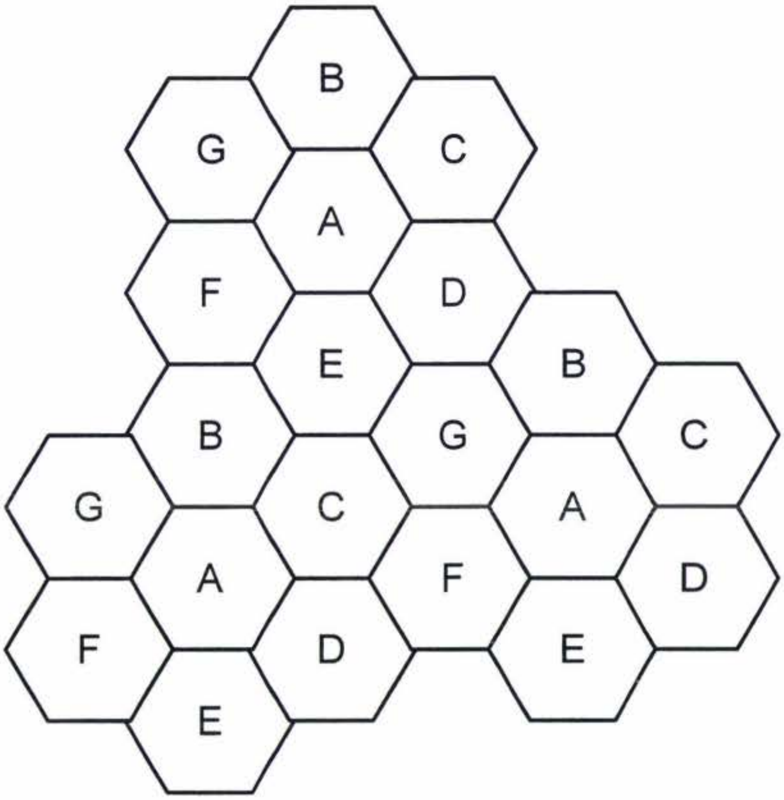


Figure 2.1 Illustration of cellular reuse concept. Cells with the same letter use the same set of frequencies. In this example, the cluster size, N, is equal to seven (cells A,B ...G), and the frequency reuse factor is 1/7 since each cell contains one-seventh of the total number of available channel

When using hexagons to model coverage areas, base station transmitters are depicted as either being in the center of the cell (center-excited cells) or on three of the six cell vertices (edge-excited cells). Normally, omni-directional antennas are used in center-excited cells and sectored directional antennas are used in corner-excited cells. Practical consideration usually do not allow base stations to be placed exactly as they appear in the hexagonal layout. Most system designs permit a base station to be positioned up to one-fourth the cell radius away from the ideal location.

To further understand the frequency reuse concept, consider a cellular system which has a total of S duplex channels available for use. If each cell is allocated a group of k channels (k<S), and if the S channels are divided among N cells into unique and

disjoint channel groups which each have the same number of channels, the total number of available radio channels can be expressed as

$$S = k\,N \tag{2.1}$$

The N cells which collectively use the complete set of available frequencies is called a cluster. If a cluster is replicated M times within the system, the total number of duplex channels, C, can be used as a measure of capacity and is given

$$C = MkN = MS \tag{2.2}$$

As seen from equation (2.2), the capacity of a cellular system is directly proportional to the number of time a cluster is replicated in a fixed service area. The factor N is called the cluster size and is typically equal to 4, 7, or 12. Due to the fact that the hexagonal geometry of Figure 2.1 has exactly six equidistant neighbors and that the lines joining the centers of any cell and each of its neighbors are separated by multiples of 60 degrees, there are only certain cluster sizes and cell layouts which are possible[15]. In order to tessellate-- to connect without gaps between adjacent cells-- the geometry of hexagons is such that the number of cells per cluster, N, can only have values which satisfy equation (2.3).

$$N = i^2 + ij + j^2 \tag{2.3}$$

where i and j are non-negative integers, and $i > j$. For example, $i = 2$, $j = 1$, then $N = 7$.

If the cluster size N is reduced while the cell size is kept constant, more clusters are required to cover a given area and hence more capacity (a larger value of C) is achieved because of higher number of user per cell. A large cluster size indicates that the ratio between the cell radius and the distance between co-channel cells is large. Conversely, a small cluster size indicates that co-channel cells are located much closer together. The value of N is related to co-channel interference while a mobile or base station can tolerate while maintaining a sufficient quality of communications. From a design viewpoint, the smallest possible value of N is desirable in order to maximize capacity over a given coverage area (i.e., to maximize C in equation (2.2)). The frequency reuse factor of a cellular system is given by 1/N, since each cell within a cluster is only assigned 1/N of the total available channels in the system [15][16].

The cellular concept was a major breakthrough in solving the problem of spectral congestion and user capacity. It offers very high capacity in a limited spectrum allocation without any major technological changes. The cellular concept is a system level idea which replaces a single, high power transmitter (large cell) with many low power transmitters (small cells), each providing coverage to only a small portion of the service area. Each base station is allocated a portion of the total number of channels available to the entire system, and nearby base stations are assigned different group of channels so that the interference between base stations (and the mobile users within those cells) is minimized. By systematically spacing base station and their channel groups throughout a area, the available channels are distributed throughout the geographic region and may be reused as many times as necessary, so long as the interference between co-channel stations is kept below an accepted level.

As the demand for service increases (i.e., as more channels are needed within a particular market), the number of base stations may be increased (along with a corresponding decrease in transmitter power to avoid added interference), thereby providing additional radio capacity with no additional increase in radio spectrum. This fundamental principle is the foundation for all modern wireless communication systems, since it enables a fixed number of channels to serve an arbitrarily large number of subscribers by reusing the channels throughout the coverage region. Furthermore, the cellular concept allows every piece of subscriber equipment within a country or continent to be manufactured with the same set of channels, so that any mobile may be used anywhere within the region [15].

Using the frequency reuse technique (or cellular structure), the overall system capacity can in principle be made as large as desired by steadily reducing the area of the cell, while controlling power levels to avoid co-channel interference--that is the interference to other users operating in another cell using the same frequency. System designs are now under development for use in cities, where very small cells called microcell, A microcell will cover an area about the size of a city block and will serve users carrying low-power pocket-size phones. More details about microcell issues will be discussed in the next chapter.

2.1.2 Interference and system capacity

Interference is the major limiting factor in the performance of cellular radio systems. It has been recognized as a major bottleneck in increasing capacity and is often responsible for dropped calls [16]. The two major types of system-generated interference are co-channel interference and adjacent channel interference.

Frequency reuse implies that in a given coverage area there are several cells that use the same set of frequencies. These cells are called co-channel cells, and the interference between signals from these cells is called co-channel interference. Unlike normal noise which can be overcome by increasing the signal-to-noise ration (SNR), co-channel interference cannot be combated by simply increasing the carrier power of a transmitter. This is because an increase in carrier transmit power increases the interference to neighboring co-channel cells. To reduce co-channel interference, co-channel cells must be physically separated by a minimum distance to provide sufficient isolation due to propagation.

In a cellular system, when the size of each cell is approximately the same, co-channel interference is independent of the transmitted power and becomes a function of the radius of the cell (R), and the distance to the center of the nearest co-channel cell (D). This distance is called reuse distance and is shown in Figure 2.2. By increasing the ratio of D/R, the spatial separation between co-channel cells relative to the coverage distance of a cell is increased. Thus interference is reduced by improved isolation of RF energy from the co-channel cell. The parameter Q, called the co-channel reuse ratio, is related to the cluster size. For a hexagonal geometry:

$$Q = \frac{D}{R} = \sqrt{3N} \quad [16] \qquad (2.4)$$

A small value of Q provides larger capacity since the cluster size N is small, whereas a large value of Q improves the transmission quality, due to a small level of co-channel interference. A trade-off must be made between these two objectives in actual cellular design.

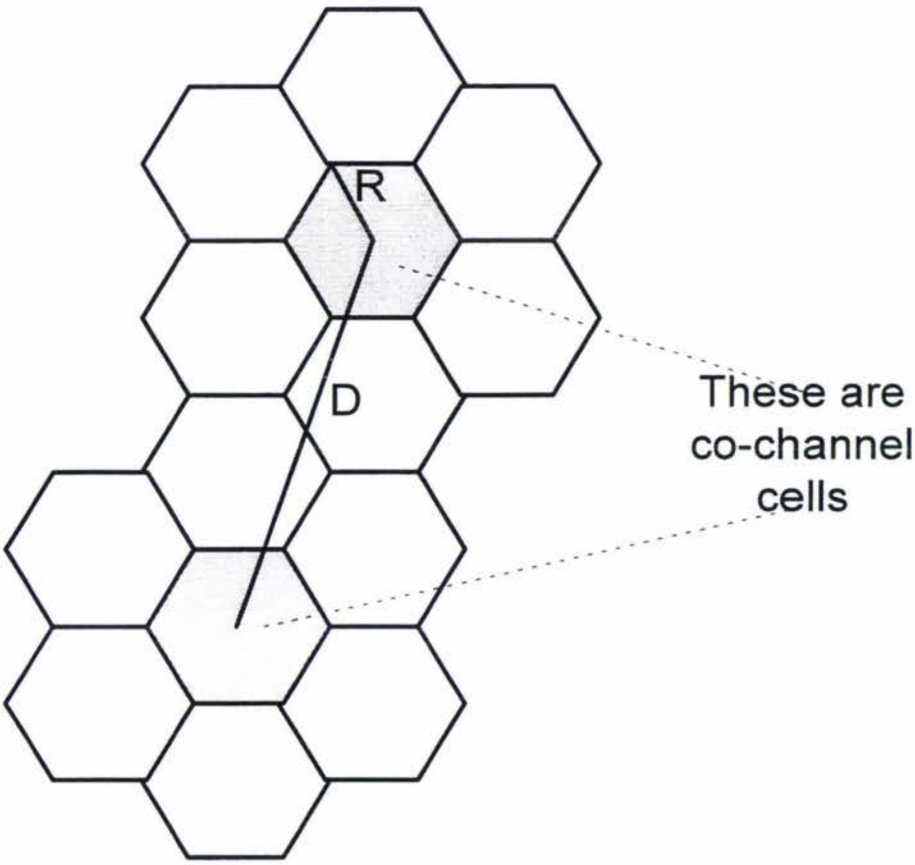


Figure 2.2 Frequency reuse distance D and cell size R.

Let M be the number of co-channel interfering cells. Then, the carrier-to-interference ratio (C/I) for a mobile receiver which monitors a forward channel can be expressed as

$$\frac{C}{I} = \frac{C}{\sum_{i=1}^{M} I_i} \qquad (2.5)$$

where C is the desired signal power from the designed base station and I_i is the interference power caused by the i th interfering co-channel cell base station. If the signal levels of co-channel cells are known, then the C/I ratio for the forward link can be found using equation (2.5). Figure 2.3 shows the different tier's co-channel interference with the cluster size equal to 7. Several tiers of co-channel interference exists. In cellular technology first tier co-channel interference is believed to be the main source of co-channel interference of the system.

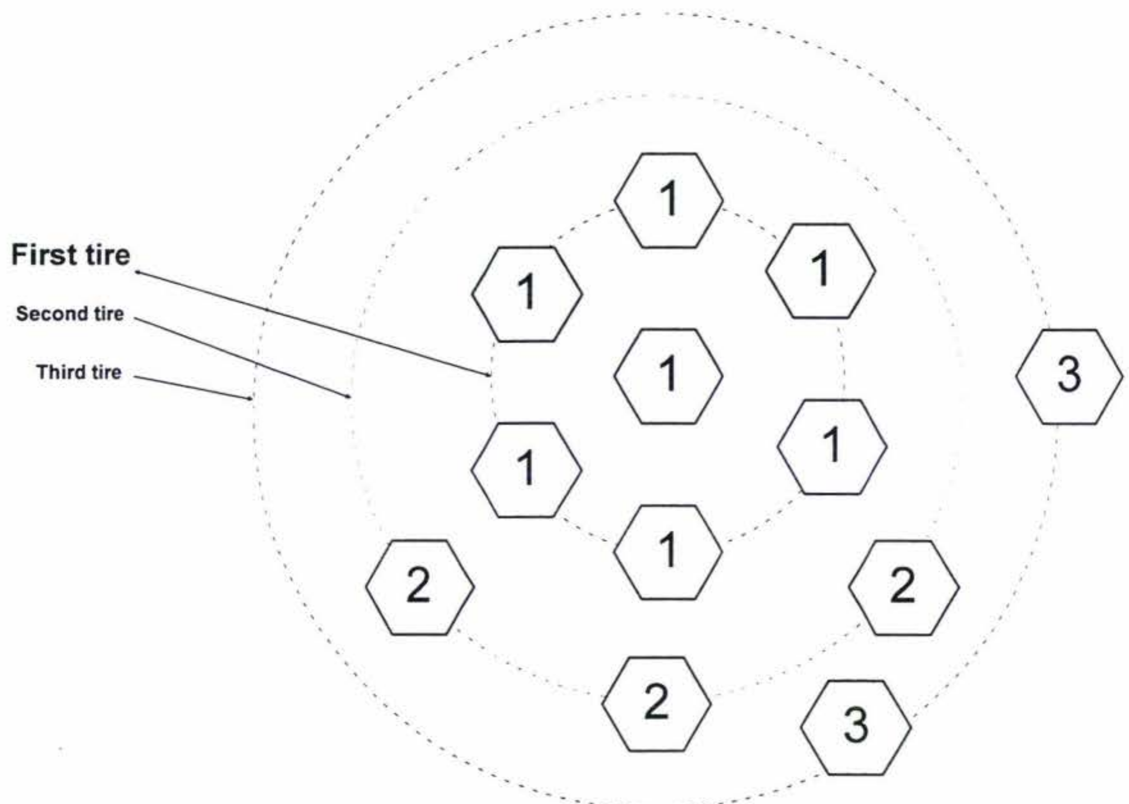


Figure 2.3 Example of co-channel interference with the cluster size equal to 7. There are six effective interfering cells of cell 1 in first tier, and there are also co-channel interfering cells in second tier and third tier and ....

Interference resulting from signals which are adjacent in frequency to the desired signal is called adjacent channel interference. Adjacent channel interference can be minimized through careful filtering and channel assignments. Since each cell is given only a fraction of the available channels, a cell need not be assigned channels which are all adjacent in frequency. By keeping the frequency separation between each channel in a given cell as large as possible, the adjacent channel interference may be reduced considerably. Thus instead of assigning channels which form a contiguous band of frequencies within a particular cell, channels are allocated such that the frequency separation between channels in a given cell is maximized. By sequentially assigning successive channels in the frequency band to different cells, many channel allocation schemes are able to separate adjacent channels in a cell by as many as N channel bandwidths, where N is the cluster size. Some channel allocation schemes also prevent a secondary source of adjacent channel interference by avoiding the use of adjacent channels in neighboring cell sites.

2.1.3 Development of Wireless Networks

The development of wireless networks has progressed through three generations. First generation cellular and cordless telephone networks are based on analog technology. All first generation cellular systems use FM modulation, and cordless telephones use a single base station to communicate with a single portable terminal. A typical example of a first generation cellular telephone system is the Advanced Mobile Phone Service (AMPS) [17] system used in the United States.

Second generation wireless systems use digital communication channels. Examples of second generation wireless systems include the Global System for Mobile (GSM) [18], the TDMA and CDMA US digital standards IS-54 and IS-95 [19], Second Generation Cordless Telephone (CT2) [20], the Digital European Cordless Telephone (DECT) standard, the European standard for wireless network, the Personal Access Communications System (PACS) [21] local loop standard, and many more.

Second generation wireless networks have introduced new network architectures that have reduced the computational burden of the mobile switching center (MSC). For example GSM has introduced the concept of a base station controller (BSC) which is inserted between several base stations and the MSC [22]. This architectural change has allowed the data interface between the base station controller and the MSC to be standard, thereby allowing carriers to use different manufacturers for MSC and BSC components. This trend in standardization and interpretability is new to second generation wireless networks.

All second generation systems use digital voice coding and digital modulation. The systems employ dedicated control channels (common channel signaling) within the air interface for simultaneously exchanging voice and control information between the subscriber, the base station, and the MSC while a call is in progress. Second generation systems also provide dedicated voice and signaling trunks between MSCs, and between each MSC and the PSTN.

In general, second generation systems have been designed to reduce the computational and switching burden at the base station or MSC, while providing more flexibility in the channel allocation scheme so that systems may be deployed rapidly and in a less coordinated manner[23].

Third generation wireless systems will evolve from mature second generation systems. The aim of third generation wireless networks is to provide a single set of standards that can meet a wide range of wireless application and provide universal access throughout the world. In third generation wireless systems, the distinctions between cordless telephone and cellular telephones will disappear, and a universal personal communicator (a personal handset) will provide access to a variety of voice, data, and video communication services.

Third generation wireless systems will use the Broadband Integrated Services Digital Network (B-ISDN) as a backbone to provide access to information network, such as the Internet and other public and private database. Third generation networks will carry many types of information (voice, data, and video), will operate in a varied region (dense or sparsely populated regions), and will serve both stationary users and vehicular users traveling at high speeds.

The terms Personal Communication System (PCS) and Personal Communication Network (PCN) are used to imply emerging third generation wireless systems for hand-held devices. International Mobile Telecommunication (IMT2000) and

Universal Mobile Telecommunication System (UMTS) are examples of future PCS systems [24][25][26].

2.1.4 Wireless LANs

Wireless LAN will be used in office environments in which users are mobile and may require moderate to high bandwidth. WLAN's will be used in environments where cable installation is expensive or impractical. Such environments include manufacturing floors, trading floors on stock exchanges, conventions and trade shows, and history buildings.

There are three technologies which are currently in use for wireless LAN (WLAN) systems: infrared light systems, licensed cellular systems operating at 1.8-1.9 GHz , and unlicensed spread-spectrum systems operating in the ISM bands (902-908 MHz, 2.4-2.5 GHz, and 5.8-5.9 GHz) [27][28].

Compared with Cellular networks which provide an excellent coverage to support wide-area mobile and portable communications service, WLANs are generally used to serve a well-defined used areas, such as a warehouse or an office building. In fact, a WLAN network with a centralised topology can be viewed as a single-cell network, and connection to another "WLAN cell" is typically provided by installation of a bridge or router via a backbone network. Furthermore, the propagation characteristics of WLAN links are such that co-channel interference between separated networks (e.g., networks installed in separate buildings or even separate floors of the same building) can be avoided. Thus cellular configurations with frequency reuse have found only minimal adoption to date in the WLAN industry. Figure 2.4 shows a typical cellular WLAN setup with Microcells interconnected through a backbone Ethernet cable.

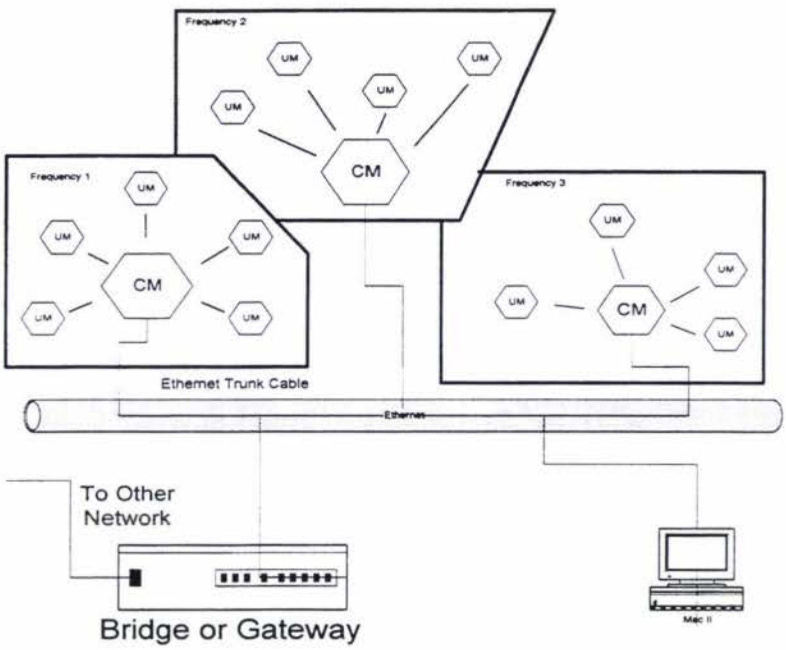


Figure 2.4 Wireless LAN in a cellular environment [29].

The growing demand for wireless data services implemented with battery operated portable computers lead to requirements for wide-area coverage supported by a cellular architecture. The backbone for such cellular networks could be a privately owned cellular architecture such as those used by ARDIS or Mobitex. It could also be a data service such as CDPD overlaid onto the cellular phone infrastructure, or a data service integral to a cellular network such as IS-95 or GSM.

The emerging widespread use of wireless LAN systems together with users' desire for such systems to interoperate has created a requirement for standards. Many standards bodied are currently defining standards for wireless systems that relate to different layers of the networking protocol stack. Of these, two important physical and data link layer standards, IEEE 802.11 and the European HIPERLAN, are described. The IEEE 802.11 [27] committee has been working on the establishment of a standard for wireless LANs. The IEEE 802.11 standard concentrate heavily on method of media access control (MAC), by which multiple users can share a wireless network while minimising the incidence of collisions. The MAC proposals offered by various companies were narrowed down to two potential approaches, CSMA/CA (Carrier-sense multiple access with collision detection and avoidance) and a version of reservation-TDMA. The CSMA/CA approach was finally selected [29].

The general architecture and terminology defined by the 802.11 committee is as follow. Two primary topologies are supported by the 802.11 standard: one in which the stations access the backbone network via access points (i.e., base station), and one in which a group of stations communicate directly with each other in an ad hoc network, independent of any infrastructure or base stations. The first topology is useful for providing wireless coverage of building or campus areas by deploying multiple access points whose radio coverage areas overlap to provide complete coverage. The stations associated with a given access point are referred to as its basic service set (BSS) in the 802.11 standard, but more commonly as the members of the access point's cell. The second topology, the one for ad hoc network, is useful for applications such as file sharing in a conference room scenario. The MAC protocol of the 802.11 standard was developed to allow these two types of topologies to coexist.

The IEEE 802.11 standard defines a single MAC protocol for use with all of the defined physical layers. The use of a single MAC protocol better enables chip vendors to achieve high-volume production, which will help keep the costs low for these systems. The primary access method, the distributed coordination function (DCF), used in the protocol is drown from the family of carrier-sense multiple access with collision avoidance (CSMA/CA) protocols. Since the radio medium does not permit the use of a collision detection (CD) mechansim, as used in the CSMA/CD protocol of Ethernet, the CSMA/CA protocol uses a random backoff to reduce the likelihood of two frames colliding. Collisions are most likely to occur during the time period immediately following the transmission of some frame, since two or more stations may be listening to a busy medium and hence transmit when it becomes free. In the CSMA/CA protocol of 802.11, the random backoff time is distributed according to a uniform distribution (in discrete slot times) where the maximum extent of the uniform range is called the contention window (CW) in 802.11. The CW parameter, that is , the range of this uniform distribution, is doubled (up to a maximum limit) each time a frame transmission is unsucessful, as determined by the absence of an acknowledgment (ACK) frame. This exponential backoff mechanism helps reduce collisions in response to increasing numbers of contending station.

The European community decided to pursue the goal of a wireless LAN that would be indistinguishable in performance from wired LAN's such as Ethernet, and also have some support for isochronous services. A committee was set up to formulate a HIPERLAN standard. Unlike for the IEEE 802.11 standard, this committee was not driven by existing products or regulations. A set of functional requirements was defined, and the committee set out to satisfy the requirements. The MAC protocol used in HIPERLAN is based on a carrier-sensing mechanism, but is quite different in its details from that used in the IEEE 802.11 standard disscussed earlier. The channel access used in HIPERLAN standard consists of three phases: prioritization, elimination, and yield [27][28].

The prioritization phase is aimed at allowing only nodes having packets of the highest available priority to contend further for channel access. This phase consists of a number of slots, with a node having a packet with priority p transmitting a burst in slot p+1 if it has heard no higher-priority burst. At the end of the first burst on the channel, the prioritization phase ends and the elimination phase begin. During the elimination phase, nodes that transmitted a burst during the prioritization phase now contend for the channel. This is achieved by each node transmitting a burst for a geometrically distributed number of slots and then listening to the channel for one time slot. If another burst is heard while listening to the channel, the node stops contending for the channel. Thus , only the node(s) with the longest burst will, in the absence of the hidden node problem, be allowed to further contend for the channel. Immediately after the longest burst and listening period of the elimination phase is the start of the yield phase. In this phase, each of the surviving nodes defers transmission for a geometrically distributed number of slots, while listening to the channel. However, if they hear any transmission, they defer transmission altogether. The purpose of the elimination phase is to bring the number of contenders down to a small number, and then the yield phase tries to ensure that only one node eventually transmits. As a result, the chances of actual collisions for data are negligibly small.

2.2 Classification of multiple access protocols

An important question when designing and standardizing cellular mobile radio system is the selection of the multiple access scheme. Channel-access methods and methods for sharing a channel among multiple users are the essential ingredients in achieving efficient operation and good performance in a wireless network. Users in a wireless network seldom need to have access to a channel for a long period of time. Thus schemes are needed for providing multi-user access, usually referred to as multiple access, to the frequency and time resources of the network in an orderly manner and in a way that minimizes transmission overhead while maximizing overall network capacity.

In any cellular design the overall capacity of each system can differ from system to system. System capacity may be capped by three limiting elements: radio capacity, control link capacity, and switch capacity [30] as shown in Figure 2.5. Radio capacity indicates that how much information can be sent between radio terminal and the base station. Control link capacity measures the capacity of transmitting information over linkages between the base station and the switches. Switch capacity measures the traffic capacity at the switching office. Capacity of the system can be improved by increasing the lowest capacity unit of the system. Here radio capacity is the fundamental element of the three, so the improvement of radio capacity by using an appropriate multiple access technique is an important consideration in cellular network design.

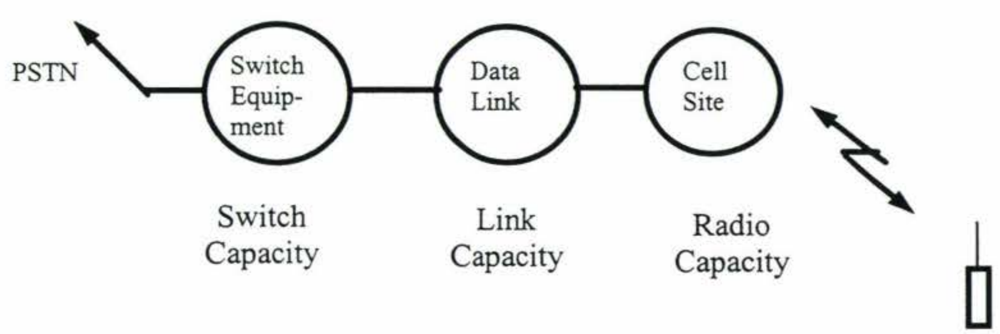


Figure 2.5 Cellular systems Capacity [30].

There are three major categories of channel-access methods for the wireless communications environment: random access, schedule access, and polling.

Random access is an optimistic protocol in which a terminal with a message will transmit it immediately, hoping that no other terminals will transmit at the same time and thus collision it. Since there is no coordination among the terminals, collision may occur. Algorithms must be developed such that collided messages are retransmitted most efficiently. Typical random access protocols for wireless networks are ALOHA, slotted ALOHA, Idle Case Multiple Access with Collision Detection (ICMA/CD) [31], Base Control Multiple Access with Collision Control (BCMA/CD) [32], Busy Tone Multiple Access (BTMA) [33]. ALOHA is one of the common types of random protocol. The maximum channel throughput using the ALOHA protocol is only 18.4%. The slotted ALOHA protocol which divides the channel into time slots is another contention based protocol. With a slotted system a terminal can transmit a packet only at the start of the slot which reduces the vulnerable period. The throughput in slotted ALOHA is 36.8%, twice that of ALOHA [13]. The random protocols like R-ALOHA, ICMA/CD, BCMA/CD, BTMA use a control channel to inform other mobiles about channel status, but there still remains the probability of every packet colliding with another packet, because the performance of these protocols depends on other parameters like separation between mobile and base station, packet size, transmission bit rate and packet detection time.

Scheduled access is a pessimistic protocol in which no one is allowed to transmit until a reservation has been made successfully on its behalf. Therefore, there is no collision. However a price is paid in the additional overhead due to the scheduling process. Scheduled access protocols are recognized as more effective multiple access protocols and are widely used in second and third generation mobile communication networks.

There are three basic multiple access methods which belong to scheduled access - Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA) and Coder Division Multiple Access (CDMA). And based on the TDMA protocol, some packet access protocols were developed. Those are R-ALOHA, Packet

Reservation Multiple Access (PRMA) protocol and Advanced Time Division Multiple Access (ATDMA) Protocol.

Another way of imposing discipline on a network of independent users is to equip one station in the network as a controller that periodically polls all the other stations to determine if they have data to transmit. In marked contrast with R-ALOHA, where control is distributed among all user terminals, polling technique utilizes very centralized control. The controller station may have a polling list giving the order in which the terminals are polled. If the polled station has something to transmit, it starts sending. If not, a negative reply is detected by the controller, which then polls the next terminal in the sequence. Polling is widely used in dedicated telephone networks for data communications, such as networks serving ATM machines and airline reservation systems. However, it has generally not been adopted in existing mobile data networks or WLANs.

2.3 Multiple access protocols for mobile radio

2.3.1 Pure ALOHA

The original ALOHA protocol is sometimes called pure ALOHA to distinguish it from subsequent enhancements of the original protocol. The concept of pure ALOHA, shown in Figure 2.6, is very simple: Users transmit whenever they have information to send. A user sends information in packets, each packet is encoded with an error-detection code. Of course, because users transmit packets at arbitrary times, there will be collisions between packets whenever packet transmissions overlap by any amount of time, as indicated in Figure 2.7. Thus after sending a packet, the user waits a length of time equal to the round-trip delay for an acknowledgment (ACK) from the receiver. If no acknowledgment is received, the packet is assumed lost in a collision and is transmitted again with a randomly selected delay to avoid repeated collisions.

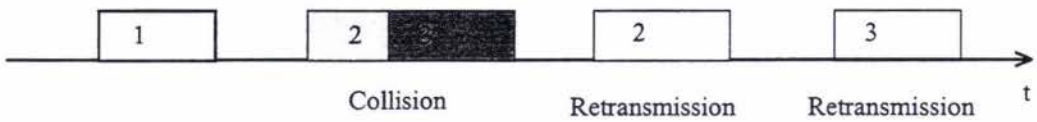


Figure 2.6 Collision mechanism in ALOHA

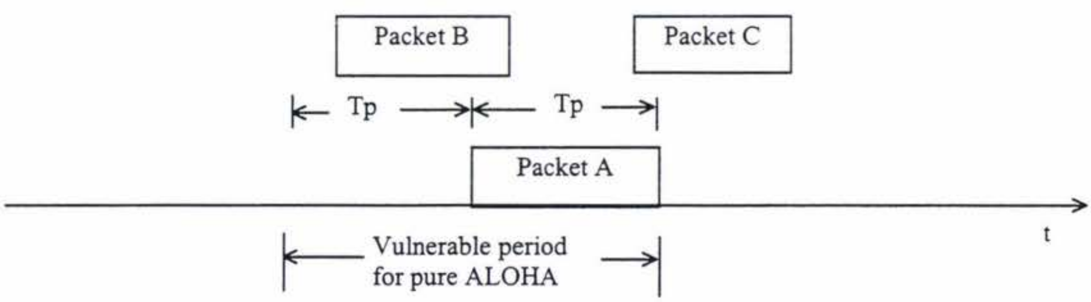


Figure 2.7 Packet collision in pure ALOHA. The vulnerable period is two times the packet interval.

2.3.2 Slotted ALOHA

To increase the efficiency of the ALOHA protocol, the slotted ALOHA scheme was proposed. In this scheme, shown in Figure 2.8, the transmission time is divided into time slots, with each slot exactly equal to a packet transmission time. All the users are then synchronized to these time slots, so that when a user terminal generates a packet of data, the packet is held and transmitted in the next time slot. (Synchronization can be accomplished by transmitting a periodic synchronization pulse from one designated station in the network) With this scheme of synchronized time slots, the interval of vulnerability to collision for any packet is reduced to one packet time from two packet time in pure ALOHA.

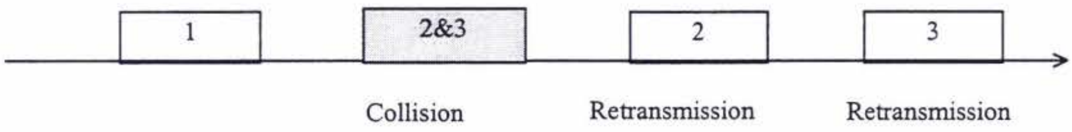


Figure 2.8 Collision mechanism in slotted ALOHA

2.3.3 TM-BCMA/CD

TM-BCMA/CD, is a mini slotted procedure based on three logical channels mapped onto two physical carriers [32]. TM-BCMA/CD uses the DOWN channel to multiplex speech/data blocks with control information. The base splits DOWN packets (base-to-mobile) into blocks with each block appended by a control packet. A data block and a control packet form a mini-slot as shown in Figure 2.9. If there are no data packet transmissions then the DOWN control channel contains CONTROL packets only. CONTROL packets consist of IDLE, BUSY or STOP packets associated with synchronization, addressing, head and tail bits.

For channel access, when mobile stations have a packet ready for transmission, they access that radio channel at the start of the next mini-slot as defined by the DOWN channel, provided that the DOWN control packet contains IDLE(I) information. Otherwise the mobiles reschedule their transmission for some future mini-slot. When the base station detects a packet's preamble at the start of a mini-slot it synchronizes bit timing and continues receiving the UP control packet until the end of CRC. If the CRC succeeds, indicating there was no collision, the base broadcasts BUSY(B) information in the following DOWN control packet. The mobile, upon receiving the BUSY packet, continues with the information packet transmission. Consequently slots are dynamic and shift during a frame depending on whether an attempt for access on a UP channel was successful or not successful. A number of attempts can be made within one slot since mobiles detect collisions within one mini-slot.

Collisions will occur if another mobile begins transmission at the start of the same mini-slot. The vulnerable interval during which packet collisions may occur is only due to the difference in propagation delay between individual mobiles and the base station. In this case, the CRC will fail and the base broadcasts a STOP(S) signal in the following DOWN control packet. Mobiles upon detecting the STOP control packet abort transmission and reschedule the packet transmission at some later mini-slot.

TM-BCMA/CD provides minimum access delay for data traffic, but can not support any significant number of voice connections because of higher contention probability. This protocol can only be used for data traffic.

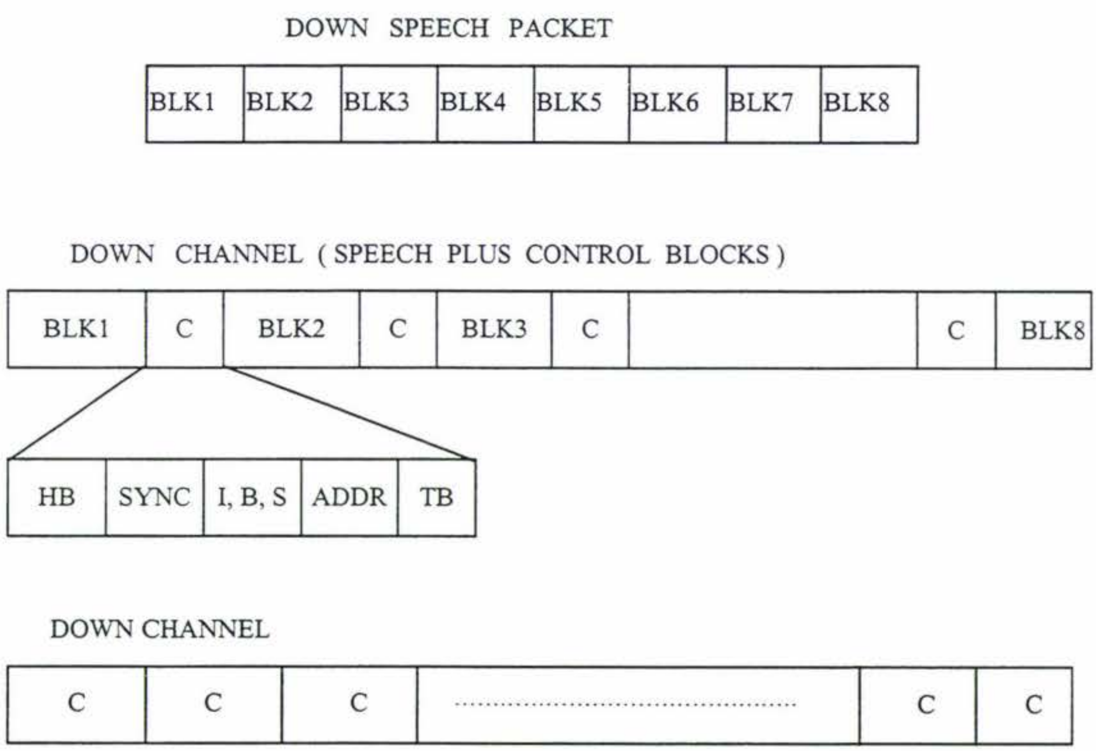


Figure 2.9 TM-BCMA/CD channel structure

2.3.4 The Frequency Division Multiple Access (FDMA)

The frequency division multiple access (FDMA) assigns individual channels to individual users. It can be seen from Figure 2.10 that each user is allocated a unique band and channel. These channels are assigned on demand to users who request service. During the period of a call, no other user can share the same frequency band or channel.

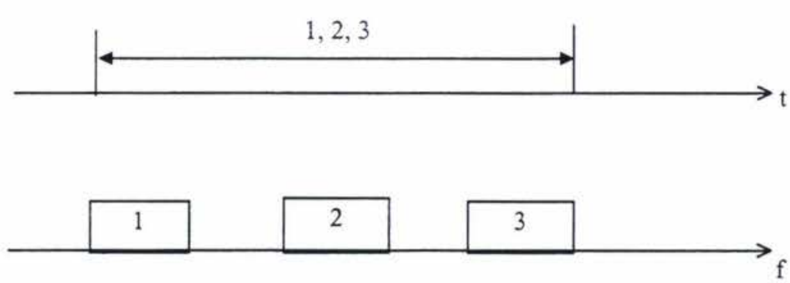


Figure 2.10 FDMA format

The FDMA scheme is widely used in first generation mobile radio systems which are called analog mobile radio systems, such as Advanced Mobile Phone Service (AMPS). FDM technique is employed in the GSM system for splitting the UP & DOWN links.

2.3.5 Time Division Multiple Access (TDMA)

Time Division Multiple Access (TDMA) divides the radio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive. It can be seen from Figure 2.11 that each user occupies a cyclically repeating time slot, so a channel may be thought of as a particular time slot that reoccurs every frame, where N time slots comprise a frame. The Packet Reservation Multiple Access (PRMA) protocol and the Advanced TDMA protocol (ATDMA) are all TDMA based protocol. These protocols are organized around the TDMA slot and frame structure. They are a scheduled access packet transmission technique, which operates in the reservation model for multimedia traffic. PRMA and ATDMA protocols will be discussed in detail in the next section. Different TDMA wireless protocols have different TDMA frame arrangements, and some are described in the next section. The features of TDMA include the following:

- TDMA shares a signal carrier frequency with several users, where each user makes use of nonoverlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth, etc.
- Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use.
- Because of discontinuous transmissions in TDMA, the hands-off process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots. An enhanced link control, such as that provided by mobile assisted hands-off can be carried out by a subscriber by listening on an idle slot in the TDMA frame.

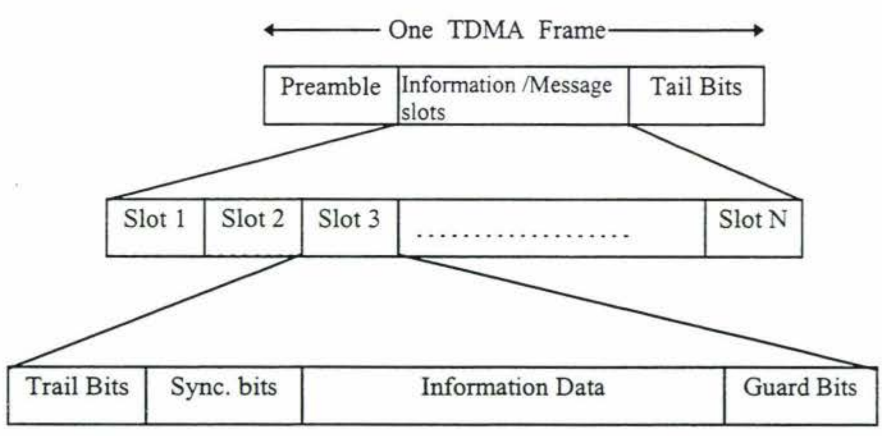


Figure 2.11 TDMA frame structure

- TDMA uses different time slots for transmission and reception, thus duplexes are not required. Even if FDD is used, a switch rather than a duplexer inside the subscriber unit is all that is required to switch between transmitter and receiver using TDMA.

- Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.
- In TDMA, the guard time should be minimized. If the transmitted signal at the edges of a time slot are suppressed sharply in order to shorten the guard time, the transmitted spectrum will expand and cause interference to adjacent channels.
- High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted. This requires the receivers to be synchronized for each data burst. In addition, guard slots are necessary to separate users, and this results in the TDMA systems having larger overheads as compared to FDMA.
- TDMA has an advantage in that it is possible to allocate different numbers of time slots per frame to different users. Thus bandwidth can be supplied on demand to different users by concatenating or reassigning time slots based on priority.

2.3.6 Code Division Multiple Access (CDMA).

Spread spectrum multiple access (SSMA) uses signals which have a transmission bandwidth that is several orders of magnitude greater than the minimum required RF bandwidth. A pseudo-noise (PN) sequence converts a narrowband signal to a wideband noise-like signal before transmission. SSMA also provides immunity to multipath interference and robust multiple access capability. SSMA is not very bandwidth efficient when used by a single user. However, since many users can share the same spread spectrum bandwidth without interfering with one another, spread spectrum systems become bandwidth efficient in a multiple user environment. It is exactly this situation that is of interest to wireless system designers. There are two main types of spread spectrum multiple access techniques; frequency hopped multiple access (FH) and code division multiple access (CDMA).

In a CDMA system, the narrowband message signal is multiplied by a random sequence which is known as the spreading signal. The spreading signal is a pseudo-noise code sequences that has a chip rate which is orders of magnitudes greater than the data rate of the message. All users in a CDMA system, as seen from Figure 2.12 use the same carrier frequency and may transmit simultaneously. Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords.

The receiver performs a time correlation operation to detect only the specific desired codeword. All other codewords appear as noise due to decorrelation. For detection of the message signal, the receiver needs to know the codeword used by the transmitter. Each user operates independently with no knowledge of the other users.

In CDMA, the power of multiple users at a receiver determines the noise floor after decorrelation. If the power of each user within a cell is not controlled such that they do not appear equal at the base station receiver, then the near-far problem occurs [23].

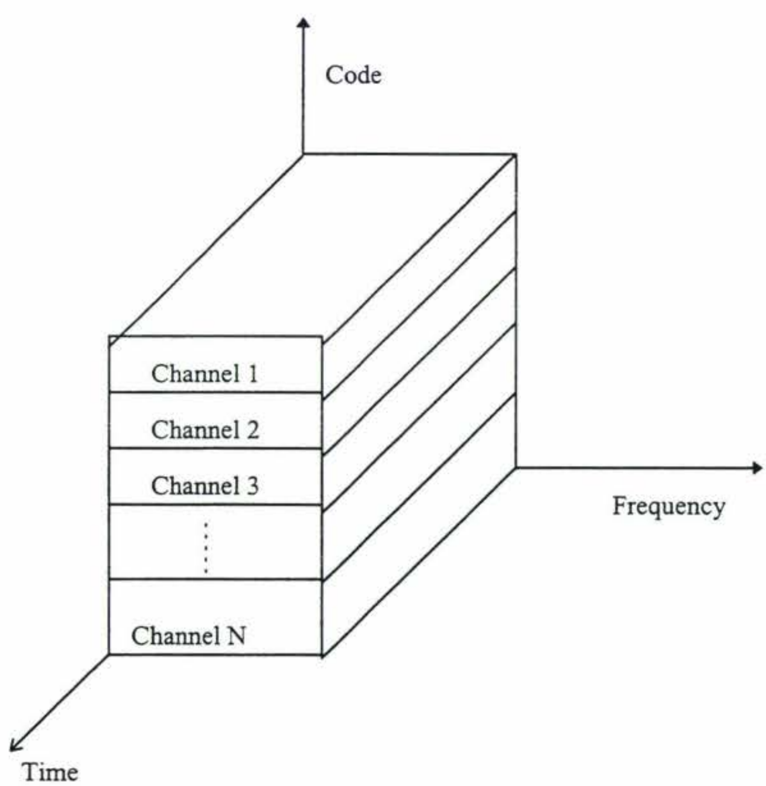


Figure 2.12 CDMA system in which each channel is assigned a unique PN code which is orthogonal to PN codes used by other users.

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will capture the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the possibility that weaker signals will be received. To combat the near-far problem, power control is used in most CDMA implementations. Power control is provided by each base station in a cellular system and enssures that each mobile within the base station coverage area provides the same signal level to the base station receiver. This solves the problem of nearby subscriber overpowering the base station receiver and drowning out the signals of far away subscribers. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link. Despite the use of power control within each cell, out-of-cell mobiles provide interference which is not under the control of the receiving base station. The features of CDMA include the following:

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.
- Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the number of users in a CDMA system raises the noise floor in a linear manner. Thus, there is no absolute limit on the number of users in CDMA. Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased.
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading.
- Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, a multipath which is delayed by more than a chip will appear as noise. A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal.

- Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft hand-off. Soft hand-off is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may chose the best version of the signal at any time without switching frequencies.
- Self-jamming is a problem in CDMA system. Self-jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of the users in the system.
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

2.3.7 R-ALOHA

R-ALOHA is a reservation based protocol, the basic idea of this protocol is that the terminal which wants to transmit packet/packets reserves a slot for a certain duration. Other terminals which are also attempting transmission will refrain from using that slot. A terminal which holds a slot must release that channel as soon as it finishes transmission[13]. The methods differ in the way reservations are made. One of the versions uses a subchannel for the implementation of reservation. This scheme requires every terminal to make advance requests before transmitting. Each frame contains one special slot as shown in Figure 2.13 which is divided into several small subslots used to make reservations. When any terminal wants to transmit data, it sends a request packet during one of the reservation subslots. If the reservation is successful i.e. if the packet was received correctly by the receiver, then a regular slot is reserved for the terminal. Another version of the R-ALOHA uses the regular slots for obtaining a reservation. When a terminal has a packet/packets to transmit, it sends a packet in an available free slot. If the packet is successfully transmitted then that terminal reserves that slot till the end of its use. In a reservation scheme all the terminals must keep track of the status of the slots. The efficiency of the reservation scheme depends on the length of channel holding time and feedback time indicating success or collision.

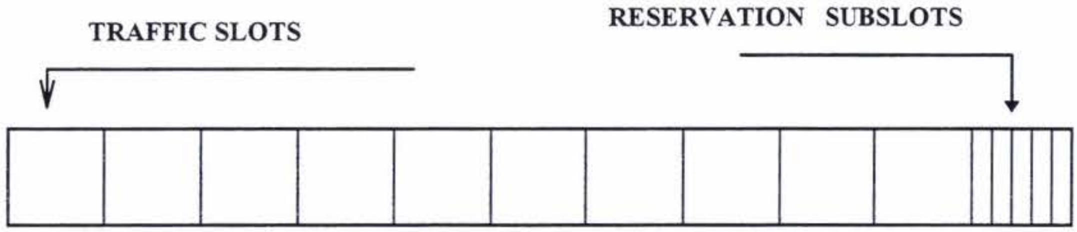


Figure 2.13 Slot structures for R-ALOHA

2.3.8 Packet Reservation Multiple Access (PRMA)

PRMA [35][36][37][38][39][40][41] is a scheduled access packet transmission technique, which operates in the reservation model for speech traffic. It uses a contention model for obtaining a reservation for a speech terminal and also for transmitting data packets. PRMA operation is organized around the TDMA slot and frame structure. Two different carriers are used, one for the UP channel and one for the down channel as shown in Figure 2.14.

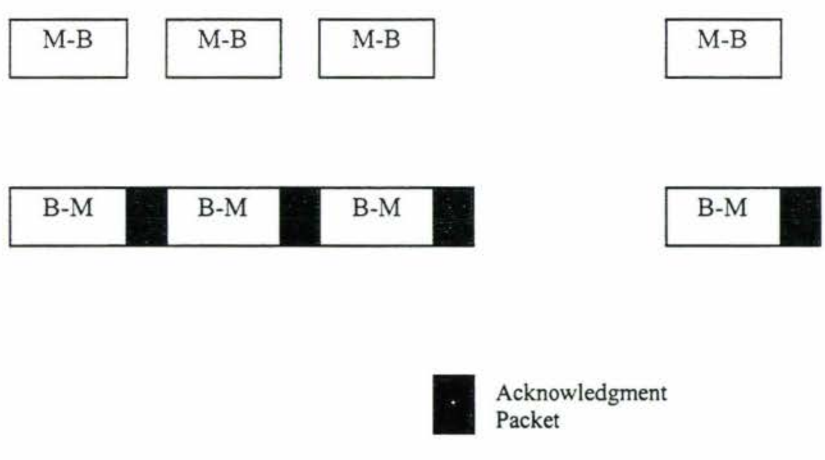


Figure 2.14 PRMA channel structure

The up channel is used to transmit data or speech packets from mobile to base using a particular slot. The down channel is used to transmit information packets from base to mobile as well as the control information contained in the ACKNOWLEDGMENT packet. PRMA can accommodate mixed traffic e.g. voice and data. Voice packets are given priority over data packets by offering reservation to speech traffic, while data packets contend on each access. The status of each slot in the up channel is continuously broadcast by the base station via an ACKNOWLEDGMENT packet. Thus each mobile is aware of the status of each slot in the current frame.

A voice source will attempt transmission at the beginning of a talkspurt and will use the next available free slot. If the voice packet is accurately detected by the base station, it gains a reservation to that mobile for the duration of the talkspurt, by means of the acknowledgment packet.

When a data packet is accurately detected by a base station no reservation is granted but the base informs the mobile about the successful packet transmission. Should the base fail to detect the packet, due either to collision or bit errors, no reservation is granted to voice sources and no acknowledgment is given to data sources.

In the case of an unsuccessful transmission a voice terminal attempts to retransmit that packet until that packet is successfully transmitted. This procedure capitalizes on the principle that voice sources may accommodate occasional packet loss, whereas data sources may accommodate packet delay. Voice packets are dropped when the speech packet dropping threshold expires. However data packets are not dropped because of longer access delay.

To retransmit a collided packet a mobile must have permission and there must be a free slot available. A mobile station retransmits with a certain probability. This value is a system variable and is related to the system operating parameters. The choice of this "permission probability" directly affects operational efficiency. It is suggested by Goodman et.al [37] that the optimum permission probability for voice is 0.25 and for data 0.125, for the system parameters used.

PRMA is a suitable option for the integrated voice and data services [35][36]. This protocol provides lower speech packet loss and also lower access delay for the data traffic. It is suitable for use in smaller cells with higher transmission bit rate.

PRMA could be an attractive option for the integrated voice/data service for mobile radio because it offers several advantages, which are[35]:

- Cell size has less effect on performance.
- Speech quality will be less affected because the speech packets are not lost in the middle of talkspurts.
- The efficiency increases with the increase of transmission bit rate which will be useful for the microcellular environment.
- Resource, i.e. slots, are used most efficiently.

2.3.9 Advanced Time Division Multiple Access (ATDMA)

The Advanced-TDMA protocol is an advancement of PRMA and thus sometimes is known as PRMA++ [42]. In recent years it is being studied for the 3rd generation mobile radio systems [26][43][44][45][46]. The ATDMA protocol is a flexible one which can be configured in different ways to suit the requirements of different types of traffic. The multiplexing structure of the protocol is shown in Figure 2.15. A request packet from a mobile station (MS) is sent via a R slot on the UP link using the slotted ALOHA algorithm. If a traffic slot is available on the UP link the base station (BS) upon receiving the request packet allocates a traffic slot (T slot) on the UP link to the requesting terminal to transmit its information packets. If no traffic slot is available on the UP link then the request is kept in the allocation queue and a T slot is allocated when the resources are available using the First In First Out (FIFO) technique. The BS sends the acknowledgment information to the MS via the A slot on the DOWN link. In the case of speech packet transmission speech packets are dropped from the transmission buffer if they cannot be transmitted within a specified duration i.e. the packet dropping threshold time. Data packets are not dropped and they are kept in the buffer until they can be transmitted. The ATDMA protocol can adapt in the different channel environment with different transmission mode. Detailed discussion about the ATDMA protocol will be provided in the next chapter.

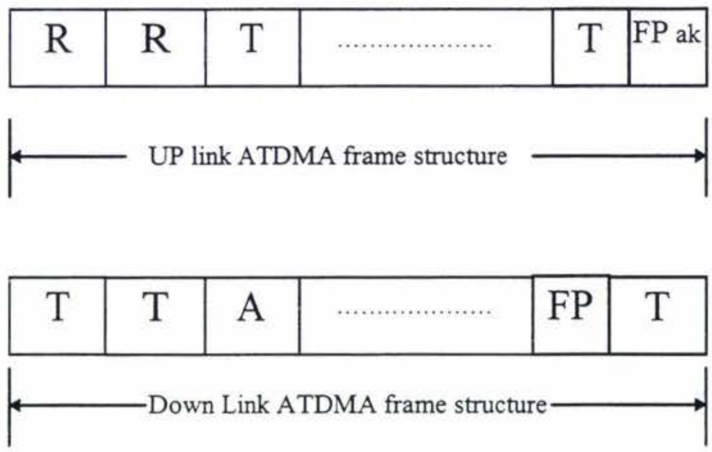


Figure 2.15 A-TDMA Frame Structure

2.4 Discussion and comparison

The TDMA and CDMA schemes have been used in second generation mobile radio systems which are based on digital transmission and hence are called digital mobile radio systems. Examples of digital mobile radio systems are the TDMA based Global System for Mobile Communications (GSM), IS-54, Pacific Digital Cellular (PDC), Digital Europe Cordless Telephone (DECT), Personal Handyphone System (PHS) and the CDMA based IS-95. Digital mobile radio systems can support more users per base station per MHz of spectrum, allowing wireless system operators to provide service in high-density areas more economically. Presently, many researchers and engineers around the world are engaged in determining multiple access schemes appropriate for third generation cellular mobile radio system UMTS. This task is a major technical challenge because a large variety of scenarios and services have to be taken into account, and because it is likely that unique optimum solutions do not exist.

There are advantages in using these access techniques, some of then include [46][47][48][49][50][51]:

One advantage of TDMA compared with FDMA and CDMA is that common radio and modem equipment, at a given carrier frequency, can be shared among N users at a base station[36]. Another advantage with respect to FDMA is that bit rates to and from each individual user terminal can be easily varied according to current user needs, by allocating more or fewer timeslots to the user. This is especially advantageous for integrated service applications.

With respect to CDMA, TDMA has the advantage of less stringent power control requirements, since interuser interference is controlled by time slot and frequency allocation instead of by processing gain resulting from coded bandwidth spreading.

Another important advantage related to FDMA and CDMA is that the time slot structure gives time for measurements of alternative slots, frequencies, and ports in order to support mobile assisted or mobile controlled hand-off.

TDMA also has some advantages with respect to FDMA and CDMA. Because user terminals have a 1/N duty cycle, they have a periodically pulsating power envelop. This presents a challenge to designers of portable RF units. Frequency and time slot assignment and management entail a certain extra complexity in TDMA systems, which is not found in CDMA system. Also, the N time higher bit rate means that TDMA may require equalization against multipath, which is generally avoided with FDMA.

ATDMA and PRMA are TDMA based packet access mechanisms. TDMA is advantageously combined with packet-type multiple access schemes in integrated voice/data applications, involving perhaps several different types of traffic with different bit rates and on/off characteristics. In these two protocols, a dynamic slot allocation procedure in which idle time slots are requested by users on a contention basis. If a user is successful in reserving one or more slots in a frame, that slot or slots is reserved for that user as long as required. When a slot is relinquished by a user (at the end of speech burst or data packet), it is open for contention and reservation by other users. It should be noted that communications (both voice and data) consist in a sequence of activity periods (we refer to talkspurts and dataspurts for voice and data, respectively) and silence periods. In PRMA and ATDMA protocol, an activity detection is implemented to reduce offered traffic by the inverse of the activity factor, while keeping the same offered traffic. This is not the case for conventional multiple access schemes.

2.5 Conclusion

The main topic in this chapter is multiple access schemes for mobile radio. Some of the typical mobile radio systems and multiple access protocols have been discussed. Comparisons of advantages and disadvantages of these multiple access methods has also been made. Generally speaking, a reservation access protocol provides some advantages over a random access protocol for transmitting realtime traffic eg. voice. A packet access protocol provides some advantages over a fixed-allocation protocol. A TDMA based protocol provides several advantage over a CDMA based protocol. From the discussion it is apparent the packet based reservation protocol such as PRMA & ATDMA can be very strong contender for third generation mobile radio systems.

CHAPTER III

ATDMA AIR INTERFACE IN A MICROCELL ENVIRONMENT

3.0 Introduction

Second generation cellular mobile radio systems are used by millions of users and their use is increasing rapidly. It is expected that the capacity offered by these systems will be exhausted in the near future. Second generation systems are primarily voice-oriented, however, some of them support low speed data services [41]. Nevertheless, broadband services are not supported. For capacity reasons and in order to be able to offer new services and system features, new initiatives of innovation are needed which will lead to third-generation cellular mobile radio systems.

The CCIR Committee TG 8/1 is contemplating a Future Public Land Mobile Telecommunication System (FPLMTS currently known as IMT-2000) that will provide a world-wide Personal Communications Network (PCN). In Europe the RACE programme was launched in 1987 and included projects to identify the enabling techniques for the Universal Mobile Telecommunication System (UMTS). The essential goals of UMTS and IMT-2000 are the same, and the system they seek is often referred to as the third generation PCN [50].

In this chapter the ATDMA protocol which has been proposed by the RACE ATDMA project as an air-interface to support UMTS will be discussed. Section 1 will introduce the ATDMA multiple access protocol structures and ATDMA system design requirements. Section 2 will briefly discuss the ATDMA function models, section 3 will discuss the advantages of the ATDMA protocol and finally microcell environment and microcell propagation issues will be discussed in section 4. The microcell model will be used to evaluate the performance of the ATDMA protocol in a cellular radio environment. Section 5 is conclusions. It is likely that the ATDMA protocol will be used as an air interface in the third generation UMTS based mobile radio networks [51][52].

3.1 ATDMA protocol structure and system requirements

UMTS, a third generation mobile system, envisaged to begin service some time after the year 2000, is expected to be able to provide any telecommunication service to anybody to anywhere at anytime by using a single identity [53].

The ATDMA protocol is being investigated as a potential air-interface for the UMTS. Its design requirements are considered from both the user and system perspective. In this section the ATDMA protocol structure, ATDMA based system's key design requirements, operating environments, and its supported services, will be discussed.

3.1.1 ATDMA protocol structure

In an ATDMA system, capacity is allocated on demand using the ATDMA protocol proposed by the RACE project [42]. The ATDMA protocol is also known as the PRMA++. Like the original PRMA scheme proposed by Goodman [54], the ATDMA protocol avoids wasting capacity by not allocating resources during the inactive period of a traffic source (during silence periods in speech or for inactive period of a data burst). The configuration of the protocol is flexible to allow customisation by the operator.

The main features of the ATDMA protocol are:

- Time-slot allocation under base station control
- Separate physical channels for access control
- Physical channels allocated when requested, and kept until released (separate request not needed for each block sent)
- Common technique used for all traffic and dedicated control channels.

The method of allocation is service dependent, as shown in the table 3.1.

Bearer type	**Access method**
Voice	Access on a speech spurt basis, release at end of talkspurt.
Constant bit rate data (low and long constrained delay)	Access at the start of a call only. For long delay services that have a burst nature and can tolerate either some clipping or a variable initial delay then packet access could be used instead of circuit switched.
Unconstrained delay (cell relay)	Access on a traffic spurt or individual ATM cell basis. Release at end of traffic spurt or interruption for a higher priority service. When a service is interrupted the BS will know that it still requires capacity, and allocate resources as they become available.

Table 3.1 Access method for each bearer service type [55]

The ATDMA protocol has the same frame and slot structure as the TDMA protocol [42]. The slots on the up-link of A-TDMA are separated into reservation slots ('R' slots), traffic slots ('T' slots) and fast paging-acknowledgement slots ('FP_{ak}' slots). Slots on the down-link are separated into acknowledgement slots ('A' slots), fast paging slots ('FP' slots) and traffic slots ('T' slots). Figure 3.1 illustrates up-link and down-link A-TDMA frame structures. A mobile transmits a reservation request, or a random access burst in a 'R' slot whenever a burst of activity commences. The requesting terminal is allocated a 'T' slot immediately if resources are available for the duration of that activity (e.g. talkspurt, data block or video frame duration). The slot allocation is acknowledged using the paired 'A' slot on the down-link. In the case of resources not being available, the reservation request is queued and an acknowledgement is sent via the 'A' slot. The mobile will then continue to monitor the 'A' slot until it receives a slot reservation. Therefore, reservation requests are not blocked when all 'T' slots are allocated (as in PRMA). The base station has centralised control over the 'T' slot allocation policy.

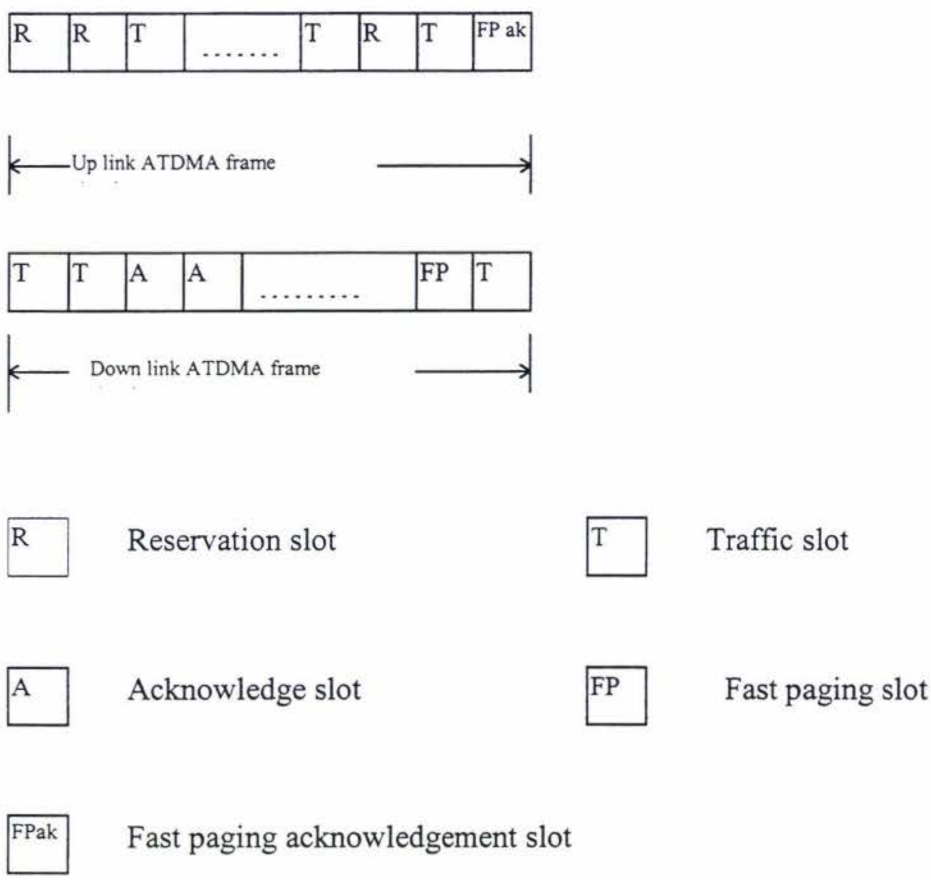


Figure 3.1 A-TDMA Frame structure

Resource reservation requests are transmitted in the 'R' slots using the ALOHA protocol and are therefore subjected to collisions. When collisions occurs all the collided terminals back off and retransmit. However in the above situation if the capture effect is used then the strongest mobile station is selected by the base station [56][57]. The mobiles which do not get the chance to transmit (which do not receive a positive acknowledgement on the paired A slot) might enter a collision resolution phase, where they would get a chance to re-transmit their reservation requests.

If an access attempt is unsuccessful then the mobile will re-transmit with a given re-transmission probability in the next available 'R' slot. The attempt for retransmission will continue until the packet is successfully transmitted or the packet dropping threshold is exceeded. When the threshold is exceeded, in the case of voice, the packet would be dropped.

Data traffic is a loss sensitive and delay insensitive source of traffic. Therefore, no data packet would be dropped. Due to the delay insensitive nature of data traffic a data terminal will not re-transmit its request packet in the current frame. The data terminal will retransmit in the next frame. Therefore, voice terminals have the higher priority during re-transmission of request packets whereas a data terminal have the lowest priority.

When the 'R' slot access attempt is successful the base station enters the 'T' slot allocation process. A terminal's request will be queued if there are no 'T' slots available. In the case of voice, the terminal will remain in the queue for the available 'T' slot allocated by the base station. If a 'T' slot is not available, the terminal will drop the packet when the delay threshold is exceeded. Data terminals do not drop packets.

When a 'T' slot is available the first mobile in the queue will be allocated the slot and this would be acknowledged through an 'A' slot on the down-link. Once a 'T' slot is successfully allocated to a mobile, the mobile transmits its packets in the reserved model. At the end of each talkspurt the allocated 'T' slot will be released by the mobile and made available to another mobile. On the other hand, data bursts are sent in blocks of M packets. When M packets of the data burst have been transmitted, the 'T' slot would be released. For the data terminals to transmit the remaining packets it would go through the slot allocation process again.

Data bursts are sent in blocks. Each transmitting data terminal will obtain a short block reservation which allows them to transmit M consecutive packets in reserved mode in M consecutive transmission frames. The length of reserved block is the same irrespective of the data segment length of a transmitting terminal. If a transmitting data terminal has more than M number of data packets to transmit then it will transmit those using multiple reservations. The idea behind the block reservation is to minimise contention during access and also give priority to voice terminals. A very small reservation block will introduce high contention whereas a larger reservation block can introduce extra delay for the voice packets thus increasing the number of dropped speech packets

3.1.2 ATDMA Operating environments

The terrestrial component of UMTS must operate in a range of environments from rural settings to indoor. This can not be optimally met with one radio interface with a fixed carrier spacing and so an ATDMA system is designed for three basic cell types: pico, micro and macro cells. Each cell type is supported with a different physical layer.

The concept of picocell, microcell, and macrocell are defined by their cell size, that is the maximum dimension of the cell. Macrocells which will support a range below 35 km are used to provide wide area cellular coverage and used as "umbrella cells" offering back-up and "gap filling" coverage for a predominantly microcell region. Microcells which can support ranges below 1 km are used in high density traffic areas, for example urban areas, for increasing the spectrum efficiency. Picocells which can support the range below 100m are largely used for indoor applications offering the full set of UMTS services including high bit rate data services.

These cell types are shown in Figure 3.2 and some of the significant parameters are presented in Table 3.2. Microcell problem will be discussed in depth in section 4.

3.1.3 Services supported by an ATDMA system

Future wireless networks must support a wide range of services from speech to high bit rate data, many of which will be asymmetric, that is, different bit rates for up and downlink. Furthermore future wireless networks like UMTS should be "future-proof" , so the basic radio access infra-structure should be able to support new types of teleservices. To achieve these goal, the ATDMA system only needs to offer a limited set of standardised, independent, uni-directional radio connection elements (referred as "bearers") based on a short list of standard types. The proposed list is given in table 3.3 [26].

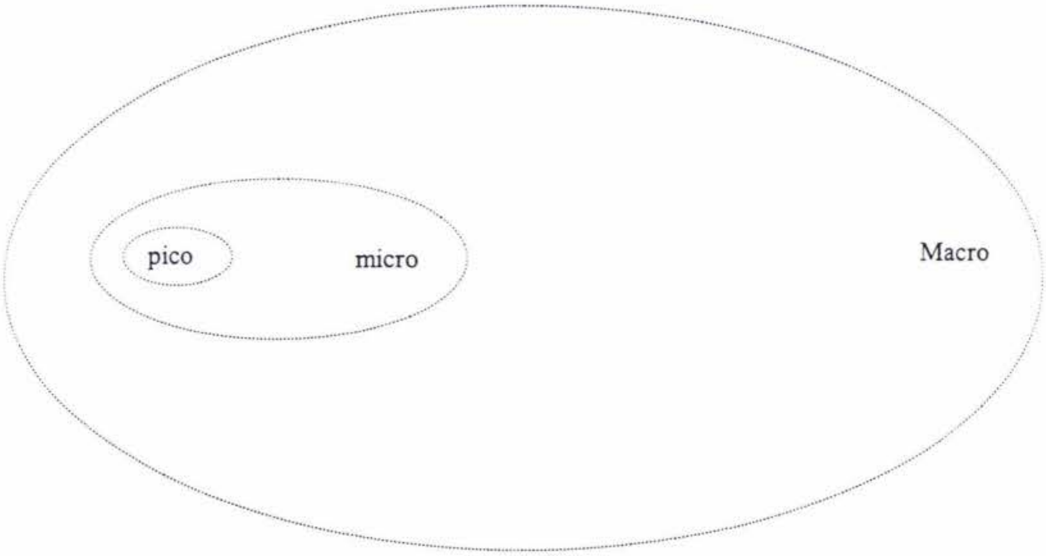


Figure 3.2 Cell types – macrocell, microcell, and picrocells

Parameter	**Picocell**	**Microcell**	**Macrocell**
Cell radius	<100m	<1 km	<35 km
BS antenna	Ceiling mounted	Below building height	On top of building or on tower.
Application	Indoor	High density outside areas	"Umbrella" cells, low density areas
Services	All services	Most services	Limited sub-set
Max. Velocity	10 Km/h	100 Km/h	250 Km/h

Table 3.2 Principal parameters for each cell type [55]

Service	Design constraint	Performance targets
Speech	Delay < 30 ms	MOS > 4.0 Decoded BER < 10^{-3} Frame error rate < 2%
Low delay data	Delay <30 ms	BER < 10^{-6} Error sec < 10 s/h
High delay data	Delay < 300 ms	BER < 10^{-6} Error sec < 10 s/h
Unconstrained delay data	8 and 53 byte cells Packet loss < 10^{-6}	Av. Delay < 50 ms 90% delay < 100 ms

Table 3.3. Parameters for ATDMA bearer service types [26]

3.1.4 Duplexing choice

It is expected that the ATDMA system will operate with two unequal bands (1885-2025 and 2110-2200 MHz) will be difficult to exploit with a frequency division duplex(FDD)-only system. However, it is also clear that large cells will require the use of FDD, so the overall system must support both frequency and time division duplexing [55].

3.1.5 Adaptive air interface

It is expected that future wireless networks will operate in different scenarios, with different types of cells and support different services. All these factors place different constraints and requirements on the radio access system design and so a single set of air interface parameters will, undoubtedly, not be jointly optimal in all the combination. This consideration has let to the ATDMA's concept of an adaptive air interface which is based on the optimisation of use of the available resources so that the links are assigned the minimum resource required to maintain the service quality.

The concept of adaptation is split into two aspects; the one related with short to medium term adaptation is called dynamic adaptation, and the one related to medium to long term adaptation is called static adaptation.

Static adaptation group describes the way the radio access system can be adapted to suit the particular needs or requirements of Users, Operators, Cell types and Service types. From the viewpoint of the base station, these adaptation models will be somewhat static, that is, they can only vary on a medium to long term basis. Furthermore, the current nature of the air interface will be defined at call set-up and will only change during inter-operator or inter-cell type handover, due to the moment of mobile between different cell types, or in special operating situations.

Dynamic adaptation is a kind of time varying adaptation to the current propagation environment and the net traffic load on the terminal and the base station. For example, the Forward Error Control (FEC) rate, slot allocation and interleaving depth could be mutually changed to minimise the required average resource allocation while still maintaining the required Quality of Service.

3.1.5.1 Adaptation to cell types

The ATDMA system addresses the issue of mixing cells by defining a flexible air interface in which a number of variants of a generic transmission system can match the propagation characteristics in the cell. As an example, the propagation delay is much smaller in a picocell than in a macrocell so higher bit rates and small overheads can be supported in picocells. This is important not only for allowing more speech traffic on a single carrier but also to support the high user bit rates which will be used in small cells.

3.1.5.2 Adaptation to Users/Traffic

It is expected that by means of appropriate mechanisms the resources available for allocation to users at different places can be dynamically arranged to match the particular expected demand at those places. In this way more capacity can be allocated to hot-spots in city centres during working hours and then spread over the residential areas afterwards. The type of resources whose allocation could be rearranged is, for example, the set of valid frequencies at a BS or the permitted working power levels. The Dynamic Channel Allocation (DCA) technique, which has been developed to avoid the need for detailed frequency planning, also offers support for dynamically allocating resources to the base station.

The ATDMA system can also adapt to changes in traffic load by short term relaxation of service quality objectives. This will introduce a degree of "soft capacity" limits to the system and so a trade-off between call blocking and minimum service quality can be made by the operator.

3.1.5.3 Adaptation to Interference

The key parameters in any cellular mobile communication system is the ratio C/I (Carrier to Interference Ratio) at the receiver. Instead of designing for the worst case of this ratio, the ATDMA system is based on an adaptive air interface so that the system can efficiently operate at different C/I values. The different dynamic adaptation strategies tend to offer the highest possible value for this ratio in all situations.

Two different approaches are followed to control this ratio: a power control algorithm which sets the carrier power at the lowest possible level while still maintaining link quality and a longer term "link adaptation" algorithm which selects the smallest level of redundancy required to offer service integrity.

Power control is used on both up and down links. The control algorithm operates on a slot basis and uses link quality measurements on the link being controlled for longer term control, and measurements on the reverse link (as an estimate of pass loss) for short term control. The range of short term power control can be adjusted depending on the correlation between path loss in the two directions. For services with multi-slot allocations, each traffic slot is controlled via an independent process.

The power control is slightly different for the up and down links, for circuit and packet switch bearers and whether or not the traffic channel is bi-directional.

The operating mode of transport (coding, modulation, interleaving, etc.) for each of the delay constrained bearer services (speech and low and long constrained delay data) is adaptive to meet changing conditions. The link adaptation process is a radio access control technique which selects the operating mode based on the need to ensure that service quality is being maintained with the minimum of assigned radio resources.

There are two different link adaptation algorithms in an ATDMA system:

- Short term link adaptation. This process operates with an update period of between 0.5 and 5 seconds and bases its decision on the observed average channel quality of the assigned radio resource while it is under control of the faster power control process. An operating mode change is performed whenever average channel quality or transmit power are observed to be outside their permitted operating ranges.

- Long term link adaptation. This process is based on the observation that the distribution of C/I is dependent on the distance from the base station and so the transport operating mode is selected accordingly. In this way mobiles close to the cell centre will be assigned the most spectrum efficient operating mode while mobiles near the cell boundary are assigned a more robust, and hence more resource consuming, operating mode.

3.1.5.4 Adaptation to source activity

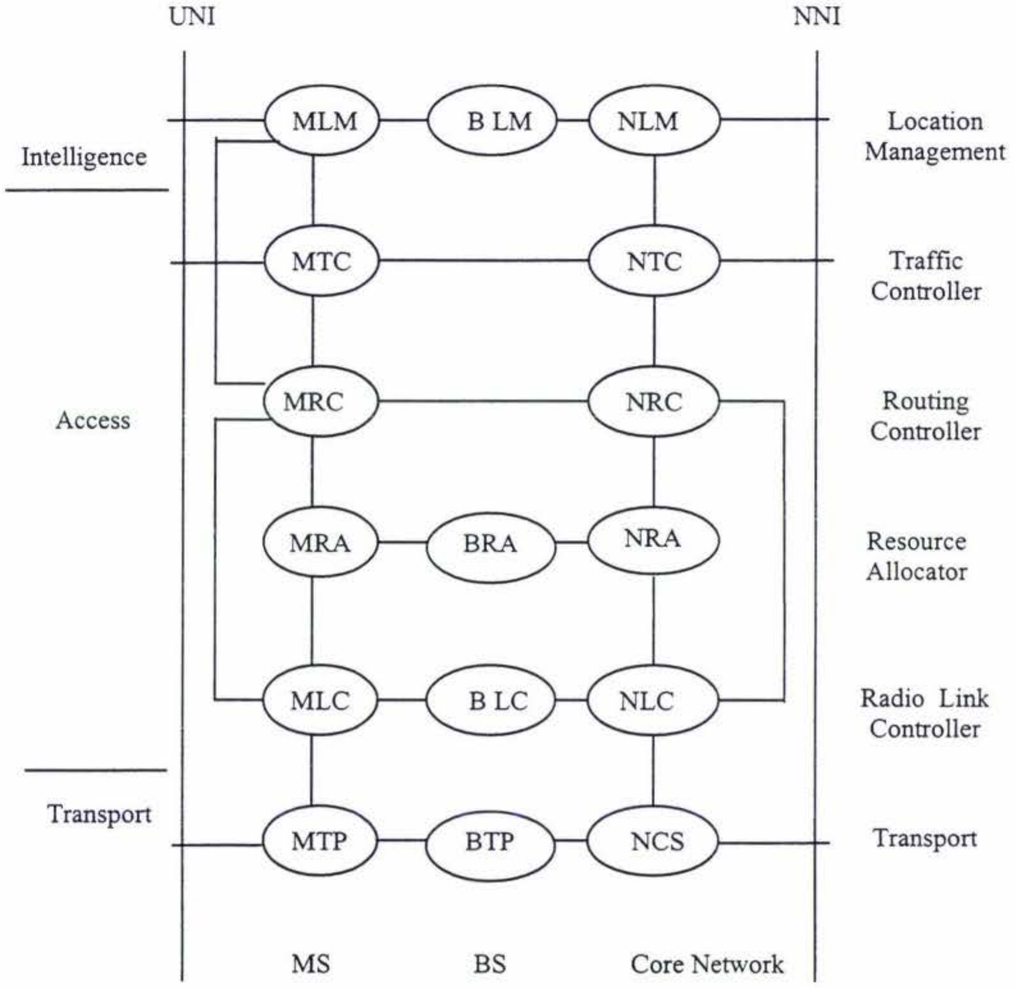
An additional aspect of interference control is the exploitation of source activity, particularly for speech services. Instead of using a discontinuous transmission technique, as in GSM, the ATDMA system uses packet access technique where radio resources are assigned on a needs basis only. With respect to interference reception, the advantage of packet assignment is that while the mean occupancy of a TDMA slot is increased, the total number of installed channels per cell can be reduced through statistical multiplexing.

3.2 The ATDMA functional model

The functional model explains how the ATDMA protocol will interact with different entities of a wireless network. The functional model discusses wireless network related issues such as power control, link adaptation, channel assignment, handover, channel release, channel quality measurements and reporting. The functional model of the ATDMA protocol is shown in Fig 3.3 while Table 3.4 shows signalling information being passed between interfaces of different functional elements. Functions of all of the logical groups in the above Figure would be distributed over some or all of the three main parts of the network (MS, BS and the fixed network). Most of the signalling and network architecture related work are based on the ATDMA functional model.

The radio resource allocator (RA) performs one of the most important aspects of the ATDMA system. It allocates radio resources among competing terminals. As in most wireless networks in ATDMA the allocation of resources is controlled within the network rather than the terminal. This allows for efficient trade-offs between competing demands and allows operator control. As a result, main RA functions are located in the base station [26].

RA architecture is illustrated in Fig 3.4. It consists of a resource allocator group which includes a core assignment function (assigns resources based on availability of slots, CIR, priority of terminal, etc.) located in the BS. It also consists of a background process setting channel between MS and BS to service different requests from other control processes such as link adaptation, ARQ, speech activity, handover and admission of new calls or handover attempts.



MTP : Mobile Transport Part, **BTP** : Base Transport Part, **NCS** : Network Combiner & Switching, **MLC** : Mobile Link Controller, **BLC** : Base Link Controller, **NLC** : Network Link Controller, **MRA** : Mobile Resource Allocator, **BRA** : Base Resource Allocator, **NRA** : Network Resource Allocator, **MRC** : Mobile Routing Controller, **NRC** : Network Routing Controller, **MTC** : Mobile Traffic Controller, **NTC** : Network Traffic Controller, **MLM** : Mobile Location Manager, **BLM** : Base Location Manager, **NLM** : Network Location Manager

Figure 3.3 ATDMA Functional Model [26][55]

Interface	From	To	Signals
TC/RC	TC	RC	Channel set-up and close request
	RC	TC	Channel set-up and close ACK measurements of call quality
TC/LM	LM	TC	Signalling channel set-up and close request
	TC	LM	Signalling channel ACK
LM/RC	RC	LM	Locate request and location update
	LM	RC	Location information
RC/LC	RC	LC	Link set-up and close, bearer set-up and close and BS search commands
	LC	RC	Link set-up and close ACK, bearer set-up and close ACK and measurement of current call quality and adjacent cells
RC/RA	RC	RA	resource reservation request
	RA	RC	Resource reservation grant and new connection detection
LC/RA	LC	RA	Resource request, change and release
	RA	LC	Resource grant
LC/TP	LC	TP	Transport command
	TP	LC	Activity detection

Table 3.4 Signalling messages passed between the interfaces of functional elements of the functional model given in Figure 3.3 [55]

All the information about the slots is kept in a table called the resource table. The resource table does not simply contain a list of free and active slots, but it also stores slot quality parameters, and calculated and expected BS loading. Using this information and the priority of the request, the core will accept, reject or queue a request. This same process should apply for acceptance of new calls, existing inactive to active calls and for incoming handover requests. Different thresholds should be used to ensure that priority is given to existing delay sensitive inactive and handover calls.

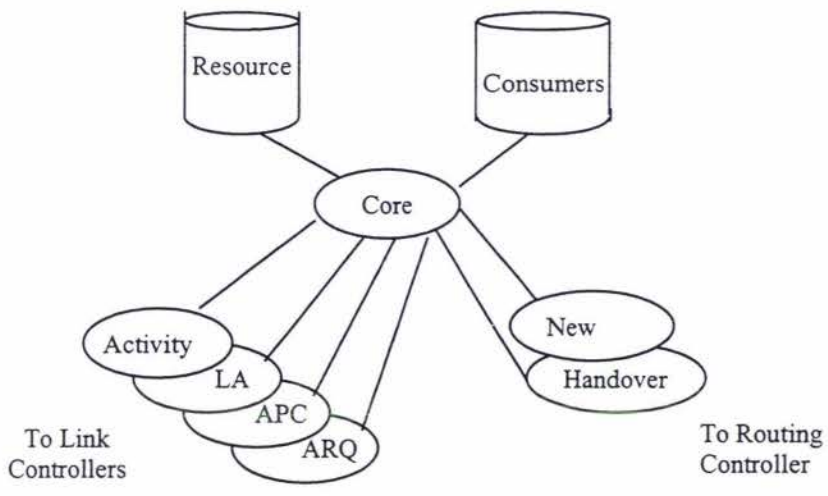


Figure 3.4 Resource Allocator Architecture [26][55]

At the start of a talkspurt, video frame or a data burst the ATDMA packet assignment protocol is used. Based on the BS loading, taking into account all current services (active and inactive), the packet assignment protocol located at the core of RA would accept, reject or queue the resources. The ATDMA protocol adapts separate resource request periods (reservation request slots) and usage periods (traffic slots) in the same ATDMA frame. Request signalling for the up-link allocations uses a fast random access request channel (R slot) and for the down-link, a common multiplexed fast paging channel (FP slot) is used. 'R' slot and 'FP' slot uses 'A' and 'FP_{ak}' slots respectively as their paired acknowledgement channels.

3.3 Advantages of the ATDMA Protocol

- Priority, quality based channel (slot) allocation is possible with a base station centred resource allocation algorithm since channel status and quality measurements are recorded at regular intervals.

- Change of transmission modes to satisfy the user requirements. In case of the deteriorating quality of a channel, a mode with lower net source coder rate and enhanced error correction capabilities can be used. Once the channel conditions become normal the mode could be changed to the initial value.

- True 'seamless' handover is a possibility with the incorporation of bi-directional DCCH for signalling during handover. This feature is a significant advancement over most second generation mobile systems where the user traffic may be stopped to pass handover signalling (e.g. in GSM some traffic channels could be reduced to set-up signalling channels for handover [58]).

- Link adaptation and Adaptive Power Control allow the system to adapt to propagation conditions making planning less critical [55].

- Maintaining inactive terminals by means of incorporating a low capacity LCCH link.

- Successful reservation requests are queued when all 'T' slots are allocated. Thus, reservation requests are not blocked as in PRMA. It also reduces contention.

- Data block reservation scheme and capture effect are applied to reduce the collision caused by contention, and also the capacity of the system is increased [59][60].

3.4 Microcells in personal communication systems

For the cellular industry to sustain growth and to meet the challenges of emerging, competing technologies, e.g. personal communications system (PCS), cellular systems must continually improve. A major attribute of personal communications system(PCS) of the future will be massive coverage, meaning both larger capacity (reaching a large number of people in a limited amount of space and spectrum), and near-ubiquity (reaching them almost wherever they are) [61].

Many techniques have been proposed to address these challenges, particularly the capacity issues. Channel-access methods, which have already been discussed earlier, such as TDMA, ATDMA, CDMA techniques, are proposed to increase the system capacity and are considered as a key technique in the design and implementation of third generation cellular mobile networks.

Microcell systems help address the gamut of network challenges facing the cellular industry. The advantages of microcell systems [62] include a significant increase in system capacity, lower power and equipment miniaturisation that reduces cost and enables flexible deployment, and technology that allows cellular operators to explore innovative applications.

A microcell [63] is a region served by a radio base unit, just as in current systems, only smaller in area. Traditional cells (referred to hereafter as macrocells) are served using high power base stations with antenna towers 50m high or more. They are limited by zoning, economics, and other factors to radii no smaller than about 1 km. Microcells, on the other hand, would be served by compact, low power base units mounted on lamp posts or utility poles, at heights of about 10m, and would have radii of 1 km or even less. They could be used to reach "dead spots" caused in traditional cellular areas by shadowing, or they could be used to served subscribers on foot carrying low-power portables with small antennas. They could also exist indoors, for example, in malls, airports, train stations, hotels, and office buildings.

With the use of microcells, people can easily enhance the system capacity just by reducing the cell size. The increase in system capacity is proportional to 1/R (R = cell radius) [61]. This leads to the potential importance of microcells. Various challenges must be met, however, to fully realise the potential. Some of them are [64]:

1. To make microcell base units, along with the connection infrastructure, simple and economical.
2. To understand the new radio propagation environment associated with microcells, and to parley that understanding into both suitable propagation models and methods for predicting radio link performance.
3. The small size of microcells poses special difficulties regarding channel assignments and subscriber hand-offs from one cell site to another. This is because

the small cell size enhances the probability of hand over compared with the larger cell size.

4. There are thorny architectural issues, such as how to combine microcells with macrocells and how to share limited spectral resources between them.

In this section microcell radio propagation characteristics and microcell propagation models will be discussed.

3.4.1 Radio propagation

The wireless networks operate at frequencies ranging from a few hundred kilohertz (mobile radio dispatch network) to a few tens of gigahertz (some wireless local area networks WLANs). However, the emerging new systems and services are those operating in a range from around 900 MHz to a few gigahertz. Cellular systems, second-generation cordless telephone, and some wireless LANs operate in the 900 MHz region, whereas the emerging new personal communications systems (PCS) such as UMTS are targeted at bands around 2 GHz. The ATDMA system is one of the proposed air interface for the UMTS and will operate around 2 GHz. The characteristics of radio propagation in this frequency range will be discussed.

The frequencies in the range of a few gigahertz have several attractive features for use in the evolving wireless information networks. At these frequencies a transmitter with power less than 1 W can provide coverage for several floors within a building, and if used outdoors it can cover the distances of the order of a few kilometres, as needed for cellular urban radio communications [65]. Furthermore, at these frequencies the size of an efficient antenna can be on the order of an inch, and antenna separations as small as several inches can provide uncorrelated received signals for achieving diversity in the received signal [66].

In a mobile radio system, communications engineers are generally concerned with two main radio channel issues: link budget and time dispersion. The link budget is determined by the amount of received power that may expected at a particular distance or location from a transmitter. It is affected by several factors such as interference from other cells and fading of the channel. It determines fundamental quantities such as transmitter power requirements, coverage areas, and battery life. Time dispersion arises due to multipath propagation whereby replicas of the transmitted signal reach the receiver with different propagation delays due to the different propagation mechanisms. The time dispersive nature of the channel determines the maximum data rate that may be transmitted without equalisation and also determines the accuracy of navigational services such as vehicle location.

3.4.2 Multipath fading

In most radio channels the transmitted signal arrives at the receiver from various directions over a multipath. Figure 3.5 represents a mobile radio scenario where the received signal arrives by several paths bounced from large objects such as buildings and local paths scattered from objects close to the receiver, such as the ground or trees.

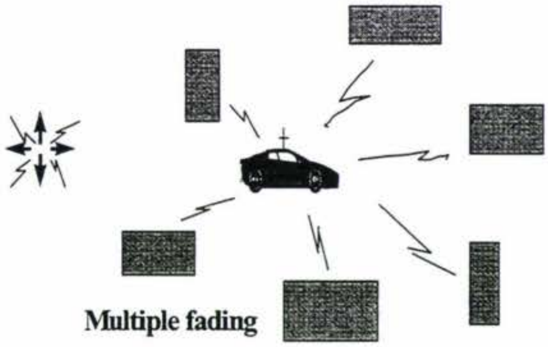


Figure 3.5 Example of mobile radio Rayleigh fading

The phase and amplitude of the signal arriving by each different path are related to the path length and the conditions of the path; this results in considerable amplitude fluctuation of the composite received signal.

At the receiver, while the average power decreases with distance, the power also fluctuates. The reason for this power fluctuation is that the relative phases of the arriving paths are changing as the receiver moves from one location to another. Therefore, there is a randomness in the summation of these paths. At certain locations all the paths are essentially in phase alignment, producing relatively large received power; and in some other locations the paths are nearly cancelling each other, producing a drastic reduction of the received power. These fluctuations constitute Envelop Fading observed by the mobile users.

Further discussion follows about multipath fading [63]. In a multipath environment, the composite received signal is the sum of the signals arriving along different paths. Except for the LOS path, all paths are going through at least one order of reflection, transmission, or diffraction before arriving at the receiver. Upon each reflection of a path from a surface, a certain fraction of the power is absorbed by the surface and the reminder of the power in that path carries beyond the reflection. If the path has been reflected K_i times before arriving at the receiver, and at each reflection the reflection coefficient is a_{ij}, the overall reflection factor is

$$a_i = \prod_{j=1}^{K_i} a_{ij} \qquad (3.1)$$

where a_{ij} is the reflection coefficient for the j th. reflection of the i th. path. Therefore, the amplitudes of the signals received from paths other than the LOS path are subject to reflection loss as well as the standard distance-attenuation factor.

If there are L paths and the distance travelled by the i th path is d_i, the amplitude and the phase of the received signal are given by

$$A_r e^{j\phi_r} = A_0 \sum_{i=1}^{L} \frac{a_i}{d_i} e^{j\phi_i} \qquad (3.2)$$

where $\phi_i = -2\pi \, f \, d_i \, / \, c$. Figure 3.6 shows a phasor diagram representing the signals arriving from different paths as well as the received signal amplitude and phase. The received power is given by

$$P_r = P_0 \left| \sum_{i=1}^{L} \frac{a_i}{d_i} e^{j\phi_i} \right|^2 \qquad (3.3)$$

The right hand side of the equation shows the magnitude-square of the vector sum of all paths. If the phase of the first path is used as the reference and the vector sum is taken with the phase of all paths relative to the first path, the result remains the same.

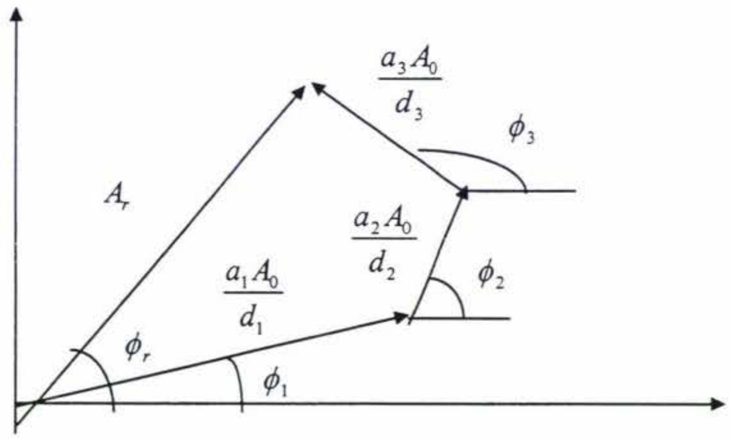


Figure 3.6 Phasor diagram for narrowband signalling on a multipath channel[63].

3.4.3 Local movement and Doppler spread [63]

It is well known from the fundamentals of physics that whenever a transmitter and a receiver are in relative motion, the received carrier frequency is shifted relative to the transmitted carrier frequency. This shifting of frequency is the Doppler effect of wave propagation between nonstationary points.

Figure 3.7 shows a typical example in which a fixed and a portable terminal are communicating over a radio link. The distance between the transmitter and the receiver is d_0 and the portable terminal is moving with speed v_m toward the fixed terminal. Let us assume the portable terminal is transmitting a signal at frequency f_c and the amplitude of the received signal is A_r. If the transmitter is stationary, the received signal is represented by r(t) = Real [$A_r e^{j2\pi f_c(t-\tau_0)}$], where $\tau_0 = d_0/c$ is the time required for the radio wave to propagate from the transmitter to the receiver with velocity c.

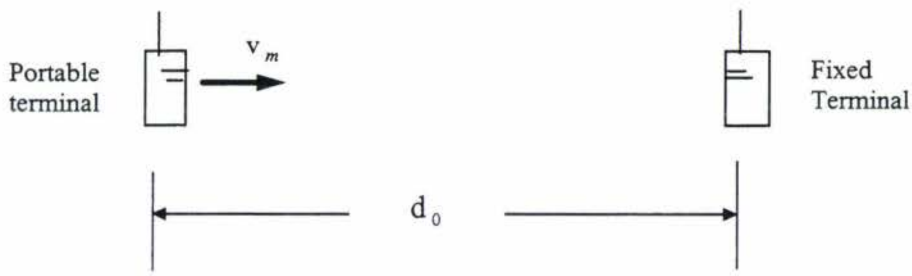


Figure 3.7 A typical example in which a fixed and a portable terminal are communicating over a radio link. The distance between the transmitter and the receiver is d_0, and the portable terminal is moving with speed v_m toward the fixed terminal.

As the transmitter moves toward the receiver, the propagation time will change with time as

$$\tau(t) = \frac{d(t)}{c} = \frac{d_0 - v_m t}{c} = \tau_0 - \frac{v_m}{c}t \qquad (3.4)$$

The receiver signal is then given by

$$r(t) = A_r e^{j2\pi f_c(t-\tau(t))} = A_r e^{j[2\pi(f_c+f_d)t-\phi]} \qquad (3.5)$$

where $\phi = 2\pi \, f_c \, \tau_0$ is a constant phase shift and

$$f_d = \frac{v_m}{c} f_c \qquad (3.6)$$

is a shift in the frequency observed at the receiver, commonly referred to as the Doppler frequency shift. The Doppler frequency shift is either positive or negative depending on whether the transmitter is moving toward or away from the receiver[67].

In a realistic environment, the received signal arrives from several reflected paths with different path distances, and the velocity of movement in the direction of each arriving path is generally different for different paths. Thus a transmitted sinusoid, instead of being subjected to a simple Doppler shift, is received as a spectrum, which is referred as the Doppler spectrum. This effect, which can be viewed as a spreading of the transmitted signal frequency, is referred to in a general way as the Doppler spread of the channel.

3.4.4 Shadowing and Path Loss

Even after the multipath fading is removed by averaging over distances of a few tens of wavelengths, nonselective shadowing still remains. Shadowing is caused mainly by terrain features of the land mobile radio propagation environment. More common is the situation where there are hills, buildings etc. between the base and mobile station. The shadowing effect decreases the received signal strength. Figure 3.8 shows an example of shadowing.

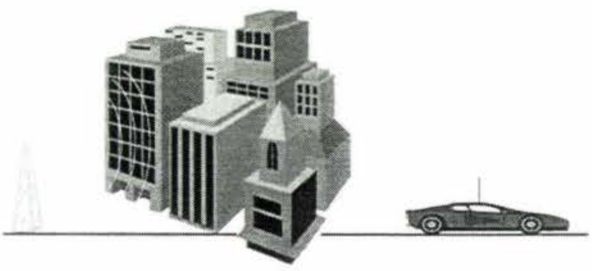


Figure 3.8 Log-normal fading

A signal influenced by fading goes up and down in signal strength. The "downs" are called fading dips. The type caused by the shadowing effect is called log-normal fading which, if the logarithm of the signal strength is taken, then the faded signal takes the form of a normal distribution around some mean value. The time between two fading dips is typically several seconds when the mobile is car—mounted and moving.

An average value of log-normal shadowing, is determined by the pass loss, which varies with the distance between the fixed station and the general location of a moving vehicle, and represents the distance-power relationship of the received signal. For indoor and urban radio channels the distance-power relationship will change with the building and street layouts, as well as with construction materials and density and height of the building in the area. Pass loss is always found by measurements. To measure the gradient of the distance-power relationship in a given area, the receiver is fixed at one location and the transmitter is placed at a number of locations with different distances between the transmitter and receiver. Either the received power or the path loss is potted in decibels against the distance on a logarithmic scale.

Based on experiment, there are many path loss prediction models. One of the most commonly used path loss models for urban propagation is the model originally developed by Okumura et al. [69] based on extensive radio propagation studies made in Tokyo. This model was further adapted for computer simulation by Hata [70]. There are also available some widely accepted models for path loss in urban radio channels, such as Lee's model, et al. [66][68][71]. More description about path loss model will be given later.

The path loss modeling methods described in this section are based on generalizations of results obtained in certain specific measurement programs. However, there is no universally accepted model for path loss. One important limitation of these modeling methods is that they do not include the specification of building characteristics. As a consequence, much attention is being given to building-specific radio propagation models such as ray tracing [67], and these techniques are emerging as the leading technique for the future. However, there are drawbacks in the use of building-specific radio propagation models: the complexity of computation, the need for large amounts of computer memory, and the enormous cost of creating a detailed electronic map. With the growing availability of electronic maps and steady increases in computation power and memory capacity of computers, it is expected that increasingly accurate building-specific radio propagation models will evolve.

3.4.5 Time dispersion

The introduction of digital transmission brings with it another problem: time dispersion. This also has its origin in reflections, but in contrast to multipath fading, the reflected signal comes from an object far away from the receiving antenna, say in the order of kilometres. Figure 3.9 show the concept of tim dispersion.

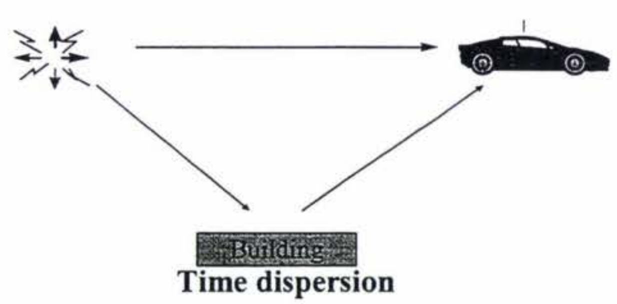


Figure 3.9. Time dispersion

The time dispersion causes Inter-Symbol Interference (ISI). ISI means that consecutive symbols interfere with each other and it gets difficult on the receiver side to decide which actual symbol is detected (or actually, sent). An example of this is shown in Figure 3.10. The sequence "1", "0" is sent from the base station. If the reflected signal arrives exactly one bit time after the direct signal, then the receiver will detect a "1" from the reflected wave at the same time as it detects a "0" from the direct wave. The symbol "1" interferes with the symbol "0" as shown in Figure 3.11.

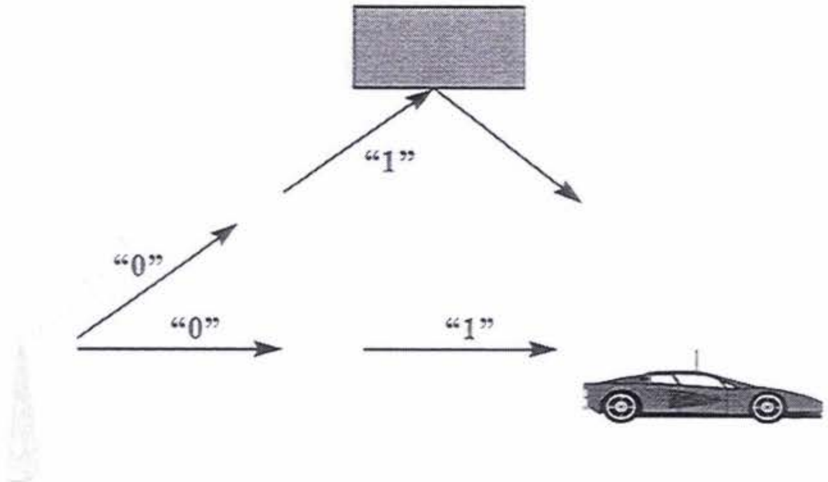


Figure 3.10 Time dispersion

Intersymbol interference (ISI) caused by multipath in bandlimited time dispersive channels distorts the transmitted signal, causing bit errors at the receiver. ISI has been recognised as the major obstacle to high speed data transmission over mobile radio channels. The solution to combatting time dispersion is to use the equalisation technique in the receivers. Equalisation is a technique used to combat ISI.

The term equalisation can be used to describe any signal processing operation that minimises ISI. Since the mobile fading channel is random and time varying, equalisers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalisers.

The general operating models of an adaptive equaliser includes training and tracking. First, a known, fixed-length training sequence is sent by the transmitter so that the receiver's equaliser may adjust its parameters properly. The training sequence is typically a pseudorandom binary signal or a fixed, prescribed bit pattern. Immediately following this training sequence, the user data is sent, and the adaptive equaliser at the receiver utilises a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate the channel. The training sequence is designed to permit an equaliser at the receiver to acquire the proper filter coefficients in the worst possible channel condition so that when the training sequence is finished, the filter coefficients are near the optimal values for the reception of user data. As user data are received, the adaptive algorithm of the equaliser tracks the changing channel. As a consequence, the adaptive equaliser is continually changing its filter characteristics over time.

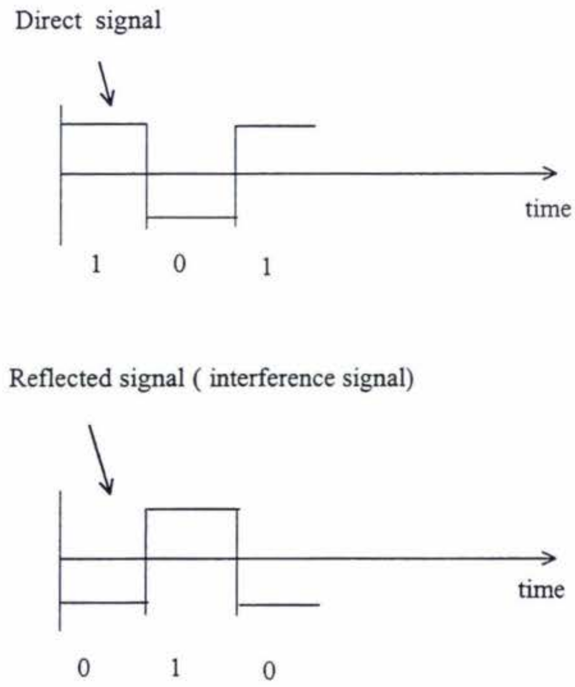


Figure 3.11 With the reflected signal arrives exactly one bit time after the direct signal, the receiver will detect a "1" from the reflected signal at the same time as it detects a "0" from the direct signal. The symbol "1" interferes with the symbol "0" .

Equalisers require periodic retraining in order to maintain effective ISI cancellation, and are commonly used in digital communication systems where user data is segmented into short time blocks. TDMA based wireless systems are particularly well suited for equalisers. TDMA systems send data in fixed-length time blocks, and the training sequence is usually sent at the beginning of a block. Each time a new data block is received, the equaliser is retrained using the same training sequence[67].

3.4.6 Channel modelling techniques

The radio channel places fundamental limitations on the performance of mobile communication systems. The transmission path between the transmitter and the receiver can vary from simple direct line of sight to one that is severely obstructed by buildings. Unlike wired channels that are stationary and predictable, radio channels are extremely random and do not offer easy analysis.

The effective design, assessment, and installation of a radio network requires an accurate characterisation of the channel. The channel characteristics vary from one environment to another, and the particular characteristics determine the feasibility of using a proposed communication technique in a given operating environment. Channel modelling technique is a technique to provide a channel model for the system designer. The model provides an accurate channel characterisation for each frequency band, including key parameters and a detailed mathematical description of the channel. Channel model enables the designer and user of a wireless system to predict signal coverage, achievable data rate, and the specific performance attributes of alternative signalling and reception schemes. Channel models are also used to determine the optimum location for installation of antennas and to analyse the interference between different systems.

There now exists an extensive body of literature on radio propagation prediction and modelling. Many researchers have developed a variety of experimentally or theoretically based models to predict radio propagation in various frequency bands, and for various physical characteristics of the transmission path. A number of prediction models have been developed that take into account antenna height, path length, earth curvature, terrain irregularity, foliage, urban streets and buildings, tunnels, and so on. Widely used propagation models include those of Bullington [64][68], Longley and Rice [66], Okumura [69], Hatt [70], and Lee [71].

The Bullington model is an early theoretical model based on a smooth earth propagation theory. It includes approximations for estimating the effects of hills and other obstructions in the radio path [68]. The interesting feature is that this method avoids multiple hills by constructing a single equivalent knife edge; Troposphere effects are accounted for by means of graphs. The prediction procedure was initially given as nomograms.

The Longley-Rice model is based on well established propagation theory with atmospheric and terrain effects allowed for by a very large data bank of empirical adjustments [66]. The propagation path is divided into three regions in this model, namely, line-of-sight, diffraction, and forward scatter region, and the field strength is predicted by applying different linear formula to the path loss in each region, and according to type of occasion.

After extensive measurements in and around Tokyo, Okumura [69] proposed the following equation for the path loss:

$$L = A_m\,(f, d) - H_b\,(h_b\,, d) - H_m(h_m\,, f) \tag{3.7}$$

where A_m is the median attenuation relative to free space in an urban area over quasi smooth terrain with a base antenna height of 200 m (quite high) and a mobile antenna height of 3 m. H_b and H_m are the height gain factors for the base and mobile antenna heights, h_b and h_m respectively. All these factors are given in graphs. For different earth profiles Okumura suggested adjustment factors for equation (3.7), also given in graphs. The drawback in Okumura's model is that it was not designed to be used on a computer, but this difficulty was overcome by Hata, who put the model to computational use.

The analytic approximation for the path loss of Okumura's model has been given by Hata[70] as a set of approximations. The propagation loss in an urban area is used as a standard formula, given by:

$$L = 69.55 + 26.16 \log f - 13.82 \log h_b - a(h_m) + (44.9\text{-}6.55 \log h_b) \log d \tag{3.8}$$

In this equation $a(h_m)$ is the correction factor for h_m. For mobile antenna heights in a small city $a(h_m)$ is given by :

$$a(h_m) = (1.1 \log f - 0.7)\, h_m - (1.56 \log f - 0.8) \qquad (3.9)$$

while in a large city, where the building height average is more than 15 m, $a(h_m)$ is given by:

$$a(h_m) = 8.29 (\log 1.54\, h_m)^2 - 1.1 \qquad f \leq 200 \text{ MHz}$$

$$= 3.2 (\log 11.75\, h_m)^2 - 4.94 \qquad f \geq 400 \text{ MHz}$$

In suburban and open areas, the propagation loss is given by:

Suburban: $L = L\,(\text{urban area}) - 2 (\log f/28)^2 - 5.4$

Open area: $L = L\,(\text{urban area}) - 4.78 (\log f)^2 + 18.33 \log f - 40.94$

Lee's model, like Okumura's , is based on measurements in specific environments. The measurements have been undertaken in North American cities [71]. Based on the measurement, an empirical model was developed which can better describe the urban microcell environment. Lee's model will be discussed in detail in the next chapter.

3.5 Conclusion

In this chapter, the concept of an ATDMA based radio access system that has been designed specifically for use in a third-generation mobile system UMTS has been introduced. Based on the ATDMA functional model, an ATDMA protocol and ATDMA frame structure that support all of the services of an ATDMA system have also been studied.

The ATDMA system offers a flexible, open, radio interface that supports a range of environments and services based on a clear division between radio link maintenance and traffic channel transmission. Priority and quality based channel allocation capabilities, an improved mechanism to satisfy user requirements (by changing transmission modes) and seamless handover capabilities, are some of the main reasons for choosing the ATDMA protocol as the access protocol over other wireless networks for third generation wireless networks. Furthermore, through the use of packet assignment, link adaptation and quality based power control, it is expected that this radio access system will achieve a significant improvement in overall system capacity when compared with existing second-generation mobile systems.

The rest of the chapter discussed the radio propagation characteristics and microcell propagation model. Microcells embedded in conventional cellular systems can provide improved coverage and performance, increased system capacity, and delivery of innovative value-added services. It is suffice to say that efforts to bring about low-cost, high-quality personal communications are being vigorously pursued in industries, universities, government agencies and standards bodies around the world. In order to implement the above goals, investigation of radio propagation characteristics and the development of microcell propagation models becomes one of the critical issues in microcell design.

CHAPTER IV

PERFORMANCE OF THE ATDMA PROTOCOL IN A CELLULAR RADIO ENVIRONMENT

4.0 Introduction

In this chapter the performance of the ATDMA protocol in a microcell environment will be studied. As described in previous chapters the PCN design engineers try to achieve the maximum spectrum efficiency by designing the best possible multiple access techniques. The ATDMA protocol was chosen as this class of multiple access technique can support third generation high capacity mobile communication networks and PCN. In this chapter, the performance of the ATDMA protocol is investigated by means of a computer simulation method. The discrete event simulation language Simscript II.5 was used to simulate the performance of the ATDMA protocol in the microcellular environment to support mixed voice and data traffic.

The objective of the simulation include:

- To investigate the performance of the ATDMA protocol in a microcell environment
- To investigate the optimum system parameters of the ATDMA protocol and then the optimum system capacity
- To investigate a method to enhance the performance of the ATDMA protocol

In this chapter, a microcell channel model will be discussed in section 1. Section 2 will study the simulation of the ATDMA protocol. Section 3 presents the simulation results and analysis of the results. Finally section 4 gives the conclusions of the study.

4.1 Microcell channel model

In a radio environment the irregular configuration of the natural terrain, the various shapes of architectural structures, changes in weather, and changes in foliage conditions make the prediction of propagation loss very difficult. In addition the signal is received while the mobile unit is motion. There is no easy analytic solution to this problem. Combining both statistics and electromagnetic theory helps to predict the propagation loss with great accuracy.

There are very few theoretical models in predicting propagation path loss, but many empirical models. In this section, a channel model will be discussed which was developed based on Lee's model [71]. The model was developed to investigate path loss and carrier to interference characteristics of a microcell propagation environment.

4.1.1 An area-to-area path loss prediction model

An area-to-area prediction is usually used to predict a path loss over a generally flat terrain without knowing the particular terrain configuration over which the actual path loss is found. This method, like Okumura's, is based on measurements in specific environments. The measurements have been taken in North American cities [71]. The general path loss equation is based on the signal power P_r received relative to the power received at 1 mile intercept P_{r0}:

$$P_r = P_{r0} - \gamma \log\left(\frac{r}{r_0}\right) - \mathrm{n} \log\left(\frac{f}{f_0}\right) + \alpha_0 \qquad \text{(dB expression)} \qquad (4.1)$$

where γ is called the path loss slope factor, r is the distance from the base, f_0 is the reference frequency, r_0 is the 1 mile intercept distance, P_{r0} is the power received at the 1 mile intercept. α_0 is the adjustment factor to account for the height gains of the base and mobile antennas and different transmission power. This concept is empirical, because P_{r0} is found from measurement. As is well known, the path loss slope changes from city to city, and one environment to another. Table 4.1 shows the parameters of γ and p_{r0} found from the empirical data at 1 mile intercept distance with reference conditions [69][73][74][75][76].

ENVIRONMENT	1MILE INTERCEPT(P_{r0})	PATH LOSS SLOPE(γ)
Free space	-45.0 dBm	20.0 dB/dec
Open area	-49.0 dBm	43.5 dB/dec
Suburban area	-61.7 dBm	38.4 dB/dec
Urban(Philadelphia)	-70.0 dBm	36.8 dB/dec
Urban(Newark)	-64.0 dBm	43.1 dB/dec
Urban(Tokyo)	-84.0 dBm	30.5 dB/dec

Table 4.1 Propagation parameters obtained using lee's modell [69][73][74][75][76].

The reference parameters which are used to get the experimental data of the 1 mile intercept are defined as follow:

Frequency $f_0 = 900$ MHz

Base-station antenna height = 30.48 m (100 ft)

Base-station power at the antenna = 10 watts

Base-station antenna gain = 6 dB above dipole gain

Mobile-unit antenna height = 3 m (10 ft)

Mobile-unit antenna gain = 0 dB above dipole gain

Then the adjustment factor α_0 can be found for different sets of conditions. α_0 is expressed as:

$$\alpha_0 = 20\log\left(\frac{h_1}{100'}\right) + 10\log\left(\frac{p_t}{10w}\right) + (g_1 - 4) + g_2 + 10\log\frac{h_2}{10'}$$

$$= 20\log h_1 + 10\log p_t + g_1 + g_2 + 10\log h_2 - 64 \qquad (4.2)$$

where new values are transmission power p_t in watts, base antenna height h_1 and mobile antenna height h_2 in feet, base antenna gain g_1, and mobile antenna gain g_2 in dB.

The value of n in Eq. (4.1) is found from empirical data. Okumura [69] indicates n = 30 dB/dec, and Young [72]indicates n = 20 dB/dec. Therefore

$$20 \text{ dB/dec} < n < 30 \text{ dB/dec} \qquad (4.3)$$

when n is valid for the frequency range from 30 to 2000 MHz and the distance range from 2 to 30 km , or approximately 1.5 to 20 miles. The value n seems dependent on the geographical location and the operating frequency ranges. In a suburban or open area with the operating frequency below 450 MHz, n = 20dB/dec is recommended. In an urban area with the operating frequency above 450 MHz, n = 30 dB/dec is recommended.

General Formula of the Model [71]

$$p_r = -45 - 20\log r - n\log\left(\frac{f}{900}\right) + \alpha_0 \quad \text{dBm (Free space)}$$

$$= -49 - 43.5\log r - n\log\left(\frac{f}{900}\right) + \alpha_0 \quad \text{dBm (open area)}$$

$$= -61.7 - 38.4\log r - n\log\left(\frac{f}{900}\right) + \alpha_0 \quad \text{dBm (suburban)}$$

$$= -70 - 36.8 \log r - n \log\left(\frac{f}{900}\right) + \alpha_0 \quad \text{dBm (Philadelphia)}$$

$$= -64 - 43.1 \log r - n \log\left(\frac{f}{900}\right) + \alpha_0 \quad \text{dBm (Newark)} \qquad (4.4)$$

$$\alpha_0 = 20 \log\left(\frac{h_1}{100'}\right) + 10\log\left(\frac{p_t}{10w}\right) + (g_1 - 4) + g_2 + 10\log\frac{h_2}{10'}$$

$$= 20 \log h_1 + 10\log p_t + g_1 + g_2 + 10\log h_2 - 64$$

where new values are p_t in watts, base antenna height h_1 and mobile antenna height h_2 in feet, base antenna gain g_1, and mobile antenna gain g_2 in dB, r in miles and f in MHz.

Based on the above discussion, the path loss characteristics of different environments were obtained as shown in Figure 4.1. Parameters used for the simulation are shown in table 4.2.

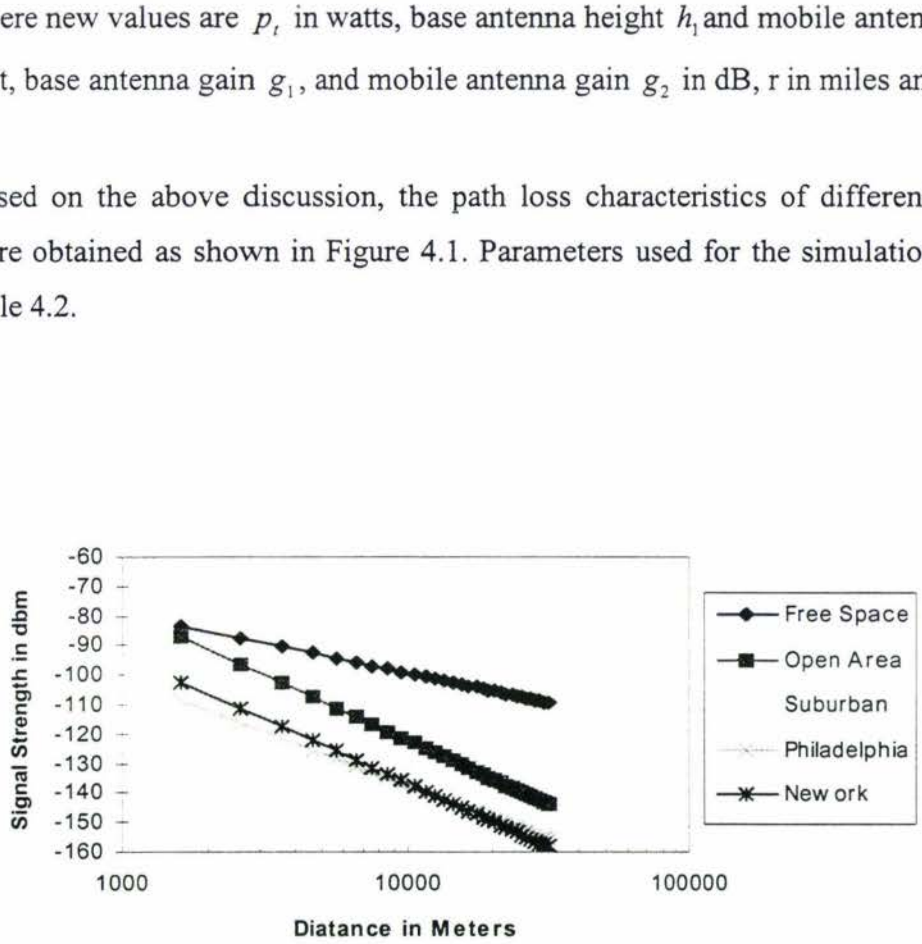


Figure 4.1 Propagation path loss in different areas.

Parameters of area to area prediction model	Value
h1 (base station antenna height)	20 feet
h2 (mobile station antenna height)	6 feet
Pt (transmitting power)	1 w
f	2 GHz
g1 (base station antenna gain)	4
g2 (mobile station antenna gain)	0

Table 4.2 Simulation parameters for area to area path loss predication model

4.1.2 Microcell prediction model

When the size of the cells is small, less than 1 km, the street orientation and individual blocks of buildings make a difference in signal reception. The street orientations and individual blocks of buildings do not make any noticeable difference in reception when this signal is well attenuated at a distance over 1 km. When the cells are small, the signal arriving at the mobile unit is blocked by the individual buildings; this weakens the signal strength and is considered as part of the path loss. Therefore another approach to predict is considered to find out the path loss characteristic in a microcell with radii smaller than 1 km. In a small cell path loss is calculated based on the dimensions of a building block. Since the ground incident angles of the waves are, in general, small due to the low antenna heights used in small cells, the exact height of buildings in the middle of the propagation paths is not important, as shown in Figure 4.2. Therefore only a two-dimensional map is used. Although the strong received signal at the mobile unit is received from the multipath reflected waves, not from the waves penetrating through the buildings, there is a correlation between the attenuation of the signal and the total building blocks, along the radio path. The larger the building blocks, the higher the signal attenuation. When the wave is not being blocked by a building it is in line-of -sight communication. From the measurement data along the streets in an open line-of-sight condition, the line-of-sight signal reception curve p_{los} is formulated. Also, from the measured signal p_{os} along the streets in Non Line-of-Sight Communication (NLOS) conditions within the cells, the additional signal attenuation α_B curve due to the portion of building blocks over the direct path is found by subtracting the received signal from p_{los}. The steps for forming an additional signal attenuation formula α_B are as follows:

1. Calculate the total blockage length B by adding the individual building blocks. For example, B = a +b +c at point A shown in Figure 4.3.
2. Measure the signal strength P_{los} for line-of-sight communication.
3. Measure the signal strength P_{os} for NLOS communication.
4. The local mean at point A is P_{os} (at A). The distance from the base to the mobile unit is d_A. The blockage length B at point A is B = a + b + c. Then the value of α_B for a blockage of B can be expressed as

$$\alpha_B(\mathrm{B} = \mathrm{a} + \mathrm{b} + \mathrm{c}) = P_{los}(\mathrm{d} = d_A) - P_{os} \qquad (\text{at } d_A)$$

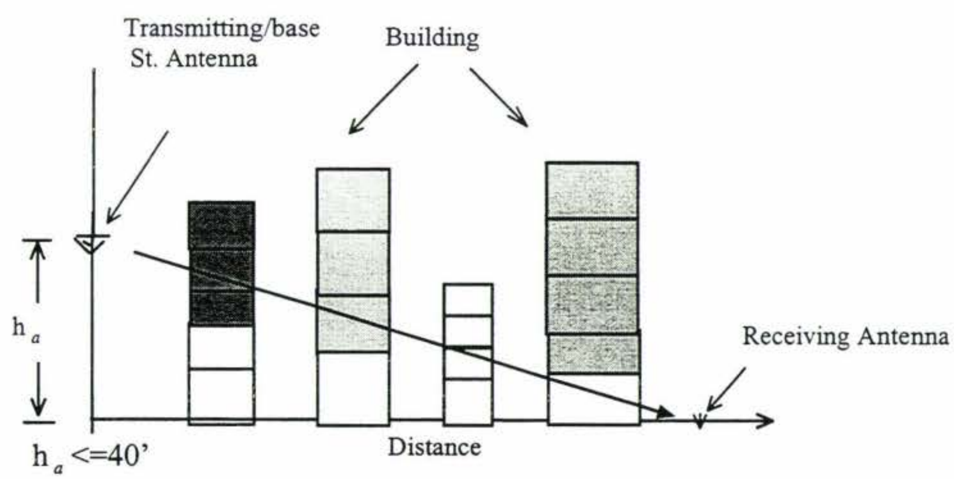


Figure 4.2 The propagation mechanics of low-antenna height at the cell site.

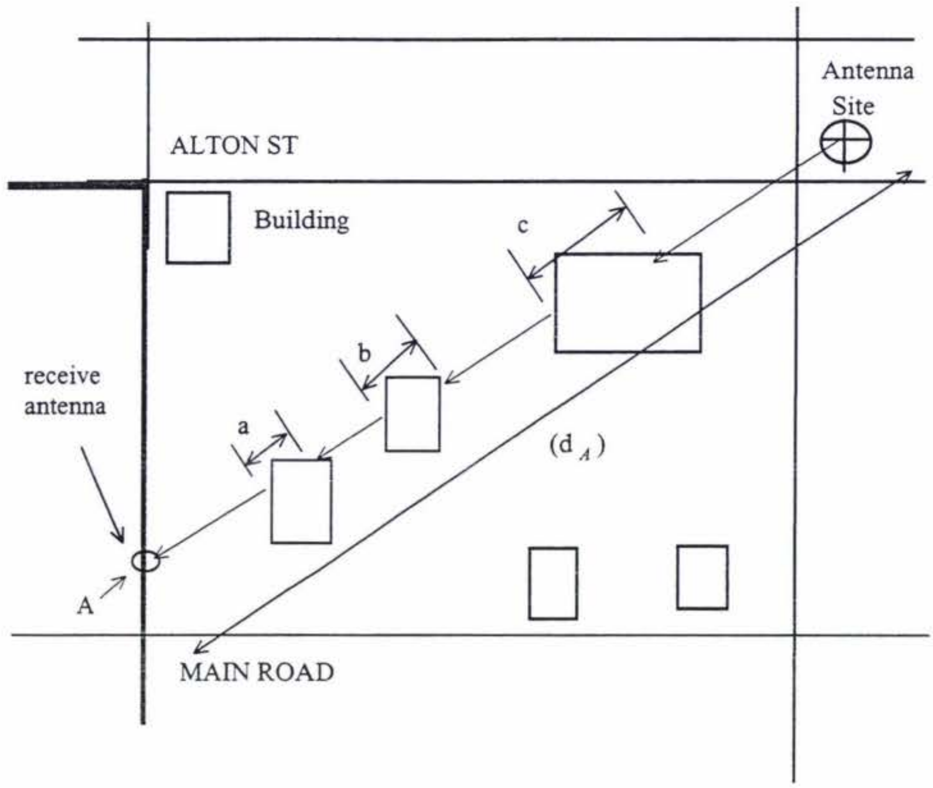


Figure 4.3 Building block occupancy at location A using the grid structure. Then B = a + b +c.

In the small-cell prediction model, two curves, P_{los} and α_B are used to predict the received signal strength. Therefore the microcell (small cell) model can be formed as

$$P_r = P_{los} - \alpha_B \qquad (4.5)$$

where P_{los} is the line-of-sight path loss and α_B is the additional loss due to the length of the total building blocks B along the paths.

The expressions to be evaluated are as follows:

$$P_{los} = P_t - 77 \text{ dBm} - 21.5 \log \frac{d}{100'} + 30 \log \frac{h_1}{20} \qquad 100' \le d < 200'$$

$$= P_t - 83.5 \text{ dBm} - 14 \log \frac{d}{200'} + 30 \log \frac{h_1}{20} \qquad 200' \le d < 1000' \qquad (4.6)$$

$$= P_t - 93.3 \text{ dBm} - 36.5 \log \frac{d}{1000'} + 30 \log \frac{h_1}{20} \qquad 1000' \le d < 5000'$$

and

$$\alpha_B = 0 \qquad 1' \le B$$

$$= 1 + 0.5 \log(B/10) \qquad 1' \le B < 25'$$

$$= 1.2 + 12.5 \log(B/25) \qquad 25 \le B < 600' \qquad (4.7)$$

$$= 17.95 + 3 \log(B/600') \qquad 600' \le B < 3000'$$

$$= 20 \text{ dB} \qquad 3000' \le B$$

where P_t is the full ERP in dBm, d is the total distance is feet, h, is the antenna height in feet. B is the length of the building block. Substitute Eq. (4.6) and Eq. (4.7) into Eq. (4.5), and the predicted received signal P_r is obtained. Figure 4.4 illustrated the microcell propagation path loss performance with different building block length.

4.1.3 Carrier-to-interference analysis of the microcell prediction model

As discussed in section 2.1.3, co-channel interference plays the major limiting role in the performance of cellular radio systems. So in this study the co-channel interference was considered as the main source of interference to calculate the carrier-to-interference ratio in a microcell. Simulation was undertaken to investigate the carrier-to-interference ratio of microcells with different cell sizes and cluster sizes.

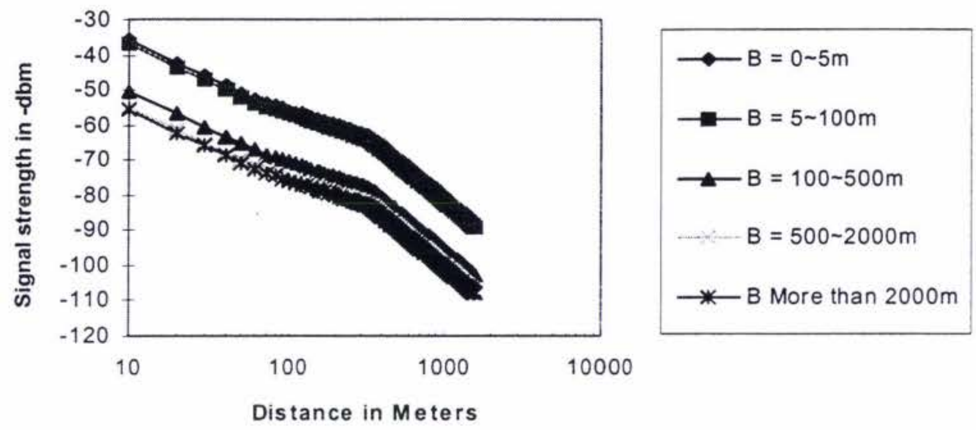
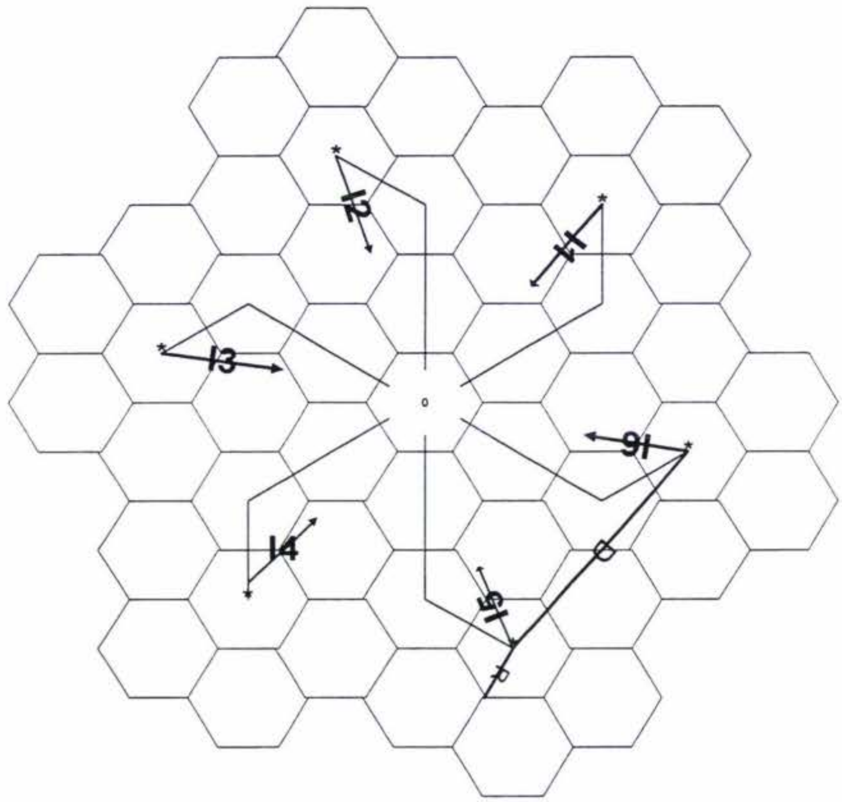


Figure 4.4 Microcell path loss characteristics with Pt = 1 w, h1(base station antenna height equals to 20 feet, and different building block length. B is the block length of the building.

According to equation 2.4 and 2.5, the carrier to interference performance of the microcell model can be obtained. For example Figure 4.5 illustrates the co-channel cells with cluster size equal to 7. The carrier-to-interference ratio is for this case:

$$\frac{C}{I} = \frac{C}{6 * I_i} \tag{4.8}$$

C is the signal strength of the mobile in the main cell (shown in Figure 4.6). It is determined by the distance between the mobile terminal and the base station.



$i=2, j=1$
$N=7$
$D/R= \sqrt{3N}$

Figure 4.5 Co-channel interference with 6 co-channel cells in the situation where cluster size equals to 7. $I_1 \sim I_6$ is the co-channel interference I_i from different cells.

I_i is the co-channel signal strength. In Figure 4.5, I_1 ......I_6 are used to show different I_i ($I_i = I_1 + I_2 + I_3 + I_4 + I_5 + I_6$). I_i is decided by the co-channel distance D.

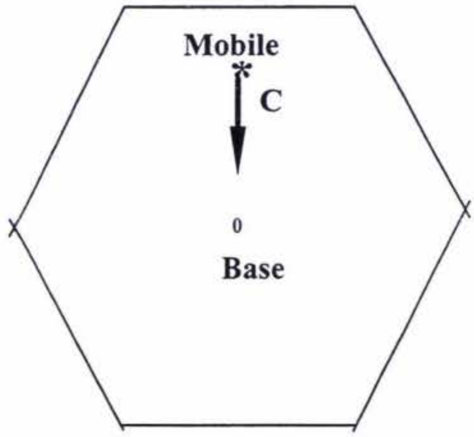


Figure 4.6 Signal strength of mobile terminal in main cell is C, which is determined by the distance between the mobile terminal and the base station

Figure 4.7 shows the carrier-to-interference ratio for two different cluster sizes in a microcell environment. The carrier-to-interference ratio discussed here is about the worst case in the hexagonal structure. Because in the practical situation, for one carrier, the number of co-channel interference produced from the co-channel signal is random. Number of cells genersting interference at anytime could be between 0 to 6 depending on the channel usage in those cells. Number of co-channel signals can very according to the allocation of channels and the access of mobile terminals. The worst case is that all 6 co-channel cells generate interference, and in this situation the smallest C/I value is obtained.

The microcell model discussed in this section will be used to investigate the ATDMA protocol's performance.

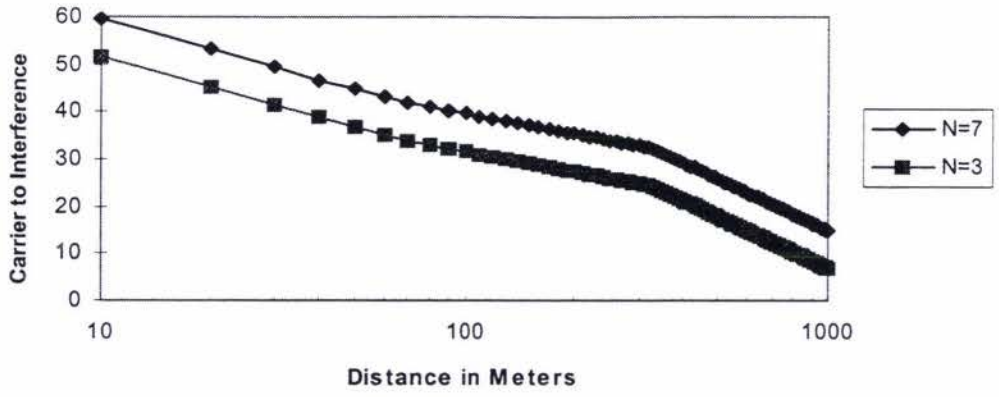


Figure 4.7 Carrier-to-interference with cell size R=1 km.

4.2 Simulation of the ATDMA protocol

A simulation model has been developed to study the performance of the ATDMA protocol in a microcell environment. Performance of the protocol was evaluated by measuring parameters such as percentage of voice packet loss, data packet delay, data block delay, etc. The model has been developed by using the discrete event simulation language SIMSCRIP II.5 [77][78].

Simulation is the computer representation of a real system under observation. Simulation has proved over the past few decades to be a powerful analytical tool with success in small and large-scale applications. There are two types of physical systems; continuous and discrete. In continuous systems the variables undergo smooth changes whereas in discrete systems the changes take place in discrete steps.

Continuous simulation represents the system model by sets of algebraic or differential equations which are solved numerically. Discrete event simulation describes the model of a system in terms of logical relationships which causes changes of state at discrete points of time. The discrete system simulation models generally trend to be both stochastic and dynamic in nature. In this approach, simulation consists of the observation and analysis of the results obtained by generating random events at different points in time in a digital computer model of the system. So the models where changes in the physical system are represented by a series of discrete event models. Time and event are the two important co-ordinates which are used in describing the simulation models. In discrete event simulation time and state relationships are represented in terms of event, activity and process. These terms are defined in the following ways:

Event: A change in the state of an entity, occurring at an instant that initiates an activity e.g. initiation or termination of a telephone call.

Activity: The state of an entity over an interval.

Process: A succession of states of an entity over one or more contiguous intervals.

The dynamic behaviour of a system is studied by tracing various system states as a function of time and then collecting the system statistics. Timing of different activities in the simulation programme is maintained by an internal clock which is incremented and maintained by the simulation programme. The simulation time can be advanced in two ways, internal oriented simulation and event oriented simulation. The first method which is a uniform time increment method where the clock is advanced from t to $t + \delta$, where δ is a uniform fixed time increment. The second method is based on a variable time increment method where the clock is incremented from time t_1 to the next event time t_2 whatever may be the value of t_2 i.e. the interval will depend on the activation of the next event. This method involves sorting of event activation times and maintaining current and future event lists. This method is mostly used in the discrete event simulation. The simulation model presented in this chapter is based on the event oriented approach.

Figure 4.8 shows the basic block diagram of the ATDMA protocol simulation model. The simulation is controlled by the MAIN model.

At the start of the simulation the necessary variables are read from the input data file and assigned to appropriate global variables by the routine READTALK. Then the appropriate global variables are initialised in the routine INITIALIZE. The ATDMA frame structure is also defined in this routine. When the frame structure is defined, aspects such as number of slots per frame, number of reservation slots and location of those slots in the frame and number of traffic slots and the location of those slots in the frame are decided. Routine INITIALIZE also activates a number of STATION process and process TIME. The process STATION represents the activity of a mobile terminal.

Some of the modules are declared as a routine and some of them are declared as a process. A process represents a sequence of actions it experiences throughout its life in a model [78]. There may exist several different processes in a program. A process has its own creation time and also an "activation time". The sequences of a process routine may be thought of as interrelated events separated by lapses of time, either predetermined or indefinite.

The process TIMER provides current and next reservation slots at any given time during the simulation. The process STATION takes care of all the traffic generators, it calls the routines TALKSPURT and SILENCE and DATA to generate voice and data packets for the simulation purpose. It also calls the routine MICROCELL to generate the C/I characteristic of the packets. The number of STATION processes activated by the INITIALIZE routine is equal to the number of mobile terminals used in the simulation. Each STATION module represents a separate mobile terminal and they are independent processes which are identified by their corresponding process ID number. The STATION activates the process MOBILE whenever it has a packet or packets to transmit.

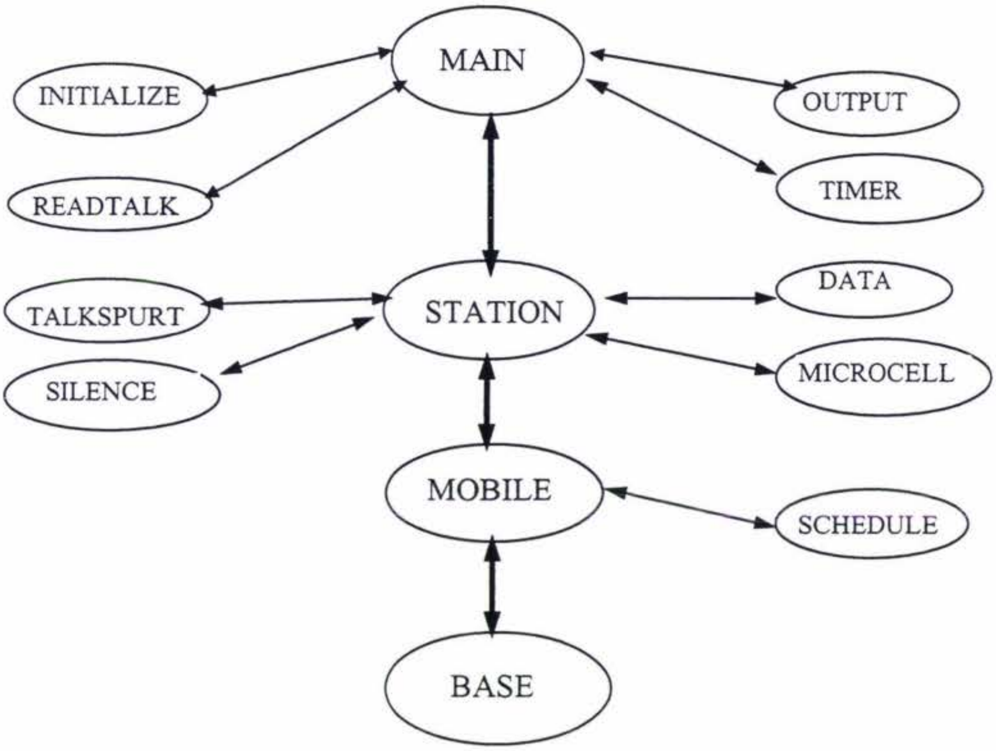
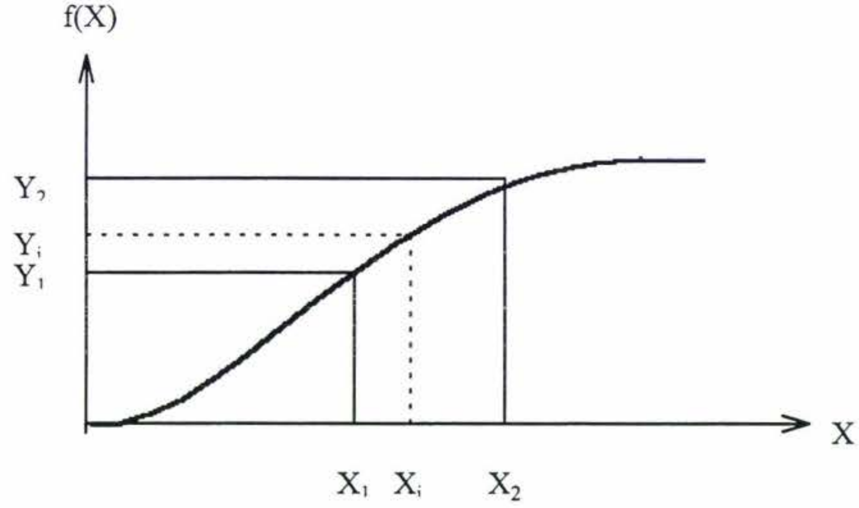


Figure 4.8 Block diagram of the simulation model

The process MOBILE executes the part of the ATDMA protocol which will be executed by the mobile terminals in a real system. MOBILE uses the routine SCHEDULER to obtain different timing involved in the protocol. The routine BASE is called by the MOBILE when a packet transmission starts. Routine BASE models the part of ATDMA protocol related to the base station in a real system. The detailed operation of the above modules are given in Figure 4.11 and Figure 4.12.

Before the detailed description of ATDMA protocol simulation, voice and data traffic generator will be discussed first. Human speech consists of speech and silence periods. In a normal telephone call without noise a speaker is found to talk for about 44% of the time [79]. However, in practical situations the duration would be higher than 44% because of external noise [80]. Since no information is transmitted during the silence periods, the channel could be released during that period so that some other terminal can get access to transmit its packets.

The talkspurts and silences were generated using a probability transformation method as shown Figure 4.9 [81]. Brady's distribution shown in Figure 4.10 has been adapted in this particular model [81]. The graph in Figure 4.10 was obtained using real data from a two way telephone conversation. Speech packets from speech a detector is strongly influenced by the choice of threshold levels [81]. In the case of a threshold level of -40 dBm, the mean talkspurt and silence periods were taken as 1.34 and 1.67 seconds respectively [79]. Any talk spurt less than 15 ms was considered as a noise and silence less than 200 ms was taken as a talkspurt [81].



F(X) : Probability density function, X_1, X_2, ... X_N = Measured abscissa, Y_1, Y_2, ... Y_N = Measured ordinate

Figure 4.9 Transformation techniques for generating talkspurt and silence length

Let f(x) be the probability density function of some measured value of X_1, X_2, ... X_N and Y_1, Y_2, ... Y_N which are the corresponding measured abscissac and ordinates respectively. In order to generate tallkspurts and silences first a random variable is generated between 0 and 1 and would be assigned the value Y_i . This particular Y_i is then used to find corresponding Y_1, Y_2, X_1 and X_2. When all the necessary parameters are found equation 4.10 is used to generate talkspurt/silence duration X_i.

$$X_i = \frac{Y_i - Y_1}{Y_2 - Y_i}(X_2 - X_1) + X_1 \tag{4.9}$$

Data traffic has been modelled with an exponential interarrival time of 'X' seconds. Then data packets were generated with an exponential distribution of mean 1.

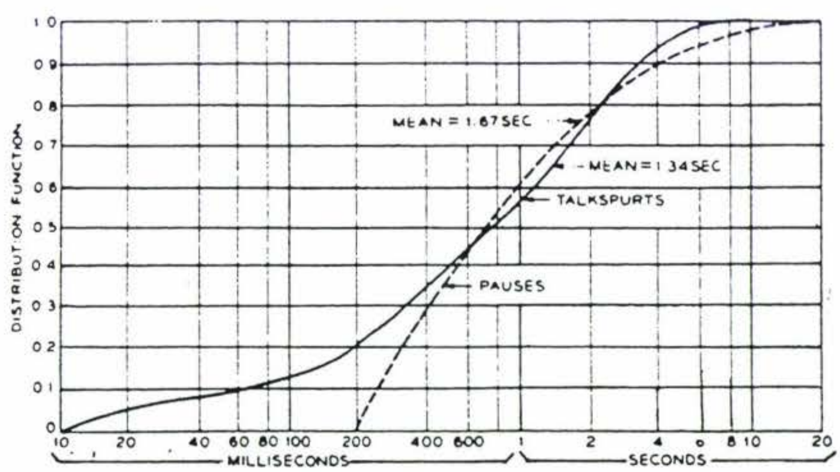


Figure 4.10 Talkspurt and silence distribution in telephone conversation [81]

Figure 4.11 shows the flow diagram of the process STATION. This process is initiated at the start of the simulation and continues until the end of the simulation. Initially the process decides what type of packet it wants to transmit. If it is a voice, the routine SILENCE is called which returns a silence period. The silence period is calculated by using Brady's statistics. For the silence duration that particular STATION remains in a wait state. After the wait state the routine TALKSPURT is called which returns a talkspurt duration using the Brady's statistics. The duration of silence and talkspurt is calculated by using two random number generators and the Brady's statistics. Then the talkspurt duration is used to calculate the number of packets needed to be transmitted during the talkspurt. If the current station needs to generate data packets, it waits for an exponential waiting time with a mean of 15 seconds (data was assigned an interarrival time of 15 seconds). Then a number of data packets are generated using another exponential distribution with a mean of 1.

Whenever any STATION has packet/packets to transmit, it activates a process MOBILE and the process STATION is suspended. The process MOBILE also returns some parameters that are used to update process STATION.

Upon re-activating the STATION, one packet is subtracted from the total number of packets to mean either one packet is successfully transmitted or one packet is dropped. In the case of data stations, packets are not dropped but, the packet count would be decreased when successful transmission occurs. Data is used as a loss sensitive traffic (in comparison voice is delay sensitive). When the total number of packets within a station reaches zero then the corresponding generator is used to re-generate more packets. In the case of a data station the data packets are sent in blocks of 'M' (size of M was varied during the study to investigate the best system performance) packets at a time. This is called data block transmission.

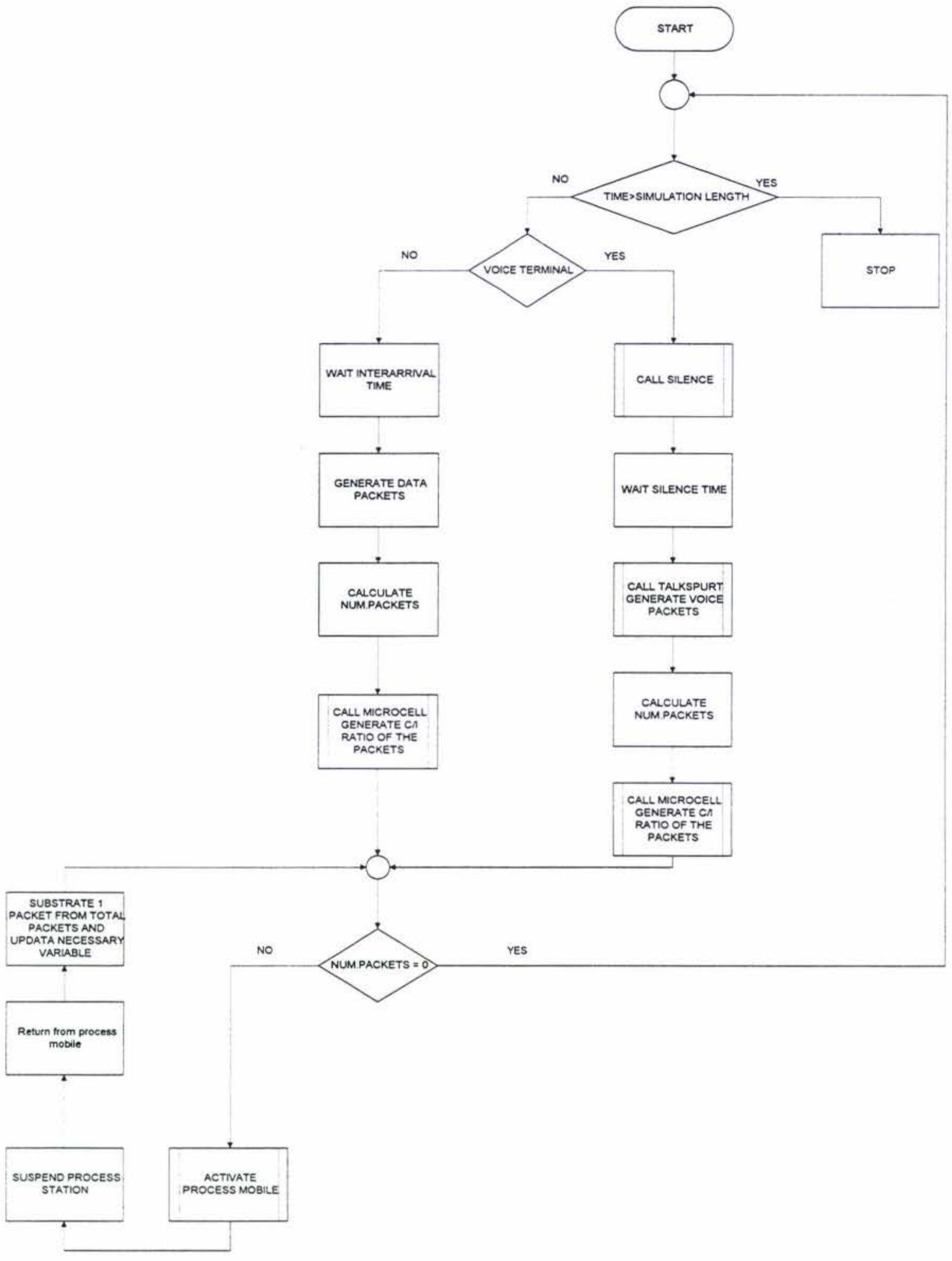


Figure 4.11 Flowchart of process STATION

Figure 4.12 shows the flow diagram of the process MOBILE which implements the major part of the ATDMA protocol. At the beginning of the process it checks whether the mobile terminal which activated the process has got any reserved slots or not. If it is a continuous transmission i.e. the mobile terminal has a reserved slot then the MOBILE process calls the routine SCHEDULER and waits for the time duration return from the routine SCHEDULER. The timing routine SCHEDULER returns the exact time when the transmission for that particular slot will start. At the end of the wait state MOBILE calls the routine BASE which supervises the transmission of the packet. The BASE station places the packet in a transmission queue.

At the end of the slot timing, the BASE checks the number of packets in the transmission queue. If the number of packets is more than one within a slot duration then a collision is detected and the BASE capture the mobile terminal which has the largest signal strength. Capture effect is performed in the situation when the strongest received signal is higher than the second strongest signal by more than the capture threshold. BASE also returns the retransmission status to the rest of the collided mobile terminals and then initiates the retransmission procedure. Normally in a reserved slot collision is not expected.

If a transmission is successful then BASE sends a positive acknowledgement to the mobile terminal. For a voice terminal, MOBILE will keep transmitting until all the packets of the current talkspurt have been transmitted. In the case of a data terminal, transmission will continue until all the packets of the current block are transmitted. If the transmission is successful and the packet is a null packet then BASE release the reservation for the particular mobile terminal.

In case of data transmission, the retransmission procedure is that the data terminal will wait for the next available free slot in the next frame. In the case of voice transmission, the process MOBILE checks whether the packet dropping delay threshold has expired or not. If it has not expired, the voice terminal will wait for the next available free slot. If the packet dropping delay threshold is over, MOBILE will add 1 to the number of voice packet loss. For the terminals which get the free slot, the process will follow the

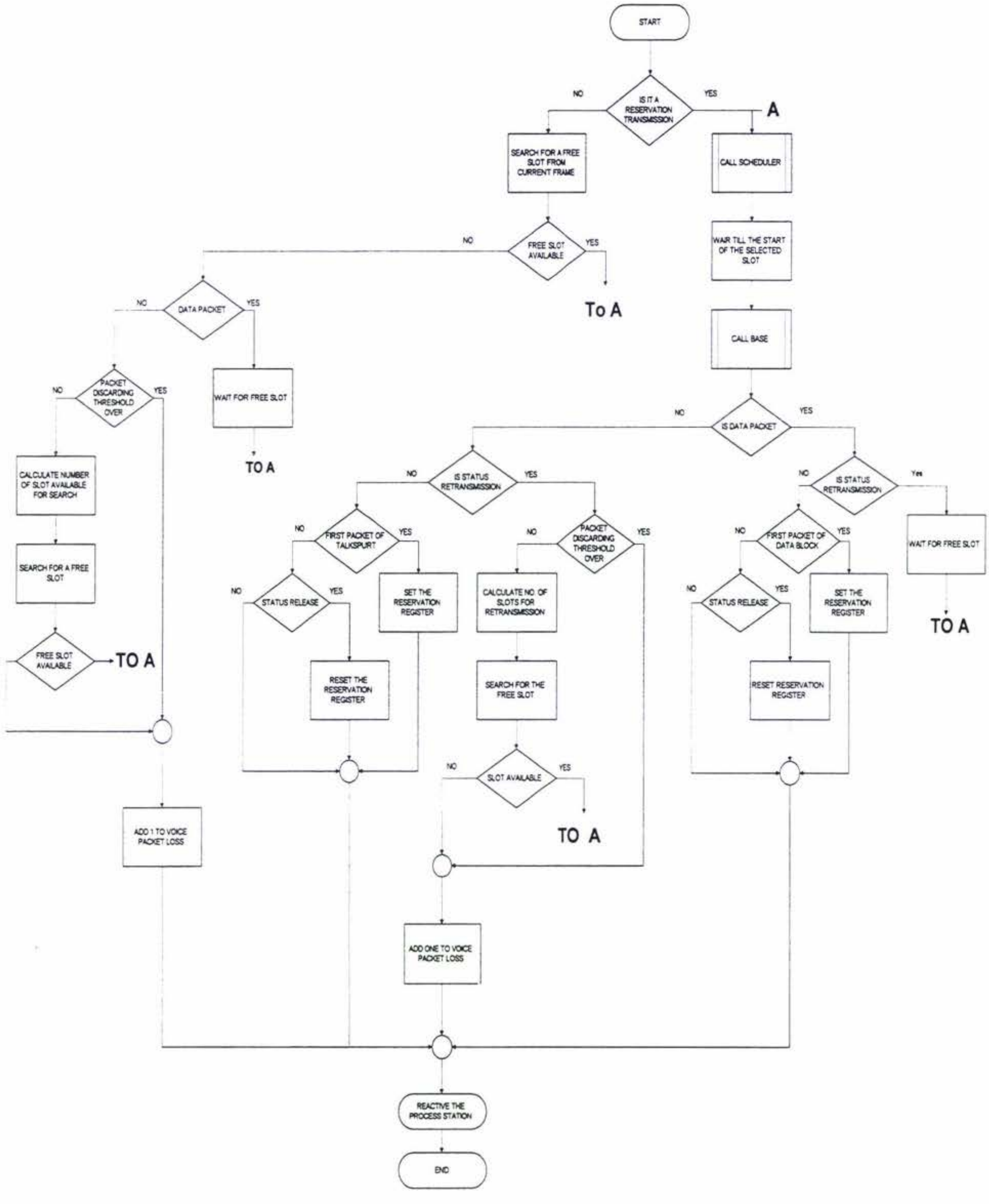


Figure 4.12 Flowchart of the process MOBILE

procedure as in the reserved transmission. The voice sources have higher permission probability for transmission than data terminals.

At the beginning of the process MOBILE if there is no reserved slot for the mobile terminal then the process starts looking for a free slot. For a data terminal it will wait until an available free slot is found. For a voice terminal, it will continue to search for the free slot before the packet discarding threshold period is over. If the packet dropping delay threshold is over, MOBILE will add 1 to the number of voice packet loss. If a free slot is available then the process will follow the same procedure as the reserved transmission. Depending on the acknowledgement from the base station the next action is taken.

4.3 Validation of the ATDMA protocol

To the best of my knowledge it appears that up until now an ATDMA protocol has not been simulated in a mixed voicd and data traffic with capture effect. Thus it is not possible to compare the results in this thesis with previous work. However, an ATDMA protocol has been simulated to support voice traffic in [39]. In order to validate the simulation model, comparetion was made with the voice only traffic. Through there are some difference between the simulation model and the model published in [39], however some of the results obtained from this simulation model is very close to the published results.

The major differences between the two work were in coder rate, slots per frame, burst information payload, interleaving, burst overhead. However, with voice only traffic without capture effect at transmission speed of 1Mb/s, the 1.935 user/slot was got using our model and this is particular value is exactly the same as that obtained in [39], thus validating the ATDMA protocol simulation model.

4.4 Simulation results

A discrete event simulation model was developed using SIMSCRIP II.5 to study the performance of the ATDMA protocol using the proposed reservation data block for an integrated voice & data traffic. The performance of the protocol is investigated by using the capture effect. Some of the parameters used for the simulation model are given in the table 4.3.

Parameter	Value
Transmission rate	1 Mb/s, 2 Mb/s
Transmission frame size	10 ms
Speech coder rate	16 kb/s
Slots per transmission frame	45, 85
Mean Data segment length	1 KB to 40 KB
Average talkspurt length	1.5 sec
Average speech activity	44 %
Speech packet dropping threshold	20 ms
Data model	Negative exponential
Exponentially distributed mean data interarrival rate	15 seconds
Channel model	Lee's model [71]
Cell type used	Microcell
Cell size	1 km
Carrier frequency	2 GHz
Cluster size	7
Antenna heights	20 ft (BS), 6 ft (MS)
Antenna gain	4 (BS), 0 (MS)
Transmitter power	1 W
Simulation length	300 s
Warm-up time	32 s

Table 4.3 Simulation parameters for the ATDMA protocol

For voice and data traffic different packet access transmission strategies are used. For voice packet transmission each mobile obtains a reserved slot for the duration of a talkspurt. For data a new block reservation scheme has been proposed here. Under the proposed scheme each transmitting data terminal will obtain a short block reservation which allows it to transmit M consecutive packets in reserved mode in M consecutive transmission frames. The length of the reserved block is the same irrespective of data segment length of a transmitting terminal. The idea behind the block reservation is to minimize collisions during access and give priority to voice packets. So the chosen number of reserved slots (R) used in one frame and the length of data block (M) are the key design parameters which can decide the system's performance.

For example if more R slots are used, more terminals get the opportunity to access the base station, then contention decreases, since the total number of slots in one frame is fixed, increase of the R slot will lead to a decrease in the number of traffic slots (T slots). The decrease of the T slots will lead to the increase of the allocation delay, resulting decrease in system performance. Whereas if the number of R slots used is too few then contention will increase and this will result in a drop in system performance.

From the point of view of data block length M, if the data block length is too short then data terminals will attempt to access the channel too frequently resulting in higher collisions and longer channel allocation time. If larger block length is used then the channel will be held by the data terminals for a longer period resulting in a higher percentage of speech packet loss.

According to the above discussion, simulation was carried on to find the optimum number of R slots and optimum data block length M and then to investigate the ATDMA protocol performance.

Simulation was first taken to find the optimum performance of the protocol at 1 Mb/s transmission rate. Figure 4.13 and Figure 4.14 show that while keeping the voice terminals constant at 45, the maximum number of data terminals that could be supported with the use of a 1% speech packet loss criteria is 66. The optimum system parameters are M=16 and R=6 at 1 Mb/s transmission rate. The maximum system capacity of the ATDMA protocol is 111 terminals (66 data terminals + 45 voice terminals). Multiplexing efficiency of 2.46 is achieved.

Delay characteristics of the ATDMA protocol are shown in Figure 4.15, With optimum system parameters data packet access delay is below 100 ms. The delay increases sharply with different values of M and R.

Figure 4.16 to Figure 4.18 show the ATDMA protocol performance with transmission bit rate equals to 2 Mb/s. With 2 Mb/s transmission bit rate, the optimum number of R slots is R=11, the optimum number of data block length is M=16, with optimum system parameters the maximum system capacity is 126 data terminals plus 85 voice terminals totalling 211 terminals, offering multiplexing efficiency of 2.48. Also the access delay characteristics shows that with optimum system parameters, data packet access delay and end to end data packet delay are below 100 ms.

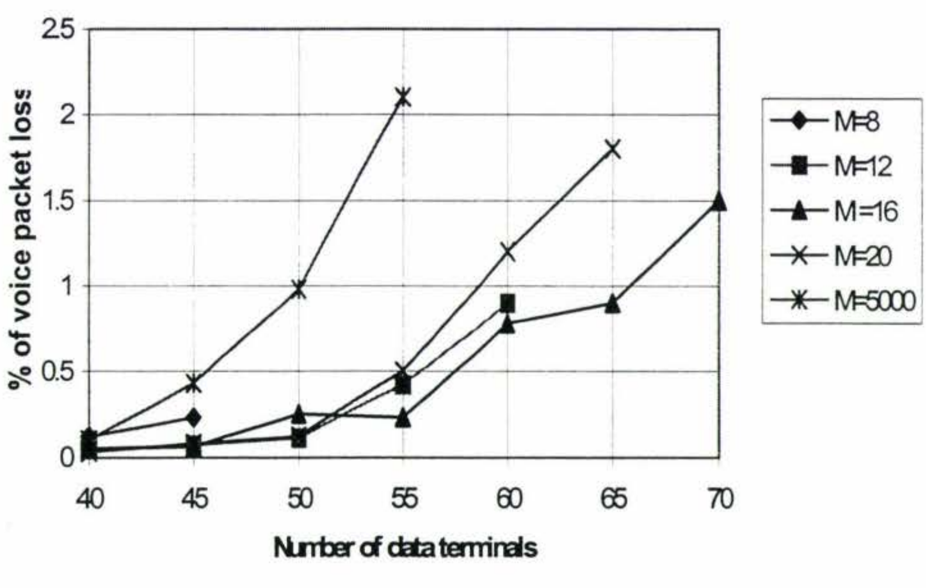


Figure 4.13 Performance of the ATDMA protocol with different data block length M. Transmission bit rate is 1 Mbs. Number of voice terminals used is 45. Number of reserved slots (R) used is 6.

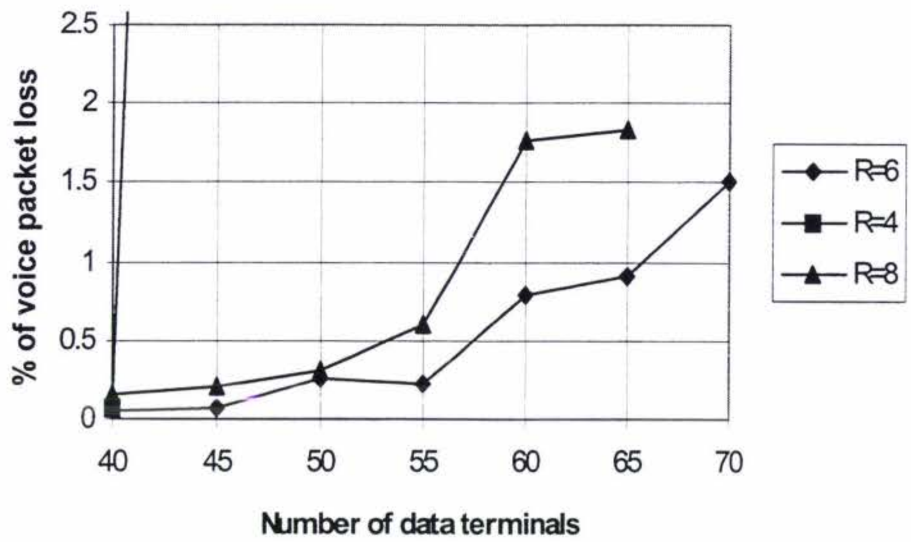


Figure 4.14 Performance of the ATDMA protocol with different numbers of R slots. Transmission bit rate is 1 Mb/s. Data block length M = 16. Number of voice terminals used is 45.

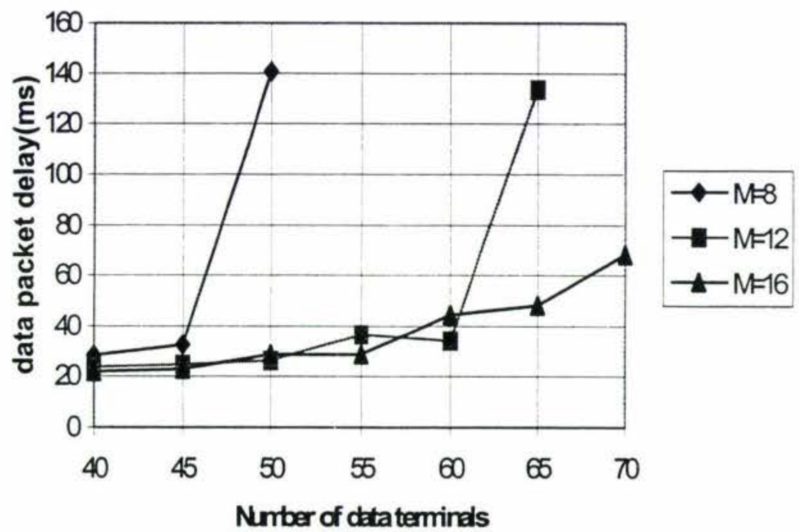


Figure 4.15 Data packet access delay performance of the ATDMA protocol. Transmission bit rate is 1 Mb/s. Number of voice terminal used is 45, Number of R slots used is 6. M referred to length of data block length.

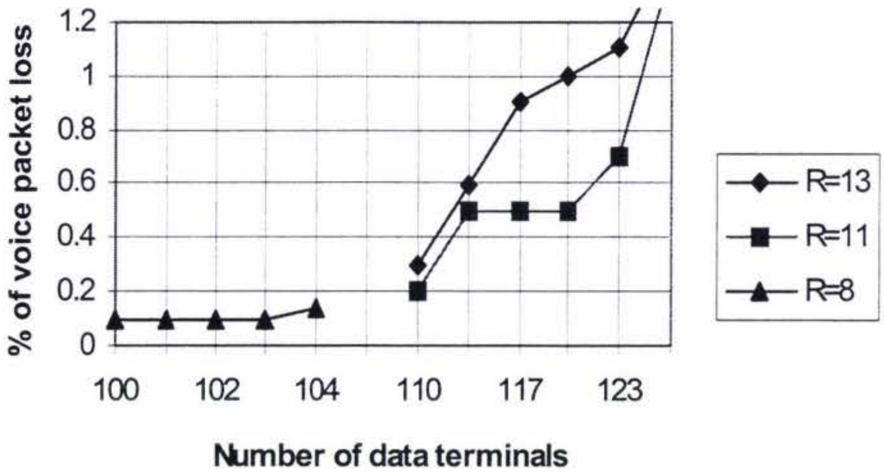


Figure 4.16 Performance of the ATDMA protocol with different numbers of R slots. Transmission bit rate is 2 Mb/s. Data block length M = 16. Number of voice terminals used is 85.

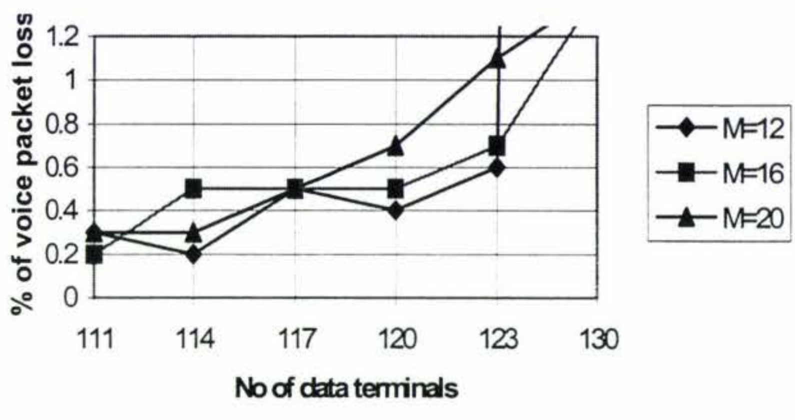


Figure 4.17 Performance of the ATDMA protocol with different data block length M. Transmission bit rate is 2 Mb/s. Number of voice terminals used is 85. Number of reserved slots (R) used is 11.

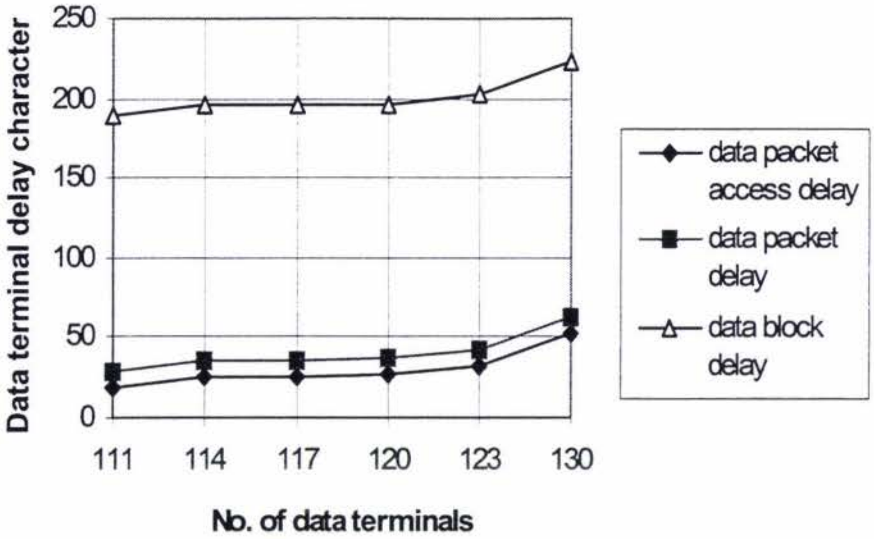


Figure 4.18 Data terminal delay performance of ATDMA protocol with optimum system performance. Transmission bit rate is 2 Mb/s. Number of voice terminal used is 85, optimum system parameters used are M=16, R=11.

Further simulation results were obtained by combining the data block reservation scheme and the capture effect. In the multiple access process, when more than one terminals try to access the base station at the same time, contention occurs. In this simulation capture effect is used. Without capture effect, when contention occurs, the base station simply discards all the access requests and the contending terminals have to try to access the network using the next available time slot. This increases the access delay and is an important factor affecting the total system performance. With the use of capture effect, when contention occurs, the terminal with the largest carrier to interference rate (C/I) is selected by the base station. Simulation results show that by combining the capture effect and data block reservation scheme, the system performance increases considerably.

Figure 4.19 to Figure 4.21 show the performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with the transmission bit rate of 1 Mb/s. Simulation results show that optimum system parameters are M=4 and R=4. The system can support up to 83 data terminals and 45 voice terminals (128 terminals). Multiplexing efficiency of 2.83 is achieved. This figure shows that 15.8% increase in system capacity can be achieved. With the use of capture effect, Less number of R slot is required compared with no capture effect, this is because with the use of capture effect R slot utilization is increased.

Figure 4.22 and Figure 4.23 show the improvement of system performance by comparing the system with capture effect and the system without capture effect. Simulation results indicate that by combining the data block reservation scheme and capture effect the ATDMA protocol capacity could be increase from 111 to 128 terminals, up by 15%. By comparing capture and no capture situations it was found that with capture effect shorter data block reservation length (M) and fewer R slots per frame can be achieved. Decreasing the value of M leads to higher priority to voice terminals, and decreasing the value of R increases the number of traffic slots per frame. Capture effect reduces the probability of collision resulting higher system capacity.

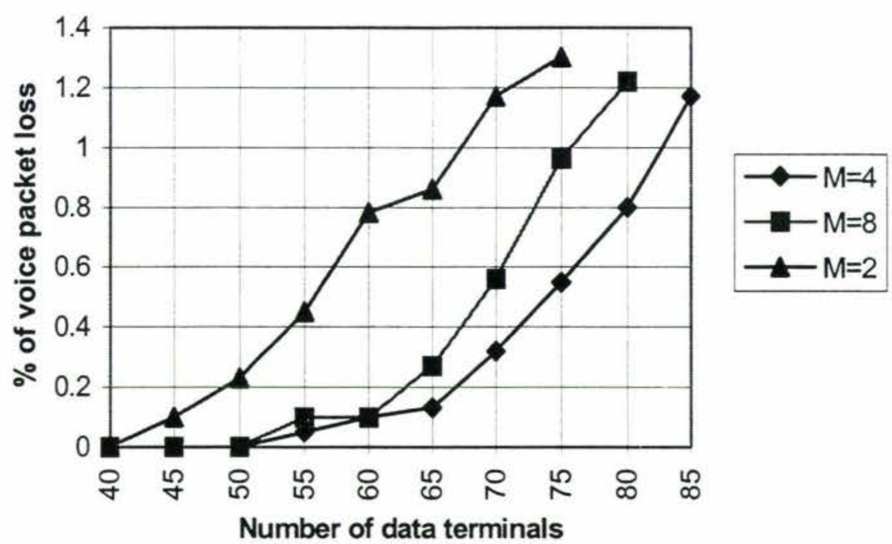


Figure 4.19 Performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with different data block length M. Transmission bit rate is 1 Mb/s. Number of voice terminals used is 45. Number of reserved slots (R) used is 4.

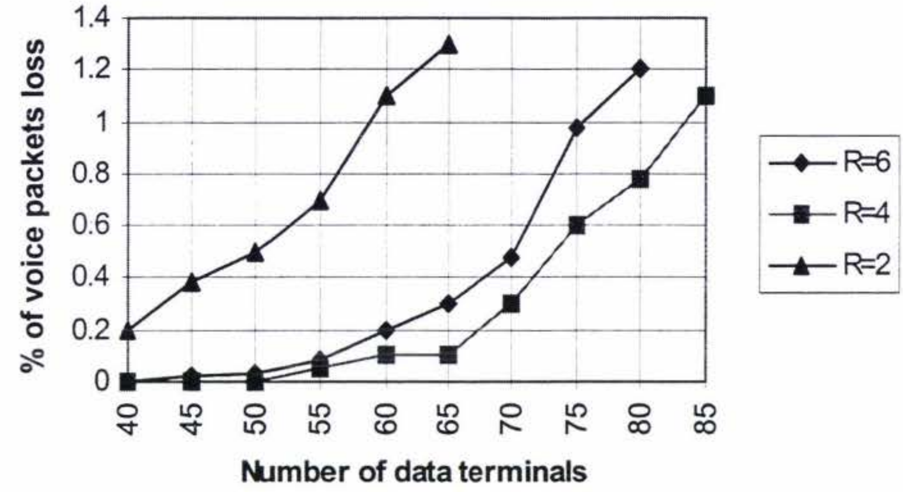


Figure 4.20 Performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with different numbers of R slots. Transmission bit rate is 1 Mb/s. Number of voice terminals used is 45. Data block length M=4 is used.

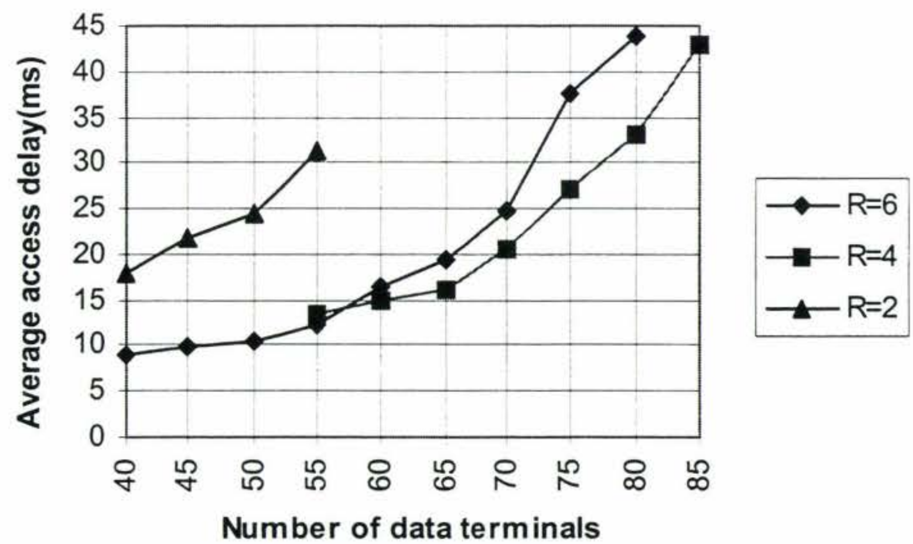


Figure 4.21 Data terminal delay performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with different numbers of R slots. Transmission bit rate is 1 Mb/s. Number of voice terminals used is 45. Optimum system parameters used are M=4, R=4.

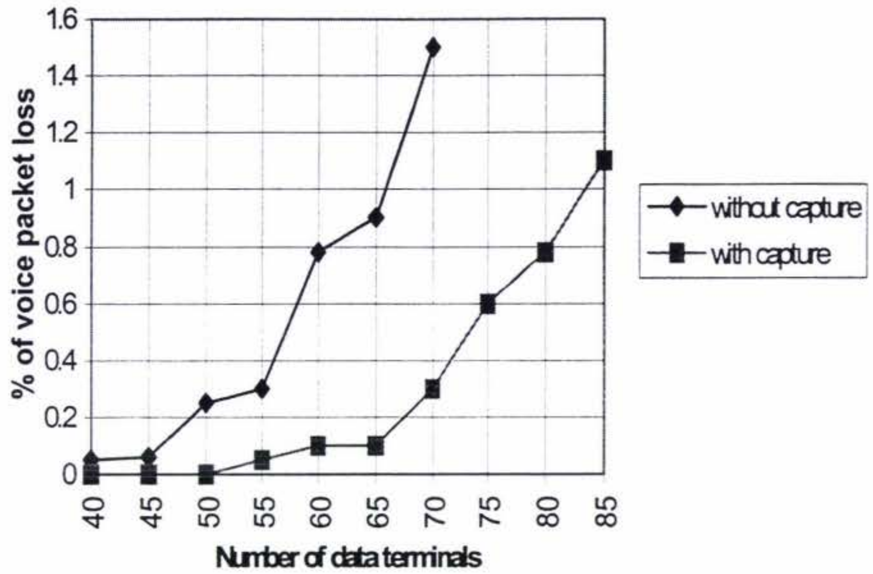


Figure 4.22 Comparing the performance of the ATDMA protocol with capture effect and without capture effect. Transmission bit rate is 1 Mb/s. Number of voice terminals used is 45.

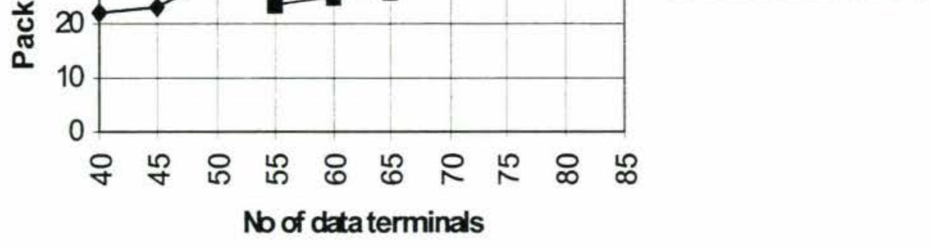


Figure 4.23 Comparing the data packet delay performance of the ATDMA protocol with capture effect and without capture effect. Transmission bit rate is 1Mb/s. Number of voice terminals used is 45.

Simulation was also used for the investigation of transmission bit rate of 2 Mb/s. Figures 4.24 to 4.27 present the simulation results of transmission bit rate equal to 2 Mb/s. By combining the data block reservation scheme and capture effect the ATDMA protocol capacity could be increased from 211 to 238 terminals. up by 13%. Data block length decrease from 16 to 8 and the number of R slots decreases from 11 to 4. Optimum parameters are M=8, R=4.

Table 4.4 shows different performance figures. The table compares the performance of the block reservation scheme and the effect of capture effect on the traffic performance of the protocol.

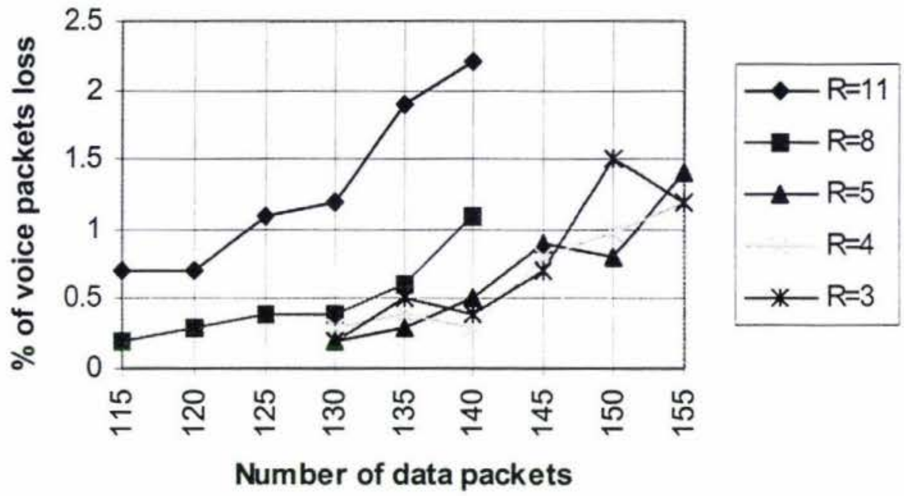


Figure 4.24 Performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with different number of R slots. Transmission bit rate is 2 Mb/s. Number of voice terminals used is 85. Data block length M is 8.

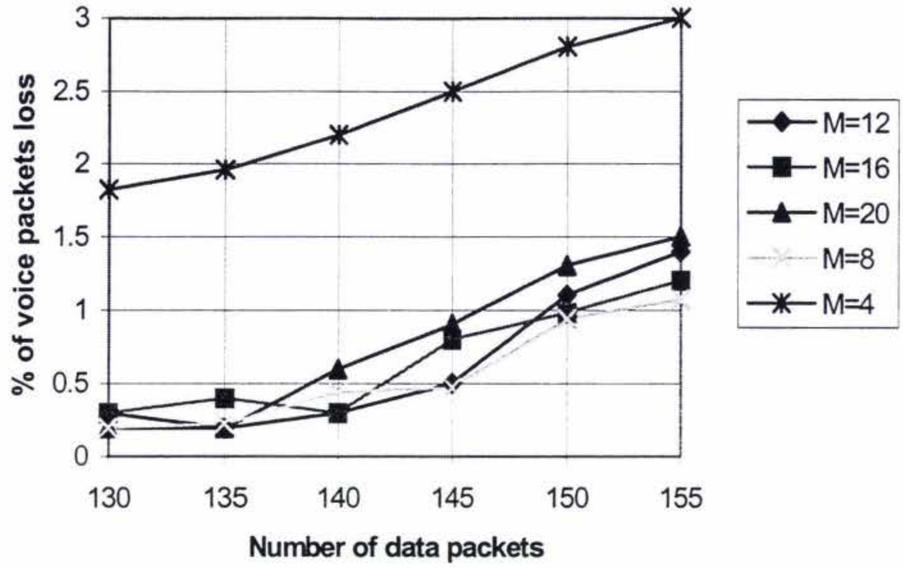


Figure 4.25 Performance of the ATDMA protocol by combining the data block reservation scheme and capture effect with different data block length M. Transmission bit rate is 2 Mb/s. Number of voice terminals used is 85. Number of reserved slots (R) used is 4.

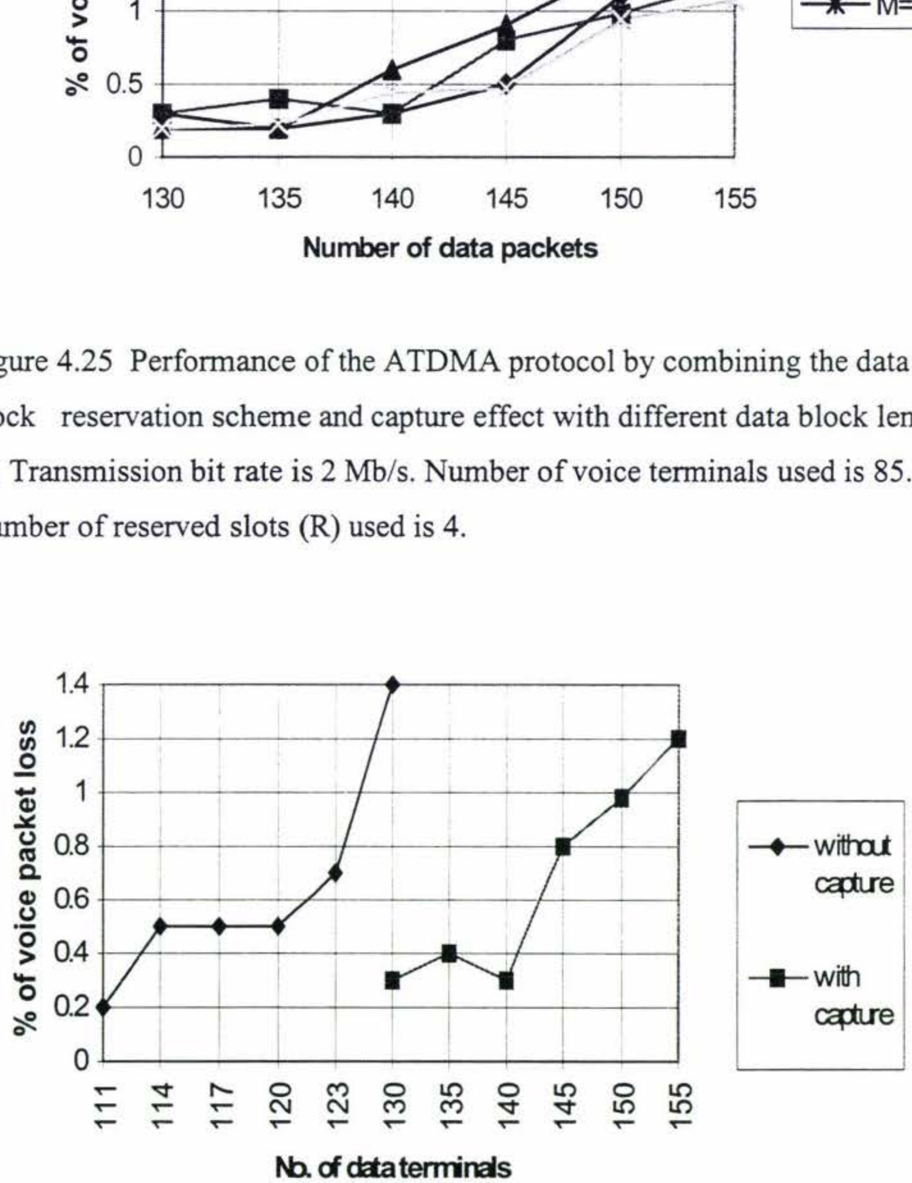


Figure 4.26 Comparing the performance of the ATDMA protocol with capture effect and without capture effect. Transmission bit rate is 2 Mb/s. Number of voice terminals used is 85.

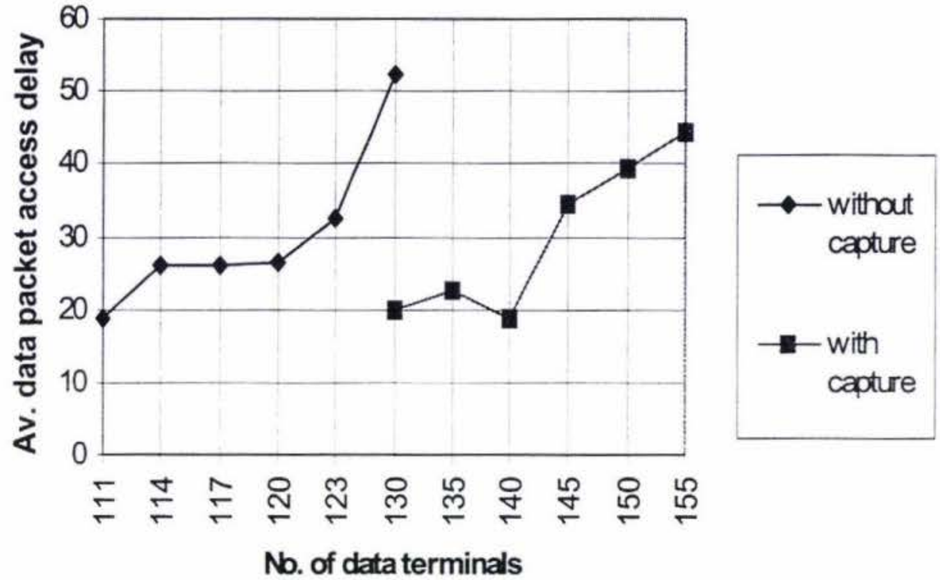


Figure 4.27 Comparing the data packet delay performance of the ATDMA protocol with capture effect and without capture effect. Transmission bit rate is 2 Mbs. Number of voice terminals used is 85.

The results show that the traffic slot utilization is nearly the same for both capture and no capture situation because the traffic slots are contention free. Using the capture effect an increase of traffic capacity can be achieved by using fewer R slots and also shorter data block length M. Using fewer R slots and shorter data block length M results in the higher utilization of R slots because of the absence of contention and a higher number of traffic slots available. The higher utilization of R slots and more available traffic slot result in the improvement of protocol capacity.

Transmission bit rate	ATDMA reservation scheme	Optimum number of M, R and T.	No. of terminals supported	Maximum R slots utilization	Maximum T slots utilization	No. of calls supported per slot
1 Mbs	Without capture effect	M=16 R=6 T=39	66 data 45 voice	22%	96%	2.42
1 Mbs	With capture effect	M=4 R=4 T=41	83 data 45 voice	36%	97%	2.84
2 Mbs	Without capture effect	M=16 R=11 T=74	126 data 85 voice	23%	98%	2.48
2 Mbs	With capture effect	M=8 R=4 T=81	153 data 85 voice	36%	98%	2.80

Table 4.4 Performance of the ATDMA protocol with optimum parameters. M refers to data block length, R refers to number of reserved slots in one frame, T refers to number of traffic slots in one frame.

Simulation was continued to study the effect of the capture ratio on the protocol's capacity. A useful parameter in analyzing the capture effect in a packet radio system is the minimum power ratio of an arriving packet, relative to the other colliding packets, such that it is received. This ratio is called the capture ratio, and is dependent upon the channel characteristics as well as the design of the modulation and coding scheme, the average C/I, and the length of the packets [56]. In this study, average received C/I ratio difference was used as the parameter of capture ratio to investigate the effect of capture effect on the ATDMA capacity. When collision occurs, the base station captures the signal with the largest C/I ratio and also signal strength of the strongest signal compared with the second strongest signal to find out the capture ratio. Optimum system parameters and protocol's traffic capacity changes with different capture ratios. Table 4.5 illustrates protocol parameters and capacity with different capture ratios. It shows that when capture ratio is increased, for maximum protocol capacity optimum length of data block length M also increases. If the capture ratio is too large, M will increase to 16, which is the same value as without capture effect situation. This means capture effect has no affect on protocol performance in this situation. Figure 4.28 shows the protocol capacity with different capture ratios. It is obvious from this figure that the effect of capture effect is related to the capture ratio. The larger the ratio, the smaller gain in traffic capacity compared no capture situation. With capture ratios equal to 6~10 dB, the capacity of the system increases 5~6 % with capture effect, compared with without capture effect.

Capture ratio	Optimum number of M, R	Capacity
0 db	M=4, R=4	128
2 db	M=6, R=6	120
6 db	M=10, R=6	117
10 db	M=12, R=6	116

Table 4.5 Optimum parameters with different capture ratio. Transmission bit rate equals to 1 Mb/s.

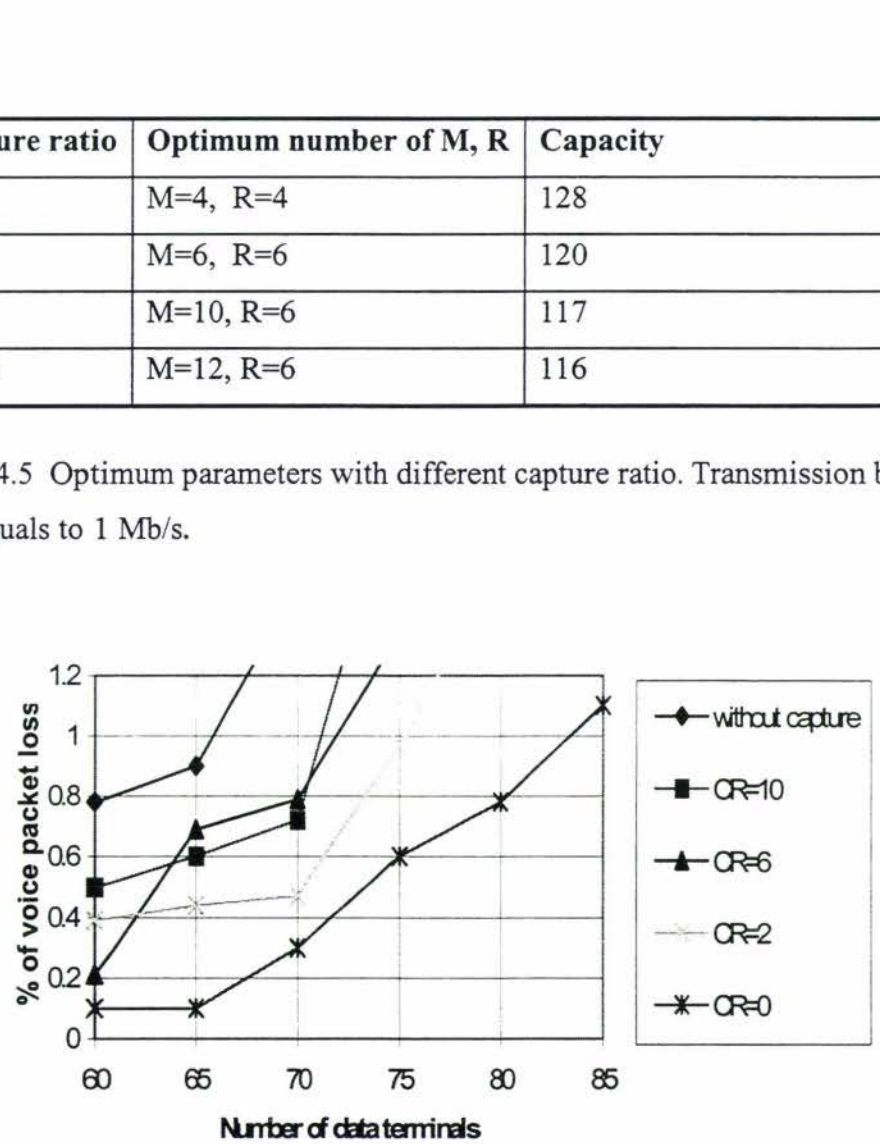


Figure 4.28 Comparing the performance of the ATDMA protocol with different capture ratios (CR). Transmission bit rate is 1 Mb/s. Number of voice terminals used is 45.

4.5 Conclusion

In this chapter the performance of the Advanced Time Division Multiple Access (ATDMA) protocol in a microcellular environment was investigated using the computer simulation method. To study the performance of the protocol a discrete event based simulation model was developed which included an ATDMA protocol model and a microcell channel model of a city area. The efficiency of the protocol in a microcellular environment and also different access techniques for voice and data traffic to improve the efficiency of the protocol were examined in this chapter.

A microcell channel model was developed based on a seven cell cluster where only the first tier co-channel interference was used. The channel model was developed for the city area where the signal arriving at the mobile unit is blocked by the individual buildings; this weakens the signal strength and is considered as part of the path loss. In small cells, the signal loss was calculated based on the dimensions of the building blocks. The exact height of buildings in the middle of the propagation path is not important.

The simulation model was used to find out the appropriate parameters for the optimum performance of the protocol. In order to improve the protocol capacity, a new modified access & transmission strategy which combine a data block reservation scheme and capture effect was developed to find out the optimum system parameters and optimum protocol performance. Data being loss sensitive, there is a need to obtain the most efficient block size which can provide the minimum packet loss for voice terminals and the minimum end-to-end delay for the data terminals. If the data block length is too short then data terminals will attempt to access the channel too frequently resulting in a higher number of collisions and longer channel allocation times. Whereas if larger block length is used, then channels will be held by the data terminals for longer periods resulting in a higher percentage of speech packet loss. Application of capture effect successfully solves the problem of collision, and also combining the data block allocation scheme and capture effect provides an opportunity to successfully utilise reservation slots and traffic slots, and to improve the protocol capacity. The simulation results show that by using the modified access strategy and by selecting appropriate parameters the traffic carrying capacity of the protocol can be increased up to 15% [82].

Further simulation was also undertaken to investigate the effect of capture ratio on the protocol performance. Simulation results show that the increase in performance depends on the capture ratio.

CHAPTER V

OVERALL CONCLUSION AND FUTURE WORK

The performance of the ATDMA protocol in a microcell environment has been investigated in this thesis. The particular interest in the conduct of the research was to develop an access strategy for the ATDMA protocol and to investigate the protocol performance in a microcell channel environment for transmitting mixed voice/data packets by using the optimum protocol parameters. A microcell propagation model which considers the first tier co-channel interference in urban areas was developed to investigate the performance of the protocol. A data block reservation scheme combining capture effect were studied to increase the performance of the protocol. By combining the data block reservation scheme and the capture effect the ATDMA protocol's traffic efficiency for a mixed voice and data traffic in an urban microcell environment was investigated by means of a computer simulation method. With the consideration of the capture ratio, the effect of capture has also been evaluated in a more practical manner. One of important finding of this work is that the ATDMA protocol's system parameters need to be optimised for specific application environment. The proposed data block reservation technique can definitely improve the traffic performance by minimising the contention for data and voice traffic. Simulation results presented here shows that by using the optimum system parameters and the capture effect the traffic carrying capacity of the protocol can be significantly improved.

The microcell environment is a subject of major interest for the third generation mobile radio systems due to its many advantages over the conventional cellular system. Microcells embedded in conventional cellular systems can provide improved coverage and performance, increase system capacity, and delivery of innovative value-added services. Microcells also have improved propagation properties, with less severe fading, and a much increased coherence bandwidth, allowing low error transmission at higher bit rates than currently available within conventional cells. Microcells require substantially lower transmitter power than conventional cells, so that battery life is increased and the size of the handset mobile telephone reduced. They only require small base stations with short antenna which can be sited unobtrusively and their coverage area can be predicted more accurately than that of conventional cells, thus facilitating network design.

Radio channel places fundamental limitations on the performance of mobile communication systems. The effective design, assessment, and installation of a radio network requires an accurate knowledge of the characteristics of the channel. Channel modelling technique is an important research area in the design and implementation of a mobile radio network. In this investigation a microcell model which is based on William Lee's model was developed to investigate the path loss and carrier to interference (C/I) characteristics of the microcell propagation environment. The model was developed based on a seven cell cluster where only the first tier co-channel interference was used. Number of interference cells for other cluster sizes will be same because of the hexagonal geometry. For different cluster sizes the main difference is the co-channel distance. Interference generated by each of the co_channel cell will depend on the usage of a particular time slot in the co-channel cells. Interfering signal strength from co-channel cells is calculated at the receiver by using the developed propagation model. The distance between a interfering cell and the interfered cell is related to cell size and the cluster size. Interference is calculated on the UP link measured at the base station. The channel model was developed for a city area where the signal arriving at the mobile unit is blocked by individual buildings; This weakens the signal strength and is considered as part of the path loss. In a small cell the signal loss was calculated based on the dimensions of the building blocks. The exact height of the building in the middle of the propagation path is not important. The microcell model was used to investigate the ATDMA protocol performance.

A mobile radio network different from other multi-access networks due to its 'hidden user' problem and its high crucial spectrum resource. So one of the most important questions when designing and standardising cellular mobile radio systems is the development of a multiple access protocol. 'Hidden user' problem can be eliminated by using a central cordinater which is used in the ATDMA protocol. The ATDMA protocol has been studied in recent years for the third generation mobile radio systems UMTS [26].

The ATDMA protocol is a flexible one which can be configured in different ways to suit the requirements of different types of traffic. It is a hybrid multiple access technique which emphasises flexible and adaptive radio resource allocation with respect to the implementation of a wide range of services with different delay and quality requirements. Those adaptation schemes could be related to cell types, user/traffic type, interference, source activity, etc. The ATDMA protocol can operate either in circuit switched mode or in packet switched model. It combines the advantages of both circuit switched and packet switched techniques. Also the ATDMA protocol has other advanced features.

A simulation model was developed to study the performance of the ATDMA protocol. The protocol was investigated by using voice and data traffic source. For voice and data traffic different access strategies are used. For voice packet transmission each mobile obtains a reservation for the duration of a talkspurt. For data a new block reservation scheme has been proposed. Under the proposed scheme, each transmitting data terminal will obtain a short block reservation which allows it to transmit M consecutive packets in reserved mode in M consecutive transmission frames. The length of the reserved block is the same irrespective of the data segment length of a transmitting terminal. The idea behind the data block reservation is to minimise the collision during access and give priority to voice packets because the voice is a delay sensitive source.

With the development of ATDMA access strategies, data packets are reserved in blocks and voice packets are reserved in talkspurts, but collision still exists. In this case capture effect was used to decrease the number of collisions and to improve the system capacity. Without capture effect when collisions occur, the base station simply discards all the terminal's allocation requirements and the contention terminals will enter a collision resolution scheme and try to access the network again within the delay threshold. With the introduction of capture effect in the ATDMA protocol, when collision occurs, the base station picks up the signal with largest C/I ratio. Use of capture effect successfully solves the problem of collision.

Simulation result shows that the key design parameters are the number of reserved slots and the data reservation block size (M). These parameters depends on the transmission conditions as well as with the transmission bit rate and the capture effect. Simulation model was used to find out the optimum value of these key parameters.

Simulation results showed that in a microcell environment, the ATDMA protocol can support 2.42 calls/slot and 2.48 calls/slot for 1 Mbs and 2 Mbs transmission rate respectively without the use of capture effect. With the use of capture effects the protocol can support 2.84 calls/slot and 2.80 calls/slot at 1 Mbs and 2 Mbs transmission rate respectively. With capture effect the capacity of the protocol increases up to 15%. Higher number of calls/slot is supported because of absence of collision. The above multiplexing efficiency figures were obtained using speech and bursty data sources.

Simulation was continued to investigate the effect of capture ratio on the performance of the protocol. It shows that the effect of the capture is related to the capture ratio. The smaller the capture ratio, the better the performance. With capture ratios equal to 6~10 dB, capacity of the protocol increases by 5~6 % with capture effect compared to without capture effect.

The ATDMA protocol capacity and several other performances have been studied in this work. Due to time constraint many more investigations could not be carried out. Useful future work would be to study the issues of dynamic data block allocation schemes. It is suggested that the length of data block could be adjusted adaptively according to the change of C/I ratio, and that the protocol performance could be improved further.

During this work, six co-channel interfering sources were taken consideration which is the worst case as discussed in chapter 2. Further simulation could be taken to investigate the effect of changed co-channel interference because in the real case co-channel cell existed is changed dynamically according to the distribution of active mobile terminal.

Simulation may need to run with different parameters to get more valuable result. For example with different cell size, different mobile distribution within the cell and also with different antenna parameters.

In this study the effect of mixed voice and data traffic was discussed. Another attractive topic may be the investigation of the effect of multimedia traffic ie. inclusion of video sources. ATDMA protocol as a third generation mobile communication network protocol have a strong potential to transmit multimedia traffic. However as this research work suggest that appropriate system parameters need to be used for achieving highest efficiency of the protocol.

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APPENDIX

A SIMULATION PROGRAM FOR THE ADVANCED TIME DIVISION MULTIPLE ACCESS (ATDMA) PROTOCOL IN A MICROCULAR ENVIRONMENT

THIS SECTION PRESENTS THE LISTING OF THE SIMULATION PROGRAM USED TO STUDY THE PERFORMANCE OF ATDMA PROTOCOL IN MICROCELL ENVIRONMENT. THE PROGRAM IS WRITTEN IN SIMSCRIPT II.5.

```
PREAMBLE

  DEFINE CAPTURE AS REAL 1-DIMENSIONAL ARRAY

  DEFINE
  CLUSTER.SIZE, BLOCK.LENGTH, ANTENNA.BASE, ANTENNA.MOBILE,      GAIN.BASE,
  GAIN.MOBILE, POWER.TRANSMITTER, N.MICROCELL, DISTANCE.CELL, FREQUENCY,
THRESHOLD
  AS REAL VARIABLES

  DEFINE RADIUM.CELL, INTER.SWITCH AS INTEGER VARIABLE
  DEFINE STRENTH AS REAL VARIABLE

  RESOURCES
   EVERY UNIT HAS A NO.LOSS
  DEFINE
   NO.LOSS AS AN INTEGER VARIABLE

  DEFINE GG AS A INTEGER VARIABLE

  DEFINE
   V.SLOTS.NO_ARRAY, RSLOT.FLG, LAST.PK_ARRAY, DATAB, FRAME, R.SLOT_ARRAY,
PACKET_ARRAY, LOS,
   V.SLOT.VALUE_ARRAY, IDEN, RESERVATION, TEMP_ARRAY,V.SLOT.COUNT_ARRAY,
   PK.COUNT, PACKET.NO_ARRAY, ARRAY_COUNT, V.NUM.DROPOUTS_ARRAY,
   DATA_PER.BLOCK, PACKET.COUNTER AND PAC.TOT
  AS INTEGER 1- DIMENSIONAL ARRAYS

  DEFINE
   A.TIME_ARRAY, DATA.START.TIME,  FRAME.START.TIME AND ACCESS.COUNTER AS REAL 1-
DIMENSIONAL ARRAYS

  PROCESSES INCLUDE  TEMPPRO, TIMER AND MOBILE


  EVERY STATION HAS
   A SIG.STRENTH,  A NUM.PACKETS, A ST_TYPE,      A COLL, A TALKSPURTS, A OWN.PACKET,
A PACK.COUNT,
   A FR.DEL,    A LAST.PACKET, A LOST, A MESSAGE.LENGTH, A RESER.SLOT, A
VIDEO.SLOTS.NO,
   A ID,  A VIDEO.SLOT.VALUE, A VIDEO.NUM.DROPOUTS, A A.TIME, A VIDEO.SLOT.COUNT AND
A SILENCES


  EVERY MOBILE HAS
   A STRENTH.SIG,  A MOB.STATION,  A TALKSPURT.TIME, A ARRIVAL.TIME, A FOUND,  A
NUM.DROPOUTS, A FILED.FLAG,
   A S.FLAG,      A STATUS,      A BUFF.SLOT,   A PE.PROB, A NEXT.LIMIT, A THIRD.LIMIT,
   A FIRST.TIME,   A R.FLAG,      A RESER.MOB,   A DEL.MOB, A LATE.F,    A MOB.LOSS,
   A SID.TXED.FLAG, A MY.SLOT,      A PACKET.COUNT, A TYPE,    A PACKET.NO,  A LT.PACKET,
   A V.SLOTS.NO,   A V.SLOT.VALUE,  A V.NUM.DROPOUTS,        A V.SLOT.COUNT AND A
TEMP.J
  AND MAY  BELONG TO THE TRANSMISSIONQ AND THE WAITINGQ

"  DEFINE TRANSMISSIONQ AS A SET RANKED BY HIGH R.FLAG
  " THE REASON FOR THIS IS SO AS TO BE ABLE TO SELECT THE WINNING MOBILE'S SID FROM A
GIVEN
  " RESOURCE ALLOCATION SLOT PERIOD

   DEFINE WAITINGQ AS A SET RANKED BY LOW ARRIVAL.TIME

   BREAK MOBILE TIES BY LOW ARRIVAL.TIME
```

```
 '' BREAK MOBILE TIES BY HIGH R.FLAG


   THE SYSTEM OWNS THE TRANSMISSIONQ
   THE SYSTEM OWNS THE WAITINGQ


   TALLY NUMBER.OF.MOBILES AS THE NUMBER,
     MINCT  AS THE MINIMUM,
     MAXCT  AS THE MAXIMUM AND
     MEANCT AS THE MEAN OF ROUND.TIME

   TALLY MEANDT AS THE MEAN OF DATA.TIME
   TALLY MEANVE AS THE MEAN OF VIDEO.TIME
   TALLY MEANFR AS THE MEAN OF FRAM.TIME

   TALLY MEANDTC AS THE MEAN OF CON_DATA.TIME
   TALLY MEANVEC AS THE MEAN OF CON_VIDEO.TIME
   TALLY MEANCTC AS THE MEAN OF CON_ROUND.TIME

   TALLY MEANTDD AS THE MEAN OF D.TIME
   TALLY MEANDB AS THE MEAN OF D.BURST_TIME

   DEFINE
    YAN, NAME1,NAME2 AND NAME3
   AS REAL, 1-DIMENSIONAL ARRAYS


   DEFINE
    ROUND.TIME, CYCLE.TIME, PACKET.TIME, PROP.COL,  TIME,  TIME1, FRAME.TIME,
FRAM.TIME,
    COL,   SCALE,   TIMEV,   TALKSPURTS, SILENCES, ONEX, ONEY, ONX, ONY,
    TOXPRT,  MEAN.TOXPRT, SIL,   MEAN.SIL,  POS,  POS1,  DI,  TP,
    TS,   LIMIT,   MESS,   MESSAGE.LENGTH, MESSAGES,  TMP,   OVER,
    POS2,   ALL,   FRAME.LENGTH,SL.TIME,  DATA.TIME,  VIDEO.TIME,  LOTS,
    FRAMES.PER.SEC,   CON_DATA.TIME, CON_VIDEO.TIME,   CON_ROUND.TIME,
    PREV.TIME,  A.TIME, D.TIME, D.BURST_TIME, TEMP.TIME  AND UTIL
   AS REAL VARIABLES

   DEFINE .SECONDS    TO MEAN DAYS
   DEFINE .MILLISECONDS TO MEAN HOURS
   DEFINE .MICROSECONDS TO MEAN MINUTES

   DEFINE
    N,     ID.NUMBER,  SID.NUM,   BIT.RATE,   INDEX,    NUM.PACKETS,
    MOB.STATION,  STOTAL,   CELL.SIZE,  WLO,    WHI,    WDELTA,
    NUM.COLLS,  NUM.MOBILES, NUM.COMPLETED,  CONT,    NUM.COLLISIONS, N.TRANS,
    CODER,   BITS,   TOTAL,    SUCCESS,   TOT,    LOST,
    NUM.OF.TOXPRT, TYPE,    NUM.SIL,    TEMP,    LATE.F,    NUM.MOB,
    RA.SLOT.NUM,  AV.RES.SLOT, WINNING.SID,  WON.FLAG,   TEMP.COUNT,  COUNT,
    RSL.NO,   PACK.DROP.RATE,    D_TNAS,   D_TNDOS,   D_SUCCESS,
    PACK.COUNT,  PACKET.COUNT,  DATA.BLOCK,  VOICE.MOB,  VID.COD.RATE,
    OVER.HEAD.BITS,    VIDEO.MOB,   HEY,    V_TNPKTS,   V_SUCCESS,
    V_NUM.LOST.PKTS, V_TNDOS,  VIDEO.BLOCKED,  V_TNAS,   V.PACKETS,  CELLSIZE,
    VID.COUNT,  Q.COUNTER,  PACKET.NO,  NO.OF.PKS,  FREE.TRAFF.SLOT.INDICATOR,
LT.PACKET,
    CURR.RES,  NEXT.RES,  QUE.LENGTH,   RES.UTILIZATION, RES.UTI,
PREV_RES.UTILIZATION,
    AR.QUE, PREV.AR_Q,   TRAFFIC.UTILIZATION,  SUCCESS.R  AND   RES.SLOTS
   AS INTEGER VARIABLES

   DEFINE
    LOSS,    SIMUL.LENGTH, REFERENCE,  COLLISION.TIME, GAMMA,
    TALKSPURT.TIME, TX.TIME,   ARRIVAL.TIME, RECOVERY.TIME,
    TIME.DIFF,  WAST,   SLOTS,   SLOT.NO,
    MAXDELAY,   INTERARRIVAL.TIME  AND  MEAN.INTERARRIVAL.TIME
   AS REAL VARIABLES
```

```
DEFINE
  FRAME.NO,  S.FLAG,   STATUS,  BUFF.SLOT, PE.PROB,  FOUND,  RESER.MOB,
  NEXT.LIMIT, THIRD.LIMIT, FIRST.TIME, TEMP.J,  RESER.SLOT, ID,   WARM,
  LAST.PACKET, R.SLOT,   R.FLAG,  DEL.MOB,  FR.DEL,  FRAME.DE, D, R, COL.LOSS,
  SLOT.LOSS,  OWN.LOSS,  MOB.LOSS,  TALKSEED,  SILSEED,  MAX.SLOT,
  TNDOS,     TNAS,  RASNUM,   ALL.USED,  TNPKTS,   NUM.LOST.PKTS,
  CUR.SLOT.NO,   ST_TYPE, NUM.MESS,  AV.PACKET, DATASEED,  VIDEO.SLOTS,
  V.SLOTS.NO,  V.SLOT.VALUE, V.SLOT.COUNT, VIDEO.SLOTS.NO, V.AMOUNT.SLOTS,
  VIDEO.SLOT.VALUE, VIDEO.NUM.DROPOUTS, V.NUM.DROPOUTS, VIDEO.SLOT.COUNT,
VIDEO.FRAME.DROPOUT,
  PREVIOUS.V_TNPKTS,  PREVIOUS.PK.COUNT  AND    NUM.SLOT
 AS INTEGER VARIABLES

" VIDEO.SLOTS : # OF VIDEO SLOTS GOING TO BE ASSIGNED
" V.SLOT.NO : # OF VIDEO SLOTS ASSIGNED
" V.SLOT.VALUE : VALUE OF THE SECOND VIDEO SLOT
" V.SLOT.COUNT : INDICATE WHICH PACKET NO IN EACH FRAME THATS BEING TRANSMITTED
WHEN VIDEO
" V.AMOUNT.SLOTS : INDICATES TO THE STATION SAYING HOW MANY PACKETS ARE
TRANSMITTED WHEN VIDEO


 DEFINE
  FRAME.START.FLAG, RA.CTS.FLAG,
  SID.TXED.FLAG, NUM.DROPOUTS AND RA.ASSIGN.FLAG
 AS INTEGER VARIABLES


 DEFINE .LOST      TO MEAN 1
 DEFINE .STARTTX    TO MEAN 2
 DEFINE .ENDPREAMBLE TO MEAN 3
 DEFINE .BUSY      TO MEAN 4
 DEFINE .IDLE      TO MEAN 5
 DEFINE .RETX      TO MEAN 6
 DEFINE .NOCOLLISION TO MEAN 7
 DEFINE .START     TO MEAN 8
 DEFINE .STOP      TO MEAN 9
 DEFINE .OK       TO MEAN 10
 DEFINE .RELEASE    TO MEAN 11

 DEFINE .WINNER    TO MEAN 12 "RESULT OF GAINING A RESOURCE
 DEFINE .LOSER     TO MEAN 13 "RESULT OF FAILING TO GAIN A RESOURCE
 DEFINE .SET      TO MEAN 14 "FLAG VALUE
 DEFINE .NOTSET    TO MEAN 15 "DITTO
 DEFINE .SIDTX     TO MEAN 16
 DEFINE .ERROR     TO MEAN 17
 DEFINE .DATATX    TO MEAN 18
 DEFINE .LASTPKT    TO MEAN 19
 DEFINE .INQUE     TO MEAN 20
 DEFINE .INQTX     TO MEAN 21
 DEFINE .RELEASEANDINQ TO MEAN 22

 DEFINE .VOICE     TO MEAN 23
 DEFINE .DATA      TO MEAN 24
 DEFINE .VIDEO     TO MEAN 25

"  DEFINE .LSIDTX    TO MEAN 26


END "PREAMBLE


MAIN

 LET HOURS.V = 1000
```

```
 LET MINUTES.V = 1000
 LET TIME1 = 1.0

 CALL READTALK
 CALL INITIALIZE

"  CALL MESSAGEBOX.R("PRESS BUTTON TO START", "COMMENCE")

 START SIMULATION

 CALL MESSAGEBOX.R("PRESS BUTTON TO END", "FINISH")
 CALL OUTPUT
 " CALL INSTA



END "MAIN











ROUTINE BASE
 GIVEN MOBILE AND ACTION
 YIELDING TR.STATUS

DEFINE
 FREE.TR.SLOT,
 TR.STATUS,
 MOBILE,
 WINNER.MOB,
 VINI,
 T.VINI,
 TEMP,
 ACTION,HH
AS INTEGER VARIABLES

 RESERVE CAPTURE AS 100
 DEFINE LARGEST AS REAL VARIABLE
 DEFINE SECOND AS REAL VARIABLE

 SELECT CASE ACTION

 CASE .SIDTX
 'TRY_AGAIN'
     IF TYPE(MOBILE) = .VIDEO
        ADD 1 TO VID.COUNT
     ALWAYS

     IF SID.TXED.FLAG(MOBILE) EQ .NOTSET
       LET SID.TXED.FLAG(MOBILE) = .SET
        ADD 1 TO COUNT

      CAPTURE(COUNT) = STRENTH.SIG(MOBILE)
      "PRINT 1 LINE WITH COUNT AND CAPTURE(COUNT) THUS
      "CAPTURE(***)=***.***

       FILE THIS MOBILE IN TRANSMISSIONQ
       WAIT (0.5*SLOTS*1000) .MILLISECONDS  " WAIT UNTIL START OF NEXT SLOT
                  "    PRINT 1 LINE WITH N.TRANSMISSIONQ THUS
                  "N.TXQ IS **
```

```
        LARGEST = CAPTURE(1)
        SECOND=0
"PRINT 1 LINE WITH LARGEST AND SECOND THUS
"LARGEST=***.*** SECOND=***.***
        FOR HH=1 TO COUNT
        DO
        IF CAPTURE(HH) GT LARGEST
        LARGEST = CAPTURE(HH)
        ALWAYS
        LOOP

FOR HH=1 TO COUNT
DO
IF CAPTURE(HH) GT SECOND
IF CAPTURE(HH) LT LARGEST
SECOND=CAPTURE(HH)
ALWAYS
ALWAYS
LOOP

       "PRINT 1 LINE WITH LARGEST AND SECOND THUS
       "LARGEST = ***.**** SECOND=***.****


       IF COUNT EQ 1
          ADD 1 TO TEMP.COUNT
          REMOVE THIS MOBILE FROM TRANSMISSIONQ
          RESER.MOB(MOBILE) = NUM.SLOT + 1
          LET ARRAY_COUNT(NUM.SLOT + 1) = R.FLAG(MOBILE)
          TR.STATUS = .INQUE
          FILE THIS MOBILE IN WAITINGQ
          ADD 1 TO Q.COUNTER
"          PRINT 1 LINE WITH TIME.V AND MEANQLENGTH THUS
"          TIME *****.***** Q.LENGTH ****.***
          ADD 1 TO AR.QUE
          ADD 1 TO RES.UTILIZATION
          COUNT = 0
          TEMP.COUNT = 0

          IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
           IF TYPE(MOBILE) = .VOICE
              LET CON_ROUND.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
           ALWAYS

           IF TYPE(MOBILE) = .DATA
              LET CON_DATA.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
           ALWAYS

           IF TYPE(MOBILE) = .VIDEO
              LET CON_VIDEO.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
           ALWAYS
          ALWAYS
      ALWAYS

      IF COUNT GT 1
           ADD 1 TO TEMP.COUNT
           IF COUNT = TEMP.COUNT
            LET COUNT = 0
            LET TEMP.COUNT = 0
           ALWAYS


         REMOVE THIS MOBILE FROM TRANSMISSIONQ

IF STRENTH.SIG(MOBILE) EQ LARGEST AND LARGEST-SECOND GT THRESHOLD
"PRINT 1 LINE WITH THRESHOLD THUS
"THRESHOLD = ***.***
```

```
  RESER.MOB(MOBILE) = NUM.SLOT + 1
  LET ARRAY_COUNT(NUM.SLOT + 1) = R.FLAG(MOBILE)
  TR.STATUS = .INQUE
  FILE THIS MOBILE IN WAITINGQ
  ADD 1 TO Q.COUNTER
  "PRINT 1 LINE WITH TIME.V AND MEANQLENGTH THUS
  "TIME *****.***** Q.LENGTH ****.***
  ADD 1 TO AR.QUE
  "ADD 1 TO RES.UTILIZATION
  "COUNT = 0
  "TEMP.COUNT = 0

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)

  IF TYPE(MOBILE) = .VOICE
  LET CON_ROUND.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
  ALWAYS

  IF TYPE(MOBILE) = .DATA
  LET CON_DATA.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
  ALWAYS

  IF TYPE(MOBILE) = .VIDEO
  LET CON_VIDEO.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
  ALWAYS

  ALWAYS

 ELSE

          RESER.MOB(MOBILE) = RESERVATION(RANDI.F(1,RASNUM,2))
          IF RESER.MOB(MOBILE) GT (RSL.NO + 3)
             IF TYPE(MOBILE) = .DATA
               TR.STATUS = .LOSER
             ALWAYS

             IF (TYPE(MOBILE) = .VOICE) OR (TYPE(MOBILE) = .VIDEO)
               LET SID.TXED.FLAG(MOBILE) = .NOTSET
"                PRINT 1 LINE WITH RESER.MOB(MOBILE) AND RSL.NO THUS
"              << MAY BE WINNER>> MOBILE(RES) ** RSL.NO **
               WAIT (SLOTS*(RESER.MOB(MOBILE) + 0.5 - RSL.NO -1)*1000).MILLISECONDS
               GO TO 'TRY_AGAIN'
             ALWAYS
          ALWAYS
          IF RESER.MOB(MOBILE) LE (RSL.NO + 3)
"              PRINT 1 LINE WITH RESER.MOB(MOBILE) AND RSL.NO THUS
"              <<LOSER>> MOBILE(RES) ** RSL.NO **
             TR.STATUS = .LOSER
          ALWAYS
ALWAYS
      ALWAYS

      IF COUNT LT 0
          WRITE AS "BASE ERROR - MOBILE SHOULD HAVE NOT ENTERED THIS STAGE",/
      ALWAYS
    ALWAYS

 CASE .LASTPKT

  ADD 1 TO TRAFFIC.UTILIZATION
  LET TR.STATUS = .RELEASE
  LET RSLOT.FLG(RESER.MOB(MOBILE)) =.NOTSET
  LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
  LET ARRAY_COUNT(RESER.MOB(MOBILE)) = 0
  LET RESER.MOB(MOBILE) = 0
  IF (TYPE(MOBILE) EQ .VIDEO) AND (V.SLOTS.NO(MOBILE) GT 1)
    ADD 1 TO TRAFFIC.UTILIZATION
```

```
    LET RSLOT.FLG(V.SLOT.VALUE(MOBILE)) = .NOTSET
    LET ARRAY_COUNT(V.SLOT.VALUE(MOBILE)) = 0
    LET PACKET.NO_ARRAY(V.SLOT.VALUE(MOBILE)) = PACKET.NO(MOBILE)
    LET V.SLOT.VALUE(MOBILE) = 0
   ALWAYS

  CASE .DATATX
   LET TR.STATUS = .OK
   ADD 1 TO TRAFFIC.UTILIZATION
   IF (TYPE(MOBILE) EQ .VIDEO) AND (V.SLOTS.NO(MOBILE) GT 1)
    ADD 1 TO TRAFFIC.UTILIZATION
   ALWAYS

  CASE .INQTX

   FOR FREE.TR.SLOT = 1 TO NUM.SLOT
   WITH RSLOT.FLG(FREE.TR.SLOT) = .NOTSET, FIND THE FIRST CASE
   IF NONE
    TR.STATUS = .INQUE
    LET RESER.MOB(MOBILE) = NUM.SLOT + 1
    LET ARRAY_COUNT(NUM.SLOT + 1) = R.FLAG(MOBILE)
   ELSE
    REMOVE THE FIRST WINNER.MOB FROM WAITINGQ
    IF (ARRIVAL.TIME(MOBILE) = ARRIVAL.TIME(WINNER.MOB)) AND (R.FLAG(MOBILE) =
R.FLAG(WINNER.MOB))
     IF LAST.PK_ARRAY(R.FLAG(MOBILE)) = .SET
       LET LAST.PK_ARRAY(R.FLAG(MOBILE)) = .NOTSET
       LET LT.PACKET(MOBILE) = .NOTSET
       TEMP = RESER.MOB(MOBILE)
       LET RESER.MOB(MOBILE) = FREE.TR.SLOT
       LET RSLOT.FLG(FREE.TR.SLOT) = .SET
       LET V.SLOTS.NO(MOBILE) = 1
"       LET V.SLOT.COUNT(MOBILE) = 1
       SUBTRACT 1 FROM Q.COUNTER
       ADD 1 TO TRAFFIC.UTILIZATION
       ARRAY_COUNT(FREE.TR.SLOT) = R.FLAG(MOBILE)
       LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
       IF FREE.TR.SLOT GE TEMP
        WAIT ((FREE.TR.SLOT - TEMP - 1)*SLOTS*1000) .MILLISECONDS
       ELSE
        WAIT ((NUM.SLOT - TEMP - 1 + FREE.TR.SLOT)*SLOTS*1000) .MILLISECONDS
       ALWAYS
       LET RSLOT.FLG(FREE.TR.SLOT) = .NOTSET
       LET  RESER.MOB(MOBILE) = 99
       LET ARRAY_COUNT(FREE.TR.SLOT) = 0

     ELSE
       LET RESER.MOB(MOBILE) = FREE.TR.SLOT
       LET RSLOT.FLG(FREE.TR.SLOT) = .SET
       SUBTRACT 1 FROM Q.COUNTER
       ADD 1 TO TRAFFIC.UTILIZATION
       ARRAY_COUNT(FREE.TR.SLOT) = R.FLAG(MOBILE)
       LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
       IF (TYPE(MOBILE) = .VIDEO) AND (VIDEO.SLOTS GT 1)
        FOR FREE.TR.SLOT = 1 TO NUM.SLOT
         WITH RSLOT.FLG(FREE.TR.SLOT) = .NOTSET, FIND THE FIRST CASE
        IF NONE
         V.SLOTS.NO(MOBILE) = VIDEO.SLOTS - 1
        ELSE
         V.SLOTS.NO(MOBILE) = 2
         ADD 1 TO TRAFFIC.UTILIZATION
         LET RSLOT.FLG(FREE.TR.SLOT) = .SET
         V.SLOT.VALUE(MOBILE) = RESER.MOB(MOBILE)
         RESER.MOB(MOBILE) = FREE.TR.SLOT
         ARRAY_COUNT(FREE.TR.SLOT) = R.FLAG(MOBILE)
         LET PACKET.NO(MOBILE) = INT.F(PACKET.NO(MOBILE)/2)
         LET PACKET.NO_ARRAY(V.SLOT.VALUE(MOBILE)) = PACKET.NO(MOBILE)
```

```
          LET PACKET.NO_ARRAY(FREE.TR.SLOT) = PACKET.NO(MOBILE)
"           V.SLOT.COUNT(MOBILE) = 1
        ALWAYS
       ELSE
        V.SLOTS.NO(MOBILE) = 1
       ALWAYS
     ALWAYS
     TR.STATUS = .WINNER
    ELSE
     FILE WINNER.MOB IN WAITINGQ
     TR.STATUS = .INQUE
     LET RESER.MOB(MOBILE) = NUM.SLOT + 1
     LET ARRAY_COUNT(NUM.SLOT + 1) = R.FLAG(MOBILE)
    ALWAYS
   ALWAYS

  CASE .RELEASEANDINQ

   LET T.VINI = 0
   FOR VINI = 1 TO NUM.SLOT
   DO
   IF RSLOT.FLG(VINI) = .NOTSET
     ADD 1 TO T.VINI
   ALWAYS
   LOOP

"  PRINT 1 LINE WITH TIME.V, Q.COUNTER AND T.VINI THUS
"  <RELEASEANDINQ> TIME : *****.***** Q.LENGTH : ***** & NO.OF FREE SLOTS: ******

   FOR FREE.TR.SLOT = 1 TO NUM.SLOT
   WITH RSLOT.FLG(FREE.TR.SLOT) = .NOTSET, FIND THE FIRST CASE
   IF NONE
    REMOVE THIS MOBILE FROM WAITINGQ
    SUBTRACT 1 FROM Q.COUNTER
    TR.STATUS = .LOSER
"    PRINT 1 LINE WITH TIME.V, R.FLAG(MOBILE) AND Q.COUNTER  THUS
"    <NO RES SLOT AVA> TIME; ******.******* ID : **   Q.COUNTER: ****
    LET RESER.MOB(MOBILE) = 0
   ELSE
    REMOVE THE FIRST WINNER.MOB FROM WAITINGQ
    IF (ARRIVAL.TIME(MOBILE) = ARRIVAL.TIME(WINNER.MOB)) AND (R.FLAG(MOBILE) =
R.FLAG(WINNER.MOB))
     TEMP = RESER.MOB(MOBILE)
     LET RESER.MOB(MOBILE) = FREE.TR.SLOT
     LET RSLOT.FLG(FREE.TR.SLOT) = .SET
     SUBTRACT 1 FROM Q.COUNTER
     ADD 1 TO TRAFFIC.UTILIZATION
     LET V.SLOTS.NO(MOBILE) = 1
     LET ARRAY_COUNT(FREE.TR.SLOT) = R.FLAG(MOBILE)
     LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
     IF FREE.TR.SLOT GE TEMP
      WAIT ((FREE.TR.SLOT - TEMP - 1)*SLOTS*1000) .MILLISECONDS
     ELSE
      WAIT ((NUM.SLOT - TEMP - 1 + FREE.TR.SLOT)*SLOTS*1000) .MILLISECONDS
     ALWAYS
     LET RSLOT.FLG(RESER.MOB(MOBILE)) =.NOTSET
     TR.STATUS = .WINNER
     LET RESER.MOB(MOBILE) = 99
     LET ARRAY_COUNT(FREE.TR.SLOT) = 0
"     PRINT 1 LINE WITH TIME.V, R.FLAG(MOBILE) AND Q.COUNTER  THUS
"     <WINNER> TIME; ******.******* ID : **   Q.COUNTER: ****
    ELSE
     REMOVE THIS MOBILE FROM WAITINGQ
     FILE WINNER.MOB IN WAITINGQ
     SUBTRACT 1 FROM Q.COUNTER
     TR.STATUS = .LOSER
"     PRINT 1 LINE WITH TIME.V, R.FLAG(MOBILE) AND Q.COUNTER THUS
```

```
"     <LOSER> TIME; ******.******* ID : **   Q.COUNTER: ****
      LET RESER.MOB(MOBILE) = 0
    ALWAYS
   ALWAYS

  DEFAULT
   WRITE AS "ERROR - UNKNOWN FLAG PASSED TO BASE",/
  ENDSELECT


'END.BASE'
 RETURN
END "BASE



ROUTINE INITIALIZE

DEFINE I, S, TEMP.VALUE, VAL AND L AS INTEGER VARIABLES
RESERVE RSLOT.FLG AS 100
RESERVE RESERVATION AS 100
RESERVE A.TIME_ARRAY AS 400
RESERVE TEMP_ARRAY AS 100
RESERVE DATA_PER.BLOCK AS 300
RESERVE LOS, PAC.TOT AND IDEN AS 100
RESERVE DATAB, FRAME, LAST.PK_ARRAY, PACKET_ARRAY, R.SLOT_ARRAY AND
PACKET.COUNTER AS 300
RESERVE DATA.START.TIME, FRAME.START.TIME, PK.COUNT, PACKET.NO_ARRAY AND
ACCESS.COUNTER AS 300
RESERVE ARRAY_COUNT AS 300
RESERVE  V.NUM.DROPOUTS_ARRAY, V.SLOT.VALUE_ARRAY, V.SLOTS.NO_ARRAY AND
V.SLOT.COUNT_ARRAY AS 300

LET RA.CTS.FLAG     = .NOTSET
LET RASNUM      = RES.SLOTS     " NUMBER OF RESERVATION SLOTS

LET MAXDELAY      = 0.032

LET HEY          = 0
LET VID.COUNT    = 0
LET TNDOS        = 0
LET TNAS         = 0
LET D_TNDOS      = 0
LET D_TNAS       = 0
LET ALL.USED     = 0
LET TNPKTS       = 0
LET NUM.LOST.PKTS = 0
LET AV.PACKET    = 0

LET VIDEO.FRAME.DROPOUT = 0
LET V_TNPKTS     = 0
LET V_TNDOS      = 0
LET V_TNAS       = 0
LET V_SUCCESS    = 0
LET NUM.LOST.PKTS = 0

LET STOTAL            = 0
LET TOTAL             = 0
LET CONT              = 0
LET SUCCESS           = 0
LET D_SUCCESS         = 0
LET TOXPRT            = 0
LET SIL               = 0
LET NUM.OF.TOXPRT     = 0
LET NUM.SIL           = 0
LET MEAN.TOXPRT       = 0
LET MEAN.SIL          = 0
```

```
LET FRAME.NO          = 0
LET OWN.LOSS          = 0
LET COUNT           = 0
LET TEMP.COUNT         = 0
LET Q.COUNTER         = 0
LET RES.UTILIZATION    = 0
LET PREV_RES.UTILIZATION = 0
LET TRAFFIC.UTILIZATION = 0
LET RES.UTI           = .NOTSET
LET BITS          = FRAME.LENGTH*CODER
LET NUM.SLOT          = BIT.RATE/((BITS+OVER.HEAD.BITS)*FRAMES.PER.SEC)
LET SLOTS          = FRAME.LENGTH/NUM.SLOT
LET REFERENCE         = FRAME.LENGTH - SLOTS
LET VIDEO.BLOCKED      = .NOTSET
LET FREE.TRAFF.SLOT.INDICATOR = .NOTSET

PRINT 5 LINES WITH BITS, NUM.SLOT, SLOTS, DATA.BLOCK, AND BITS THUS
BITS     : ****
NUM.SLOT  : **
SLOT SIZE :**.*********MS
DATA.BLOCK:*******
PACKET SIZE: *****


FOR I = 1 TO 300
DO
 FRAME(I) = .NOTSET
 DATAB(I)  = .NOTSET
 DATA_PER.BLOCK(I) = 0
LOOP

FOR I = 1 TO NUM.SLOT
DO
 RSLOT.FLG(I) = .NOTSET
LOOP

IF RASNUM = 1
  RESERVATION(1) = 1
  RSLOT.FLG(1) = .SET
  WRITE AS " OOPS ONLY ONE RESERVATION SLOT IS USED",/
  GO TO 'CONT'
ALWAYS

LET TEMP.VALUE = INT.F(RASNUM/2)

FOR I = 1 TO TEMP.VALUE
DO
  RESERVATION(I) = I
  RSLOT.FLG(I) = .SET
LOOP

LET I = TEMP.VALUE + 1

WHILE I LE RASNUM
DO
  'RE_CALL'
  VAL = RANDI.F(INT.F(NUM.SLOT/2),NUM.SLOT,2)
  FOR L = (TEMP.VALUE + 1) TO I
  WITH RESERVATION(L) = VAL, FIND THE FIRST CASE
  IF NONE
    RESERVATION(I) = VAL
    RSLOT.FLG(RESERVATION(I)) = .SET
    ADD 1 TO I
  ELSE
    GO TO 'RE_CALL'
  ALWAYS
LOOP
```

```
 FOR I = (TEMP.VALUE + 1) TO RASNUM
 DO
   'NEXT'
   FOR L = (I + 1) TO RASNUM
   WITH RESERVATION(L) LT RESERVATION(I), FIND THE FIRST CASE
   IF NONE
   ELSE
     LET S = RESERVATION(I)
     LET RESERVATION(I) = RESERVATION(L)
     LET RESERVATION(L) = S
     GO TO 'NEXT'
   ALWAYS
 LOOP

 "LET RSLOT.FLG(NUM.SLOT) = .SET
 "LET RSLOT.FLG(NUM.SLOT - 1) = .SET


 FOR I = 1 TO RASNUM
 DO
   PRINT 1 LINE WITH I AND RESERVATION(I) THUS
   RESERVATION SLOT ** IS SLOT.NO: ***
 LOOP

 'CONT'

 "CREATE EVERY UNIT(NUM.MOBILES)

 PRINT 1 LINE WITH NUM.MOBILES THUS
 NUM.MOBILES  : **

 FOR I = 1 TO NUM.MOBILES
 DO
  ACTIVATE A STATION NOW
  LET ID(STATION) = I

  IF I GT (VOICE.MOB + VIDEO.MOB)
    ST_TYPE(STATION) = .DATA
  ALWAYS

  IF (I GT VOICE.MOB) AND (I LE (VOICE.MOB + VIDEO.MOB))
    ST_TYPE(STATION) = .VIDEO
  ALWAYS

  IF I LE VOICE.MOB
    ST_TYPE(STATION) = .VOICE
  ALWAYS

 LOOP

 CURR.RES = RESERVATION(1)
 NEXT.RES = RESERVATION(2)

 ACTIVATE A TIMER NOW
 ACTIVATE A TEMPPRO NOW

 END " INITIALIZE



 ROUTINE MICROCE YIELDING CARRIER


      DEFINE DISTANCE AS INTEGER VARIABLES
```

```
DEFINE POWER.TRANS,
BLOCK.CHECK, CARRIER  AS REAL VARIABLES

DEFINE POWER.FFREE,
POWER.OOPEN,
POWER.SSUBURBAN,
POWER.PPHILADELPHIA,
POWER.NNEWARK  AS REAL VARIABLES

DEFINE POWER.LOS.DISTANCE,
POWER.FREE.DISTANCE,
POWER.OPEN.DISTANCE,
POWER.SUBURBAN.DISTANCE,
POWER.PHILADELPHIA.DISTANCE,
POWER.NEWARK.DISTANCE,
INTER.FREE.DISTANCE,
INTER.OPEN.DISTANCE,
INTER.SUBURBAN.DISTANCE,
INTER.PHILADELPHIA.DISTANCE,
INTER.NEWARK.DISTANCE AS REAL VARIABLES

DISTANCE = UNIFORM.F(10,RADIUM.CELL,2)


LET POWER.FFREE=0
LET POWER.OOPEN=0
LET POWER.SSUBURBAN=0
LET POWER.PPHILADELPHIA=0
LET POWER.NNEWARK=0
LET POWER.FREE.DISTANCE = 0
LET POWER.OPEN.DISTANCE = 0
LET POWER.SUBURBAN.DISTANCE = 0
LET POWER.PHILADELPHIA.DISTANCE = 0
LET POWER.NEWARK.DISTANCE = 0


LET POWER.TRANS = 10*LOG.10.F(1000*POWER.TRANSMITTER)
LET DISTANCE.CELL = ((3*CLUSTER.SIZE)**0.5)*RADIUM.CELL
LET A0 = 20*LOG.10.F(ANTENNA.BASE) + 10*LOG.10.F(POWER.TRANSMITTER) + GAIN.BASE +
GAIN.MOBILE + 10*LOG.10.F(ANTENNA.MOBILE) -64
LET POWER.FFREE = -45 - 20*LOG.10.F(DISTANCE.CELL/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
LET POWER.OOPEN = -49 - 43.5*LOG.10.F(DISTANCE.CELL/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
LET POWER.SSUBURBAN = -61.7 - 38.4*LOG.10.F(DISTANCE.CELL/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
LET POWER.PPHILADELPHIA = -70 - 36.8*LOG.10.F(DISTANCE.CELL/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
LET POWER.NNEWARK = -64 - 43.1*LOG.10.F(DISTANCE.CELL/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0

IF BLOCK.LENGTH <= 0.3
  BLOCK.CHECK = 0
ELSE
 IF BLOCK.LENGTH < 8.3
   BLOCK.CHECK = 1 + 0.5*LOG.10.F(BLOCK.LENGTH/3.3)
 ELSE
    IF BLOCK.LENGTH < 200
      BLOCK.CHECK = 1.2 + 12.5*LOG.10.F(BLOCK.LENGTH/8.3)
    ELSE
         IF BLOCK.LENGTH < 1000
          BLOCK.CHECK = 17.95 + 3*LOG.10.F(BLOCK.LENGTH/200)
         ELSE
            BLOCK.CHECK = 20
         ALWAYS
    ALWAYS
 ALWAYS
```

```
      ALWAYS

   IF RADIUM.CELL < 1600

        IF  DISTANCE < 67
          LET POWER.LOS.DISTANCE = POWER.TRANS - 77 - 21.5*LOG.10.F(DISTANCE/33) +
30*LOG.10.F(ANTENNA.BASE/20)
        ELSE
          IF DISTANCE < 333
            LET POWER.LOS.DISTANCE = POWER.TRANS - 83.5 - 14*LOG.10.F(DISTANCE/67) +
30*LOG.10.F(ANTENNA.BASE/20)
           ELSE
            LET POWER.LOS.DISTANCE = POWER.TRANS - 93.3 - 36.5*LOG.10.F(DISTANCE/333) +
30*LOG.10.F(ANTENNA.BASE/20)
           ALWAYS
        ALWAYS



      LET INTER.FREE.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK - POWER.FFREE - 7.78
      LET INTER.OPEN.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK - POWER.OOPEN -
7.78
      LET INTER.SUBURBAN.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.SSUBURBAN - 7.78
      LET INTER.PHILADELPHIA.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.PPHILADELPHIA - 7.78
      LET INTER.NEWARK.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.NNEWARK - 7.78


   ELSE
      IF DISTANCE <=1600

          IF  DISTANCE < 67
            LET POWER.LOS.DISTANCE = POWER.TRANS - 77 - 21.5*LOG.10.F(DISTANCE/33) +
30*LOG.10.F(ANTENNA.BASE/20)
          ELSE
            IF DISTANCE < 333
              LET POWER.LOS.DISTANCE = POWER.TRANS - 83.5 - 14*LOG.10.F(DISTANCE/67) +
30*LOG.10.F(ANTENNA.BASE/20)
             ELSE
              LET POWER.LOS.DISTANCE = POWER.TRANS - 93.3 - 36.5*LOG.10.F(DISTANCE/333) +
30*LOG.10.F(ANTENNA.BASE/20)
            ALWAYS
           ALWAYS


        LET INTER.FREE.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK - POWER.FFREE -
7.78
        LET INTER.OPEN.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK - POWER.OOPEN -
7.78
        LET INTER.SUBURBAN.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.SSUBURBAN - 7.78
        LET INTER.PHILADELPHIA.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.PPHILADELPHIA - 7.78
        LET INTER.NEWARK.DISTANCE = POWER.LOS.DISTANCE - BLOCK.CHECK -
POWER.NNEWARK - 7.78



      ELSE

         LET POWER.FREE.DISTANCE = -45 - 20*LOG.10.F(DISTANCE/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
         LET POWER.OPEN.DISTANCE = -49 - 43.5*LOG.10.F(DISTANCE/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
```

```
        LET POWER.SUBURBAN.DISTANCE = -61.57 - 38.4*LOG.10.F(DISTANCE/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
        LET POWER.PHILADELPHIA.DISTANCE = -70 - 36.8*LOG.10.F(DISTANCE/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0
        LET POWER.NEWARK.DISTANCE = -64 - 43.1*LOG.10.F(DISTANCE/1600) -
N.MICROCELL*LOG.10.F(FREQUENCY/900) + A0

        LET INTER.FREE.DISTANCE = POWER.FREE.DISTANCE  - POWER.FFREE - 7.78
        LET INTER.OPEN.DISTANCE = POWER.OPEN.DISTANCE  - POWER.OOPEN - 7.78
        LET INTER.SUBURBAN.DISTANCE = POWER.SUBURBAN.DISTANCE -POWER.SSUBURBAN
- 7.78
        LET INTER.PHILADELPHIA.DISTANCE = POWER.PHILADELPHIA.DISTANCE  -
POWER.PPHILADELPHIA -7.78
        LET INTER.NEWARK.DISTANCE = POWER.NEWARK.DISTANCE  - POWER.NNEWARK - 7.78

      ALWAYS

    ALWAYS

    IF INTER.SWITCH=11
    CARRIER=INTER.FREE.DISTANCE
    ALWAYS

    IF INTER.SWITCH=21
    CARRIER=INTER.OPEN.DISTANCE
    ALWAYS
    IF INTER.SWITCH=31
    CARRIER=INTER.SUBURBAN.DISTANCE
    ALWAYS
"PRINT 1 LINE WITH DISTANCE AND CARRIER THUS
"======****    ****.**** ====
END


PROCESS MOBILE

DEFINE
 TR.STATUS,
 ACTION,
 L
 AS A INTEGER VARIABLES

 IF RESER.MOB(MOBILE) = 0
  LET ARRIVAL.TIME(MOBILE) = TIME.V
 ALWAYS

 LET LT.PACKET(MOBILE) = .NOTSET

 IF RESER.MOB(MOBILE) = 0
  GO TO 'GET_RES'
 ALWAYS

 'STPOINT'

 FOR L = 1 TO RASNUM
 WITH RESERVATION(L) = RESER.MOB(MOBILE) , FIND THE FIRST CASE
 IF NONE
   LET SLOT.NUM = RESER.MOB(MOBILE)
   IF SLOT.NUM = (NUM.SLOT + 1)
    SLOT.NUM = NEXT.RES + 1
    RESER.MOB(MOBILE) = NEXT.RES
    LET ACTION = .INQTX
   ELSE
    IF (LAST.PACKET(STATION) EQ .SET)
     IF TYPE(MOBILE) = .DATA
      LET ACTION = .LASTPKT
```

```
      ALWAYS
      IF PACKET.NO(MOBILE) = 1
        LET ACTION = .LASTPKT
      ALWAYS
     ELSE
      LET ACTION = .DATATX
     ALWAYS
    ALWAYS
  ELSE "WAIT TILL WE CAN TRANSMIT THE SIDX
   'GET_RES'
   SLOT.NUM = RESERVATION(RANDI.F(1,RASNUM,2))
   RSLOT.FLG(SLOT.NUM) = .SET
   RESER.MOB(MOBILE) = SLOT.NUM
   IF LAST.PACKET(STATION) = .SET
    LET LAST.PK_ARRAY(R.FLAG(MOBILE)) = .SET
   ALWAYS
   LET ACTION = .SIDTX
  ALWAYS

'RETRY'
  IF (NUM.DROPOUTS(MOBILE) GE 1)
'RET'
   IF RESER.MOB(MOBILE) EQ (NUM.SLOT + 1)
     RESER.MOB(MOBILE) = NEXT.RES
     SLOT.NUM = NEXT.RES + 1
     IF ((TYPE(MOBILE) NE .DATA) AND (NUM.DROPOUTS(MOBILE) EQ (PACK.DROP.RATE - 1))
     AND (CURR.RES GT NEXT.RES)) AND ((LT.PACKET(MOBILE) = .SET) OR
     ((LAST.PK_ARRAY(R.FLAG(MOBILE)) = .SET) AND (STATUS(MOBILE) = .INQUE)))
       ACTION = .RELEASEANDINQ
"       FOR J = 1 TO NUM.SLOT
"       DO
"         PRINT 1 LINE WITH J, ARRAY_COUNT(J), PACKET.NO_ARRAY(J) AND  RSLOT.FLG(J) THUS
"         SLOT ** OCCUPIED BY MOB : *** AND HAS PACKETS : ****** LEFT/// **'
"       LOOP
     ELSE
       ACTION = .INQTX
     ALWAYS
   ELSE
     SLOT.NUM = RESER.MOB(MOBILE)
     RSLOT.FLG(SLOT.NUM) = .SET
     LET ACTION = .SIDTX
   ALWAYS
  ALWAYS

'GETTX'
  CALL SCHEDULER GIVING SLOT.NUM YIELDING TIME.DIFF

  WAIT TIME.DIFF .MILLISECONDS "FOR BEGINNING OF SLOT

  CALL BASE GIVING MOBILE AND ACTION YIELDING TR.STATUS
  LET STATUS(MOBILE) = TR.STATUS


  IF TYPE(MOBILE) = .VOICE

   SELECT CASE STATUS(MOBILE)

   CASE .WINNER


    IF (PACKET.NO(MOBILE) = 1) AND (RESER.MOB(MOBILE) NE 0)
     LET RSLOT.FLG(RESER.MOB(MOBILE)) =.NOTSET
     LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
     LET ARRAY_COUNT(RESER.MOB(MOBILE)) = 0
     LET RESER.MOB(MOBILE) = 0
    ELSE
     LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
```

```
ALWAYS

IF RESER.MOB(MOBILE) = 99
  RESER.MOB(MOBILE) = 0
 ALWAYS

IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
 ADD 1 TO TNAS
 ADD 1 TO SUCCESS
 LET ROUND.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
ALWAYS

CASE .LOSER, .RETX, .ERROR
ADD 1 TO NUM.DROPOUTS(MOBILE)
IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
 ADD 1 TO TNDOS
ALWAYS

IF (NUM.DROPOUTS(MOBILE) GE PACK.DROP.RATE)
 IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
  ADD 1 TO LOSS
  ADD 1 TO SLOT.LOSS
  ADD 1 TO MOB.LOSS(MOBILE)
 ALWAYS
 LET SID.TXED.FLAG(MOBILE) = .NOTSET
ELSE
 " WE ARE GOING TO GIVE IT ANOTHER BASH...
 LET SLOT.NUM        = RESER.MOB(MOBILE)
 LET SID.TXED.FLAG(MOBILE) = .NOTSET
 GO TO 'RETRY'
ALWAYS


CASE .INQUE

IF CURR.RES LT NEXT.RES
 GO TO 'RET'
ALWAYS

ADD 1 TO NUM.DROPOUTS(MOBILE)

IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
 ADD 1 TO TNDOS
ALWAYS

IF (NUM.DROPOUTS(MOBILE) GE PACK.DROP.RATE)
 IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
  ADD 1 TO LOSS
  ADD 1 TO SLOT.LOSS
  ADD 1 TO MOB.LOSS(MOBILE)
 ALWAYS
"   LET SID.TXED.FLAG(MOBILE) = .NOTSET
ELSE
 " WE ARE GOING TO GIVE IT ANOTHER BASH...
 GO TO 'RETRY'
ALWAYS

CASE .OK

LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
 ADD 1 TO SUCCESS
ALWAYS

CASE .RELEASE
IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
 ADD 1 TO SUCCESS
```

```
  ALWAYS

 DEFAULT

  PRINT 1 LINE WITH TR.STATUS THUS
  ERROR IN MOBILE - UNKNOWN RETURN STATUS ** FROM BASE, RETRYING
  LET SLOT.NUM = RASNUM
  LET SID.TXED.FLAG(MOBILE) = .NOTSET
  GO TO 'RETRY'

 ENDSELECT

 LET V.NUM.DROPOUTS(MOBILE) = 0
 LET R.SLOT_ARRAY(R.FLAG(MOBILE))  = RESER.MOB(MOBILE)
 LET PACKET_ARRAY(R.FLAG(MOBILE)) = PACKET.NO(MOBILE)
 LET FRAME.DE = DEL.MOB(MOBILE)
 LET PK.COUNT(R.FLAG(MOBILE)) = 0

 IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
  IF MOB.LOSS(MOBILE) GT 0
   ADD MOB.LOSS(MOBILE) TO NUM.LOST.PKTS
  ALWAYS
 ALWAYS

ALWAYS

IF TYPE(MOBILE) = .DATA

 SELECT CASE STATUS(MOBILE)


 CASE .WINNER

  IF (PACKET.NO(MOBILE) = 1) AND (RESER.MOB(MOBILE) NE 0)
   LET RSLOT.FLG(RESER.MOB(MOBILE)) =.NOTSET
   LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
   LET ARRAY_COUNT(RESER.MOB(MOBILE)) = 0
   LET RESER.MOB(MOBILE) = 0
  ELSE
   LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
  ALWAYS

  ADD 1 TO PACKET.COUNT(MOBILE)
  IF RESER.MOB(MOBILE) = 99
    RESER.MOB(MOBILE) = 0
  ALWAYS

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO D_TNAS
   ADD 1 TO D_SUCCESS
   LET DATA.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
  ALWAYS

 CASE .LOSER, .RETX, .ERROR

  ADD 1 TO NUM.DROPOUTS(MOBILE)

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO D_TNDOS
  ALWAYS

  LET SID.TXED.FLAG(MOBILE) = .NOTSET
  WAIT (FRAME.LENGTH*1000).MILLISECONDS

  GO TO 'RETRY'
```

```
 CASE .INQUE

  IF CURR.RES LT NEXT.RES
   ACTION = .INQTX
   GO TO 'RET'
  ALWAYS

  ADD 1 TO NUM.DROPOUTS(MOBILE)

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO D_TNDOS
  ALWAYS

"   LET SID.TXED.FLAG(MOBILE) = .NOTSET
  ACTION = .INQTX
  GO TO 'RETRY'

 CASE .OK
   LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
   ADD 1 TO PACKET.COUNT(MOBILE)
  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO D_SUCCESS
  ALWAYS

 CASE .RELEASE
  ADD 1 TO PACKET.COUNT(MOBILE)
  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO D_SUCCESS
  ALWAYS

 DEFAULT
  PRINT 1 LINE WITH TR.STATUS THUS
  ERROR IN MOBILE - UNKNOWN RETURN STATUS ** FROM BASE, RETRYING
  LET SLOT.NUM = RASNUM
  LET SID.TXED.FLAG(MOBILE) = .NOTSET
  GO TO 'RETRY'

 ENDSELECT

 LET V.NUM.DROPOUTS(MOBILE) = 0
 LET R.SLOT_ARRAY(R.FLAG(MOBILE))  = RESER.MOB(MOBILE)
 LET PACKET_ARRAY(R.FLAG(MOBILE)) = PACKET.NO(MOBILE)
 LET FRAME.DE = DEL.MOB(MOBILE)
 LET PK.COUNT(R.FLAG(MOBILE)) = PACKET.COUNT(MOBILE)

ALWAYS

IF TYPE(MOBILE) = .VIDEO

 SELECT CASE STATUS(MOBILE)

 CASE .WINNER


  IF (PACKET.NO(MOBILE) = 1) AND (RESER.MOB(MOBILE) NE 0)
   LET RSLOT.FLG(RESER.MOB(MOBILE)) =.NOTSET
   LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
   LET ARRAY_COUNT(RESER.MOB(MOBILE)) = 0
   LET RESER.MOB(MOBILE) = 0
   IF V.SLOTS.NO(MOBILE) GT 1
    LET RSLOT.FLG(V.SLOT.VALUE(MOBILE)) =.NOTSET
    LET PACKET.NO_ARRAY(V.SLOT.VALUE(MOBILE)) = PACKET.NO(MOBILE)
    LET ARRAY_COUNT(V.SLOT.VALUE(MOBILE)) = 0
    LET V.SLOT.VALUE(MOBILE) = 0
    V.SLOTS.NO(MOBILE) = 1
   ALWAYS
  ELSE
```

```
   LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)
  ALWAYS

   LET V.NUM.DROPOUTS(MOBILE) = NUM.DROPOUTS(MOBILE)

   IF V.SLOTS.NO(MOBILE) GT 1
    PACKET.NO_ARRAY(V.SLOT.VALUE(MOBILE)) = PACKET.NO(MOBILE)
   ALWAYS


"    IF V.SLOTS.NO(MOBILE) GT 1
"      IF (R.FLAG(MOBILE) = 35) AND (RESER.MOB(MOBILE) = 5) AND V.SLOT.VALUE(MOBILE) = 28
"         PRINT 1 LINE WITH V.SLOTS.NO(MOBILE), TIME.V,
RESER.MOB(MOBILE),V.SLOT.VALUE(MOBILE),
"         FRAME.NO AND PACKET.NO(MOBILE) THUS
"         <WINNER>VSN: ** TIME : ******.****** RES1: ** RES2: ** FRAME.NO: ******* PACKET.NO:
********
"       ALWAYS
"    ALWAYS


   IF RESER.MOB(MOBILE) = 99
     RESER.MOB(MOBILE) = 0
   ALWAYS

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   LET VIDEO.TIME = (TIME.V - ARRIVAL.TIME(MOBILE))*HOURS.V
   ADD 1 TO V_TNAS
   ADD 1 TO V_SUCCESS
  ALWAYS

 CASE .LOSER, .RETX, .ERROR
  ADD 1 TO NUM.DROPOUTS(MOBILE)

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO V_TNDOS
  ALWAYS

  IF (NUM.DROPOUTS(MOBILE) GE 20)
   LET RESER.MOB(MOBILE) = 0
   LET PACKET.NO(MOBILE) = 1
   LET V.NUM.DROPOUTS(MOBILE) = 0
   IF MOBILE IN WAITINGQ
    REMOVE THIS MOBILE FROM THE WAITINGQ
   ALWAYS
   IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
    ADD 1 TO VIDEO.FRAME.DROPOUT
    ADD 1 TO LOSS
    ADD 1 TO SLOT.LOSS
    ADD 1 TO MOB.LOSS(MOBILE)
   ALWAYS
   LET SID.TXED.FLAG(MOBILE) = .NOTSET
  ELSE
   " WE ARE GOING TO GIVE IT ANOTHER BASH...
   LET SLOT.NUM        = RESER.MOB(MOBILE)
   LET SID.TXED.FLAG(MOBILE) = .NOTSET
   GO TO 'RETRY'
  ALWAYS


 CASE .INQUE

  IF CURR.RES LT NEXT.RES
   GO TO 'RET'
  ALWAYS

  ADD 1 TO NUM.DROPOUTS(MOBILE)
```

```
  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO V_TNDOS
  ALWAYS

  IF (NUM.DROPOUTS(MOBILE) GE (20 - V.NUM.DROPOUTS(MOBILE)))
   LET RESER.MOB(MOBILE) = 0
   LET PACKET.NO(MOBILE) = 1
   LET V.NUM.DROPOUTS(MOBILE) = 0
   REMOVE THIS MOBILE FROM THE WAITINGQ
   IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
    ADD 1 TO VIDEO.FRAME.DROPOUT
    ADD 1 TO LOSS
    ADD 1 TO SLOT.LOSS
    ADD 1 TO MOB.LOSS(MOBILE)
   ALWAYS
"    LET SID.TXED.FLAG(MOBILE) = .NOTSET
  ELSE
   " WE ARE GOING TO GIVE IT ANOTHER BASH...
"    LET SID.TXED.FLAG(MOBILE) = .NOTSET
    GO TO 'RETRY'
  ALWAYS

 CASE .OK


  LET V.NUM.DROPOUTS(MOBILE) = 0
  LET PACKET.NO_ARRAY(RESER.MOB(MOBILE)) = PACKET.NO(MOBILE)

  IF V.SLOTS.NO(MOBILE) GT 1
   PACKET.NO_ARRAY(V.SLOT.VALUE(MOBILE)) = PACKET.NO(MOBILE)
  ALWAYS

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
    ADD 1 TO V_SUCCESS
  ALWAYS

 CASE .RELEASE

  LET NUM.DROPOUTS(MOBILE) = 0
  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO V_SUCCESS
  ALWAYS

 DEFAULT
  PRINT 1 LINE WITH TR.STATUS THUS
  ERROR IN MOBILE - UNKNOWN RETURN STATUS ** FROM BASE, RETRYING
  LET SLOT.NUM = RASNUM
  LET SID.TXED.FLAG(MOBILE) = .NOTSET
  GO TO 'RETRY'

 ENDSELECT

 LET R.SLOT_ARRAY(R.FLAG(MOBILE))  = RESER.MOB(MOBILE)
 LET PACKET_ARRAY(R.FLAG(MOBILE)) = PACKET.NO(MOBILE)
 LET FRAME.DE = DEL.MOB(MOBILE)
 LET PK.COUNT(R.FLAG(MOBILE)) = 0

 IF V.SLOTS.NO(MOBILE) GT 1
    V.AMOUNT.SLOTS = 2
 ELSE
    V.AMOUNT.SLOTS = 1
 ALWAYS

 IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
  IF MOB.LOSS(MOBILE) GT 0
```

```
      ADD MOB.LOSS(MOBILE) TO V_NUM.LOST.PKTS
      LET VIDEO.BLOCKED = .SET
     ALWAYS
    ALWAYS

   ALWAYS

   IF RESER.MOB(MOBILE) = (NUM.SLOT + 1)
     IF LAST.PK_ARRAY(R.FLAG(MOBILE)) = .SET
      REMOVE THIS MOBILE FROM WAITINGQ
      SUBTRACT 1 FROM Q.COUNTER
      GO TO 'GET.STATION'
     ALWAYS
     SUBTRACT 1 FROM PACKET.NO(MOBILE)
     LET NUM.DROPOUTS(MOBILE) = 0
     LET MOB.LOSS(MOBILE) = 0

     IF PACKET.NO(MOBILE) EQ 1
      LET LT.PACKET(MOBILE) = .SET
     ELSE
      LET LT.PACKET(MOBILE) = .NOTSET
     ALWAYS


     IF TIME.V GT SIMUL.LENGTH
      CALL OUTPUT
     ALWAYS

     GO TO 'STPOINT'
   ALWAYS

  IF RESER.MOB(MOBILE) NE (NUM.SLOT + 1)
"   LET V.SLOT.COUNT_ARRAY(R.FLAG(MOBILE)) = V.SLOT.COUNT(MOBILE)
   LET V.SLOTS.NO_ARRAY(R.FLAG(MOBILE)) = V.SLOTS.NO(MOBILE)
   LET V.SLOT.VALUE_ARRAY(R.FLAG(MOBILE)) = V.SLOT.VALUE(MOBILE)
   LET V.NUM.DROPOUTS_ARRAY(R.FLAG(MOBILE)) = V.NUM.DROPOUTS(MOBILE)
   LET A.TIME_ARRAY(R.FLAG(MOBILE)) = ARRIVAL.TIME(MOBILE)
'GET.STATION'
    IF LAST.PK_ARRAY(R.FLAG(MOBILE)) = .SET
     LET RESER.MOB(MOBILE) = 0
     LET R.SLOT_ARRAY(R.FLAG(MOBILE))  = RESER.MOB(MOBILE)
     LET LAST.PK_ARRAY(R.FLAG(MOBILE)) = .NOTSET
    ALWAYS
    REACTIVATE THE STATION CALLED MOB.STATION(MOBILE) NOW
  ALWAYS

END "MOBILE





ROUTINE OUTPUT


OPEN UNIT 3 FOR OUTPUT, NAME IS "THRES6.TXT"
USE UNIT 3 FOR OUTPUT

PRINT 69 LINES WITH
DATA.BLOCK,
NUM.SLOT,
RES.SLOTS,
PACK.DROP.RATE,
SIMUL.LENGTH,
```

```
TALKSEED,
SILSEED,
BIT.RATE,
BITS,
CODER,
NUM.MOBILES,
VOICE.MOB,
(NUM.MOBILES - VOICE.MOB - VIDEO.MOB),
VIDEO.MOB,
FRAME.NO,
((RES.UTILIZATION/(FRAME.NO*RES.SLOTS))*100),
(RES.UTILIZATION/FRAME.NO),
((TRAFFIC.UTILIZATION/(FRAME.NO*(NUM.SLOT - RES.SLOTS)))*100),
QUE.LENGTH/(FRAME.NO),
VIDEO.SLOTS,
TNDOS,
TNAS,
((TNDOS/TNAS)*100),
SUCCESS,
(SUCCESS+NUM.LOST.PKTS),
((TOXPRT/(TOXPRT+SIL))*100),
(TNAS/SIMUL.LENGTH),
NUM.LOST.PKTS,
((NUM.LOST.PKTS/(SUCCESS+NUM.LOST.PKTS))*100),
MEANCTC,
MEANCT,
(MEANCT - MEANCTC),
AV.PACKET,
D_TNDOS,
D_TNAS,
((D_TNDOS/D_TNAS)*100),
D_SUCCESS,
MEANDTC,
MEANDT,
MEANDT+FRAME.LENGTH*1000,
MEANDT+DATA.BLOCK*FRAME.LENGTH*1000,
(MEANDT - MEANDTC),
MEANTDD,
MEANDB,
V_TNDOS,
V_TNAS,
V_SUCCESS,
(V_SUCCESS+V_NUM.LOST.PKTS),
V_NUM.LOST.PKTS,
MEANVEC,
MEANVE,
(MEANVE - MEANVEC),
MEANFR
AS FOLLOWS

      ````````````````````````````````````````````````
      `       SCHEDULED ACCESS : CAPVOIDAT              `
      ````````````````````````````````````````````````
        DATA BLOCK : **
      ` NO OF SLOTS PER FRAME: **                `
      ` NO OS RES SLOT: **                  `
      ` NO OF TRIES BEFORE DROP : ***             `
      ````````````````````````````````````````````````
      ` SIMULATION LENGTH: **** SECS              `
      ` TALKSEED:      **                 `
      ` SILSEED :      **                 `
      ` BIT RATE:  ********** BITS/S            `
      ` PACKET SIZE:   ***** BITS             `
      ` CODER RATE    ****** BITS/SEC           `
      ` NUMBER OF MOBILES : *****               `
      ` NUMBER OF VOICE MOB : ****              `
```
```

```
        ' NUMBER OF DATA MOB  : ****                   '
        ' NUMBER OF VIDEO MOB : ****                   '
        ' TOTAL NUMBER OF FRAMES : *******             '
        ' RESERVATION UTILIZATION  :  ***.****%         '
        ' RESERVATION SLOTS PER FRAME *****.***         '
        ' TRAFFIC SLOT UTILIZATION :  ***.****%         '
        ' NO.IN WAITING.Q    :  ******.*******/FRAME '
        ' MAX.NO.OF VID.SLOTS PER VIDEO TERMINAL : *** '
        ''''''''''''''''''''''''''''''''''''''''''''''''
        '              VOICE                '
        ''''''''''''''''''''''''''''''''''''''''''''''''
        ' TOT VOICE DROPOUTS   *******              '
        ' NUM SUCCESS ASSGNTS(VOICE)  *******          '
        ' D.OUTS VS ASSGNTS(VOICE) **.****%           '
        ' SUCCESSFUL PACKETS(VOICE) ******            '
        ' TOTAL NUM PACKETS(VOICE)  ******            '
        ' VOX ACTIVITY        ***.****%              '
        ' ASSIGNMENTS/SEC   ****.***/S              '
        ' NUM OF LOST PACKETS(VOICE)***********        '
        ' AV.LOST PACKETS(VOICE)  ***.****%           '
        ' CONTENTION DELAY(VOICE) ***.*** MS          '
        ' AV. ACCESS DELAY(VOICE) ***.*** MS          '
        ' ALLOCATION DELAY(VOICE) ***.*** MS          '
        ''''''''''''''''''''''''''''''''''''''''''''''''
        '              DATA                 '
        ''''''''''''''''''''''''''''''''''''''''''''''''
        ' AV.DATA PACKET SIZE : *****               '
        ' TOT DATA DROPOUTS    ********             '
        ' NUM SUCCESS ASSGNTS  *******              '
        ' DROPOUTS VS SUCCESS  **.****%             '
        ' SUCCESSFUL PACKETS(DATA) ******            '
        ' CONTENTION DELAY(DATA) ***.*** MS          '
        ' AV. ACCESS DELAY(DATA) ***.*** MS
        ' AV.PACKET DELAY(DATA)***.*** MS            '
        ' AV.BLOCK DELAY(DATA)  ***.*** MS
        ' ALLOCATION DELAY(DATA) ***.*** MS          '
        ' TOTAL END-TO-END DELAY *********.**** MS    '
        ' END-TO-END DATA BURST DELAY ********.**** MS '
        ''''''''''''''''''''''''''''''''''''''''''''''''


SKIP 3 LINES
STOP
END "OUTPUT


ROUTINE READTALK

DEFINE METRA, AND INDEX AS INTEGER VARIABLES

  OPEN UNIT 11 FOR INPUT, NAME IS "TALKDATA"
  USE UNIT 11  FOR INPUT

  RESERVE NAME1(*) AS 60
  RESERVE NAME2(*) AS 60
  RESERVE NAME3(*) AS 60
  RESERVE NAME4(*) AS 60

  FOR METRA = 1 TO 15, READ NAME1(METRA) AND NAME2(METRA)
  FOR METRA = 16 TO 26, READ NAME1(METRA) AND NAME2(METRA)
  LET EOF.V = 1

  CLOSE UNIT 11

  OPEN UNIT 11 FOR INPUT, NAME IS "DATA6"
  USE UNIT 11 FOR INPUT
```

```
   LET EOF.V = 0
   FOR INDEX = 1 TO 22, READ NAME3(INDEX)
   LET EOF.V = 1

   LET INDEX = 0
   LET INDEX = INDEX + 1
   LET BIT.RATE = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET CODER = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET D = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET R = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET THRESHOLD=NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET SIMUL.LENGTH = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET NUM.MOBILES = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET FRAME.LENGTH = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET WARM = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET TALKSEED = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET SILSEED = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET RES.SLOTS = NAME3(INDEX)        " NO. OF RESERVATION SLOTS FOR EACH FRAME
   LET INDEX = INDEX + 1
   LET PACK.DROP.RATE = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET DATASEED = NAME3(INDEX)
   LET INDEX = INDEX + 1
   LET INTERARRIVAL.TIME = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET VOICE.MOB = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET VIDEO.MOB = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET DATA.BLOCK = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET OVER.HEAD.BITS = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET FRAMES.PER.SEC = NAME3(INDEX)
   LET INDEX = INDEX+1
   LET VIDEO.SLOTS = NAME3(INDEX)


   CLOSE UNIT 11
"PRINT 1 LINE WITH THRESHOLD THUS
"========THRESHOLD= ***.***
   OPEN UNIT 11 FOR INPUT, NAME IS "MICRDATA"
   USE UNIT 11 FOR INPUT
   LET EOF.V = 0
   FOR INDEX = 1 TO 11, READ NAME4(INDEX)
   LET EOF.V = 1

   LET INDEX = 0
   LET INDEX = INDEX + 1
   LET RADIUM.CELL = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET FREQUENCY = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET CLUSTER.SIZE = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET BLOCK.LENGTH = NAME4(INDEX)
```

```
   LET INDEX = INDEX + 1
   LET INTER.SWITCH = NAME4(INDEX)

   LET INDEX = INDEX + 1
   LET ANTENNA.BASE= NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET ANTENNA.MOBILE = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET POWER.TRANSMITTER = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET GAIN.BASE = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET GAIN.MOBILE = NAME4(INDEX)
   LET INDEX = INDEX + 1
   LET N.MICROCELL = NAME4(INDEX)
N.MICROCELL = 30
   PRINT 1 LINE WITH RADIUM.CELL, FREQUENCY, CLUSTER.SIZE,
BLOCK.LENGTH,INTER.SWITCH,
   ANTENNA.BASE, ANTENNA.MOBILE, POWER.TRANSMITTER, GAIN.BASE,
GAIN.MOBILE,N.MICROCELL
   AS FOLLOWS
   **** **** **** **** **** **** **** **** **** **** ****
   CLOSE UNIT 11

END "READTALK



ROUTINE SCHEDULER GIVEN SLOT.NO YIELDING TIME.DIFF

" CALCULATES THE TIME DIFFERENCE BETWEEN CURRENT TIME AND SHEDULED TIME TO
TRANSMIT

DEFINE ST.TIME AND SLOT.TIME AS REAL VARIABLES

 LET SLOT.TIME = (SLOT.NO - 1)* SLOTS
 LET ST.TIME = FRAME.TIME + SLOT.TIME " TELLS WHAT TIME YOU WOULD TRANSMIT

 IF ST.TIME LE TIME.V
  LET ST.TIME = ST.TIME + FRAME.LENGTH
 ALWAYS

 LET TIME.DIFF = ST.TIME - TIME.V

 IF TIME.DIFF LT 0                    " THIS IS NOT NEEDED DONE ONLY FOR RELIABILITY
  LET ST.TIME = ST.TIME + FRAME.LENGTH   " SAME
  IF ST.TIME LT TIME.V                 " SAME
   WRITE AS "##WARNING WILL ROBINSON - DANGER DANGER",/
  ALWAYS
  LET TIME.DIFF = ABS.F(ST.TIME - TIME.V)        " TIME.DIFF SHOULD BE AT THIS STAGE
POSITIVE
 ALWAYS

 LET TIME.DIFF = ABS.F(TIME.DIFF*1000)

" IF STATUS(MOBILE) = .INQUE
"   PRINT 1 LINE WITH SLOT.NO, FRAME.TIME, TIME.V AND TIME.DIFF THUS
"   >>>>>> SLN: *** FRTIME: *****.******* TIME : *******.******** TIME.DIFF: **********.*******
" ALWAYS

 RETURN
END "SCHEDULER



ROUTINE SILENCE
```

```
DEFINE P AS A REAL VARIABLE
DEFINE COUNT AS AN INTEGER VARIABLE

LET P = RANDOM.F(SILSEED)

FOR COUNT = 16 TO 26
DO
  IF P > NAME1(COUNT) AND P < NAME1(COUNT+1)
    LET ONY = (P-NAME1(COUNT))/(NAME1(COUNT+1)-NAME1(COUNT))
    LET ONX = NAME2(COUNT+1)-NAME2(COUNT)
    LET SILENCES = (ONY*ONX)+NAME2(COUNT)
  ALWAYS
LOOP
END "SILENCE



PROCESS STATION
PRINT 1 LINE WITH TIME.V THUS
STATION BE ACTIVTED AT TIME ***.***
DEFINE REFERE AS A REAL VARIABLE

WHILE TIME.V LE SIMUL.LENGTH
DO

 IF ST_TYPE(STATION) = .VOICE

   CALL SILENCE
   LET SILENCES(STATION) = SILENCES
   LET SIL          = SILENCES + SIL
   ADD 1 TO NUM.SIL

   WAIT SILENCES(STATION)*1000 .MILLISECONDS
   CALL TALKSPURT
   LET TALKSPURTS(STATION)  = TALKSPURTS

   LET NUM.PACKETS(STATION) = TALKSPURTS(STATION)/FRAME.LENGTH + 1

   CALL MICROCE YIELDING STRENTH
   LET SIG.STRENTH(STATION) = STRENTH

   IF (TIME.V LT SIMUL.LENGTH) AND (TIME.V GT WARM)
     LET TOT = TOT + NUM.PACKETS(STATION)
     ADD NUM.PACKETS(STATION) TO TNPKTS
   ELSE
     LET TOT = 0
   ALWAYS
   LET TOXPRT = TALKSPURTS + TOXPRT
   ADD 1 TO NUM.OF.TOXPRT
   LET PACK.COUNT(STATION) = 0

 ALWAYS

 IF ST_TYPE(STATION) = .DATA

    IF DATAB(ID(STATION)) = .SET
      D.BURST_TIME = (TIME.V - DATA.START.TIME(ID(STATION)))*HOURS.V
      D.TIME = (((TIME.V -
DATA.START.TIME(ID(STATION)))*HOURS.V)/DATA_PER.BLOCK(ID(STATION)))
"      PRINT 1 LINE WITH ((TIME.V - DATA.START.TIME(ID(STATION)))*HOURS.V),
DATA_PER.BLOCK(ID(STATION)) AND
"      (((TIME.V - DATA.START.TIME(ID(STATION)))*HOURS.V)/DATA_PER.BLOCK(ID(STATION)))
THUS
"      TIME *****.**** PACKETS ****** RATIO ****.*****
      DATAB(ID(STATION)) = .NOTSET
    ALWAYS
```

```
    WAIT EXPONENTIAL.F(INTERARRIVAL.TIME*1000,DATASEED) .MILLISECONDS
    LET MESSAGES = 0

    UNTIL (MESSAGES > 0.01) AND (MESSAGES < 10)
    DO
        LET MESSAGES = EXPONENTIAL.F(1, DATASEED)
    LOOP
    LET MESSAGE.LENGTH(STATION) = ((MESSAGES*100000)/(BITS))*FRAME.LENGTH
    LET NUM.PACKETS(STATION) = INT.F(MESSAGE.LENGTH(STATION)/FRAME.LENGTH)
    LET NUM.PACKETS(STATION) = NUM.PACKETS(STATION) +
INT.F(NUM.PACKETS(STATION)/DATA.BLOCK) +1

    CALL MICROCE YIELDING STRENTH
    LET SIG.STRENTH(STATION) = STRENTH


    IF (TIME.V LT SIMUL.LENGTH) AND (TIME.V GT WARM)
     LET AV.PACKET = AV.PACKET + NUM.PACKETS(STATION)
    ALWAYS
    ADD 1 TO NUM.MESS
    LET MESS = MESSAGE.LENGTH(STATION) + MESS
    LET PACK.COUNT(STATION) = 0
    IF NUM.PACKETS(STATION) GT 0
     LET DATAB(ID(STATION)) = .SET
     LET DATA.START.TIME(ID(STATION)) = TIME.V
     LET DATA_PER.BLOCK(ID(STATION)) = NUM.PACKETS(STATION)
    ALWAYS
 ALWAYS

 IF ST_TYPE(STATION) = .VIDEO

 'VIDEO.GEN'

  IF FRAME(ID(STATION)) = .SET
    FRAM.TIME = (TIME.V - FRAME.START.TIME(ID(STATION)))*HOURS.V
    FRAME(ID(STATION)) = .NOTSET
  ALWAYS
  WAIT (0.1*1000) .MILLISECONDS
  LET FRAME.START.TIME(ID(STATION)) = TIME.V
  FRAME(ID(STATION)) = .SET
  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD 1 TO HEY
  ALWAYS
  CALL VIDEO
  LET NUM.PACKETS(STATION) = TRUNC.F((V.PACKETS*48*8)/BITS) + 1

  CALL MICROCE YIELDING STRENTH
  LET SIG.STRENTH(STATION) = STRENTH

  IF (TIME.V GT WARM) AND (TIME.V LT SIMUL.LENGTH)
   ADD NUM.PACKETS(STATION) TO V_TNPKTS
  ALWAYS

  LET PACKET.COUNTER(ID(STATION)) = NUM.PACKETS(STATION)
  LET PACK.COUNT(STATION) = 0
ALWAYS

UNTIL NUM.PACKETS(STATION) LE 0
DO
 IF ST_TYPE(STATION) EQ .DATA AND PACK.COUNT(STATION) EQ DATA.BLOCK
  LET PACK.COUNT(STATION) = 0
  LET RESER.SLOT(STATION) = 0
 ALWAYS

 IF (TIME.V LT SIMUL.LENGTH) AND (TIME.V GT WARM)
  LET REFERE = (REFERENCE)*1000
  WAIT REFERE .MILLISECONDS
```

```
      ADD 1 TO STOTAL
      ADD 1 TO OWN.PACKET(STATION)
     ELSE
      LET REFERE = (REFERENCE)*1000
      WAIT REFERE .MILLISECONDS
     ALWAYS

     CREATE A MOBILE
     LET MOB.STATION(MOBILE)  = STATION
     LET R.FLAG(MOBILE)       = ID(STATION)
     LET V.SLOTS.NO(MOBILE)   = VIDEO.SLOTS.NO(STATION)
     LET V.SLOT.VALUE(MOBILE) = VIDEO.SLOT.VALUE(STATION)
     LET TYPE(MOBILE)         = ST_TYPE(STATION)
     LET RESER.MOB(MOBILE)    = RESER.SLOT(STATION)
     LET PACKET.NO(MOBILE)    = NUM.PACKETS(STATION)
     LET DEL.MOB(MOBILE)      = FR.DEL(STATION)
     LET SID.TXED.FLAG(MOBILE) = .NOTSET
     LET FILED.FLAG(MOBILE)   = .NOTSET
     LET V.NUM.DROPOUTS(MOBILE) = VIDEO.NUM.DROPOUTS(STATION)
     LET NUM.DROPOUTS(MOBILE) = 0
     LET ARRIVAL.TIME(MOBILE) = A.TIME(STATION)
     LET MOB.LOSS(MOBILE)     = 0
     LET PACKET.COUNT(MOBILE) = PACK.COUNT(STATION)
     LET STRENTH.SIG(MOBILE) = SIG.STRENTH(STATION)

     ACTIVATE THIS MOBILE NOW
     SUSPEND

     LET A.TIME(STATION) = A.TIME_ARRAY(ID(STATION))
     LET RESER.SLOT(STATION) = R.SLOT_ARRAY(ID(STATION))
     LET NUM.PACKETS(STATION) = PACKET_ARRAY(ID(STATION))
     LET VIDEO.SLOTS.NO(STATION) = V.SLOTS.NO_ARRAY(ID(STATION))
     LET VIDEO.SLOT.VALUE(STATION) = V.SLOT.VALUE_ARRAY(ID(STATION))
     LET VIDEO.NUM.DROPOUTS(STATION) = V.NUM.DROPOUTS_ARRAY(ID(STATION))

     LET PACK.COUNT(STATION) = PK.COUNT(ID(STATION))
     SUBTRACT 1 FROM NUM.PACKETS(STATION)

     IF NUM.PACKETS(STATION) EQ 1
      LET LAST.PACKET(STATION) = .SET
      LET PACK.COUNT(STATION) = 0
     ELSE
      LET LAST.PACKET(STATION) = .NOTSET
     ALWAYS

    IF ST_TYPE(STATION) = .DATA
     IF PACK.COUNT(STATION) EQ (DATA.BLOCK - 1)
       LET LAST.PACKET(STATION) = .SET
     ALWAYS
    ALWAYS

    IF TIME.V GT SIMUL.LENGTH
     CALL OUTPUT
    ALWAYS
  LOOP
LOOP
END "STATION
ROUTINE TALKSPURT

DEFINE Q AS A REAL VARIABLE
DEFINE ADD AS AN INTEGER VARIABLE

LET Q = RANDOM.F(TALKSEED)

FOR ADD = 1 TO 15
DO
  IF Q > NAME1(ADD) AND Q < NAME1(ADD+1)
```

```
    LET ONEY = (Q-NAME1(ADD))/(NAME1(ADD+1)-NAME1(ADD))
    LET ONEX = NAME2(ADD+1) - NAME2(ADD)
    LET TALKSPURTS = (ONEY*ONEX)+NAME2(ADD)
   ALWAYS
 LOOP

 END "TALK



 PROCESS TEMPPRO

 LET PREVIOUS.PK.COUNT = 0
 LET PREVIOUS.V_TNPKTS = 0
 LET PREV.AR_Q     = 0
 LET PREV.TIME     = 0.0

"OPEN UNIT 2 FOR OUTPUT, NAME IS "CAPTEMP6.TXT"
"USE UNIT 2 FOR OUTPUT

 WHILE TIME.V LT SIMUL.LENGTH
 DO
  IF TIME.V GT 0
   PRINT 1 LINE WITH TIME.V, ((AR.QUE - PREV.AR_Q)/(TIME.V - PREV.TIME)), (V_TNPKTS -
PREVIOUS.V_TNPKTS),
   ((TNPKTS + AV.PACKET + V_TNPKTS) - PREVIOUS.PK.COUNT), FRAME.NO AND SUCCESS.R
THUS
   AT TIME : *****.**** Q.LENGTH *****.**** VID.PACKETS: ******** TOT.PACKET: ************
FRAME.NO : ****** SUCCESS.'R': **
  ALWAYS

  LET PREV.AR_Q = AR.QUE
  LET PREV.TIME = TIME.V
  LET PREVIOUS.PK.COUNT = TNPKTS + AV.PACKET + V_TNPKTS
  LET PREVIOUS.V_TNPKTS = V_TNPKTS
  WAIT (20).SECONDS
LOOP

PRINT 1 LINE WITH (TNPKTS + AV.PACKET + V_TNPKTS) AND AV.PACKET THUS
TOTAL PACKETS TRNSMITTED : *********** DATA ********


END "TEMPPRO



PROCESS TIMER

" DEVELOP THE FRAMES THROUGHT THE SIMULATION
DEFINE I AND VINI AS INTEGER VARIABLES

UNTIL TIME.V > SIMUL.LENGTH
DO
 LET FRAME.NO   = FRAME.NO +1
 LET QUE.LENGTH = QUE.LENGTH + Q.COUNTER
 LET SUCCESS.R = RES.UTILIZATION - PREV_RES.UTILIZATION
 LET PREV_RES.UTILIZATION = RES.UTILIZATION
 LET FRAME.TIME  = TIME.V

 FOR I = 1 TO NUM.SLOT
 DO
   LET RES.UTI = .NOTSET
   FOR VINI = 1 TO RASNUM
   WITH I= RESERVATION(VINI), FIND THE FIRST CASE
   IF NONE
   ELSE
    CURR.RES = I
```

```
    IF RESERVATION(VINI +1) GT RESERVATION(VINI)
     NEXT.RES = RESERVATION(VINI +1)
    ELSE
     NEXT.RES = RESERVATION(1)
    ALWAYS
   ALWAYS
   SL.TIME = TIME.V
   RSL.NO = I
   WAIT (SLOTS*1000) .MILLISECONDS
 LOOP

LOOP

END "TIMER
```