AN INVESTIGATION INTO ADVANCE TIME DIVISION MULTIPLE ACCESS BASED PERSONAL COMMUNICATION NETWORKS

This Thesis is Presented in Partial Fulfilment of the Requirements for the Degree of Master of Technology in Production Technology at Massey University

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ABSTRACT

This thesis examines and simulates a statistically multiplexed multiple access technique known as Advanced Time Division Multiple Access (ATDMA). The simulations were carried out in a multimedia traffic environment. Parameters that could optimise the network performance in terms of quality, reliability and capacity have been examined using a simulation model. This thesis also examines network architecture and signalling related issues.

The simulation results were analysed to propose a suitable ATDMA frame structure in terms of the frame length and the organisation of traffic and reservation slots. The simulation results indicated that the performance of the ATDMA based system can be enhanced when delay insensitive data is transmitted as blocks of packets of a specific size. The simulation results also indicated that the performance of the ATDMA based system can be further enhanced when a video terminal is allocated a single traffic slot as opposed to multiple traffic slots. Further simulations have been carried out to determine the up-link traffic channel capacities and control channel capacities. This thesis also examined aspects that could further enhance the performance of an ATDMA based system.

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LIST OF ABBREVIATIONS

ACCH	Associated Control Channel
AGCH	Access Grant Channel
AMPS	Adavanced Mobile Phone System
APC	Automatic power Control
A-TDMA	Advanced-Time Division Multiple Access
ATDMA-PL	Advanced-Time Division Multiple Access-Physical Layer
ATM	Asynchronous Transfer Unit
ATM-NTU	ATM Network Transfer Unit
BC	Bearer Controller
BCCH	Broadcast Control Channel
B-ISDN	Broadband ISDN
BLC	Base Link Controller
BLM	Base Location Manager
BRA	Base Resource Allocator
BS	Base Station
BSS	Base Station System
BSC	Base Station Controller
BTP	Base Transport
BTS	Base Transceiver Station
CC	Common Controller
CCCH	Common Control Channel
CDMA	Code Division Multiple Access
CCSS7	Common Channel Signaling System Number 7
CL	Connectionless
CO	Connection-Oriented
CS	Convergence Sublayer
CT-2	Second Generation Cordless Telephones
DCCH	Dedicated Control Channel
DECT	Digital European Cordless Telephone
ETSI	European Telecommunications Standards Institute

FBR	Fixed Bit Rate
FIFO	First Come First Out
FPLMTS	Future Public Land Mobile Telecommunication System
HB	Header Bits
GEOS	Geostationary Satellite
ID	Identification Number
IN	Intelligent Network
ISDN	Integrated Services Digital Network
ISDN-UP	Integrated Services Digital Network User Part
LA	Link adaptation
LC	Link Controller
LE	Local Exchange
LEOS	Low Earth Orbit Satellite
LCCH	Leash Control Channel
LM	Location Manager
LSD	Least Significant Digit
MAC	Medium Access Control
MAP	Mobile Application Part
ME	Measurement Entity
MEOS	Medium Earth Orbit Satellite
MLC	Mobile Link Controller
MLM	Mobile Location Manager
MRA	Mobile Resource Allocator
MRC	Mobile Routing Controller
MS	Mobile Station
MSC	Mobile Switching Center
MSD	Most Significant Digit
MSCP	Mobility and Service Control Point
MSDP	Mobility and Service Data Point
MSS	Mobile Satellite Services
MT	Mobile Terminal
MTC	Mobile Traffic Controller

MTP	Message Transfer Part
MTP	Mobile Transport
MSU	Mobile Switching Unit
NADC	North American Digital Cellular
NCS	Network Combiner & Switching
NLC	Network Link Controller
NLM	Network Location Manager
NLP	Network Layer Protocol
NNI	Network to Network Interface
NRA	Network Resource Allocator
NRC	Network Routing Controller
NTC	Network Traffic Controller
OSI	Open System Interconnection
PCN	Personal Communication Network
PRMA	Packet Reservation Multiple Access
PSTN	Public Switched Telephone Network
RA	Resource Allocator
RACH	Random Access Channel
RAMA	Resource Auction Multiple Access
RC	Routing Controller
RF	Radio Frequency
RF-NTU	RF Network Transfer Unit
RGCH	Access Grant Channel
RLL	Radio Link Layer
RLP	Radio Physical Layer
RSS	Radio Support System
SAAL	Signaling ATM Adaptation Layer
SAR	Segmentation and Reassembly
SC	Slot Controller
SCCP	Signaling Connection Control Part
SCP	Service Control Point
SNL	Signaling Network Layer

SS 7	Signaling System Number 7
STP	Signaling Transfer Points
TB	Tail Bits
TC	Transmission Controller
TCH	Traffic Channel
TX	Transient Exchange
TDM	Time Division Multiplexed
TDMA	Time Division Multiple Access
QoS	Quality of Service
UMTS	Universal Mobile Telecommunication System
VCI	Virtual Channel Identifier
VPI	Virtual Parth Identifier
WARC'92	World Administrative Radio Conference 1992

CHAPTER 1
Introduction

1.1 General Introduction

Over the recent years, second generation hand-held mobile telephones have become an essential element of the rapidly expanding telecommunication industry [1, 19, 20]. Second generation mobile phones are capable of supporting voice, fax, data and low bit rate video/multimedia services [7]. By contrast, a much versatile set of services could be supported over Asynchronous Transfer Mode (ATM) based fixed networks [8, 11, 12]. With the introduction of third generation Personal Communication Networks (PCN), the traditional boundary between fixed and wireless networks would be crossed [13]. Currently, Future Public Land Mobile Telecommunication System (FPLMTS) [3] and Universal Mobile Communication System (UMTS) [18, 15] are the largest projects concerned with the issues of PCN development. According to the European Telecommunications Standards Institute (ETSI) and its dedicated sub-technical committee SMG5, UMTS has been specified as the third generation mobile communication system based on FPLMTS [18]. With the introduction of fully operational PCN, a person carrying a personal communicator could send, receive, store and process many kinds of information, in essence all the services possible through the fixed networks, regardless of user location. Services provided over PCN would be in the form of a phone call, text message, facsimile or video. PCN should be able to support Constant Bit Rate (CBR), Variable Bit Rate (VBR), real time or non real time traffic. There are also a few end-user requirements PCN developers must keep in mind in order to make the personal telephone a success. Some of them include service capabilities, size of the personal communicator, purchase and usage cost, quality of service and the independence of mobility.

Currently wireless communication consists of cellular and cordless systems. Cellular systems such as Time Division Multiple Access (TDMA), Global System for Mobile Communication (GSM) [2, 3], Advanced Mobile Phone System (AMPS) [7] and Code

Division Multiple Access (CDMA) [4, 5] provide radio coverage mostly by high power base stations that support both portable and vehicular based units [1]. Terminal mobility is supported by call handover between base stations within the network. Cordless systems such as Second Generation Cordless Telephones (CT2) and Digital European Cordless Telephone (DECT) [19, 20] offer limited coverage by personal base stations or low power public base stations with no handover capability [1]. Personal communication systems will converge cellular and cordless systems along with some of the broad-band services provided over fixed networks into a single universal network. In such a system, a single terminal can be used in a variety of environments. Therefore, terminals would be equipped to handle telephone, pager, fax, answering machine, digital diary and even good quality video.

Introduction of personal communication systems will considerably increase the amount and the variety of traffic supported in the air interface making PCN developers' task much more challenging. With the increased use of mobile facilities the available radio spectrum for second generation mobile communication is becoming scarce. What makes the researchers' task that much more challenging is that they need to develop a network that supports all of the existing services plus next generation services with more users and with no degradation to the quality of service (QoS). Therefore, efficient use of the radio spectrum is one of the highest priority areas of PCN development. Due to the dynamic nature of PCN traffic, a packet based system that incorporates dynamic channel allocation could be a suitable candidate for a PCN air interface. A packet based system allocates resources only when it is actually required. Some of the reservation based packet access techniques being proposed are Packet Reservation Multiple Access (PRMA) [6], Resource Auction Multiple Access (RAMA) [10, 44] and Advanced Time Division Multiple Access (ATDMA) [14]. In addition GSM [48] and CDMA based systems are also being proposed by PCN developers from different parts of the world [9, 17].

1.2 Thesis Structure

The main objective of this research was to study a PCN based multimedia system and its signalling related issues. A statistically multiplexed radio access mechanism known as

ATDMA was studied by means of a simulation model developed in Simscript II.5. The model has been used to study the traffic efficiency of the protocol. Simulation results were also used to study the signalling aspects of a possible ATDMA based system.

Chapter 2 presents an overview of the PCN. This chapter examines the current situation in mobile communications and the evolution of PCN. This chapter also examines some PCN design issues and their requirements. Requirements such as Quality of Service (QoS), delay, throughput and packet loss are discussed for different service types. Finally, current research activities in PCN are addressed in terms of multiple access techniques and network architecture. This chapter also presents a brief discussion on the UMTS architecture, one of the largest projects in third generation mobile communication systems.

Chapter 3 presents the PCN architecture and its signalling structure. This chapter examines the PCN architecture in terms of wireless and fixed networks. The wireless network studied here is based on the ATDMA access protocol whereas the fixed network is based on ATM. Therefore, the above PCN consists of Mobile Stations (MSs), Base Stations (BSs) and a fixed network. This chapter also examines some signalling related issues. Some of these include signalling and user information transfer in PCN (in terms of the OSI structure) and burst structures of ATDMA. The chapter also present the basic structure of control and traffic plains for the above network based on the ATDMA functional model and Common Channel Signalling System Number 7 (CCSS7).

Chapter 4 presents the simulation of the ATDMA protocol. The main aim of the simulation was to find out appropriate parameters that could be used to optimise the ATDMA network performance in terms of capacity, quality, reliability and durability. Some of the parameters of interest are reservation request slots ('R' slots) to traffic slots ('I' slots) ratios, data block size, traffic capacities and the organisation of 'R' and 'I' slots in an ATDMA frame. This chapter also investigates up-link signalling volumes generated for various channels.

Chapter 5 presents an overall conclusion on the work done in ATDMA/ATM based PCN design. Some suggestions are also presented for future PCN development.

CHAPTER 2
Overview of Personal Communication Networks

2.1 Introduction

With the introduction of Personal Communication Networks (PCN), wireless communication is entering a new era. Despite the increasing demand and service requirements of second generation cellular systems, personal communication will emerge with multimedia capability. Today, people want to talk to and leave voice or written messages to other people regardless of their locations. Personal communication system must therefore offer mobility, integrated services, advance customer control and friendly user interfaces. Personal communication will put stringent requirements on the network that support all these services since, the network must not only provide advanced and high quality services, it must also handle call control, service control and mobility for a much higher user capacity than a second generation cellular system.

2.2 Personal Communication Networks (PCN) and its Evolution

The PCN evolution process probably started with the introduction of fixed, analog phone networks. After many successful years with analog phones, the business community wanted more than the features of fixed telephones. Instead of leaving messages and waiting for a long time for a reply (e.g. when the called person is not in the office), the flexible cordless phone system was introduced [7]. A separate paging system was then introduced to provide more flexibility and mobility. Then, to further increase the service capabilities, one way voice mail was introduced. This system had its share of problems due to its one way nature (e.g. people never returned calls at least in real time). Although at this point in time cordless phones were very popular, specially in residential and small business markets, bigger businesses were looking for more flexibility, mobility and security [7]. As a result, first generation analog mobile networks were introduced during the early 1970's. They supported two way voice calls but security, quality of transmission, system capacity and high costs made network providers develop and introduce the second generation cellular mobile concept (e.g. GSM, IS-54, IS-95, PDC). Cellular systems divide the coverage areas into cells and reuse the frequencies in different cells. If the cells are small the capacity of the overall system could be high. The distinguishing feature of second generation cellular systems compared to the previous radio system was the use of many base stations with relatively small cell radii. Therefore, second generation mobile systems support much more capacities than its predecessor due to the smaller frequency re-use distances. When second generation mobile systems progressed from analog to digital, further capacity increases were possible since multiple channels were supported in single carrier frequency. At present they provide voice, data and low speed video services. In the mean time, with the introduction of optical fiber systems, a variety of high bit rate services have been possible through the fixed networks. This resulted in the demand for a universal mobile communication system that could provide any service that is possible through the fixed network regardless of the user location [13]. As a result, work on Future Public Land Mobile Telecommunication System (FPLMTS) and Universal Mobile Telecommunication System (UMTS) began. UMTS is predicted to provide services some time after year 2000 [61].

New generation of mobile communication systems are currently under development with the aim to support over a seamless radio infrastructure, not only the diverse offerings of second generation systems, but also a much wider range of broad-band services. If these networks are developed from the existing networks it would avoid the massive investment required to deploy a new network over a relatively short period of time. Such an evolution would allow the network providers to maintain or increase the profitability during the evolution phase, while providing more services. The introduction of the third generation systems are expected to start at the turn of the century [18, 15]. Fig 2.1 shows the evolution of mobile services that has taken place since the early 1970's. This figure also shows future predictions. As shown in Fig 2.1, by the turn of the century channel bit rates around 2 Mbps will be available, while within the next 15 years 155 Mbps [1] channels would be practical. Bandwidths of such a nature are available at present through the fixed network, but maintaining them through the wireless network would need extremely sophisticated wireless network architectures. When high bit rate services are possible through the wireless networks, a personal communicator handset would provide most of the services possible through the fixed network. Such a handset is shown in Fig 2.2. One of the major advantages of PCN, the independence of mobility or anywhere anytime concept of PCN is illustrated in Fig 2.3.

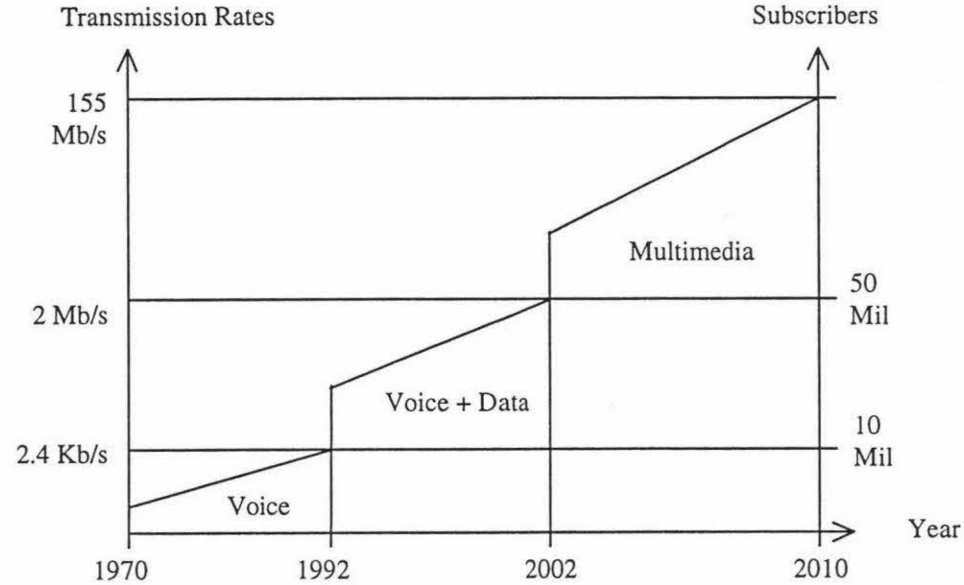
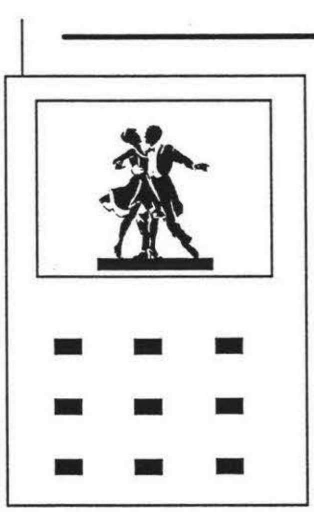


Fig 2.1 Evolution of Communication Services



- Digital voice
- E-mail and file transfers
- Facsimile at group IV speeds or higher and paging
- Radio-determined vehicle location capabilities
- Message leave and retrieve capabilities (e.g. answering machine)
- News services (including articles and pictures)
- Multimedia capabilities
- Video telephony (up to 10 frames per second)
- Dial up video in a slow scan mode (up to 10 frames/sec)

Fig 2.2 Eventual Personal Communicator handset with its service capabilities

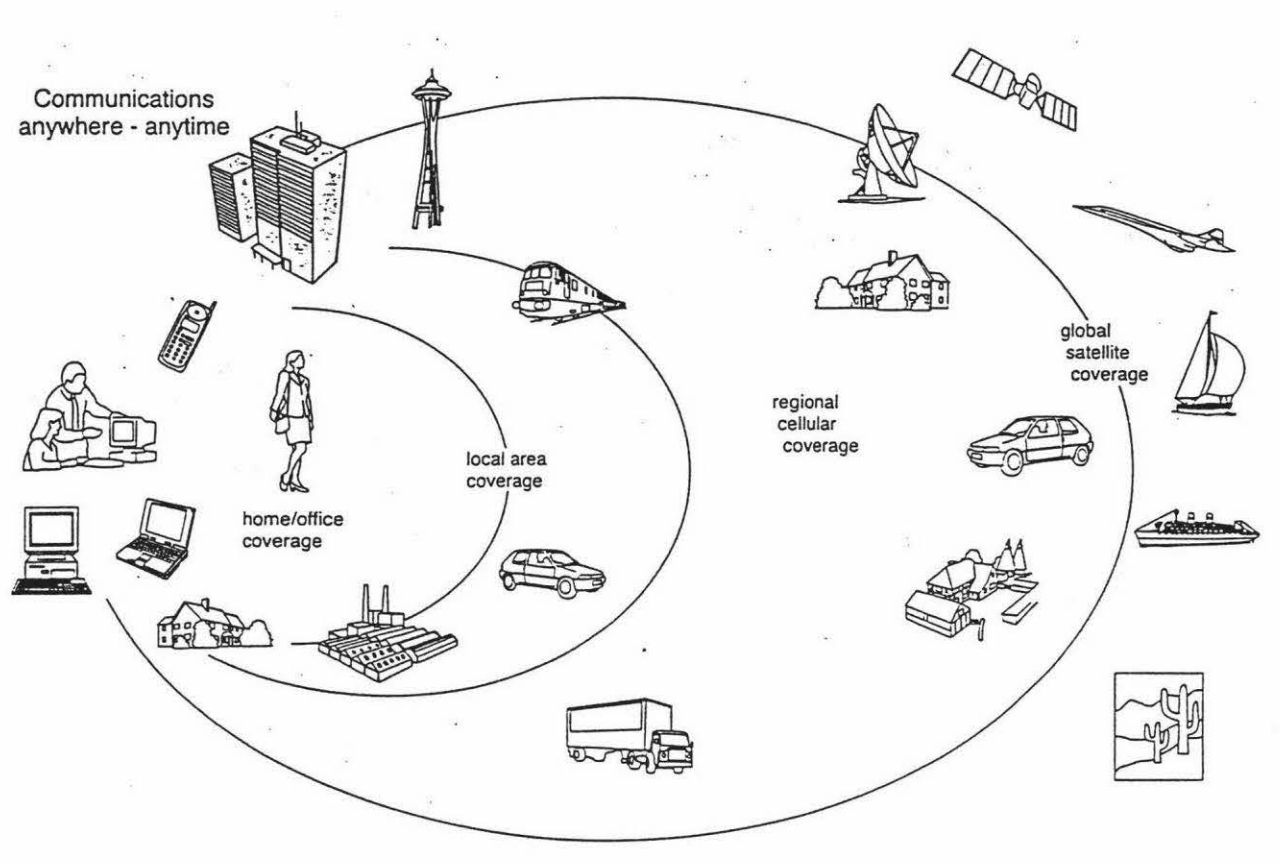


Fig 2.3 Anywhere Anytime Concept of PCN [13]

2.3 Current State of Wireless Tele-Communication

Today, wireless communication is available via many different systems. Some of them are given in Table 2.1 [40]. Two of the main wireless communication technologies currently being used are the cellular and cordless systems. Network control in a cellular system is done with the help of sophisticated digital signal processing algorithms and computational techniques. They reduce interference, make handover decisions, provide channel information to Mobile Stations (MS), perform power control and link adaptation [20]. By contrast, cordless phones independently decide how to communicate to the rest of the world [20]. Neither the centralised cellular approach nor the independent cordless approach could handle the control of the next generation mobile systems. One possible solution is to have a packet based distributed control management technique [21].

Analog cellular	Nordic mobile telephone Advanced mobile phone service Total access communication systems NETZ C, D Nippon advanced mobile telephone system
Digital cellular	GSM IS-54 (D-AMPS) IS-95 (Qualcomm CDMA) Japanese digital cellular
Cordless communications (CT)	Analog domestic cordless telecommunications (CT) Improved CT 1 Common air interface CT 2 system New CT 3 systems Digital CT at 900 MHz Digital European CT (DECT)
Wireless LAN	IEEE 802.11 MOBITEX High Performance Radio LAN (HIPERLAN) ALTAIR (proprietary wireless LAN)
Private mobile radio (PMR)	Analog 12.5 KHz/FM Trans-European trunked radio (TETRA)
Mobile satellite	Aeronautical in-flight Land: EUTELTRACS, PRODAT Marttime: Inmarsat A, B, C Proposed: IRIDIUM, GLOBALSTAR, ODYSSEY, CONSTELLATION ELLIPSAT
Paging	Many entrenched systems European radio message system

Table 2.1 Some existing or soon to be deployed mobile radio systems [40, 43].

Personal communication networks consist of fixed and wireless parts. Information transfer through wireless is much more challenging than through the fixed networks due to its dynamic nature and limited resources. Although fixed networks have many advantages over the wireless networks (e.g. cost, reliability, quality, etc.), wireless communication is gradually replacing some of the wireline communication services due to its independence of mobility [19, 20]. Currently second generation digital wireless communication networks are already in existence while third generation PCNs are expected to emerge by the turn of the century [18, 15]. When the next generation mobile networks are in full operation, a universal network will have the capability of supporting many different services over a single network.

2.4 Universal Mobile Telecommunication System (UMTS)

This section presents a brief description of UMTS, since it is one of the largest projects that investigates issues related to the third generation mobile communication systems (The other major third generation wireless system that is developed in parallel with UMTS is FPLMTS. FPLMTS functionalites and objectives are closely aligned with UMTS). By the turn of the century UMTS is expected to support existing mobile telecommunications services at transmission rates up to 2 Mbps [18, 15]. Some other objectives of UMTS are listed below [18]:

- Integration of all communication technologies into a single network
- Quality of service comparable to that achieved through fixed networks
- Provide a unique UMTS user number that is independent of network or service provider
- Service capabilities to half of the European population
- Seamless global coverage
- Creation of direct satellite access
- High radio spectrum utilisation
- Low cost for the end user
- Friendly user interfaces

ATDMA and CDMA are investigated as the access techniques for UMTS. On the other hand, investigation on the fixed part of the network is mainly based on the ATM technology [15]. Integration of UMTS into B-ISDN means that UMTS should take B-ISDN functionality into account. This requires changes or additions to the B-ISDN functionality. As an example mobility related functions should be incorporated in B-ISDN. Due to capacity limitations of the wireless medium, all services possible through B-ISDN might not be supported through the wireless network. The ultimate goal of UMTS is to cross the above barrier. UMTS also adapts the Intelligent Network (IN) concept to provide a way to add functionality to an existing network without changing too much of the original functionality. This way less cost is incurred, since original networks are updated using IN concept to support UMTS features. One of the proposed UMTS architectures are shown in Fig. 2.4

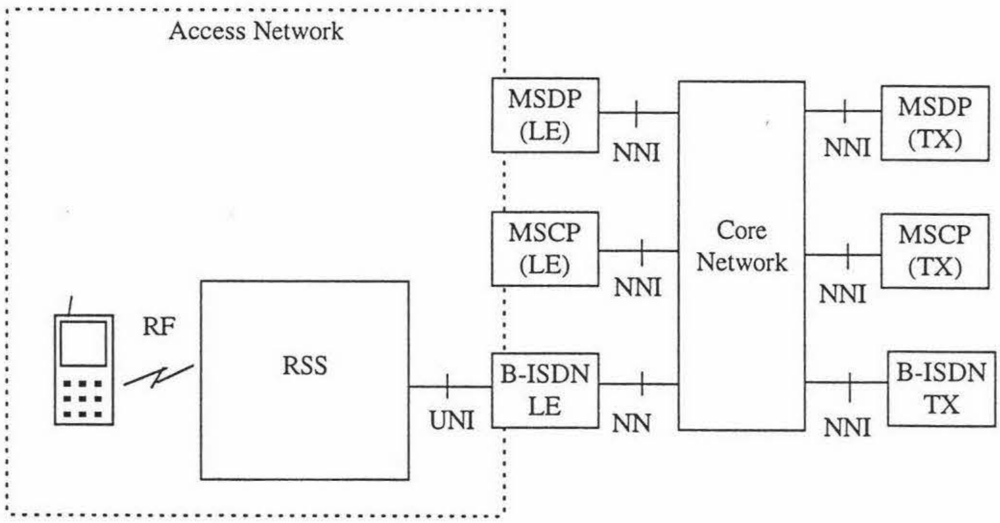


Fig 2.4. UMTS Network Architecture

The UMTS Access Network includes mobile terminal, Radio Support System (RSS) entities, B-ISDN Local Exchange (LE) (on the UNI side), the access side of any Mobile Service Control Point (MSCP) and Mobility and Service Data Point (MSDP). B-ISDN is used to provide direct connections between RSS and entities beyond B-ISDN LE [61].

Radio Support System (RSS) is assumed to have two major interfaces. They are the interface handling the wireless network and the interface handling the local exchange on the core network. In some environments RSS would have a tree structure made up of one or more base stations and a single centralised entity [61].

The UMTS Core Network has two interfaces, one on the access network side and the other on the fixed network side. As shown in Fig. 2.4, they consist of MSCP and MSDP on the local exchange side and the Transient Exchange (TX) side. The core network is assumed to be based on standard B-ISDN.

2.5 PCN Services and Requirements

Emerging next generation wireless systems will be required to transport traffic with different performance requirements and characteristics. Some of these services along with their requirements are illustrated in Table 2.2. Most of the high bit rate services listed in Table 2.2 could be supported over B-ISDN networks and most of them should be extended for wireless networks. In case of real time traffic (i.e. voice and video), stringent delay requirements have to be met in a wireless network. Therefore, transmission protocols have to be developed to give delay sensitive sources a higher priority. Information lost during transmission will also affect the QoS. In most packet based networks, voice services can tolerate speech packet loss of up to 5% [7]. Advanced signal processing techniques could be used to compensate for the lost segments. In most cases due to the high compression ratios, even a small amount of information loss affects the quality of video services [37]. Video services normally tolerate higher delay thresholds than voice. This would allow the re-transmission of video information if it is not received properly at the receiver. It is important to remember that the requirements of services are always application dependent. If PCN is to be successful then the quality of the services (information loss, interference, fading and security are some of the factors affecting the QoS of wireless networks), should be at least close to that of fixed networks and at a reasonable cost to the user. It is extremely difficult to achieve a high QoS through the wireless networks due to unfavourable channel conditions (e.g. co-channel interference, adjacent channel inference, fading). However, network providers should try to achieve a reasonable QoS by means of powerful channel/error correction coding techniques, signal processing, and modulation techniques.

Teleservice Type	Service Type	Throughput (kb/s)	Target bit error rate
Telephony	CO/CBR	8 - 32	10E-3
Teleconference	CO/CBR, CO/VBR	32	10E-3
Voice mail	CL best effort packet	32	10E-3
Digital Audio	CO/CBR	128 - 512	10E-3
Video telephony	CO/VBR , CO/CBR	64	10E-7
Video conference	CO/VBR, CO/CBR	384 - 768	10E-7
Digital HDTV	C)/CBR	15000 - 20000	10E-7
Remote terminal	CL best effort packet	1.2 - 9.6	10E-6
User profile editing	CL best effort packet	1.2 - 9.6	10E-6
Telefax (group 4)	CL/CBR	64	10E-6
Voiceband data	CL/CBR	64	10E-6
Database access	CO/CBR, CO/VBR	2.4 - 768	10E-6
Message broadcast	CL/CBR	2.4	10E-6
Unrestricted digital information	CO/CBR, CO/VBR	64 - 1920	10E-6
Navigation	CO/CBR	2.4 - 64	10E-6
Location	CL/CBR	2.4 - 64	10E-6

CO : Connection Oriented, **CL** : Connectionless, **CBR** : Constant Bit Rate, **VBR** : Variable Bit Rate

Table 2.2 A Subset of Proposed Teleservices for Third-Generation Mobile Communications Systems [1]

2.6 Design Issues of a PCN

Wireless personal communication systems encompass many technologies and services optimised for different applications. Such a PCN in a multi-service environment is shown in Fig. 2.5. As in any communication network when a PCN is developed, one should try to achieve the best possible service for all users of the network. However, the scale of PCN development is large, so tradeoffs between network efficiency, service performance and terminal cost have to be realised. Some of the resulting PCN design challenges are highlighted below:

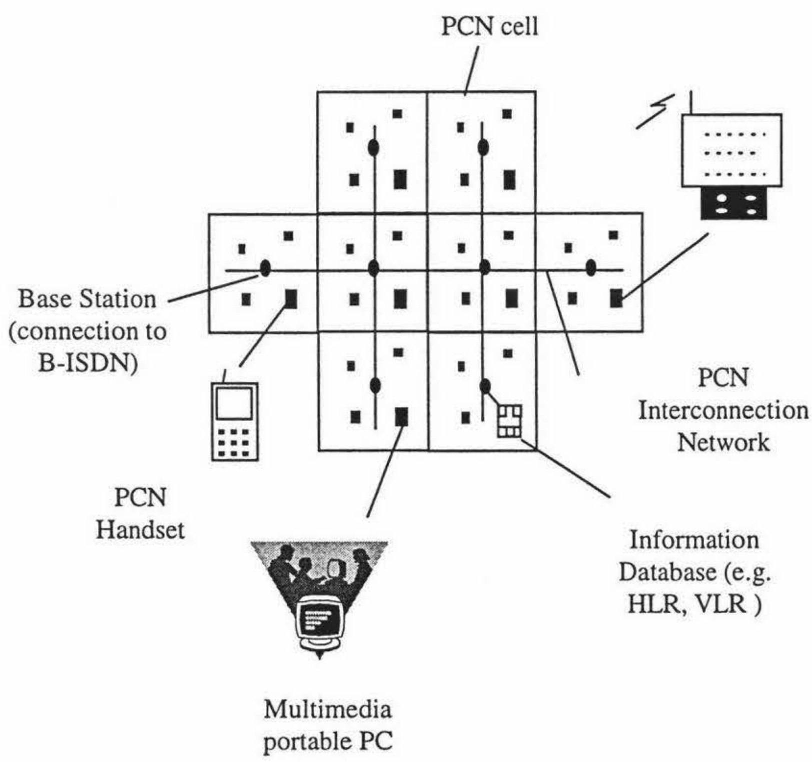


Fig 2.5. View of next Generation Multi-service PCN

- **The cost of the network would depend on the complexity of the fixed and radio equipment and on the network architecture.** Complexity of the personal communicator technology would also affect the cost, power consumption (battery size, weight and operating time) and the complexity of control functions required for network interaction. The economics of the network would also depend on the number of radio circuits per fixed radio transceiver and per radio channel frequency.

During PCN development all these aspects needs to be taken care of in order to keep the cost to a minimum.

- **A high degree of compatibility with future networks and existing networks.** A PCN should be able to support variable bit rate services at high transmission speeds from different network providers. When personal communication networks are developed, upgrading the existing networks, network infrastructure cost would be low. It would also allow network providers to continue providing services during the upgrading process.

- **High quality-of-service (QoS) by means of no distortion and negligible transmission delay for real time services.** Most of the non-real time services are delay insensitive, but they could be loss sensitive. Loss sensitive services are to be supported by incorporating better error correction capabilities and/or allowing more re-transmission attempts.

- **Privacy and security by encryption of the radio link.** Better security could also be provided with the use of frequency hopping techniques [20].

- **The need for high radio spectrum utilisation efficiency, since it affects the amount of spectrum needed and system cost.** Multiple access techniques are one of the major areas affecting the spectrum utilisation. Spectrum utilisation also depends on frequency planning methods, signal processing techniques, modulation techniques, diversity techniques, power control techniques and so on. Some of these issues will be briefly discussed in the next section of this chapter.

- **Signalling to incorporate all these services and efficient use of the signalling bandwidth.** Mobility (handover), channel access, channel release, power control, link adaptation related issues need to be addressed by a proper signalling infrastructure. The signalling aspects, specially on the wireless network should be taken care of in such a way that a minimum volume of signalling is generated. This is because the available bandwidth on the radio portion of the network is limited.

2.7 Research Areas in PCN

In order to introduce third generation mobile telecommunication systems by the turn of the century, researchers from all over the world are investigating different aspects of the PCN. Two key areas that have been studied in this thesis are:

- Multiple access techniques
- Network architecture

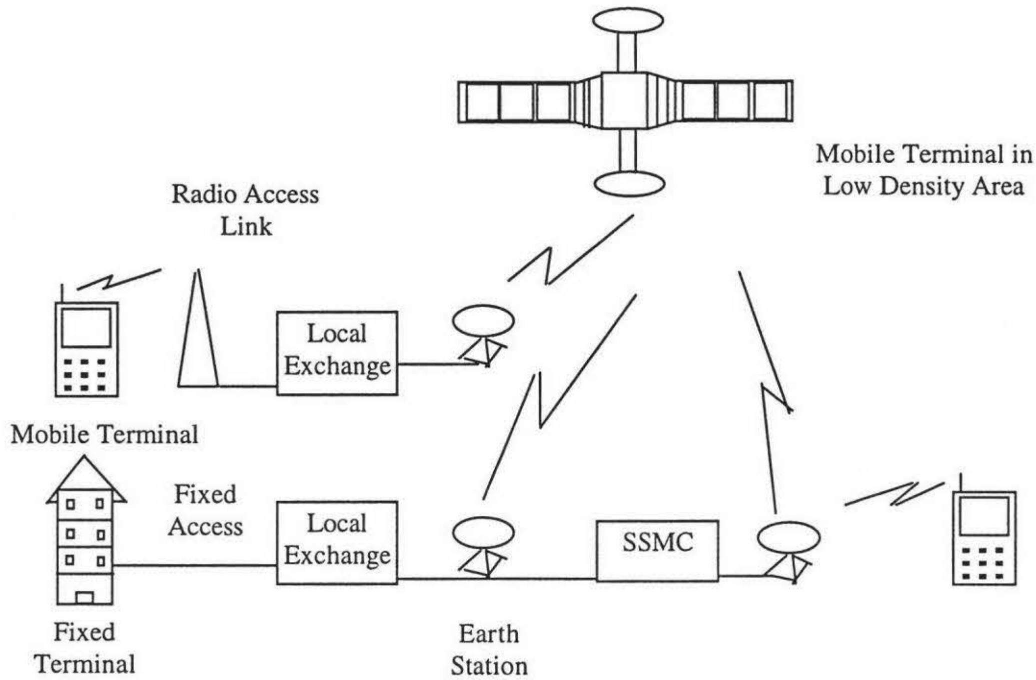
Unpredictable channel conditions, limited channels and the mobility of terminals are some of the many reasons why the implementation of wireless networks is challenging. The capacity, efficiency and performance of a personal communication system are limited mainly due to the limitations of the wireless network. Therefore, designing a suitable multiple access technique is an essential part of PCN development. Suitable burst structures are also need to be developed to suit the adapted multiple access technique. In order to provide these bursts, reliable transmission error correction and source coding techniques have to be incorporated. The quality of a radio channel would also be affected by channel fading. This is handled by using appropriate modulation techniques. Fading related matters are also handled by the incorporation of the training sequence [48] in traffic bursts. Training sequences are an important aspect of PCN design since it also allows radios to synchronise the receivers with the burst. Interference is another aspect affecting the efficiency of wireless networks. This should be handled by using low power transmitters and efficient frequency planning techniques. Security is another important aspect of PCN design. Therefore, efficient encryption techniques have to be developed. It is important to note that most of the above research areas concerns factors that interact.

Network architecture is another important aspect of PCN design. Some of the research areas of network architecture include, interworking with existing and future networks, interworking between wireless and fixed networks, integration of Intelligent Networks (IN), database management and satellite related issues. Suitable signal processing and decision making algorithms should be located at the BS or the MSC level of the fixed network. They would perform decision making, in case of handover (mobility), power control, link adaptation, resource allocation and release. When a fully operational PCN has been designed and developed, it should be able to fulfil requirements such as the ones listed below [41].

- **Provide seamless (without any interruptions in transmission) multimedia services** across fixed and mobile environments through a suitable broad-band digital network as the backbone network

- **Flexible service provision**, allowing different service providers to customise their service offering through integration with Intelligent Networks (IN). As an example, a call can be delivered anywhere and through any subscriber unit. The network can track a subscriber's location and update the locations with the help of subscriber's assigned personal ID numbers. These functions are independent of the unit they use. Some times the assigned personal number will be provided on a smart card. The IN will recognise the card, but not the unit.

- **Universal services capabilities through integrated satellite components.** There are some situations where providing radio coverage with cellular wireless networks is either not economically viable or physically impractical. In such situations, Mobile Satellite Services (MSS) could be adapted as shown in Fig. 2.6. As shown in Fig .2.6 MSS could be adapted over less densely populated areas. This would be cheaper than having either a separate fixed network infrastructure or a normal mobile network infrastructure. Therefore, MSS is an important part of a PCN in providing global coverage. Some of the MSS systems proposed include Low Earth Orbit Satellite (LEOS), Medium Earth Orbit Satellite (MEOS) and Geostationary Satellite (GEOS) systems [19, 62]. LEOS require more, but less expensive satellites to cover the earth. They can also be used for smaller coverage areas thereby increasing the capacity of the network. LEOS also have low transmission delays [19]. Since the operational characteristics of GEOS are completely opposite to that of LEOS, with long transmission delays (e.g. 0.5 seconds [19]), GEOS are not recommended for wireless mobile communication needs. MEOS fall between these two extremes.



SSMC : Satellite System Management Center

Fig 2.6. Mobile Satellite Services for Low Density Areas

2.8 Conclusions

In this chapter, some background information regarding the PCN has been presented. Although second generation mobile networks were introduced not that long ago, the need for a new mobile network with multimedia capabilities is growing rapidly. The two main reasons for this are the increase in demand and the need for a variety of high bit rate services that could not be supported by second generation networks. PCN should be capable of supporting different services with completely different service requirements. Some of the different services and their requirements along with some design issues in providing such services have been highlighted in this chapter. This chapter also discusses UMTS, since it is one of the largest projects that investigates the issues of third generation mobile communication systems. Finally, some research areas in the development of a PCN are examined.

CHAPTER 3
Advanced-Time Division Multiple Access (ATDMA) Protocol Based Personal Communication Network (PCN) Structure

3.1 Introduction

Personal Communication Network (PCN) should be able to support a much broader range of services than the second generation mobile communication networks. Therefore, PCNs would be capable of supporting broad-band services with the independence of mobility. This chapter describes the ATDMA based PCN structure which has been studied, simulated and proposed in this study. This chapter also discusses some of the ATDMA signalling issues.

In this study an ATDMA protocol has been selected for the wireless network. The user traffic transported through the wireless network should be mapped on to the fixed network infrastructure through an appropriate interface. The same should be done for the signalling (control) traffic of the PCN. During this study, different networks are adopted for user and control traffic. As a result, the fixed network architecture examined during this project was based on an ATM network for user traffic, whereas an ATM/CCSS7 network was examined for control (signalling) traffic.

3.2 Multiple Access Techniques for PCN's

A radio channel is fundamentally a point-to-point broadcast communication medium. The objective of wireless communication is to provide communication channels on demand between a Mobile Station (MS) and a Base Station (BS) that connects users to the fixed network infrastructure. The communication channels are assigned according to the adopted multiple access technique.

Based on the way communication channels are assigned, there are different types (classes) of multiple access techniques. The three main categories of multiple access techniques are Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA) and Code Division Multiple Access (CDMA). In TDMA signals can be transmitted in non overlapping slots in a round-robin fashion. Therefore, signals occupy the same frequency band but are easily separated in time. The TDMA concept is shown in Fig 3.1.1. FDMA signals occupy non-overlapping frequency bands which can be easily separated by appropriate bandpass filters. Therefore, signals can be transmitted simultaneously without interfering with each other. The FDMA concept is shown in Fig 3.1.2. In CDMA, different users employ signals that have very small cross correlation. Therefore, correlators can be used to extract individual signals from a mixture of signals even though they are transmitted simultaneously and in the same frequency band. CDMA concept is shown in Fig. 3.1.3. Network designers have to decide in favour of one or a combination of the previously mentioned techniques in order to facilitate multiple access. Preference for one or a combination of access methods over others depends largely on the overall system characteristics (e.g. digital or analog, etc.). No single access method is universally preferable. Thus, system considerations should be carefully weighed out before the design decision is made. Whatever the access technique adopted, the efficient utilisation of wireless networks is one of the most important aspects of achieving a universal personal communications network.

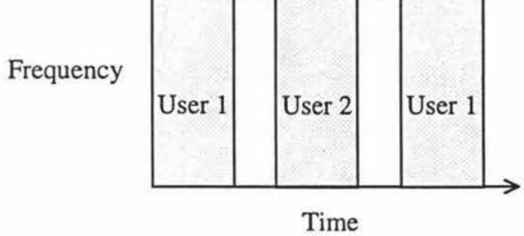


Fig. 3.1.1 TDMA

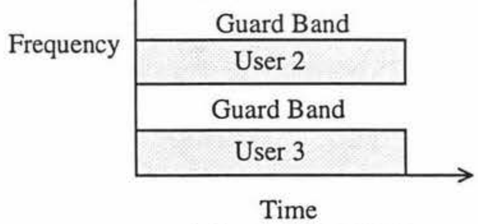


Fig. 3.1.2 FDMA

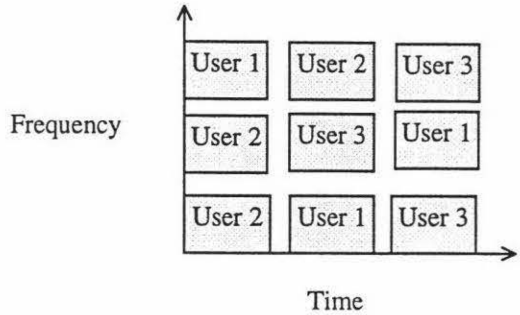


Fig. 3.1.3 CDMA

Packet switched networks are suitable for PCN, since resources are dynamically allocated to users. For example, a resource used by a particular terminal could be used by another terminal during the silence period of the initial terminal. By contrast, in a circuit switched system the channel holding time is the call duration. Packet transmission protocols can be divided into two classes. They are random access (i.e. contention based), and scheduled access or slotted access with or without reservation [29].

ALOHA is one of the common types of random protocols. The maximum channel throughput of ALOHA is only 18.4%. The slotted ALOHA protocol which divides the channel into time slots is another contention based protocol. Since in slotted ALOHA terminals can transmit only at the start of the slot, the vulnerable period is reduced and as a result, maximum channel throughput is increased to 36.8%. These protocols can not directly be applied for mobile radio applications for carrying speech traffic because of the 'hidden users' problem. The hidden user problem is one of the limitations of a mobile radio network. In some situations, due to the absence of line-of-sight, all the mobiles cannot hear each other. By contrast, terminals on the fixed network who share the common channel, would know which channel has current control.

Several random access protocols have been proposed for mobile radio applications. Random access protocols such as Idle Cast Multiple Access with Collision Detection (ICMA/CD) [71], Base Control Multiple Access with Collision Detection (BCMA\CD) [64] and Burst Tone Multiple Access (BTMA) [65] use a control channel to inform mobiles about the channel status. In these systems even with the control channel, packets are vulnerable to collisions.

Reservation based protocols try to avoid collisions. The ATDMA technique is an example of a reservation based access technique. In ATDMA, a frame consists of reservation slots and traffic slots. Access to a traffic slot is done through a reservation slot. Once a traffic slot is assigned, it will be used until all the packets of a talkspurt is transmitted. Then the traffic slot is released and made available for other users in the system. In such a system, collisions only occur during initial access. In most such systems, access to the reservation slot (or transfer of initial access information) is based on the slotted- ALOHA principle.

The reservation schemes are preferred over random access techniques because of the reduced probability of collisions. During this study, a statistically multiplexed multiple access technique known as Advanced-Time Division Multiple Access (ATDMA) is used. Some of the many attractive features of ATDMA include, adaptive channel coding schemes, seamless channel capabilities (traffic channels would not be reduced to transmit signalling information), ease of inter-connectivity with ATM networks and the ability to support multimedia type of traffic on demand.

3.3 Advanced Time Division Multiple Access (ATDMA)

During the course of this study ATDMA has been studied as the multiple access technique for the wireless network. Based on the World Administrative Radio Conference 1992 (WARC 92) decision, it is clear that with two unequal bands (1885 - 2025 and 2110 - 2200 MHz) it will be difficult to exploit with Frequency Division Duplexing (FDD) only systems[61]. Therefore, the overall system must transport FDD and Time division Duplexing (TDD).

The slots on the up-link of ATDMA are separated into reservation slots ('R' slots), traffic slots ('I' slots) and fast paging-acknowledgment slots ('FP_{ak}' slots). Slots on the down-link are separated into acknowledgment slots ('A' slots), fast paging slots ('FP' slots) and traffic slots ('I' slots). Fig 3.2.1 and Fig 3.2.2 illustrate respective up-link and down-link ATDMA frame structures. Mobiles transmit reservation requests, or random access bursts in 'R' slots whenever a burst of activity commences. If the reservation request is successful and if traffic slots are available, the mobile would be allocated with a 'I' slot(s) immediately. The slot allocation is acknowledged in the paired 'A' slot on the down-link. In case of resources not being available, the reservation request is queued and acknowledged on the 'A' slot. Therefore, the mobile will continue to monitor the 'A' slot until it receives a slot reservation. Therefore, reservation requests are not blocked when all 'I' slots are allocated (as in PRMA). Base station has centralised control over the 'I' slot allocation policy.

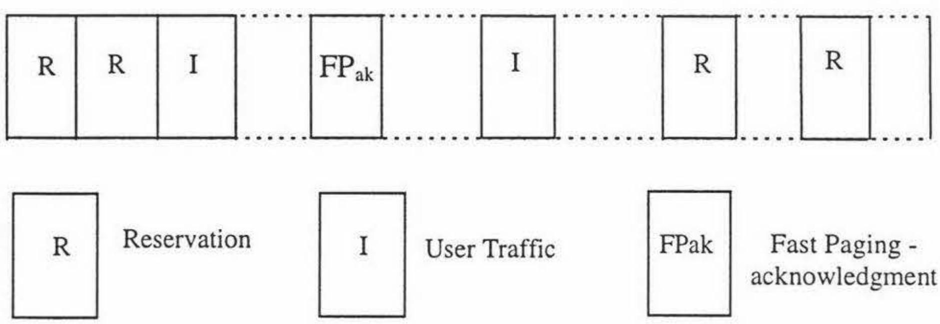


Fig. 3.2.1 ATDMA up-link Frame structure

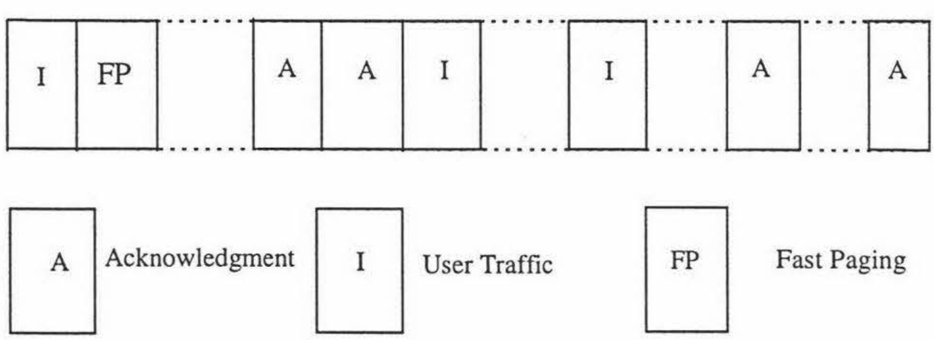


Fig. 3.2.2 ATDMA down-link Frame structure

Resource reservation requests are transmitted on the 'R' slots using the ALOHA mechanism and are therefore subjected to collisions. When collisions do occur, either the capture effect will allow one mobile to gain access or no mobile will be successful. The mobiles which do not get the chance to transmit (which do not receive a positive acknowledgment on the paired 'A' slot) might enter a collision resolution phase, where they would get a chance to re-transmit their reservation requests. The problems caused by the ALOHA based 'R' slot allocation process is examined thoroughly in the next two chapters.

If an access attempt is unsuccessful then the mobile will re-transmit with a given re-transmission probability in the next available 'R' slot until access is successful. In the mean time, packets would be dropped as the packet dropping threshold is exceeded. Packets would be dropped only for delay sensitive traffic sources. When the threshold is exceeded, in the case of voice, the packet would be dropped, whereas in video, the entire video frame would be dropped. Data traffic is modelled as a delay insensitive source of traffic. Thus, no data packet would be dropped. A data terminal would not get a chance to re-transmit its resource request during the current frame due to its delay insensitive nature. Data traffic is modelled with a low permission probability. Thus, voice and video terminals have the highest priority during re-transmission of resource request whereas data terminals have the lowest priority.

When the 'R' slot access attempt is successful, the mobile enters the 'I' slot allocation process and will be queued if there are no 'I' slots available. In the case of voice, the mobile will remain queued, dropping packets if the delay threshold is exceeded, until an 'I' slot becomes available or all the packets in a talkspurt are dropped, in which case, the mobile returns to the silent state. In the case of video, the terminal will remain in the queue until frame drop threshold is exceeded, in which case, the entire video frame would be dropped. In the case of data terminals, no packet would be dropped. When 'I' slots become available the first mobile in the queue may be allocated the slot and this would be acknowledged through an 'A' slot on the down-link. It is also possible to allocate different priorities when 'I' slot allocation is done. Once an 'I' slot is successfully allocated to a mobile, it enters a reservation mode, then the remaining packets of the talkspurt, video frame or the data block are transmitted on the reserved slot, one packet at a time in each transmission frame. In the cases of voice and video, when the last packet of a talkspurt or a video frame has been transmitted, the slot will be released and made available for other terminals. Data bursts are transmitted in blocks of N packets at a time. After transmitting N packets of the data burst the 'I' slot will be released. For the data terminal to transmit its remaining packets it should re-initiate the 'I' slot allocation process. This process would continue until all the packets of the data burst are transmitted.

ATDMA is one of the access protocols that could support adaptive channel coding where transmission could be chosen from a set of coders of varying robustness to suit the quality of the channel [16]. For each bearer service type (a bearer is an independent uni-directional radio connection) there is a transport mode, that is by a certain configuration of the transport chain, (modulation depth, error control code, and source coder) which guarantees a given performance for a given maximum signal-to-noise in an interference limited condition. Different bearer services and corresponding parameters are shown in Table 3.1 and in Table 2.2. The modes are selected by the "link adaptation" process, which changes modes to satisfy the necessary performance parameters. The change of mode may also result in a corresponding change in radio resources allocated to the traffic channel. If the quality of the channel is good (high Carrier to Interference Ratio (CIR)) a coder with higher net source coder rate can be used. If the CIR decreases, a coder with a lower net source coder rate and better error correcting capabilities can be used. Each mode of transmission has an upper and lower quality threshold. If the average of channel quality taken over an updated period is below the lower threshold, a change to a more robust mode of operation is made. If the averages are greater than the upper threshold, a less robust mode is used. The link adaptation algorithm interacts with power control to ensure that minimum power is used to enable a burst of particular target quality to be received properly. Although channel coding techniques were not investigated during this study, block and convolution coding methods adopted in GSM [48] could be used for ATDMA, since similar modulation techniques (Gaussian Minimum Shift Keying (GMSK) and binary offset Quadrature Amplitude Modulation (QAM)) are used in the two systems [61].

Service	Design Constraint	Performance Targets
Speech	Delay < 20 ms	Decoded BER , 10E-3
Video	Delay < 200 ms	Decoded BER < 10E-6
Delay Insensitive Data	Zero Packet Loss	Av. delay < 50 ms 90% Delay < 100 ms

Table 3.1 Parameters for assumed list of ATDMA bearer types.

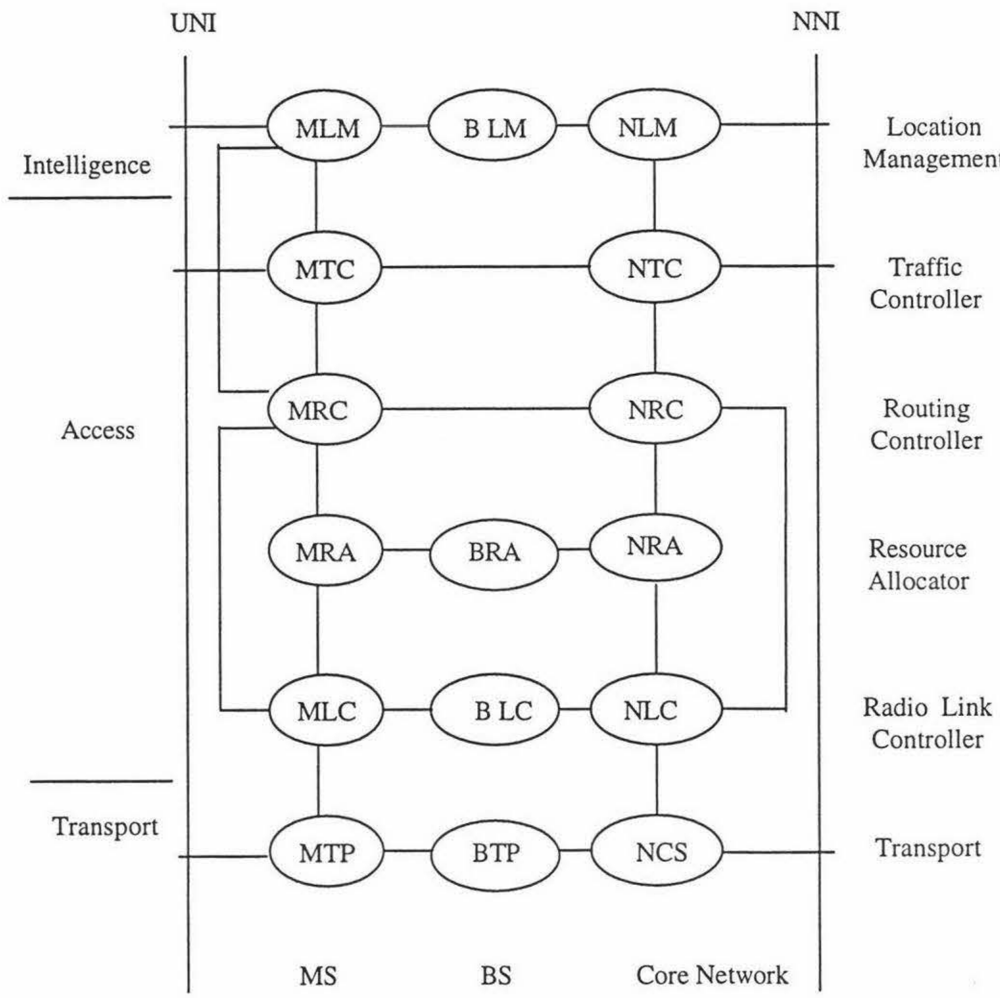
3.4 ATDMA Functional Model

The functional model defines how the ATDMA protocol interacts with different entities of a wireless network. The functional model includes 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 3.2 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 was based on the ATDMA functional model. Therefore, it is important to discuss the operation of the above functional model in detail.

3.4.1 Transport (TP)

Handles all radio and fixed network transmission functions. This group models the radio transmission functions between the mobile (MTP) and the base (BTP), and then through the fixed connection to the central network combiner and switching group (NCS). The transport mechanism takes care of all the information passed between different elements of the functional model. The transport mechanism for ATDMA is developed with the idea of incorporating ATM on the fixed network.

In accordance with the OSI model the transport group could be defined by a three layer structure. The upper network layer of the radio part supports user and control (signalling) information. Fig. 3.4.1 presents a view of this protocol model in the user plane while Fig. 3.4.2 presents the control plane. Two different protocol models (specially through the fixed network) are needed since user and signalling information are modelled to be transported in two different networks. User data is transported in an ATM based network, whereas the control data could be transported in an ATM/CCSS7 based network. ATM/CCSS7 has powerful switching implications on the overall network [24] since, ATM switches are used as Signalling Transfer Points (STPs) [52] in the CCSS7 network.



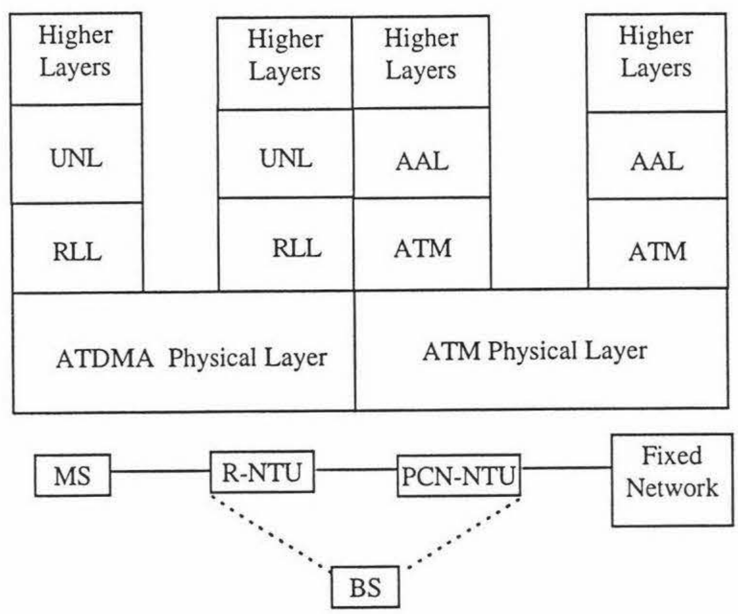
MTP : Mobile Transport Part, **BTP** : Base Transport Part, **NCS** : Network Combiner & Switching, **MLC** : Mobile Link Controller, **BLC** : Base Link Controller, **NLC** : Network Link Controller, **MRA** : Mobile Resource Allocator, **BRA** : Base Resource Allocator, **NRA** : Network Resource Allocator, **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

FIG. 3.3 ATDMA Functional Model

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 3.2 Signalling messages passed between the interfaces of functional elements of functional model given in Fig. 3.3 [16, 61]

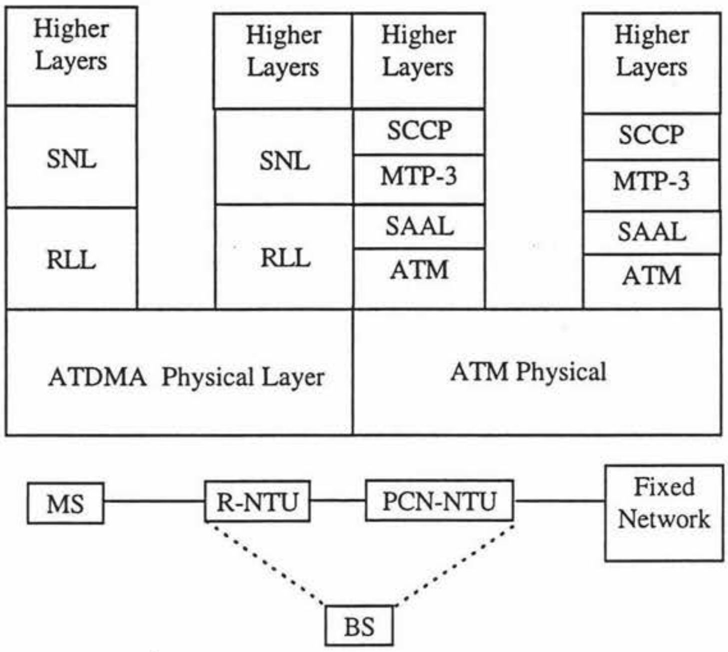
Some of the signalling generated by the wireless network would terminate at the BS level whereas, the others would be transported beyond the BS. As an example, channel set-up and release related functions are performed by BRA, therefore, there is no need for these information to be transported beyond the BS level. However, if the resource set-up or release on the wireless network requires similar functions to be performed on the fixed network, the outcome of the activities on the wireless network have to be passed to NRA. Then RAs use internal signalling to inform LCs regarding resource allocation or release. They would in turn instruct the transport mechanism to start or stop transmission. Activities involved in resource set-up and release are shown in the form of flow diagrams (Fig 3.14.1 and Fig 3.14.2). Similarly, as shown in Fig 3.14.6, during link adaptation if resources assigned on the fixed network need to be changed, then those information would be passed between BRA and NRA. As shown in Fig 3.14.5, power control related information do not need to be transported beyond BS. Functions related to handover (e.g. updating VLR, HLR) and authentication are passed between MRC and NRC, resulting in information travel beyond the BS (Fig. 3.14.4).



RLL : Radio Link Layer, **UNL** : User Network Layer, **RLL** : Radio Link Layer, **ATM-Layer** : Asynchronous Transfer Mode - Layer, **AAL** : ATM Adaptation Layer, **MS** : Mobile Station, **BS** : Base Station, **R-NTU** : Radio - Network Transfer Unit, **PCN-NTU** : PCN Network Transfer Unit

Figure 3.4.1 ATDMA user protocol model

ATDMA Physical Layer (ATDMA-PL): Transports ATDMA bursts over the radio channel. This layer performs link controller related functions such as power control, time advance, measurements on received signal strength, time alignment and quality of bursts (ATDMA packets) [61]. The characteristics of the physical layer depends upon the cell type (e.g. microcellular, picocellular) but, it is independent of the supported service type. Therefore, the ATDMA physical layer will offer a fixed payload to the link layer. In an ATDMA system, this concept is achieved by maintaining a constant size payload for each ATDMA burst.



RLL : Radio Link Layer, **SNL** : Signalling Network Layer, **RLL** : Radio Link Layer, **ATM-Layer** : Asynchronous Transfer Mode - Layer, **SAAL** : Signalling ATM Adaptation Layer, **SCCP** :Signalling Connection Control Point, **MTP -3** : Message Transfer Part - 3, **SCCP** : Signalling Connection Control Point, **MS** : Mobile Station, **BS** : Base Station, **R-NTU** : Radio - Network Transfer Unit, **PCN-NTU** : PCN Network Transfer Unit

Figure 3.4.2 ATDMA signalling protocol model

Radio Link Layer (RLL): This layer takes care of transmission and reception of individual burst over the radio channel. This layer would also perform some link adaptation and provide measurements on burst quality and error detection [16].

Signalling Network Layer (SNL): Conversion of signalling messages (e.g. segmentation of signalling information, rate adaptation) into ATDMA radio bearer types and support for content-based routing to various locations in the fixed network [16]. This layer is in parallel with UAL. Signalling on the layers above SNL could be done using Q.2931 m (this is the mobile version of Q.2931). Q.2931 is the modified access signalling protocol for B-ISDN. Q.2931 describes messages, protocols and features for user network signalling across the ATM UNI.

User Network Layer (UNL): This is the user plane of the network layer. It adapts user traffic into ATDMA bearer types. In other words, this layer performs the adaptation of user traffic (segmenting into suitable packet sizes, rate adaptation) in order to be transported through the wireless network (e.g. ATM format to ATDMA format).

Signalling ATM Adaptation Layer (SAAL): This supports AAL functions for signalling messages. SAAL in particular adapts AAL5. AAL5 is used because it supports variable bit rate services with error correction capabilities above the ATM layer. Therefore, SAAL Convergence Sublayer (CS) consists of a Common Part Convergence Sublayer (CPCS) and a Service-Specific Convergence Sublayer (SSCS) [53]. SAAL is used to encapsulate Q.2931 [54, 55, 56] messages into ATM cells in the signalling network (with help of AAL5 SAR) [50]. SAAL would also support the exchange of variable length messages above the ATM layer (hence, the need for AAL5) [53, 57]. SAAL also adapts higher layer signalling into ATM cells [50]. SAAL services are incorporated with ATM services on the data link layer.

Message Transfer Part - 3 (MTP-3) : This is incorporated to support CCSS7 functions. It provides functions and procedures related to signalling message routing and network management between signalling points, which are nodes of the signalling network [27]. When signalling originates or arrives at a STP, the choice of the particular signalling link on which it is to be transmitted is made by the message routing function. On the other hand, the purpose of the signalling network management function is to provide reconfiguration of the signalling network in case of signalling link or signalling point failure [63]. The idea is that, when a failure occurs, the reconfigurations are carried out so that the messages are not lost, duplicated or delays become excessive.

Signalling Connection Control Part (SCCP): SCCP improves the MTP-3 services and performs function equivalent to OSI's network layer. One of the improvements provided to MTP-3 by SCCP is the ability to handle messages with global title addresses (such as free phone numbers) that are not directly useable for routing to MTP [63]. SCCP is also used to set up temporary and permanent signalling connections (a virtual channel through the signalling network).

ATM layer: RLL information is mapped into this layer to be transported through the fixed network. The packet structure consists of a 48 byte payload and a 5 byte header. In general this layer performs generic flow control, cell header generation/extraction, cell VPI/VCI translation and cell multiplexing and de-multiplexing.

ATM Adaptation Layer (AAL): This layer is adopted for ATM to support many kinds of services with different traffic characteristics and system requirements. AAL consists of the Convergence Sublayer (CS) and the Segmentation and Reassembly (SAR) layers. SAR segments higher layer information into sizes suitable for the information field in an ATM cell. It also reassembles the contents of ATM cell information fields into higher layer information. On the other hand, CS handles user message identifications such as timing/ clock recovery.

3.4.2 Link Controller (LC)

The LC maintains the quality of the bearers forming that link while using minimum resources and consequently, causing minimum disruption to other links. The quality of the bearers are maintained by the LC by changing transmission modes. When the bearer qualities deteriorate (low CIR) a mode with lower net source coder rate and better error correction capabilities will be adopted. Once the channel quality improves the bearer will be shifted to the original mode. LC would also provide measurement information (e.g. bearer quality) for all MS. The architecture of LC group is shown in Fig 3.5.

Measurement Entity (ME) interfaces with the transport group's receiver chain's monitoring points and collects and processes all quality measurements associated with a link for use by all of its client control processes (e.g. slot controller, bearer controller, common controller).

Common Controller (CC) performs all functions that do not relate to an individual bearer. Its main responsibility is to set common timing advances (for synchronisation needs) for all active slots used by the MS for a given BS.

Slot Controller (SC) offers dedicated control for each active slot by implementing a closed loop Automatic Power Control (APC) scheme. Channel quality measurements in terms of CIR are frequently exchanged between MLC's and BLC's slot controllers (this information is passed with the help of ACCHf channels). If this ratio is below some threshold then BLC's SC calculates the new power that is required to restore the CIR. This mechanism is called APC.

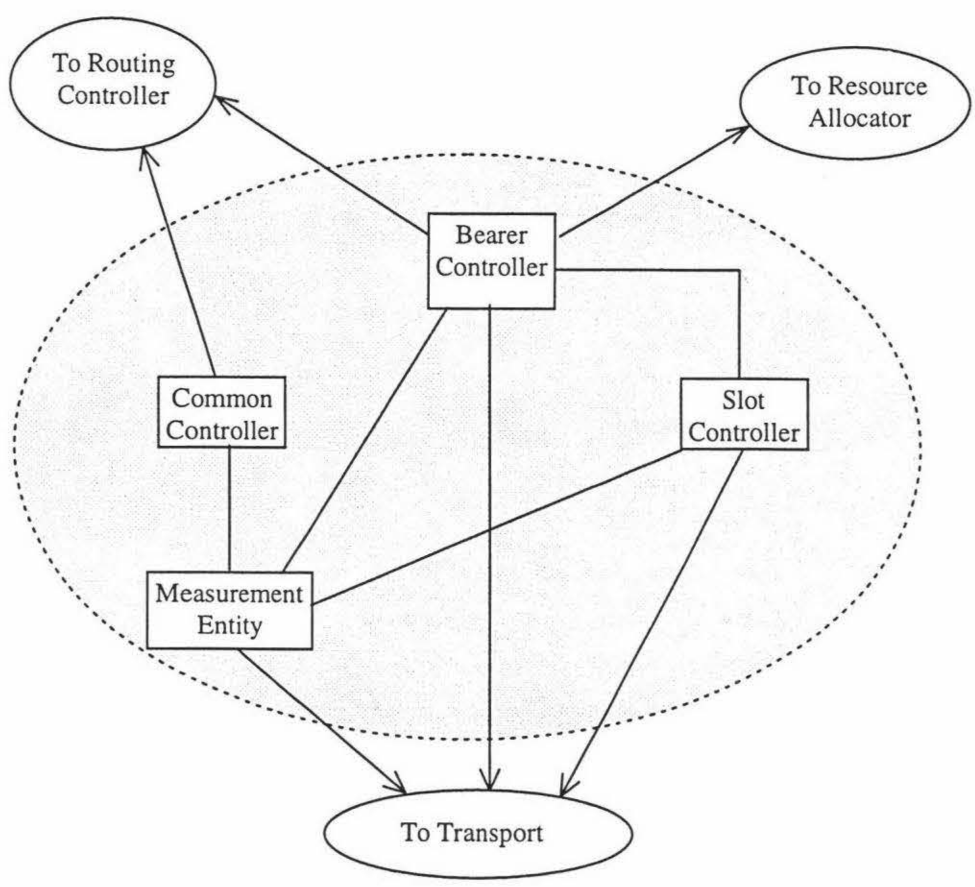


Fig 3.5 Link Controller Architecture

Bearer Controller (BC) is created and closed by the LC for each TCH and DCCH. BC functions would vary with bearer type but generally, it is responsible for the overall quality (maintains CIR) of the bearer. It intervenes whenever SC fails to maintain the service quality. The BC also implements link adaptation (LA) by requesting different number of slots per frame for new modes when necessary (e.g. requesting more resources). The request is made to RA.

3.4.3 Routing Controller (RC)

The Functions of RC include control of handover and selecting a suitable base station. LC will provide the required measurements to RC, when needed, to make the handover decisions. After the handover decision has been made, RC will configure LCs to manage the new link. Usually mobile initiated handover is used, since handover can still be performed and the call continued even if the previous link is suddenly lost. However, network and BS driven handover is maintained to allow handover for other reasons (e.g. closing down of a BS).

Path loss, received signal strength, link quality, cell type, cell loading and distance are used in the cell selection algorithm. One of the unique advantages of the ATDMA control structure is that during handover, a bi-directional Dedicated Control Channel (DCCH) will be established to carry the RC signalling so that existing calls need not be interrupted [16].

Call set up and handover are treated in a similar way in RC. During call set up, when a new call has been selected, a request is passed to that base stations' Resource Allocator (RA) which will accept or reject the call based on the current resource allocation priorities in the target cell. RC architecture is illustrated in Fig 3.6. During resource allocation a priority scheme could be applied for terminals. In such a scheme existing inactive voice calls trying for an 'I' slot for their new talkspurt would have a higher priority over handover voice calls. This would be followed by terminals trying for a traffic slot for their new video frames, handover video terminals and delay insensitive data terminals (both handover and inactive) respectively.

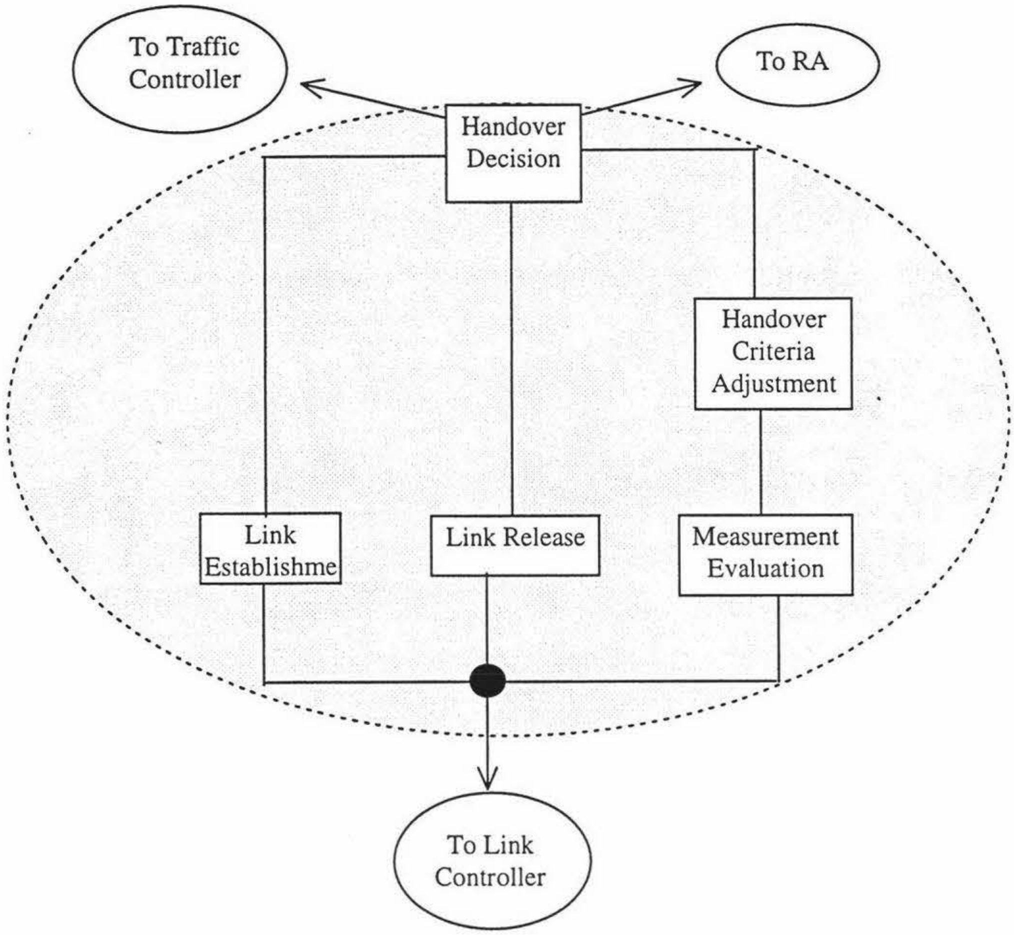


Fig 3.6 Routing Controller Architecture

3.4.4 Radio Resource Allocator (RA)

The RA performs one of the most important aspects of the ATDMA protocol, 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 [61].

RA architecture is illustrated in Fig 3.7. 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 the MS and the 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.

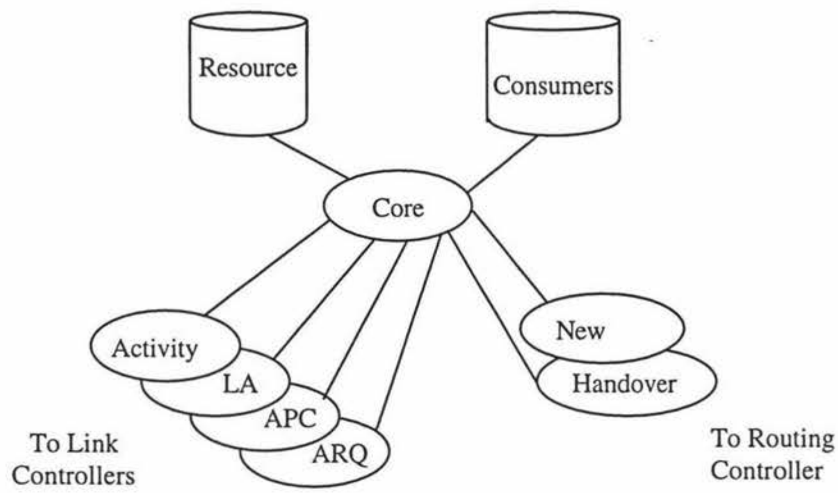


Fig. 3.7 Resource Allocator Architecture [13]

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, 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 calls (e.g. calls going through the silence mode) and for incoming handover requests. Different thresholds should be used to ensure that priority is given to existing delay sensitive inactive and handover calls.

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 acknowledgment channels.

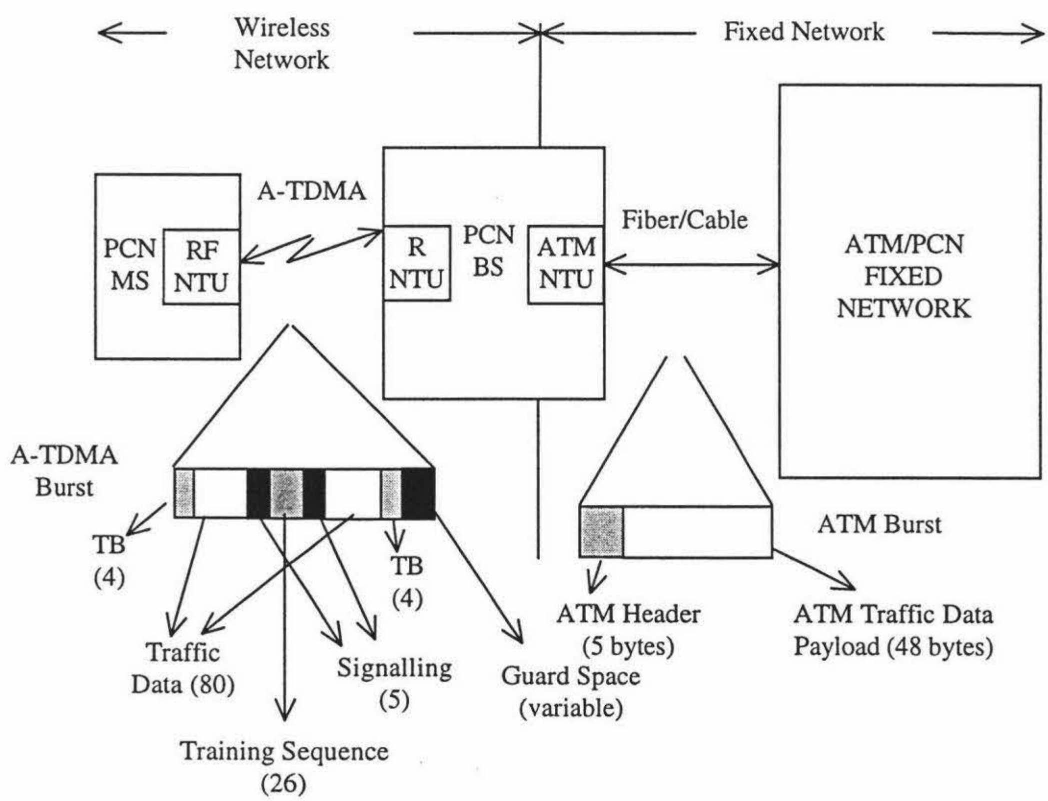
3.4.5 Location Manager (LM) and Traffic Controller (TC)

Most of the data bases used in the LM functions should be located in the fixed network. This needs to be done to keep the network cost to a minimum. In detail, databases could be connected to a MSC to be used by all MSs attached to all BSs (Fig 3.13.1). These databases include the Visitor Location Register (VLR), the Home Location Register (HLR), active and inactive terminal lists and any other database used for authentication processes. They are controlled by the Service Control Points (SCP) of the CCSS7/ATM based signalling network. Any queries of mobile stations LM (MLM) and base stations LM (BLM) are serviced by SCP (or NLM in the Fig 3.3).

The TC is responsible for call control functions on the fixed network. The main responsibilities include VCI/VPI set-up and release.

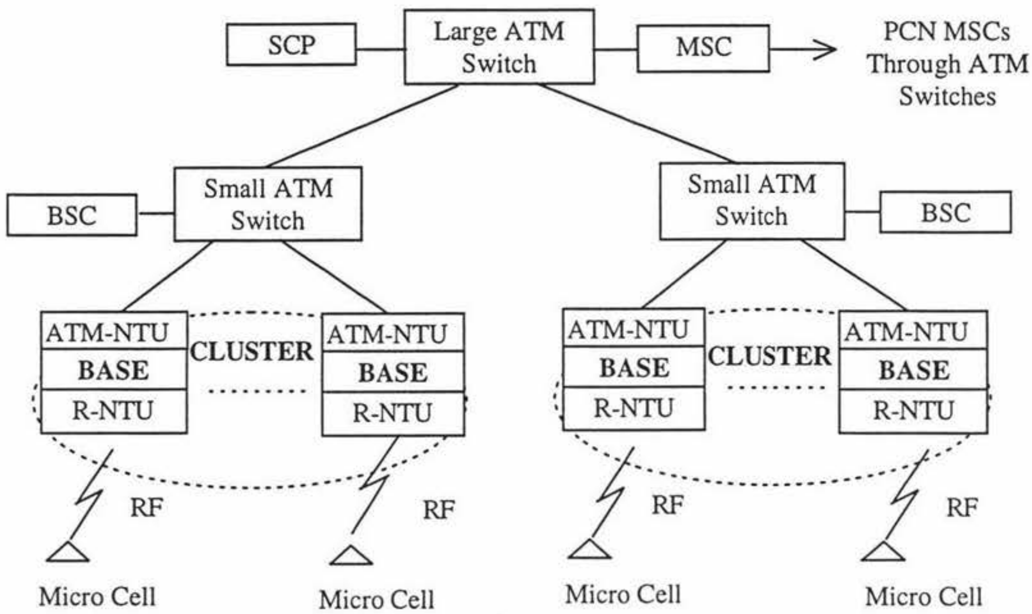
3.5 ATDMA based PCN Architecture

A PCN network should be developed in accordance with a variety of PCN requirements (as presented in section 4 and 5 of chapter 2). Due to the stringent service requirements caused by different traffic types with varying requirements and the dynamic nature of PCN traffic, a packet based network could be suitable for PCN development [21]. Packet based networks provide higher capacities since they allocate resources only when necessary. The wireless segment of the network is based on the ATDMA protocol. Traffic from the wireless network should be passed through the fixed network using appropriate protocols. Since ATM is a broad-band transmission protocol based on the B-ISDN architecture, it is expected to support a wide range of applications (many traffic types with varying requirements) through high-speed and flexible multimedia communication capabilities [22, 23, 24, 45]. Flexible bandwidth allocation, efficient multiplexing of traffic from different sources, end-to-end connectivity of broad-band services over wireless systems and fixed systems and the suitability of available ATM switching equipment for inter-cell switching are some of the numerous reasons why next generation mobile systems should adapt packet switched ATM as a backbone network. Therefore, an ATM compatible fixed network has been proposed for this study. An ATDMA/ATM packet interface structure for PCN is illustrated in Fig. 3.8 while Fig. 3.9 illustrates a typical PCN organisation structure. Most of the inter-working functions between ATDMA and ATM would be located at the BS so that all the information passed through the fixed network would be of ATM format. Therefore, the ATDMA compatible OSI stack received at BS-NTU should be mapped into an appropriate ATM compatible OSI stack at the ATM-NTU (The resulting PCN user and signalling protocol models are shown in Fig. 3.4.1 and Fig. 3.4.2, respectively) .



PCN-BS : Personal Communications Network-Base Station, **R-NTU** : Radio-Network Transfer Unit, **PCN-BS** : Personal Communication Network-Base Station, **TB** : Tail Bit **ATM-NTU** : Asynchronous Transfer Mode-Network Transfer Unit

Fig. 3.8 ATDMA/ATM packet interface Structure for PCN



R-NTU : Radio-Network Transfer Unit, **ATM-NTU** : Asynchronous Transfer Mode-Network Transfer Unit, **RF** : Radio Frequency, **BSC** : Base Station Controller, **MSC** : Mobile Switching Center, **SCP** : Service Control Point

Fig. 3.9 Typical Organisation of ATM-Based Backbone Network

According to the diagram shown in Fig 3.8, during a normal burst packets of 160 bit (as in Fig 3.8, 2*80 data bits) payloads are transmitted over the radio channels. In a similar structure proposed in [25, 26], 48 byte ATM cell payloads (or a suitable integer submultiple of an ATM cell (e.g. 24 or 16)) are proposed as the basic data unit within the wireless PCN. This would result in a transparent interface to an ATM backbone network. However, the proposed packet format in [25, 26] is not essential on the wireless PCN. Network designers could use any number of bytes per packet since they can implement ATM adapters at the BS (R-NTU) to form ATM packets before forwarding them to the ATM Network Termination Unit (ATM-NTU).

When packetising ATDMA user data into ATM packet format, it is important to use the fixed network resources in an efficient manner. To achieve this, ATM packets should contain more user information bits and less filling bits. An ATDMA burst would carry 160 bits of user information. These should be mapped to the 384 bits of the ATM payload. It is possible to map up to two ATDMA bursts (from different users) into one ATM cell. This concept is illustrated in Fig 3.10. The control information field would carry information such as call origination/termination addresses, type of information and channel control related information. VPI/VCI for the ATM cell and generic flow control functions would be provided by the ATM header. When packetising information from two channels over one ATM cell, it is important for the two channels have the same VPI values.

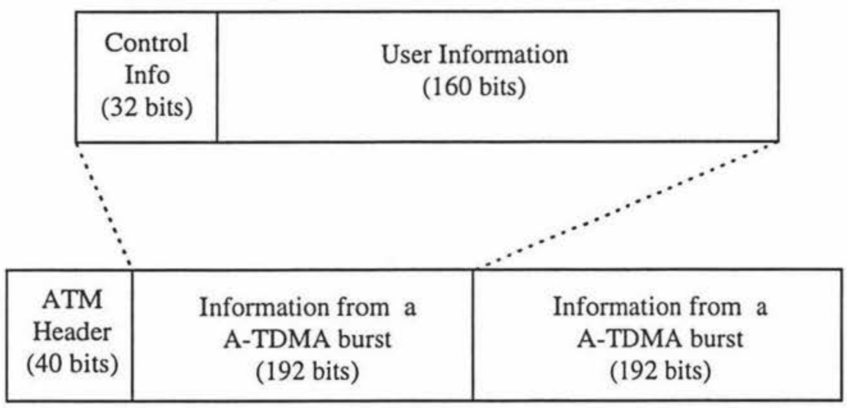


Fig. 3.10. Packetising ATDMA information in an ATM cell

3.6 ATDMA/ATM Signalling

Signalling is one of the most important aspects of any communication network. In a packet switched network information is packetised into appropriate packets but without proper signalling infrastructure information packets can not be transported. Also, if a proper signalling structure is not adopted, the user data quality would deteriorate (e.g. low CIR). Therefore, a signalling structure for the ATDMA/ATM system needs to be developed based on the ATDMA functional model.

3.6.1 Signalling Requirements for an ATM/ATDMA based PCN

The establishment, maintenance and release of radio channels as well as Virtual Channel Connections (VCC) and Virtual path Connections (VPC) on the ATM network for transfer of information.

- Channel establishment must be on demand (e.g. for the duration of a talkspurt, video frame or a data burst) and should comply with the requested connection characteristics (e.g. bandwidth, quality of service).

- Priority based channel establishments must be handled since inactive to active traffic slot allocation (e.g. silence period to start of a talkspurt) must be much faster than an initial call set-up.

- Depending on the requirements of the traffic source, the signalling network should be able to negotiate during call set-up (e.g. if required, allocation of slots with high CIR) .

- Link adaptation and power control process related signalling should be handled.

- The measurement of the quality of the link in terms of CIR and perform appropriate handover functions during active (e.g. during talkspurts) as well as during inactive (e.g. during silences) periods.

- The handling of mobility and power control related functions.

3.6.2 Signalling System Number 7 (SS7)

As some features of SS7 are incorporated in the ATM based signalling network, it is important to briefly explain the SS7 structure in a wireless network environment. The basic parts of SS7 protocol and the corresponding OSI layers are illustrated in Fig 3.11.

Message Transfer Part (MTP) 1 corresponds to the OSI physical layer and it defines physical, electrical, and functional characteristics of the signalling link connecting SS7 components. In the architecture studied on this project, this layer corresponds to the ATM physical layer (as shown in Fig. 3.4.2).

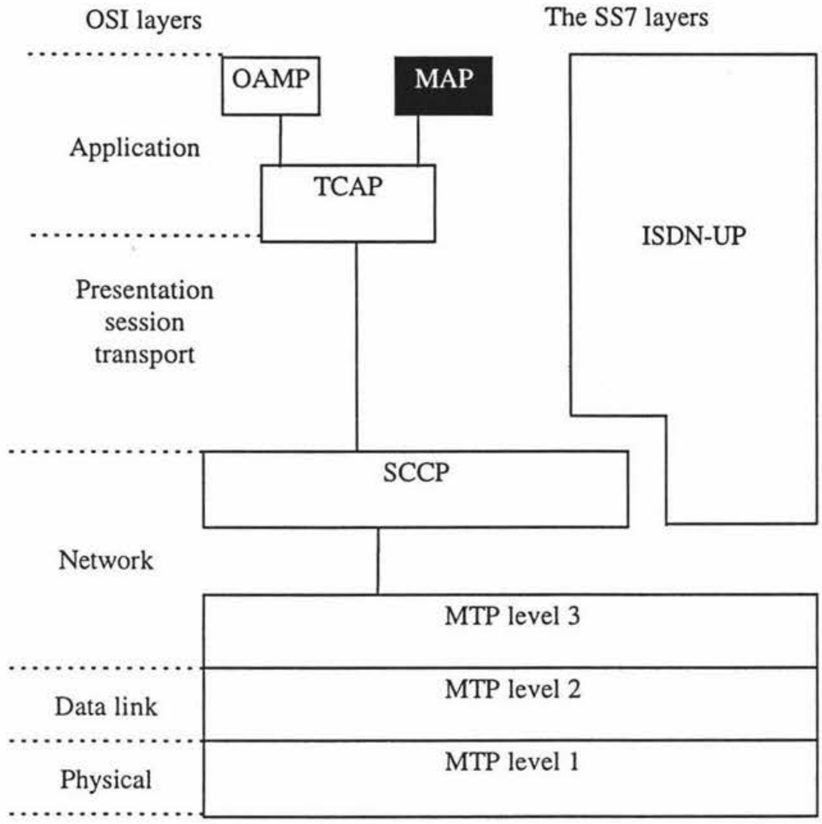


Fig. 3.11. SS7 [27, 28,46] signalling protocol

MTP 2 corresponds to the OSI data link layer and is used for reliable transfer of signalling messages between two directly connected signalling ports [28]. As shown in Fig. 3.2.2, this layer consists of ATM and Signalling ATM Adaptation Layer (SAAL). At this particular level, the signalling information carried through wireless network has to be converted into SS7 format in order to be transported in the fixed network. The opposite is the case when the signal is destined to the MS from any part of the fixed network. These conversions should be done at the BS level along with the data traffic which needs to be converted into ATM format.

MTP 3 corresponds to the OSI network layer and provides functions and procedures related to message routing and network management [46, 28].

The Signalling Connection Control Part (SCCP) provides a variety of different network layer services to meet the needs of the network service part (network service part consists of physical, data link and network layers).

The Transaction Capabilities Application Part (TCAP) provides mechanisms for transaction-oriented (as opposed to connection-oriented) applications and functions [28].

The Integrated Services Digital Network User Part (ISDN-UP) establishes circuit switched connections (e.g. call set-up/release) [27]. It also passes signalling information to each switching point involved in a call connection.

The mobile Application Part (MAP) is implemented on top of SS7. IS-41 [66] protocol that helps the applications of TCAP are located in MAP [27]. Some of the applications supported by IS-41 and TCAP include mobility management and database management. As in GSM, MAP could define the operation for ATDMA between the Mobile Switching Center (MSC) and the telephone network as well as the MSC, the Home Location Register (HLR), the Visitor Location Register (VLR), and the equipment identity register [58].

3.6.3 Control Channels for an ATDMA based System

In order to operate multiple traffic channels smoothly in any communication system a well structured signalling protocol is needed. This section concerns the issues of control channels on the wireless ATDMA network. The control channels adopted are listed in Table 3.3

The Dedicated Control Channel (DCCH) is temporary allocated for signalling during call set-up, backward or forward handover. A DCCH channel will always exist with an associated ACCH channel (as is the case with Traffic Channels (TCH)). ACCH is used for power control, link adaptation and resource change requests.

The Common Control Channel (CCCH) consists of Fast Paging Control Channel (FPCH), Access Grant Control Channel (AGCH) and Random Access Channel (RACH). The RACH (transmitted on 'R' slot) is shared by all users for channel access on the up-link whereas the FPCH is used for paging during channel access on the down-link. It is important to remember that the RACH and the FPCH are accompanied by respective acknowledgment channels on the down-link and the up-link (They are transmitted on 'A' and FP_{ak}, respectively). AGCH ('A' slot) is used to inform the MS of the allocated slot by BS during up-link channel allocation. The FP_{ak} slot is used on the down-link to perform similar functions.

Control Channel Type	Associated	Associated	Dedicated	Common	Broadcast
Functional group	LCCH	ACCH	DCCH	CCCH	BCCH
Link controller	Time advance measurements and commands ------- Adjacent cell measurements	APC and LA commands Link measurements			
Routing controller			Handover execution		
Resource allocator		Bandwidth change		Resource request and grants initial access	Frame structure and other cell information
Traffic controller			call set-up		

Table 3.3 Functions of Control Channels [16, 61]

The Broadcast Control Channel (BCCH) is used in the down-link for broadcasting synchronisation, frequency correction, cell characteristics and status.

The Leash Control Channel (LCCH) is a permanent supervisory control channel that is used to keep control of each mobile that has a connection. This channel is active even during an inactive period of the parent channel (e.g. silences, between video frames or data bursts). This channel occupies a low but guaranteed bandwidth. The LCCH is essential because even after a long period of inactivity from the MS [61] it maintains time advances for request slot bursts.

The Associated Control Channel (ACCH) consists as a pair of channels for each TCH and DCCH [61]. ACCHf (forward ACCH) carries information in the same direction as TCH or DCCH on the up-link. ACCHf information is carried in the 10 signalling bits associated with a normal burst (as in Fig 3.12). ACCHr carries ACCH information on the down-link. Thus, each up-link TCH or DCCH would be associated with an ACCHf on the up-link whereas an ACCHr would be used on the down-link.

3.6.4 Operation of The ATDMA Control Channels

Mobile initiated access and set-up (initial) - At the start of a voice, video or a data call the mobile will make a request using a Random Access Channel (RACH). This then results in the setting up of a DCCH and a LCCH. The DCCH is used to pass authentication and call set-up information between the MS and the BS. The LCCH is a permanent control channel that is used to keep control of the mobile.

Mobile initiated access and set-up (within a call) - Traffic slot allocation and set-up in-between talkspurts, video frames and data blocks should behave in the same way as the initial access but higher priority is given to active terminals. In both situations 'I' slots are allocated after a successful 'R' slot allocation. During the initial access a RACH is used to request a DCCH and a LCCH whereas during subsequent accesses (when the status of a terminal becomes active from inactive), RACH would only be used to request a DCCH since a LCCH already exists. During channel access, priorities should be given to existing voice and video terminals over data terminals and all new calls. Also higher priorities would be given to existing voice and video calls over handover calls [16, 61].

Such priority information could be passed to the BS by the RACH (traffic types could be used by the BS to find out priority).

Admission control - When a channel request has been made it would be passed to the base stations Resource Allocator (RA) to admit the call. After examining parameters such as the loading of the cell and the expected loading (based on the number of active and inactive terminals in the system) of the cell, the new call is accepted or rejected. The call would be accepted if the particular channel quality does not degrade below a pre-defined threshold. It would also be added to the list of current calls to ensure that capacity is reserved.

Idle Period - The Idle period is the time between the last DCCH function and access of 'I slots. Once the functions of DCCH are performed, it may be released until a suitable 'I' slot (traffic channel) is available. The link would be maintained by using a low capacity LCCH. This would help to allow timing advances to be maintained in order to precede with quick access and report measurements. If a handover is needed in this period MS would detect this (with the help of LCCH) and a DCCH would be set up for this purpose very quickly.

Activation of a call - When the mobile station is ready to communicate, the transfer of information could begin as soon as the call is answered at the terminating end. The LC at the terminating end will be informed that the down-link bearer should be activated. Then LC at the terminating end would pass its request to the RA. When RA allocates a traffic slot, a Fast Paging (FP) slot would be used to inform the MS at the terminating end. Once this sequence is completed the communication can begin.

Resource release - When there is no information to transmit on a currently active physical bearer, Mobiles Transport (MTP) would detect this situation and inform Mobiles Link Controller (MLC). As a result, the physical bearer would transmit background noise or an 'end of transmission' indicator. The Base Link Controller (BLC) would then be notified that the bearer is no longer needed. Following this, the LCs at both the ends would inform their respective RAs and the slot occupied by the bearer would be released.

Silence periods - When a silent period is detected, background noise information could be transmitted. It is also possible to transmit a packet that would indicate the end of a transmission. This will also serve to indicate the end of a talkspurt, data burst or a video frame [16]. The LCs at both ends will inform the RA that the slot is no longer needed and inform the transport to stop transmitting. Then LCCH will maintain the link in a similar way as in the idle period. At the start of the next takspurt, video frame or a data block, a new 'I' slot will be searched using the standard access procedure.

Link control - Automatic Power Control (APC) and link adaptation (LA) techniques will be used to maintain the link quality. Measurements of link quality and received power will be used to control the transmit power [16] (this information is passed with the help of ACCH signalling). If a mode change is made requiring more resources (e.g. more 'I' slots), a request to the RA would be made. However, priority would be given to channel allocations at the start of a talkspurt or a video frame over new calls.

Handover - As the mobile moves further away from its base station, path loss would increase and CIR would decrease. This would be looked after by the APC and the LC. In this case, the service quality would be maintained by increasing the transmitted power. This inefficient situation can be detected by the handover algorithm in the Mobile Routing Controller (MRC) [16]. The detection mechanism is mainly based on path loss measurements of adjacent base stations. RC will allow an optional period of parallel connections to two or more base stations to be supported within a common handover algorithm. A handover trigger will normally be made by the mobile station. In this way handover can still be performed and the call continued even if the previous link is suddenly lost. However, support for a BS or a MSC driven handover is maintained to allow handover for other reasons (e.g. shutting down of a BS, distribute MSs evenly among neighbouring BS). One unique feature of the ATDMA concept is that during handover a dedicated bi-directional control channel (DCCH) will be established to carry all RC signalling so that the existing call need not be interrupted. This is a significant advancement, since in the second generation mobile networks (e.g. GSM), user traffic is reduced to carry handover related signalling [48].

Handover During Idle or Silent Periods During idle and silent periods, LCCH maintains the link between the MS and the BS. During this period if a handover is needed (due to the deteriorating link quality), MS would detect this situation and a DCCH would be set up quickly for this purpose. Then the handover procedure would be carried out as stated earlier. During the handover procedure, all appropriate registers including the HLR and the VLR would be updated. Hence, the MSC would know the new location of the mobile for future correspondences. Once the handover has been completed, the DCCH may be released and once again a LCCH would be set up.

3.6.5 Proposed ATDMA Burst Structures

ATDMA is a Time Division Multiplexed (TDM) frame and a slot based structure. The frame and slot size would depend on many factors. These include end-to-end delay, channel characteristics, throughput, etc. Frequency division multiplexing is used for UP and DOWN links. One carrier frequency would be divided in time to make way for separate slots or channels. The number of channels (slots) supported by a carrier frequency depends on the transmission bit rate. Some of these channels (slots) would be used to transmit user data ('I' slots would be used for TCH) while the others would be used to transmit access information ('R' slots would be used for RACH).

A normal burst in ATDMA would carry 160 bits of coded data [(source coder rate of 16 Kbps) * (frame duration of 10 ms)]. The packet structures for a normal burst at a transmission speed of 'T Mbps' is shown in Fig 3.12. This structure applies to TCH and DCCH. As in GSM, this burst could support two physical channels (80 bits of data and 5 bits of signalling supports one channel). On the other hand, these bits could be combined to support one channel (160 bits of data and 10 bits of signalling). As the transmission speed increases, the only difference to the packet structure comes in the form of extra guard space. This is because, in a given fixed time interval, more bits would be transmitted at higher speeds (the guard space is kept constant at 13 μ s) thus requiring more bits for protection against overlapping of adjacent bursts.

Header Bits (HB) and Tail Bits (TB) are added for equalisation purposes. They allow a Viterbi-type equaliser to the start and the end of a known state or help a decision feedback type to read just the feedback coefficients [61]. Considering the channel access delay, 4 bits are used for HB and TB.

The training sequence is a fixed bit sequence known to both the mobile and the base station, which lets radios synchronise their receivers with the burst. Although all the timings are well defined, a training sequence is needed to combat the affects of multi-path fading (on top of the diversity techniques). In other words, when different versions of the same signal arrive at the receiver with slight delay variations to each other (due to different propagation paths of the same signal), there would be overlapping signals at the receiver with nearly the same timing and power levels. The result would be a delay spread of the recovered data at the receiver. In order to separate different signals and make recovered data clear, the training sequence is used. An equaliser (which is a part of a receiver) is needed with the training sequence to clean up the distorted data. An equaliser is a filter that mix different signals together into a single non-ambiguous signal. The equaliser does this by first looking at the distorted training sequence in each time slot it sees and then adjusting its own filter characteristics to get the original, clean training sequence back again. Once the training sequence is recovered the coded data bits need to be recovered as well. The training sequence essentially lets the equaliser demodulate the bit content of the data section in the burst. Generally the training sequence is placed in the middle of the burst to minimise the fluctuations which the receiver has to cope up with .

Signalling bits carry information on the Associated Control Channel (ACCH). An ACCH will always exist with Traffic Channels (TCH) and Dedicated Control Channels (DCCH) [61].

Guard space is needed for reasons such as timing alignment, time-dispersion due to multi-path propagation and power ramping [48, 61]. Although no information is transmitted through the guard space, it is included in the burst as overheads.

HB 4	Data 80	Sig 5	Training sequence 26	Sig 5	Data 80	TB 4	Guard 13T

Fig 3.12 Normal burst in ATDMA(for transmission speed of 'T' Mbps)

The RACH channel would follow a different burst structure to a normal burst. This structure is generally known as the random access burst. RACH is based on the slotted ALOHA mechanism and mobiles use such a channel to carry access information (e.g. initial access, handover, start of a new burst during a call). RACH would carry shorter segments of coded data compared to a normal burst. Also since RACH is a one-off burst synchronisation of RACH is of prime importance (RACH would incorporate a longer training sequence). Even if the MS and BS distance is at a maximum (e.g. MS at the border of the cell) the shortened burst should not overlap into any adjacent bursts. This should be done by incorporating longer guard spaces. The other difference to the packet structure compared to a normal burst is in the form of the synchronisation sequence. The synchronisation sequence would have the same significance as the training sequence, but the synchronisation sequence length would be much longer than the length of the training sequence. A longer synchronisation sequence is required, because the receiver needs more information to synchronise the new signal properly [48].

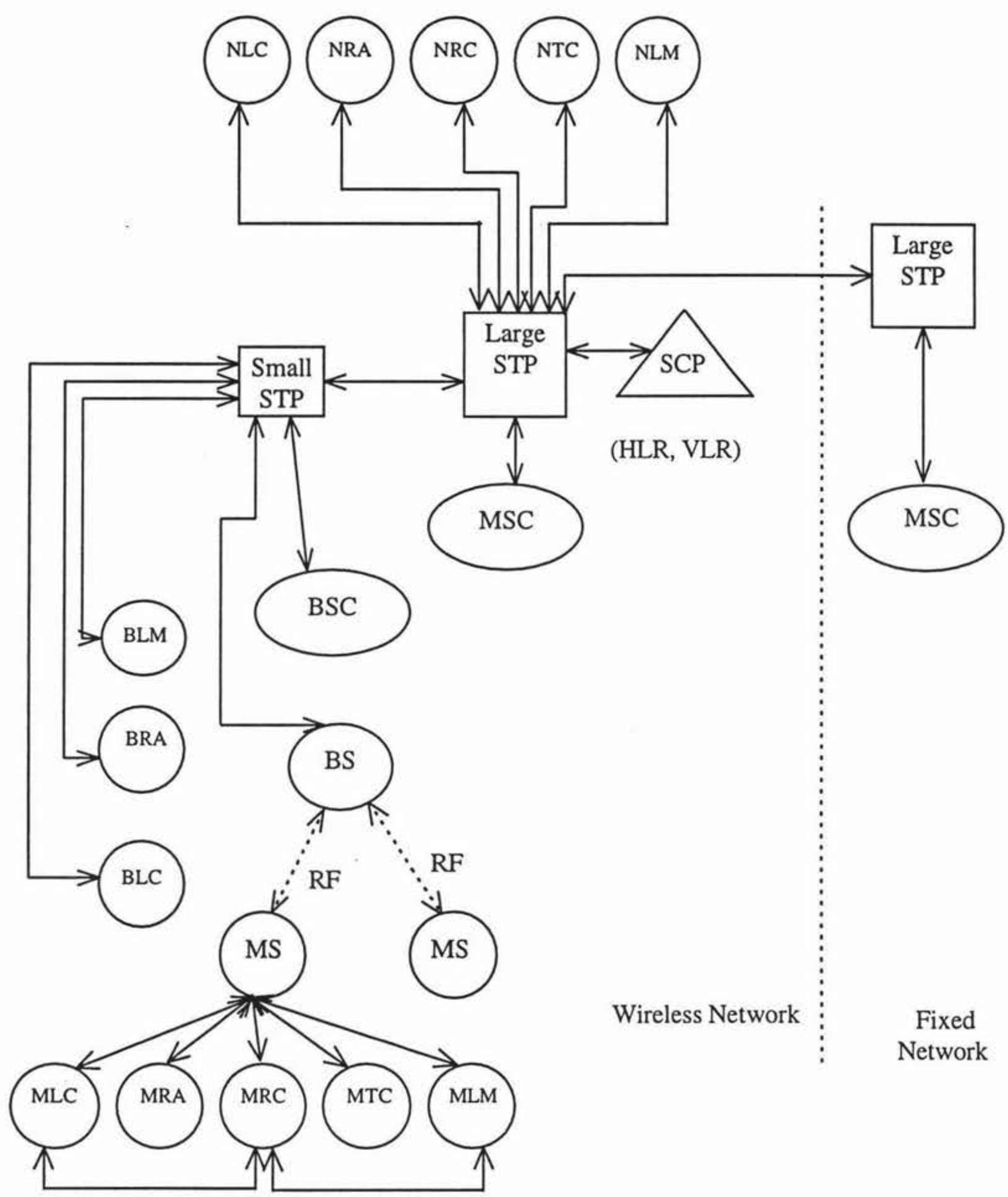
3.6.6 Signalling Structure for ATDMA /ATM PCN

A PCN system consists of wireless and fixed portions. MS to BS and vice versa the control signals are transmitted via control channels. Therefore, some percentage of the available radio spectrum will be occupied by control channels. Some of these control signals are transmitted using in-band signalling (e.g. ACCH) in the developed simulation model, while the others are transmitted as separate control channels (e.g. RACH, LCCH, DCCH, BCCH).

The control signals would be transmitted through an ATM/SS7 based network while user traffic would be transmitted in an ATM network during its transmission through the fixed PCN (as described in section 3.4.1). It is important to remember that the signalling packets have ATM and the necessary SS7 features incorporated into them (e.g. SAAL, SCCP and MTP-3). In this case STP points for CCSS7 would be ATM switches. User and signalling packets would be generated at the R-NTU (Fig. 3.8). PCN could be connected to any other network if necessary, but conversions have to be made at appropriate interfaces. The proposed signalling and control plane is shown in Fig 3.13.1 while Fig 3.13.2 illustrates the traffic plane.

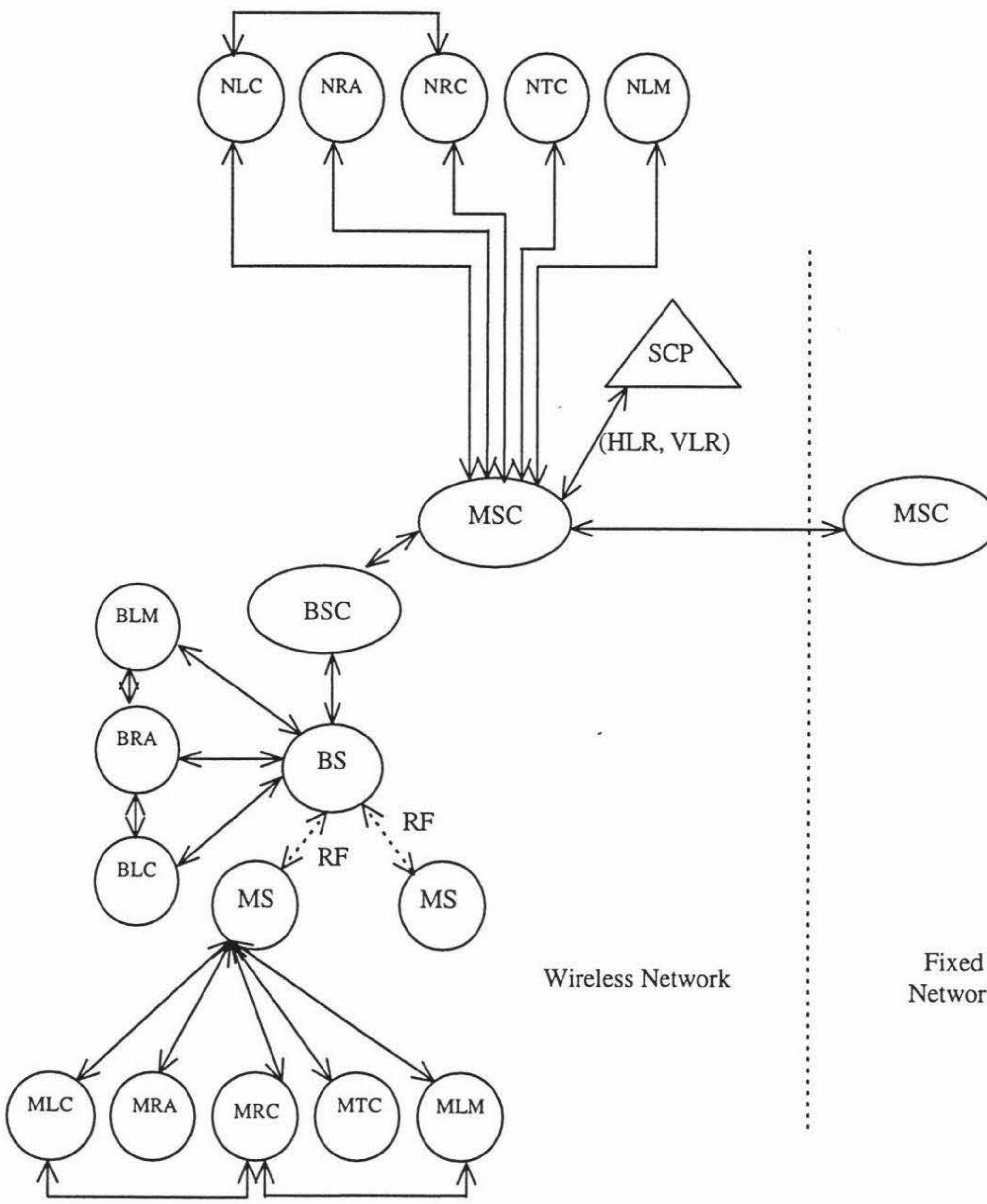
The proposed signalling structure is based on the functional model given in Fig. 3.3. The functions of LM, TC, RA, RC, LC are distributed among MS, BS and the fixed network. MS, BS and even MSC would use the above functional models to transfer information. Control information passed through such a network are illustrated by the flow charts in Fig 3.14.1 through to Fig 3.14.6. In the signalling transfer network presented in Fig. 3.13.1, elements of the functional model along with BSs and BSC are connected to a small STP. STPs perform the transfer of signalling information. The small STP is connected to a large STP along with the MSC (MSC is used for switching of user data). MSC could be connected to other MSCs through STPs. They could be a part of the PCN based ATM network or any other network (e.g. PSTN). If not ATM, then appropriate actions need to be taken to ensure smooth transfer of information. SCP provides data bases such as HLR, VLR of all the MSs. The operation of different elements of the functional model was explained in section 3.4, whereas the transfer of information between the elements of the functional model is detailed in Table 3.2.

Based on the above ATDMA/ATM/SS7 structure, it is possible to present signalling structures for resource access, resource release, call set-up and access (initial and subsequent), handover, power control and link adaptation. These are given in Fig 3.14.1 through to Fig. 3.14.6, respectively, while, Fig 3.15.1 and Fig. 3.15.2 presents VCC connection and release on the fixed network. These figures also represent involved logical channels and ATDMA functional groupings when applicable.



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, **MSC** : Mobile Switching Center, **BSC** : Base Station Controller, **STP** : Signalling Transfer Point, **SCP** : Service Control Point

Fig. 3.13.1 Proposed PCN Signalling Transfer Network



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, **MSC** : Mobile Switching Center, **BSC** : Base Station Controller, **STP** : Signalling Transfer Point, **SCP** : Service Control Point

Fig. 3.13.2 Proposed PCN Traffic Transfer Network

BS	MSC	Action
(MTP)		Activity detection (i.e. start of a talkspurt, video frame, data burst, new data block)
↓		
(MLC)		Intention of getting a slot for the particular resource is passed to mobile stations RA
↓		
(MRA)		
(MRA) $\xrightarrow{\text{RACH, DCCH}}$	(BRA)	Resource request
(MRA) $\xrightarrow{\text{RACH,}}$	(BRA) ⟶ (NRA)	Repeat resource request. BRA would look for a channel on the wireless network whereas, NRA would do the same for the fixed network.
(MRA) $\xleftarrow{\text{AGCH, DCCH}}$	(BRA) ⟶ (NCS)	Information about the allocated channels would be passed to the LCs . Then LCs could maintain the quality of the bearers (specially on the wireless network).
↓	↓ ↓	
(MLC)	(BLC) (NLC)	Channel information is then passed to the TPs and NCS for the packet transmission to begin.
↓	↓ ↓	
(MTP)	(BTP) (NCS)	

Note :

* Resource request procedure could be repeated if collision occurs

Fig. 3.14.1 Message flow for Activity Detection & Resource Assignment Procedure in an ATDMA System

MS	BS	MSC	Action
(MTP) ↓			Detects resource release (i.e. end of talkspurt or a video frame is detected by the reception of background noise information)
(MLC) →	(BLC) →	(NLC)	MS link controller transmits BSs link controller background noise. Then LCs would know that the link is no longer needed. This would be acknowledged to the TPs and NCS.
↓ (MTP)	↓ (BTP)	↓ (NCS)	
(MLC)	(BLC)	(NLC)	LCs informs the RAs that the slot(s) is no longer needed. Then the slot reserved for the resource could be released by BRA. This information would also be passed to MRA and NRA. Then NRA would release the connection on the fixed network.
↓ (MRA)	↓ (BRA)	↓ (NRA)	

Fig. 3.14.2 Resource Release in an ATDMA System

MS **BS** **MSC**	**Action**
MRA $\xrightarrow{\text{RACH(R)}}$ **BRA**	Request DCCH
MRA $\xleftarrow{\text{AGCH(A)}}$ **BRA**	Grants DCCH
MRA $\downarrow$	MRA reports successful DCCH set-up to MLC.
MLC $\xleftrightarrow{\text{DCCH}}$ **BLC$_{new}$** $\longrightarrow$ **NLC**	MLC is used to inform MRC and NRC of successful DCCH assignment
$\downarrow$ **MRC**	
MRC $\xleftrightarrow{\text{DCCH}}$ **NRC**	Now DCCH is used by MRC and NRC to perform functions such as authentication
MRA $\xleftrightarrow{\text{DCCH}}$ **BRA$_{new}$**	'I' slot allocation
MLC $\xleftrightarrow{\text{DCCH}}$ **BLC$_{new}$**	DCCH release (transmit background noise)
MTP $\xleftrightarrow{\text{TCH}}$ **BTP** $\longleftrightarrow$ **NCS**	Transfer of information

Fig. 3.14.3 Message Flow for Access Set-Up (initial and subsequent) Procedure

MS **BS_{old}** **BS_{new}** **MSC**	**Action**
MLC $\updownarrow$ **MRC**	MRC receives measurement Information from MLC about the current link and adjacent base stations. MRC will direct MLC to measure specific channels of adjacent base stations, thus MRC would already know the most promising candidate for handover [61].
MRC $\leftarrow$ - - - - - - **NRC**	Handover trigger by network's routing controller
MRC $\downarrow$ **MLC** $\downarrow$ **MRA**	DCCH set-up to request MLC and MRA
MRA $\xrightarrow{\text{RACH}}$ **BRA_{new}**	DCCH request
MRA $\xleftarrow{\text{RGCH}}$ **BRA_{new}**	Grants DCCH
MRA $\downarrow$ **MLC** $\xrightarrow{\text{DCCH}}$ **BLC_{new}** $\longrightarrow$ **NLC**	MRA reports successful DCCH set-up to MLC.
MLC $\downarrow$ **MRC**	MLC is used to inform MRC and NRC of successful DCCH assignment
MRC $\xleftrightarrow{\text{DCCH/ D channel}}$ **NRC**	Now DCCH is used by MRC and NRC to perform handover related functions such as authentication
MRA $\xleftrightarrow{\text{DCCH}}$ **BRA_{new}**	'I' slot allocation with new BS
MLC $\xleftrightarrow{\text{DCCH}}$ **BLC_{new}**	DCCH release
MLC $\Leftrightarrow$ **BLC_{old}** **MLC** $\longleftrightarrow$ **BLC_{new}**	Old and new TCH links are used in parallel
MLC $\Leftrightarrow$ **MLC**	Old TCH link released

Fig. 3.14.4 Message Flow for Forward Handover Procedure

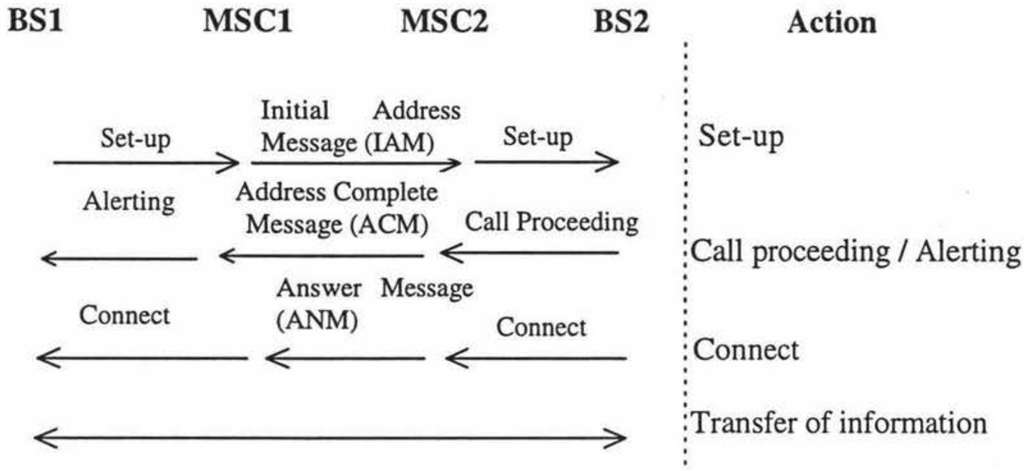
MS		BS	Action
MTP $\downarrow$ MLC		BTP $\downarrow$ BLC	Mobile and base stations' LCs take measurements of down and up-link received signal strength and quality respectively.
MLC	$\xrightarrow{\text{ACCHf}}$	BLC	MS reports its measurements
		BLC $\updownarrow$ BRA	As in Note *
MLC	$\xleftarrow{\text{ACCHr}}$	BLC	Informs new reciprocity factor and power index
MLC $\updownarrow$ MTP			Reciprocity factor is used to adjust the up-link power

Note * : BLC averages the measurement reports and determines the new preferred up-link transmit power through an algorithm that attempts to maintain received signal quality to a target value. BLC also calculates the measured relationship between up and down-link channels [61]. The requested transmit power would be compared with the current maximum allowed transmit power known to BRA and then BLC issues the new power settings and the reciprocity factor[61].

Fig. 3.14.5 Message Flow for Up-link Power Control procedure

MS	**BS**	**MSC**	**Action**
	BTP ↓ **BLC**		Reports current channel quality for all the allocated slots
MLC $\xrightarrow{\text{ACCHf}}$	**BLC**		Reports transmit power for these slots due to APC
	BLC ↓ **BRA**		If a change to a new mode is required, then request for resource changes are made to BRA.
MRA $\xleftrightarrow{\text{ACCHr/ACCHf}}$	**BRA** ⟷	**(NRA)**	BRA allows changes and inform MRA. MRA then acknowledge this message. BRA also informs NRA of change of resources.
MRA ↓ **MLC** ↓ **MTP**	**BRA** ↓ **BLC** ↓ **BTP**	**(NRA)** ↓ **(NLC)** ↓ **(NCS)**	RAs instruct LCs to change mode, which would intern instruct TPs and NCS.

Fig. 3.14.6 Message Flow for link adaptation for an up-link bearer

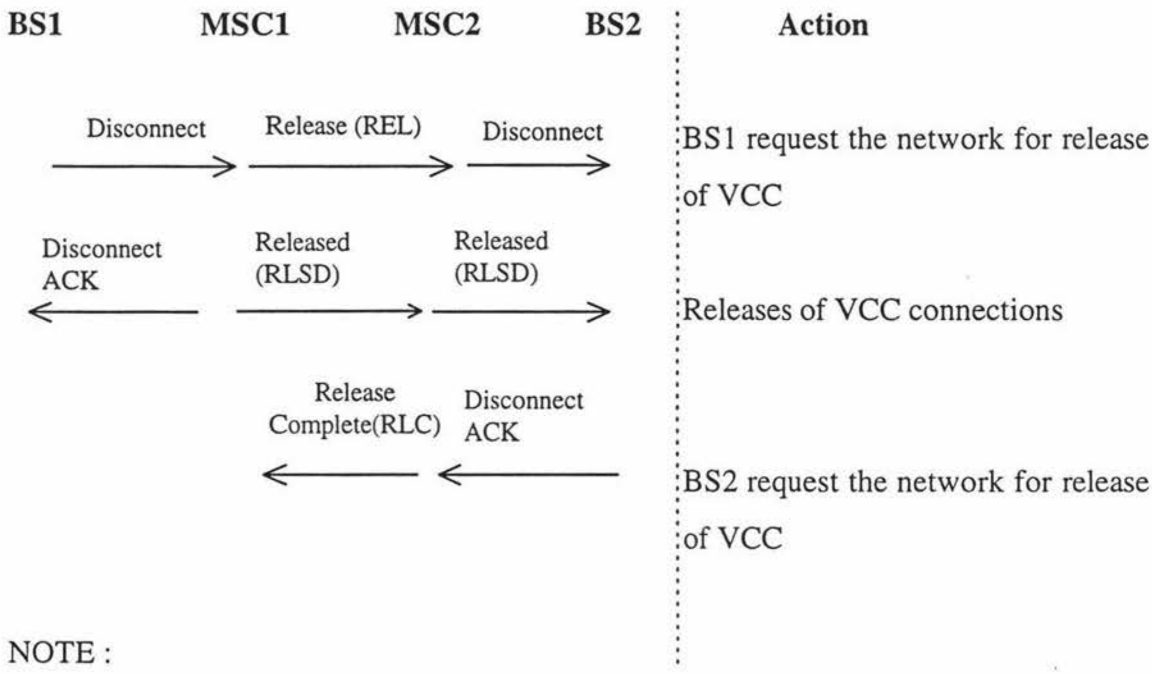


NOTE :

VCC = Virtual Channel Connection

BS1 transmits a call set-tup message (requesting for a VCC) to MSC1 and enters call initiated state. The set-up message contains information of the requested call. MSC1 transmits an IAM message to MSC2. IAM contains the called party number and type of information required. Finally, MSC2 informs BS2 of call set-up. If the requested channel quality measurements are met and access to the requested service is authorised and available, BS2 acknowledges MSC2 with a call proceeding state (or alerting). Then MSC2 transmits an ACM. This message is used for several purposes. They include, acknowledgment to MSC1 to indicate that the connection is established and also to indicate that the called party was found to be idle and is being alerted. If the called party accept, BS2 would transmit a connect message to MSC2. This information is transferred from MSC2 to MSC1 through an ANM. Then MSC1 informs BS1 of the situation. Then a VCC would be set up for transfer of information

Fig. 3.15.1 VCC Connection Set-up



NOTE :

VCC = Virtual Channel Connection

BS1 would transmit a disconnect request to MSC1 requesting for a channel release. On receipt of this message all the MSCs involved would start releasing the switched path supporting the VCC. Each MSC (in the above example only two MSCs are involved) would transmit a release message to the succeeding MSC and finally, to the terminating BS (BS1 in the above example).

Fig .3.15.2. VCC Release initiated by calling party

3.7 The Advantages of the ATDMA Protocol

- **Priority, quality based channel (slot) allocation** is possible with a base station centered 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 deteriorating channel quality, 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 of 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 [48]).

- **Link adaptation and APC** allow the system to adapt to propagation conditions making planning less critical [61].

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

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

3.8 Conclusions

In this chapter, ATDMA based PCN architectures have been studied. Priority and quality based channel allocation capabilities, an improved mechanism to satisfy user requirements (by change of transmission modes) and seamless handover capabilities, are some of the reasons for choosing ATDMA as the access protocol over the wireless network. Based on the ATDMA functional model given in Fig. 3.3 [16, 61], an ATDMA protocol that supports all of the specified ATDMA services have also been studied. This chapter also examined the mapping of ATDMA packets into ATM format. ATM's packet switched nature, flexible bandwidth allocation for bursty traffic, efficient multiplexing of traffic from different sources and ease of inter-connectivity with future networks made ATM the perfect backbone network for PCN design. The corresponding user and signalling traffic plains are presented in an OSI structure in Fig. 3.4.1 and in Fig. 3.4.2, respectively.

The rest of the chapter examined various aspects of signalling in an ATDMA/ATM based PCN. Initially, this section discussed the control channels that are being used in an

ATDMA system. Then control channel burst structures and traffic burst structures have been explained. Since ACCH is strictly associated with TCH and DCCH [16], ACCH is incorporated in a normal burst. The information field of the proposed RACH is much smaller than that of a normal burst since more protection is provided to the RACH burst structure by means of incorporating more bits for guard space and synchronisation. As RACH is a one-off burst, more synchronisation bits are needed to combat the affects of multipath fading. More guard space is needed to avoid the overlapping of the RACH burst into adjacent bursts. Guard space is calculated based on the maximum distance between a MS and a BS. Radio signal needs to travel twice the distance between stations (MS and BS) in order to have an influence on the link. It would travel one way from the BS to the MS, where the MS synchronises with the system timing, and then the other way from the MS back to the BS. The guard space is incorporated so that a random access burst does not collide with normal bursts in the same cell [48]. This chapter also presents traffic and control plains for an ATDMA/ATM based PCN, based on the ATDMA functional model given in Fig. 3.3 and the traffic and control protocols given in section 3.4.1. Control information passed in such a system is graphically illustrated in Fig 3.14.1 through to Fig 3.15.2.

CHAPTER 4
Study of Advanced-Time Division Multiple Access (ATDMA) Protocol

4.1 Introduction

One of the most important aspects of a PCN is the air interface of the network. As described in previous chapters, implementing multimedia services in a wireless environment is much more challenging than in a fixed network because of the limited bandwidth availability, fading, interference, etc. Designing a communication network over a wireless medium is challenging task. The challenge for the PCN designer is to try and achieve the 'maximum spectrum efficiency' by designing the best possible multiple access technique for the wireless medium. 'Maximum spectrum efficiency' implies increasing the number.of.users/Hz/Km^2. Since ATDMA is a statistically multiplexed radio access mechanism that suits the next generation mobile systems, this project has investigated its traffic capacity in a multimedia environment by using a simulation technique. The simulation model was developed using a discrete event simulation language known as Simscript II.5.

The change of transmission modes in an ATDMA system (due to change in source coder rates) would vary the number of coded source bits per data packet. In order to keep the link quality high and the packet size constant, extra error correction coding bits are added. This way, the gross coder rate does not change and for this reason link adaptation features are not incorporated in the simulation model (in order for the link adaptation features to be incorporated in a simulation model, channel models have to developed) . In the simulation, voice, data and video traffic sources have been used to find out the efficiency of the ATDMA protocol. Several slot allocation strategies have been tried for different traffic types to find out the optimum access strategy for the multimedia traffic. Simulation based up-link channel capacity calculations have also been presented in this chapter.

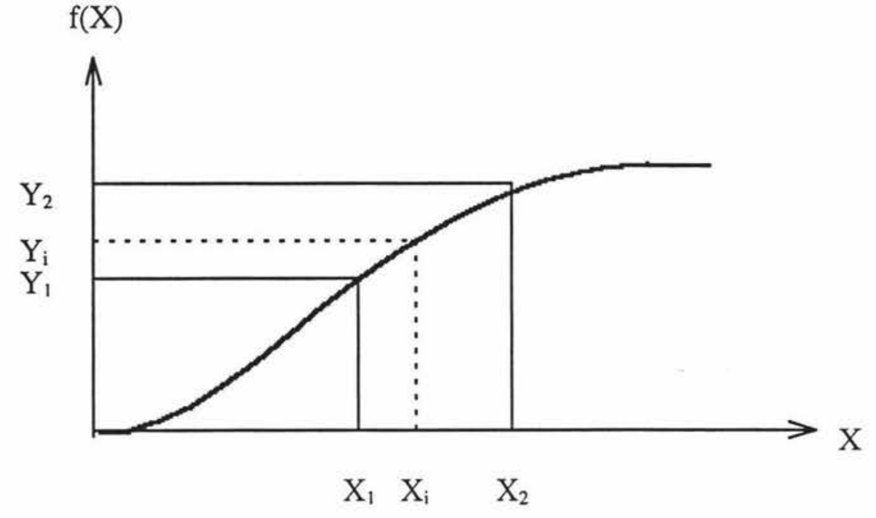
4.2 Simulation of Traffic Generators

To simulate multimedia applications, voice, video and data generators were used in the simulation model. In the following sections these voice, data and video traffic generators will be described.

4.2.1 Voice Traffic Generator

Human speech consists of talkspurts and silent periods. During a normal telephone call, without noise, a typical speaker is found to talk for about 44% of the time [31]. However, in practical situations, this duration would be higher than 44% because of external noise. Since no information is transmitted during the silent 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 Fig 4.1 [31]. Brady's distribution shown in Fig 4.2 has been adopted in this particular model [32]. The graph in Fig. 4.2 was obtained using real data from a two way telephone conversation. Speech packets from a speech detector is strongly influenced by the choice of threshold levels [33]. In case of a threshold level of -40 dBm, the mean talkspurt and silence periods were taken as 1.34 and 1.67 seconds, respectively [31]. Any talkspurt less than 15 ms was considered as a noise and a silence period less than 200 ms was taken as a talkspurt [33].



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

Fig. 4.1 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 corresponding measured abscissa and ordinate 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 Yi is then used to find corresponding Y_1, Y_2, X_1 and X_2. When all the necessary parameters are found equation 4.1 is used to generate talkspurt/silence duration X_i.

$$X_i = \frac{Y_i - Y_1}{Y_2 - Y_1}\left(X_2 - X_1\right) + X_1 \qquad \text{............} \qquad [4.1]$$

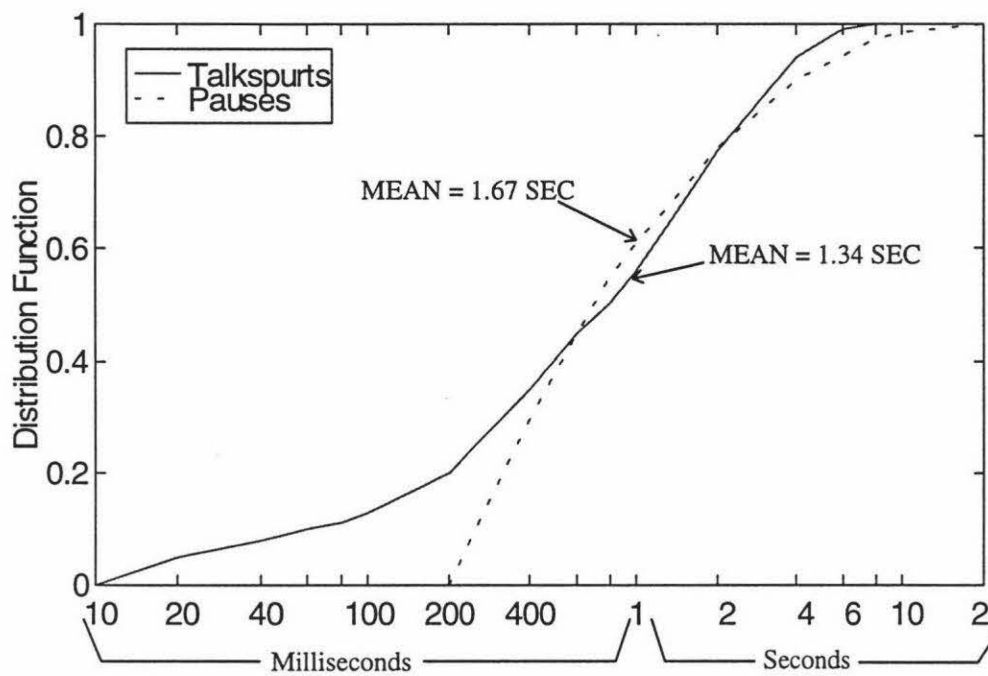


Fig. 4.2 Talkspurt and silence distribution in telephone conversation [34]

4.2.2 Video Traffic Generator

Variable Bit Rate (VBR) coders may perform better than conventional Fixed Bit Rate (FBR) coders because they generate output depending on the information content. FBR coders generate constant output irrespective of image content. When FBR coders are used, image quality suffers in areas of high activity while unnecessary amounts of information are transmitted to represent, for example uniform backgrounds. In case of VBR the output of the system varies with the image content. VBR coders generate 'bursty' traffic which is suitable for transmission by packet switched networks. For the simulations, a low variable bit rate video traffic generator has been modelled to cater for mobile environments. It is based on the model proposed in [36]. The model in [36] is for a low bit rate video packet generator for an ATM network. The proposed VBR coder is designed for ATM networks with a variable bit rate of 1-5 Mbps and a image size of 512*512 pixels at 25 frames/sec. The model developed during this study uses 64*64 pixel images at 10 frames/sec. A particular image could be further subdivided into sub-blocks. Sub-blocks of size 8*8 pixels were used for the study. The content of each subblock might change from the previous image to the current image. If the content of a sub-block is changed, that sub-block is regarded as being active. A video frame (image) could generate a maximum of 64 active sub-blocks. Each active sub-block is coded into 48 bytes. The VBR video traffic generator, generates video traffic at a maximum bit rate of 64*48*10*8 = 245.76 kbps. The following section discuses the video traffic generation process.

4.2.2.1 Video Packet Generation Process

The entire video frame (image) is subdivided in to a number of sub-blocks. The numbers of active sub-blocks is calculated after processing the entire video frame. The call generation process can be represented by a simple active/inactive (one cell/no cell produced) process. This cell production process is represented by [4.2] and [4.3] [36,37].

$$P_a(t) = A_a . t^{D_a} \qquad \text{-------------------------} \qquad [4.2]$$

$$P_i(t) = A_i . t^{D_i} \qquad \text{-------------------------} \qquad [4.3]$$

Where:

t = Time

$P_a(t)$ = Probability that the cell is in an active state at time 't' given that the state of the previous cell was inactive

$P_i(t)$ = Probability that the cell is in an inactive state at time 't' given that the state of the previous cell was active

A_i, D_i = Depends on the mode value. Experimentally (original values were changed to get the desired average mode values) calculated values for these are indicated in Table 4.1.

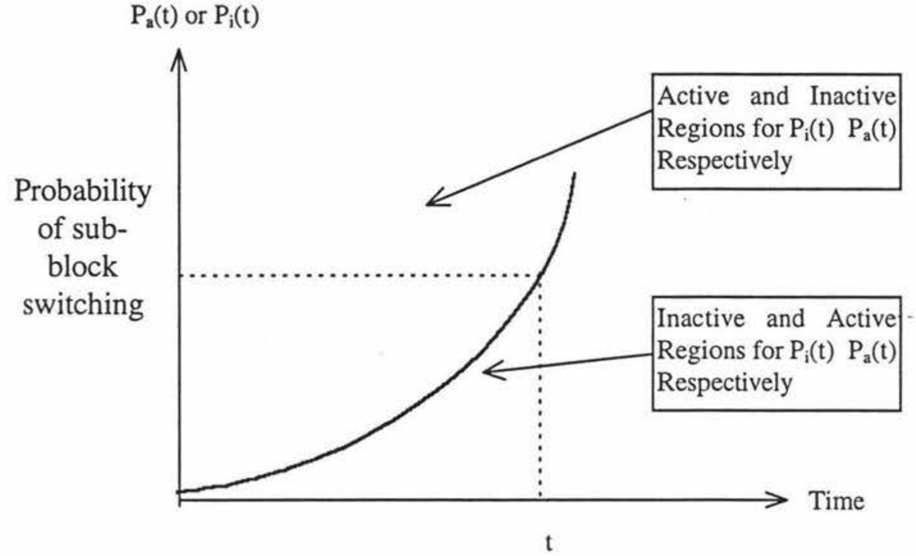
Activity Mode	A_a	D_a	A_i	D_i
1 (1Mbps)	$1.34*10^{-7}$	8.040	$5.7*10^{-1}$	0.000
2 (2 Mbps)	$1.005*10^{-2}$	4.126	$2.9*10^{-1}$	0.000
3 (3 Mbps)	$4.556*10^{-1}$	1.049	$1.7*10^{-1}$	$3.629*10^{-3}$
4 (4 Mbps)	$2.492*10^{-1}$	$7.529*10^{-2}$	$8.792*10^{-1}$	$1.875*10^{-1}$
5 (5 Mbps)	$2.5*10^{-2}$	0.000	$3.427*10^{-1}$	$8.784*10^{-1}$

Table 4.1 Constant activity fractal mode parameters

The first sub-block of each frame was randomly chosen either as an active or an inactive sub-block. Then a real variable is randomly selected (take 'u' to be the value of the random variable) from a uniform distribution between 0 and 1. Depending on the state of the previous sub-block, the video traffic generator compare 'u' with the appropriate plot obtained from either equation [4.2] or [4.3]. As shown in Fig 4.3, this comparison would allow the video traffic generator to decide if the current sub-block is going to be active or not.

The next important part of the protocol is the mode switching strategy within a frame. The mode is changed (e.g. from mode 3 to mode 5) after processing 8 sub-blocks (Each section consists of 8 sub-blocks). The mode switching strategy was implemented according to equation [4.4].

The Standard deviation, σ, was calculated from a uniform distribution between 0.5 and 12. Typical coder outputs for section sizes 4 and 8 (section sizes 4 and 8 indicate that the modes are changed after processing 4 and 8 sub-blocks, respectively) are illustrated in Fig. 4.4.1 and 4.4.1, respectively.



$P_a(t)$ = Probability that the cell is in an active state at time 't' given that the state of the previous cell was inactive

$P_i(t)$ = Probability that the cell is in an inactive state at time 't' given that the state of the previous cell was active

Fig. 4.3 Video packet generation technique using equation [4.2] and [4.3]

$$L_n = \begin{cases} |L_0| & n=0 \\ |L_{n-1} + \gamma_n| & n>0 \end{cases} \qquad 1 \le L_n \ge 5 \text{ ----------} \qquad [4.4]$$

Where:

L_0 = Starting or average activity level

L_n = Activity level of image 'section' n

γ = Normal distribution with zero mean and standard deviation σ

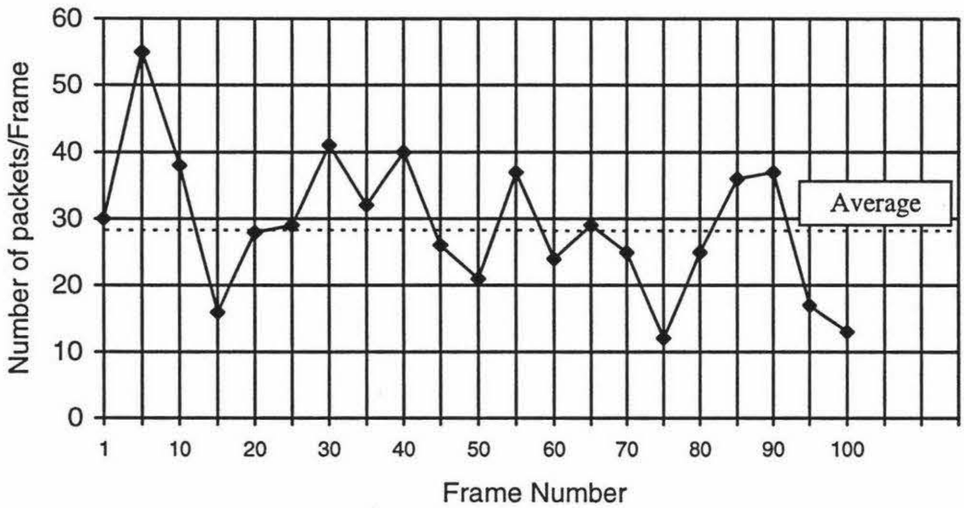


Fig. 4.4.1 Number of packets generated for different picture frames (Section size = 4)

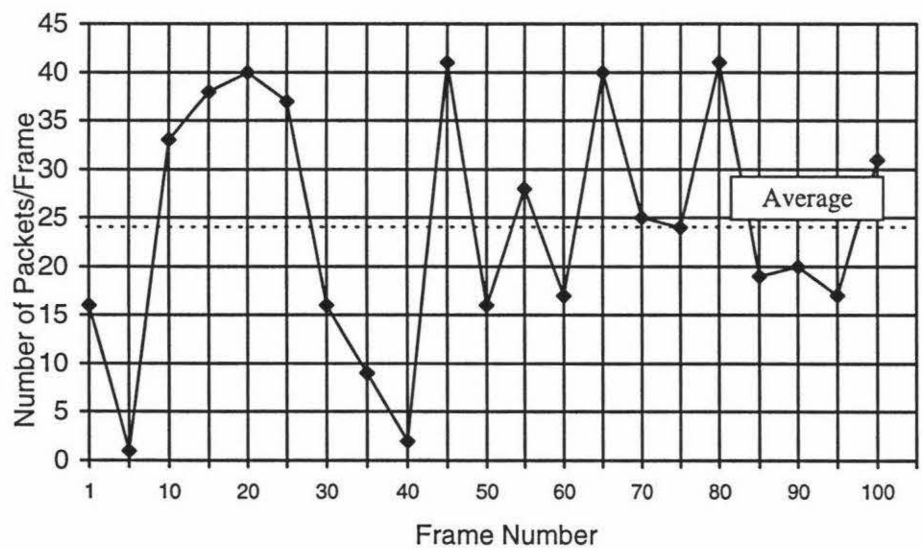


Fig. 4.4.1 Number of packets generated for different picture frames (Section size = 8)

On top of the intraframe coding (coding of adjacent sub-blocks), a correlation technique needs to be adopted for interframe coding (coding of adjacent frames). As mentioned in [38, 39], it is possible to adapt an Auto-regressive (AR) model to take care of interframe correlation. The AR model that suits the simulation work carried out during this study is a second order AR model. This model is given in equation [4.5].

$$X_n = E[x(n)] + Y_n$$
$$Y_n = a_1 Y_{n-1} - a_2 Y_{n-2} + be(n) \qquad \text{------------------------------} \quad [4.5]$$

Where a_1 a_2 and b are constants, e(n) is normally distributed gaussian noise with zero mean and variance 1, E[x(n)] is the mean bit rate (same as L_0) and Y(n) is the average bit rate of frame n.

It was not possible to calculate constants a_1 a_2 and b for low bit rate video since they were calculated using real image data. Values given in papers are for higher bit rates and are not applicable in a mobile environment.

4.2.3 Data Traffic Generator

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

4.3 Simulation of the ATDMA protocol

Simulation is a computer model based representation of a real system. Simulation is a powerful analytical tool that can be used to study the behaviour of a communication system prior to building testbeds. This project used a discrete event simulation [30] model to study different characteristics of an ATDMA based mobile network for multimedia traffic. Discrete event simulation describes the model of a real system in terms of logical relationships which cause changes of states at discrete points in time. The discrete simulation model generally tends to be both stochastic and dynamic in nature. In such an approach, results are obtained by generating scheduled events at different points in time in the simulation model. In other words, the changes in the physical system are represented by a series of discrete changes or events at specific instants in time. In discrete event simulation, time and state relationships are represented in terms of event, activity and process. These terms are defined as follows:

An event : In the simulation model a change in the state of an entity, occurring at an instant that initiate an activity; e.g. initiation or termination of a call [30].

An activity : The state of an entity over an interval.

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

Timing of different activities in the simulation is maintained by an internal clock which is incremented and maintained by the simulation program. The simulation time could either be implemented as interval-oriented or event-oriented. In an interval-oriented simulation, the clock is advanced in uniform, fixed time increments. In an event-oriented simulation, a variable time increment method is adopted based on the time between events. This method involves sorting of event activation time and maintaining current and future event lists. The ATDMA simulation model is based on the event-oriented method.

The basic block diagram for the ATDMA model is illustrated in Fig 4.5. The simulation is controlled by the module MAIN. At the start of the simulation the necessary variables are read from the input 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 frame structure is defined, aspects such as number of slots per frame, number of reservation slots and location of them in the frame are decided along with traffic slots and their locations within the frame. Routine INITIALIZE also activates the processes STATION, TIMER and TEMPPRO. TIMER keeps a track of the current and next reservation slots at any given time during the simulation while TEMPPRO is used to print packet information at a pre-defined regular interval. The process STATION takes care of all the traffic generators, it also activates the process MOBILE. MOBILE implements part of the ATDMA protocol. MOBILE also calls the routine SCHEDULE (which implements timing functions) and the routine BASE. Routine BASE models rest of the ATDMA protocol.

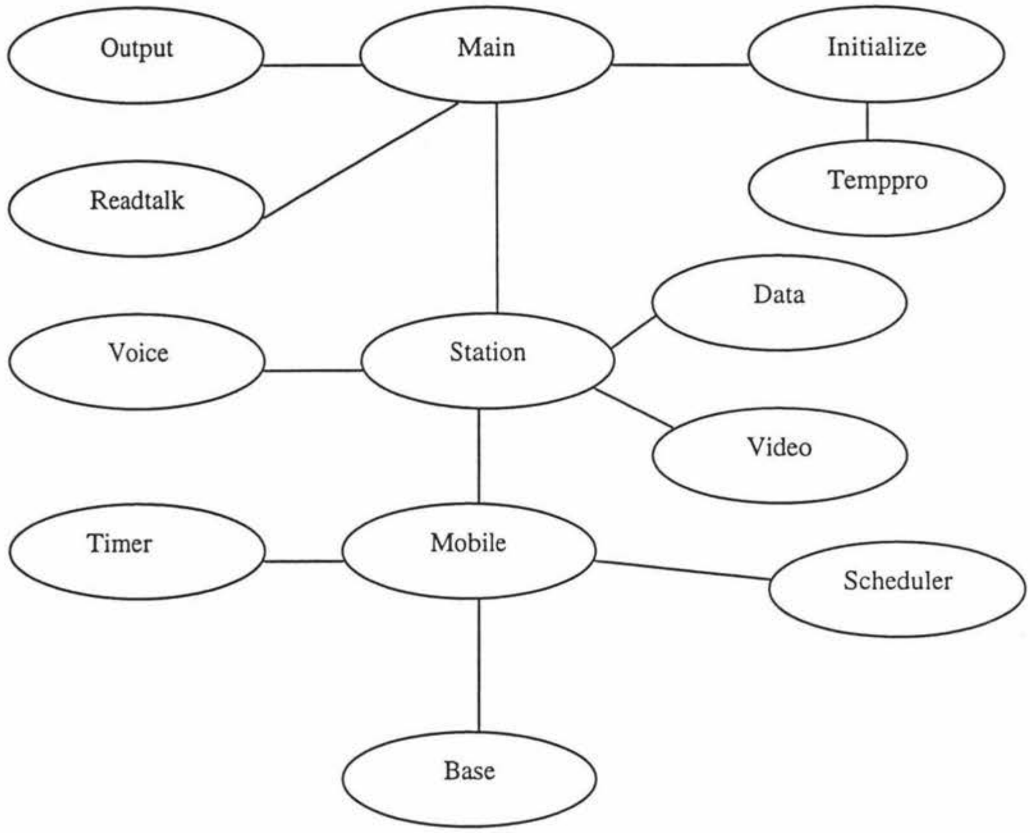


Fig. 4.5 Block diagram of the simulation Model

In simulating the operation of ATDMA the three most important modules used are STATION, MOBILE and BASE, these three modules are discussed below.

4.3.1 Process STATION

This process is initiated by routine INITIALIZE and would continue until the end of the simulation. Process STATION is shown in Fig 4.6. The process STATION represents the mobile terminals. At the beginning of the process it selects terminal type (voice, video or data) based on the ID of the terminal. If the selected terminal is voice, routine SILENCE would be called. The SILENCE routine returns a silence period. The station would then remain in a wait state for the period of time returned by the routine SILENCE. After the waiting period, routine TALKSPURT is called. This returns the talkspurt duration (talkspurts and silences are generated using Brady's distribution). 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, initially it waits for an exponentially distributed waiting time (the mean waiting time is varied during the study). Then data packets are generated using another exponential distribution with mean 1.

If the current station needs to generate video packets, initially, it would be in a wait state for 100 ms. Then the number of active sub-blocks are calculated for the current video frame by the routine VIDEO (this process was explained in sections 4.2.2 and 4.2.2.1). These active sub-blocks are then used to calculate the total number of video packets needed to be transmitted as in equation [4.6]

$$\text{Total video packets} = ((\text{Active sub-blocks}) * 48 * 8/160) + 1 \qquad \text{------------} \qquad [4.6]$$

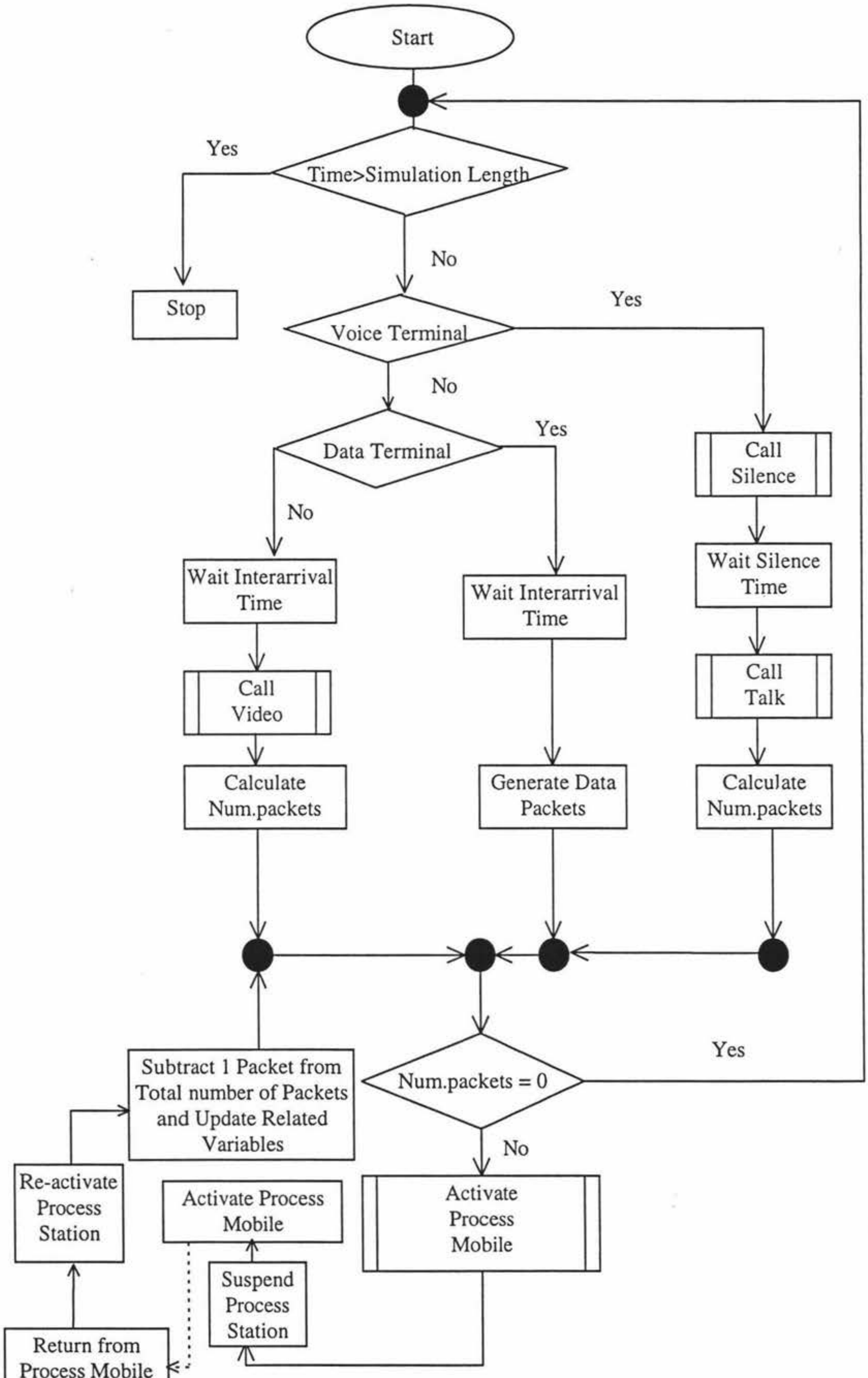


Fig. 4.6 Flow diagram of the process STATION

Whenever a station has packets to transmit, it activates the process MOBILE; it passes the necessary parameters to the process MOBILE and suspends the process STATION. The process STATION is re-activated by the corresponding MOBILE process once the necessary events are executed in the process MOBILE. Process MOBILE also returns some terminal associated parameters that are used to update the process STATION.

Upon re-activating the STATION, one packet is subtracted from the total number of packets to indicate that either one packet is successfully transmitted or one packet is dropped. The packet count could be decreased during a talkspurt. In the case of data stations packets are not dropped but the packet count is decreased when a successful packet transmission occurs. Data is designed as a loss sensitive source of traffic (in comparison voice is delay sensitive). When the delay of a video frame exceeds the threshold all the packets of that particular video frame are dropped. This is done because it is extremely difficult to re-construct a reasonable quality picture once a packet in a video frame has been lost. When the total number of packets within a station reaches zero, then the corresponding generator is used to re-generate more packets. In case the of a data station, the data packets are send in blocks of 'M' (the size of M was varied during the study) packets at a time . This is to give voice and video higher priority over data since they are delay sensitive.

4.3.2 Process MOBILE

The mobile process is described in the form of flow charts in Fig 4.6.1 - 4.6.3. MOBILE is initiated by the process STATION and it models most of the ATDMA protocol related activities. At the beginning of this process it checks the status of a particular packet. If the status indicates that the terminal is in search of a reservation (or 'R') slot, an 'R' slot number in which the terminal may transmit its reservation request would be assigned. This does not guarantee that the 'R' slot would be allocated for that particular terminal. The 'R' slot allocation uses the slotted ALOHA mechanism and as a result, there could be collisions during resource reservation requests. Routine BASE decides whether a terminal is successful or not in seizing the allocated 'R' slot. If a particular terminal is the only contender for the 'R' slot, then it seizes the slot. If a collision occurs, then the collided terminals should re-transmit their reservation requests. The selection of 'R' slots for re-transmission is also performed by the process MOBILE.

The slot number in which the terminal is transmitting (e.g. the 'I' slot) or supposed to transmit (e.g. the 'R' slot) would be passed to the routine SCHEDULER. This information is used by the routine SCHEDULER to perform the waiting time calculations. Then the routine BASE is called to monitor the transmission. Routine BASE returns the status (which indicates whether the terminal is currently performing data transmission, releasing slot and if trying for an 'R' or an 'I' whether successful or not) of transmission back to the process MOBILE. Then the rest of the functions of process MOBILE would be executed depending on the terminal type.

If a terminal was initially trying for a 'R' or an 'I' slot, then depending on the value passed by the routine BASE, the outcome would be known. If the outcome is unsuccessful (due to collisions in the case of 'R' slots or due to unavailability of traffic slots in the case of a mobile trying to get a 'I' slot) then the process MOBILE determines whether the delay threshold has been exceeded. A delay threshold of 20 ms (duration of 2 frame lengths) has been used for voice terminals and a delay threshold of 200 ms (duration of 20 frame lengths) has been used for video terminals. Video terminals should tolerate longer delays because it is difficult to reconstruct a video frame if a packet of a video frame is lost. In case of video traffic, if the delay exceeds the threshold, then the entire video frame is discarded whereas in voice, one packet is dropped and an attempt will be made to transmit the remaining packets. If the delay has not exceeded the threshold value, then a re-transmission attempt will be made. If the packet just transmitted is the first packet of a talkspurt or a video frame, then appropriate registers would be updated. If the packet is the last packet, then appropriate registers would be re-set. Finally, if the packet is a successful voice or video transmission, then once again, appropriate registers would be updated.

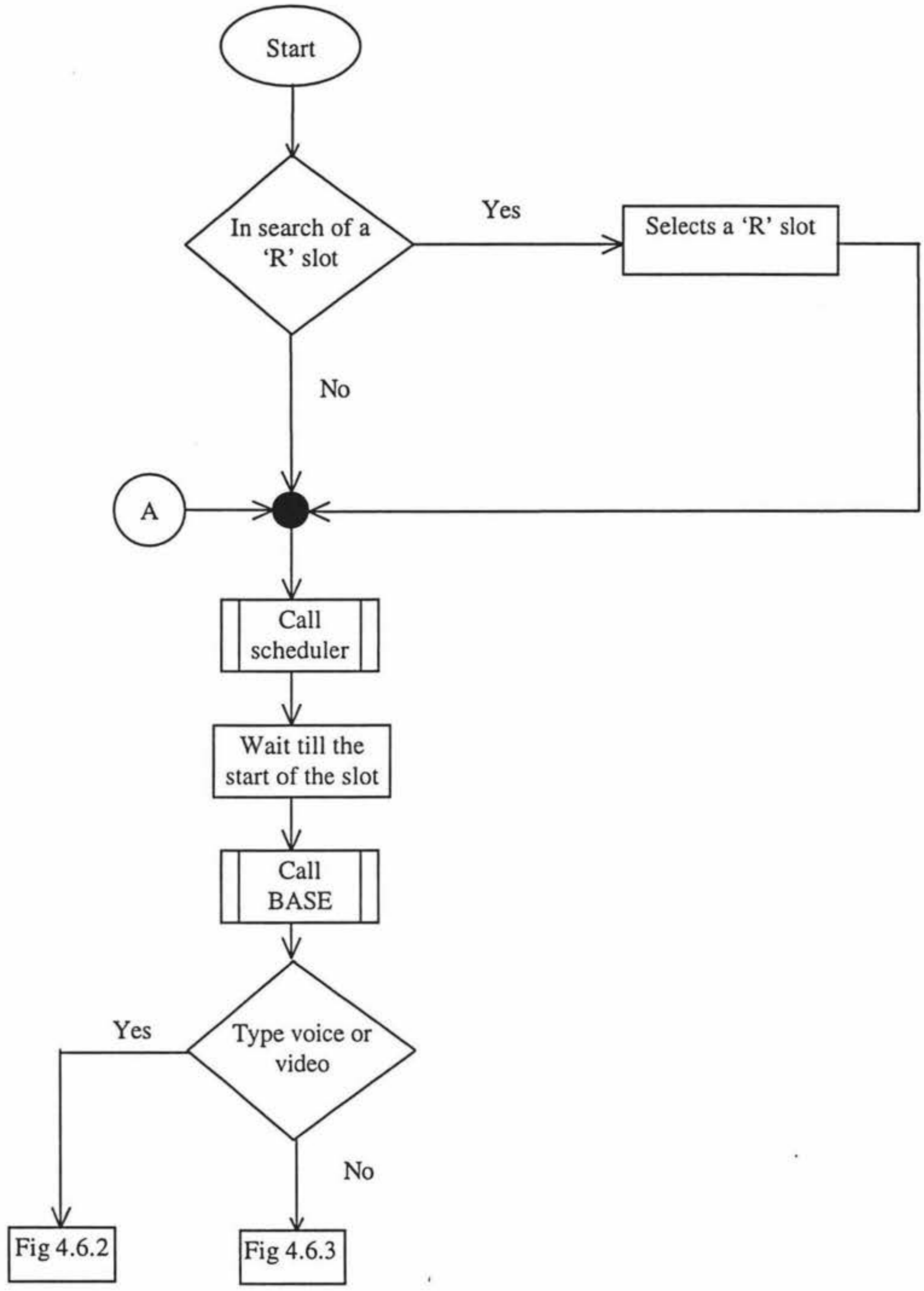


Fig. 4.6.1 Flow diagram for the first stage of process MOBILE

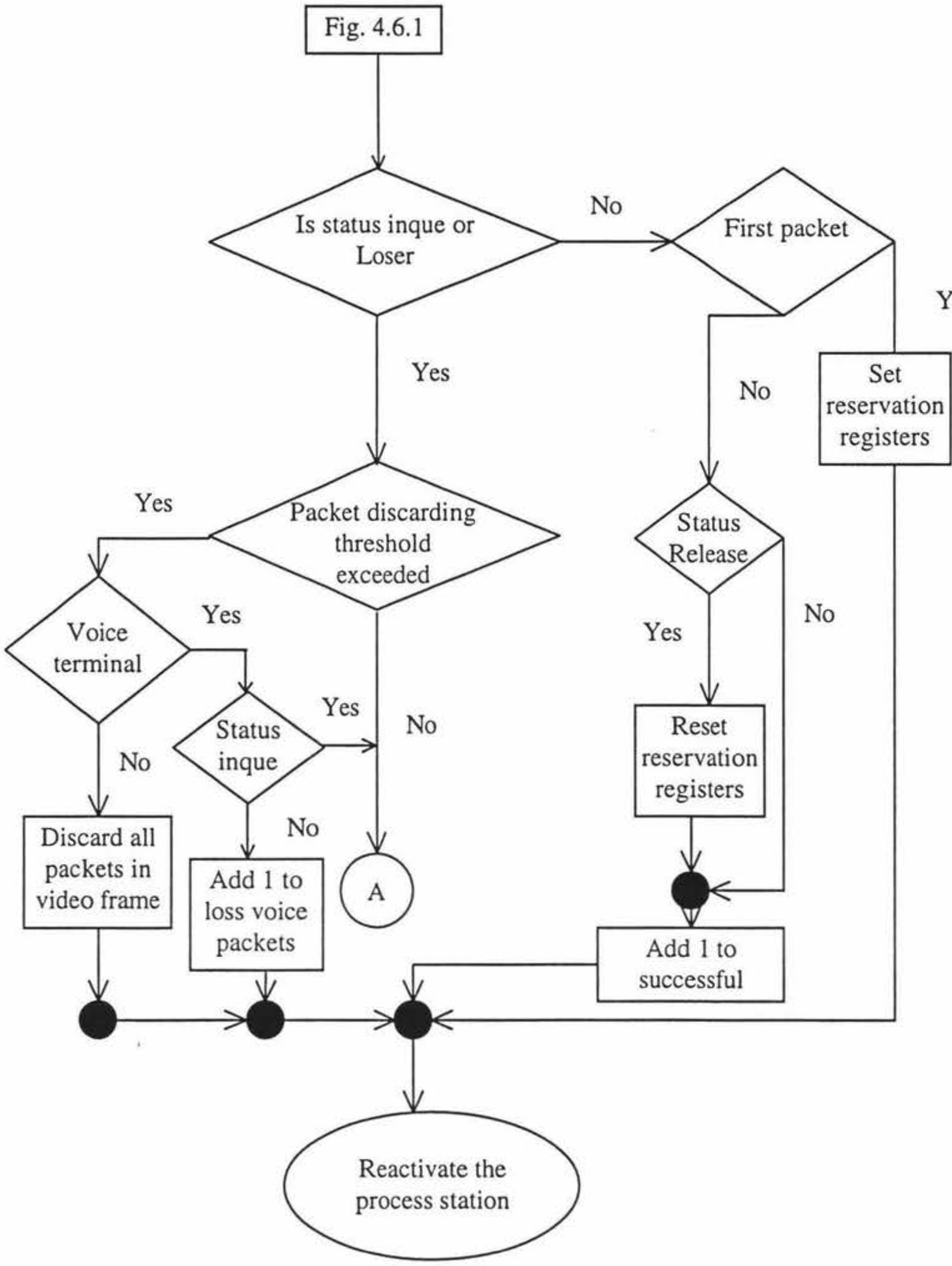


Fig. 4.6.2 Flow diagram for the voice and video portions of process MOBILE

Data packet transmission is similar to voice and video transmission but the data terminals do not have any delay thresholds. The other difference is that data terminals transmit their packets in bursts of 'M' packets at a time. If data is send in large bursts, then the allocation delay for other terminals in the system would increase. On the other hand, if data is send in small bursts, the contention delay for other terminals would increase. Therefore, a suitable value for 'M' had to be chosen in order to optimise the network performance. Finally, after updating some registers, process MOBILE would re-activate the process STATION.

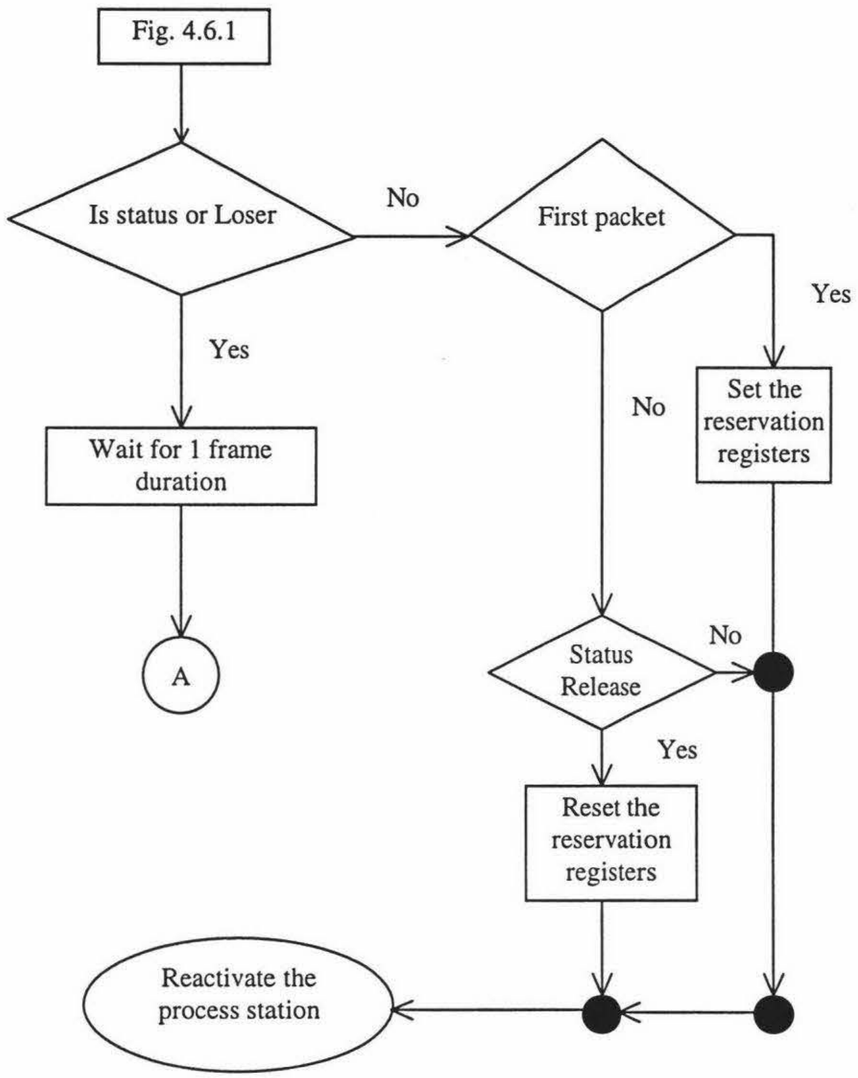


Fig. 4.6.3 Flow diagram for the data portion of process MOBILE

4.3.3 Routine BASE

This Routine is called by the process MOBILE when a terminal has packets to transmit. Initially this routine determines what appropriate function should be performed (e.g. trying to access 'R', transmitting on 'I', in the waiting queue, etc.). If the terminal is trying to transmit on the selected 'R' slot, then the slotted ALOHA algorithm is performed to determine whether there is going to be a collision in accessing the slot. If there is no collision then the terminal would transmit their reservation request on the 'R' slot. As a result, they would be admitted to the waiting queue. If a terminal is unsuccessful, a re-transmission attempt would be made. During re-transmission, voice and video terminals have higher priority than the data terminals. Therefore, voice and video terminals may select an 'R' slot in the current frame, whereas, data terminals must have a compulsory waiting time of one frame duration.

The waiting queue is also serviced by the routine BASE, based on the arrival time of the call (waiting queue is serviced as a FIFO). If the current terminal is heading the waiting queue and also if a 'I' slot is available, then the slot would be allocated to the first terminal in the waiting queue. Video terminals may be allocated multiple 'I' slots if available. All unsuccessful terminals will try once again when the waiting queue is serviced next (if there are 'N' 'R' slots in the an ATDMA frame, the waiting queue would be serviced 'N' times in each transmission frame). Routine BASE also updates the appropriate registers depending on whether the packet is the first or the last packet of a transmission burst. Once the functions of BASE are performed, the status of the transmission is passed back to the process MOBILE.

4.4 Validation of the Simulation Model

In order to validate the simulation model, it is important to validate the traffic generators. The two main generators that need validating are the voice and the video models. They are validated by gathering the required statistics from simulation models and comparing them with the standard data patterns. Then the ATDMA protocol would be validated collecting simulation results and comparing them with similar results published [14].

4.4.1 Validation of the Voice Traffic Generator

In order to validate the voice traffic generator, the model was simulated with 20 voice terminals for a period of 300 seconds (simulation time). The simulations were repeated for different sets of random number generator values. A random number known as 'talkseed' was used to generate talkspurts while 'silseed' was used to generate silences. These random numbers were used to calculate Y_i of equation 4.1 and then, the talkspurt/silence duration. The use of different random numbers results in different traffic arrival patterns. Table 4.2 shows the ratio of total talkspurts to the sum of total talkspurts and silences for different sets of 'talkseed' and 'silseed' combinations. These results indicate that irrespective of the random numbers used, the voice activity changes between 43.5% to 45%, thus, satisfying Brady's distribution.

Talkseed, Silseed	Voice Activity %
1, 2	43.53
2, 3	44.41
5, 3	44.83
8, 7	45.45
9, 2	44.55

Table 4.2 Voice activity for different random number generator values for a simulation length of 300 seconds. During this simulation 20 simultaneous voice terminals were used.

4.4.2 Validation of the Video Traffic Generator

In order to validate this particular model, a video frame of size 512*512 pixels was considered (with a sub-block size of 16*16). The frame interarrival time was 0.04 (25 frames per second). Average bit rate statistics for each mode was gathered from 10, 000 consecutive frames. For each mode, the number of active sub-blocks were calculated along with the total amount of sub-blocks generated for that particular mode. As the statistics for the total amount of time spent in each mode was also calculated, it was possible to calculate the generated bit rates for each mode. These bit rates are given in Table 4.3.1. This table indicates that modes 1, 2, 3, 4 and 5 generated average bit rates close to 1 Mbps, 2 Mbps, 3 Mbps, 4 Mbps and 5 Mbps, respectively. Results obtained from the simulation was very close to the results presented in the original papers [36, 37]. Since the results were consistent with what has been published one can assume that the developed low bit rate fractal model to be valid. In order to develop a traffic generator for the mobile environments, the same model has been scaled down to a 64*64 sub-blocks/frame (with a sub-block size of 8*8) and an interarrival time of 0.1 seconds (10 frames every second). The bit rates generated for different modes in this particular situation is given in Table 4.3.2.

Mode	Average Bit Rate(Mbps)
1	1.0495
2	1.829
3	2.719
4	4.2237
5	5.13781

Table 4.3.1 Average bit rates for each mode (these values were calculated by simulating 10,000 consecutive frames of the developed video protocol for a picture size of 512*512 pixels at 25 frames/sec)

Mode	Average Bit Rate(kbps)	Min. Bit Rate (kbps)	Max. Bit Rate (kbps)
1	26.3314	3.840	57.6
2	49.4933	3.840	80.64
3	78.912	3.840	107.52
4	96.8997	30.72	145.92
5	121.698	34.560	203.52

Table 4.3.2 Average, minimum and maximum bit rates for each mode (these values were calculated by simulating 10,000 consecutive frames of the developed video protocol for a picture size of 64*64 pixels at 10 frames/sec)

4.4.3 Validation of the ATDMA Protocol

ATDMA protocol traffic performance was obtained from the developed simulation model. These results were compared with the results published by other authors [14]. Although there were some differences between the two models, the published results were very similar to the results obtained from the simulation model. Some of the simulation parameters used in the simulation for this particular case is given in Table 4.4. The normal ATDMA burst structure for the bit rate given in Table 4.4 is shown in Fig.

4.7 (all parameters of the burst structure have been explained in detail in section 3.6.5 of chapter 3). In the burst structure given in Fig. 4.7 160 bits of coded data carry information from a single channel.

The major differences between the two models were in source coder rate, slots per frame, burst information payload and burst overhead. The corresponding values for these in [14] were 13 kbps, 72, 66b, 59b, respectively. Beside this, in the other work, the authors divided the frame into two sub-frames of 5 ms each. They have also used diagonal interleaving and 4 modes of link adaptation.

Carrier bit rate	1.8018 Mbps
Speech coder rate	16 kbps
Frame length	10 ms
Down link acknowledgment delay	3 time slots
Slots per frame	77
Speech packet delay threshold	20 ms
Speech activity	43.5% - 45%
Burst information payload	160 b
Burst overhead	74 b
Simulation length	300 s

Table 4.4 Simulation parameters for validation

HB 4	Coded Data 80	Sig 5	Training sequence 26	Sig 5	Coded Data 80	TB 4	Guard 24

Fig. 4.7 Normal burst in ATDMA(for transmission speed of 1.8018 Mbps)

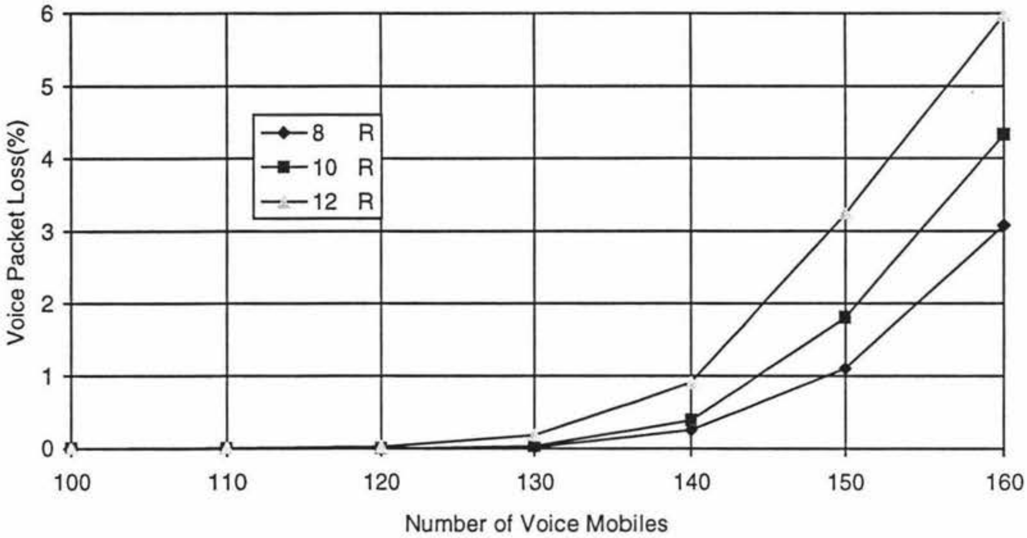


Fig. 4.8.1.1 Voice packet loss characteristics in an ATDMA environment (simulated model)

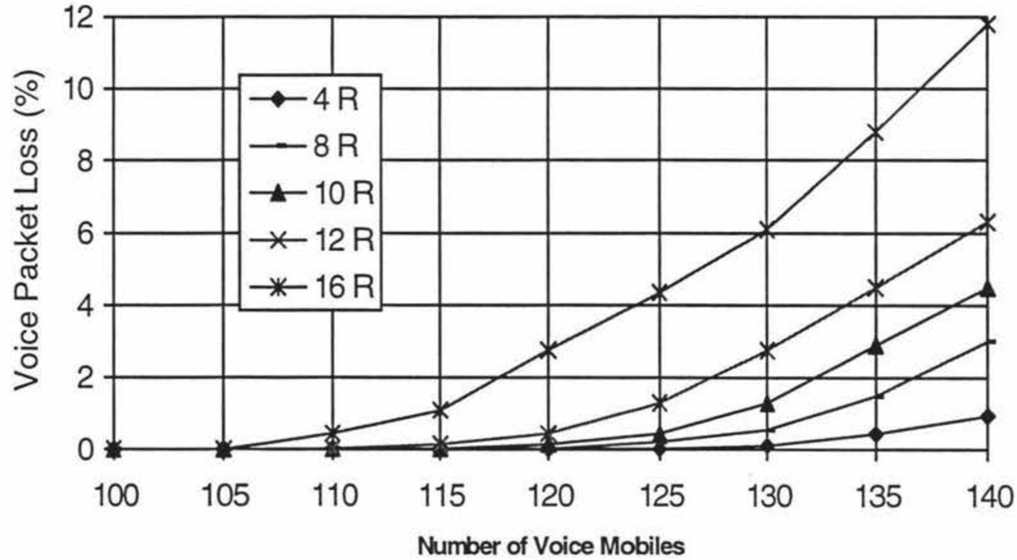


Fig. 4.8.1.2 Voice packet loss characteristics in an ATDMA environment (published model [14])

The performance results obtained for the voice transmission, according to the simulation parameters given in Table 4.4 are shown in Fig. 4.8.1.1, Fig. 4.8.2 and 4.8.3. Fig. 4.8.1.1 indicates packet loss for different combinations of 'R' slots. Fig. 4.8.1.2 shows published packet lost characteristics [14]. Speech packet loss of 1% is taken as the threshold criteria for the maximum possible conversations. Fig. 4.8.1.1 shows that the 8 'R' slots/frame can accommodate the highest number of terminals whereas, according to the published results, the above ratio was 4 'R' slots/frame. The difference is due to the 5 ms frame structure of the published results compared to the 10 ms simulated frame structure used in this study. By using the 8 'R' slots/frame a maximum of 149 simultaneous voice conversations were supported, yielding a statistical multiplexing factor of 1.935 users per slot. This particular value is same as the published results in [14], thus, validating the developed ATDMA model.

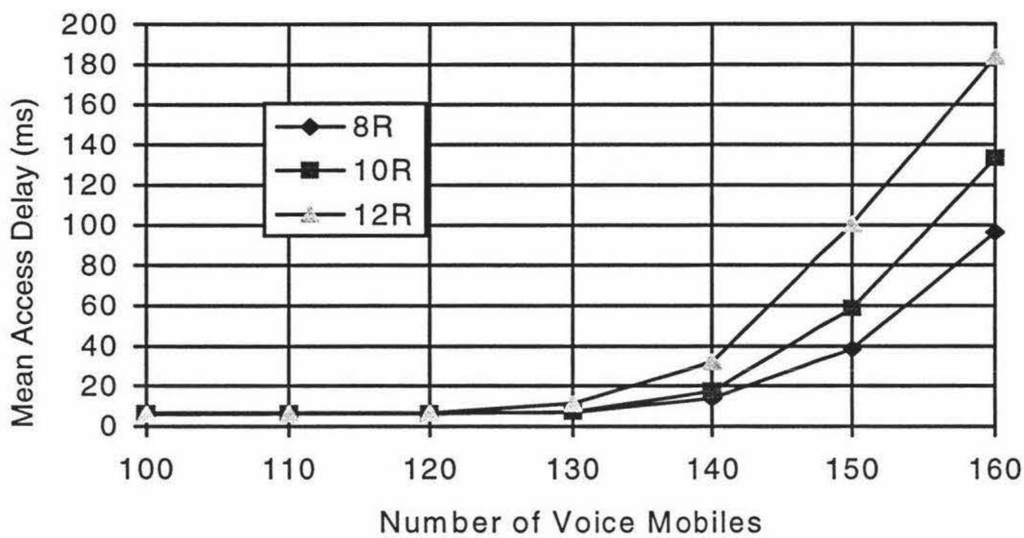


Fig. 4.8.2 Voice packet delay characteristics in an ATDMA environment

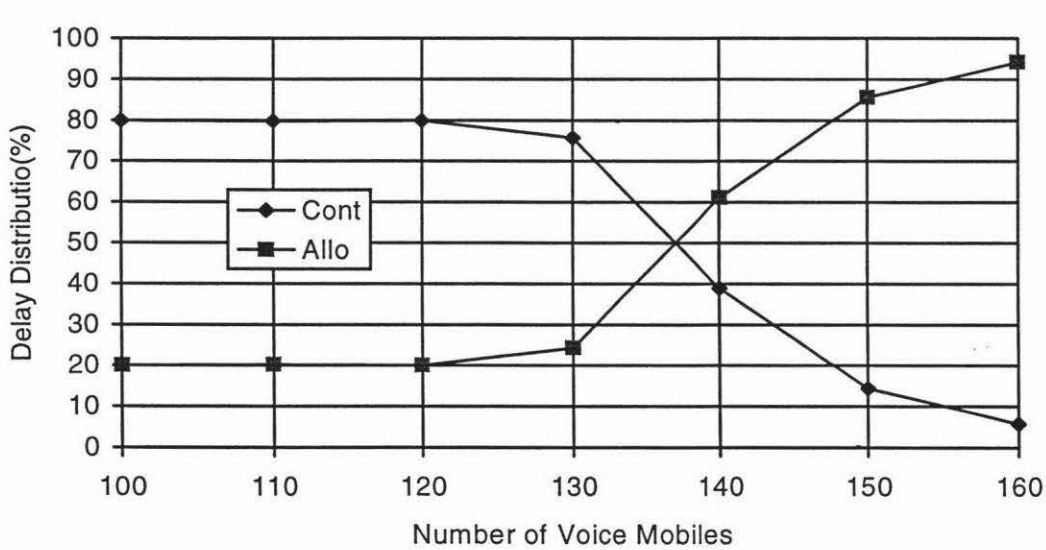


Fig. 4.8.3 Voice packet delay distribution with 8 'R' slots

4.5 Simulation results

In this section more simulation results will be presented which relates to the traffic performance of the ATDMA protocol. Fig. 4.8.2 shows the mean access delay characteristics for different values of 'R' slots/frame. The mean access delay in this particular situation is defined as the time it takes for a mobile to access and transmit its first voice packet from the time a voice activity is detected. This delay consists of contention and allocation delays. The contention delay is due to the collision resolution process in the 'R' slots while the allocation delay is due to the queuing process. Contention delay is the time between the initial collision in an 'R' slots and successfully transmitting a reservation request in a another 'R' slot. Once a successful 'R' slot allocation is done, the requesting mobile would be placed in the allocation queue and then 'I' slot(s) would be allocated in a FIFO fashion. Therefore, the allocation delay is defined as the time spent in the allocation queue. As shown in Fig 4.8.3, under normal operating conditions for any combination of 'R' and 'I' slots, contention delay is the dominant factor at low loads. As the load is increased the delay is mainly due to the allocation process. The delay at a low load is mainly caused by the collisions during the access phase and as the load increases, the delay is mainly due to the unavailability of 'I' slots. This behaviour can be seen in Fig. 4.8.3.

When implementing an ATDMA based system it is important to use the correct combination of 'R' and 'I' slots. If an ATDMA frame is implemented with few 'R' slots then, even at low loads, mean access delay would be very high (thus throughputs would drop quickly) due to the collision resolution process. For example when the simulations were carried out according to Table 4.4 and with 4 'R' slots, even with 80 voice terminals, voice packet access delay was 678.89 ms. Also if too many 'R' slots are incorporated within a frame, even at low to medium loads, it would be difficult to find a free 'I' slot . This, would increase the allocation delay and one again decrease the throughput.

Fig. 4.8.4 shows the 'R' and 'I' slot utilisations. Results were initially obtained for voice traffic. The utilisation figure is similar for multimedia traffic as shown in the latter parts of this chapter. This figure shows the low utilisation of 'R' slots. This situation arises due to collisions in 'R' slots and also because of the low access attempts of 'R' slots. This situation can be explained analytically in the following way. Assume that the ATDMA model supports 160 voice terminals according to simulation parameters of Table. 4.4. Speech has average talkspurt duration of 1.67 seconds and a silence duration of 1.34 seconds. Therefore, each terminal would try for an 'R' slot once in every 3.01 seconds. Therefore, on average during one ATDMA frame (frame duration is 10 ms) the 160 terminals would try for (10^{-2}*160/3.01) 0.53 'R' slots. This figure would be slightly higher if collisions occurred in 'R' slots. As a result, as shown before, the best performance values were obtained when 8 'R' slots were used. In order to overcome the problem of collisions, a technique similar to the capture effect needs to be implemented in such a way so that a mobile with a relativly high power gains access to the 'R' slot during collisions [14].

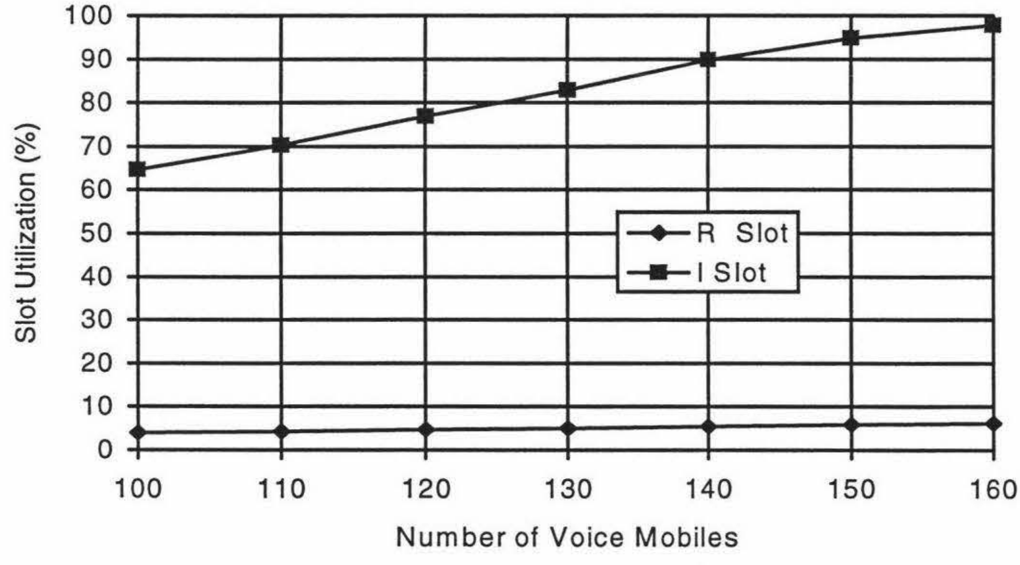


Fig. 4.8.4 Slot utilisation in a voice based ATDMA system with 8 'R' slots

Before the performance of ATDMA for multimedia traffic could be investigated, voice traffic is used to find out the best frame length for ATDMA and the locations of 'R' and 'I' slots within a frame. Table 4.5 indicates the statistical multiplexing factor of number of users per slot for different transmission rates when frames of lengths 10 ms and 16 ms were considered. The slight capacity advantage in the 10 ms structure is not adequate enough for choosing the above structure. The real advantage of using the 10 ms frame structure is in the low mean access delays. As an example, at a transmission rate of 321 kbps and voice packet loss close to 1%, the 10 ms structure had a mean access delay of 35 ms whereas, the 16 ms structure had a mean access delay of 59 ms. The reason for the shorter delays in the 10 ms structure lies in the shorter re-transmission times. Such comparisons would always be true for any transmission rate. Therefore, all the simulation results presented are based on a 10 ms frame structure. In the above simulations, the ratio of 'R' to 'I' slots have been kept around 13 %. According to Table 4.6, the 'R' slot to 'I' slot ratio of 13% generally produced the highest statistical multiplexing values of users per slot over all transmission speeds. As a result, a fixed 'R' to 'I' ratio of 13% has been used throughout this study. Table 4.6 also shows, that at 2 Mbps the ATDMA protocol supports more users for a low 'R' slot to 'I' slot ratio since there are 9 'R' slots present in the ATDMA frame. Therefore, terminals have several chances to access an 'R' slot in a given frame when collision occur; further increasing the 'R' slots would simply decrease the amount of 'I' slots available. This would result in an increase in the packet loss and a decrease in the system capacity. This is because, there are a limited amount of 'I' slots available even at low transmission speeds, due to the higher 'R' slot to 'I' slot ratio. This situation is also shown in Table 4.6.

Transmission bit rate	R slots (10 ms)	R slots (16 ms)	I slots (10 ms)	I slots (16 ms)	users/slot (10 ms)	users/slot (16 ms)
321 kbps	2	3	13	14	1.53	1.52
704 kbps	4	5	28	31	1.78	1.77
1008 kbps	6	7	39	43	1.83	1.82
2006 kbps	11	13	74	84	1.89	1.89

Table 4.5 Statistical multiplexing factor of number of users per slot for different frame lengths and transmission speeds

Transmission bit rate	R slot/I slot = 10 %	R slot/I slot = 13 %	R slot/I slot = 16 %
321 kbps	1.49	1.68	1.7
704 kbps	1.75	1.81	1.8
1008 kbps	1.81	1.83	1.82
2006 kbps	1.945	1.92	1.85

Table 4.6 Statistical multiplexing factor of number of users per slot for different 'R' slot and 'I' slot ratios and transmission speeds

Now, let us examine which 'R' and 'I' slot combination would provide the best network performance characteristics as far as the packet losses and the mean access delays are concerned. The frame structures under investigation are shown in Fig 4.9.1 through to Fig 4.9.3. The simulations were carried out at a transmission rate of 2006 kbps (thus 11 'R' slots and 74 'I' slots). At this point it is important to mention how the 'R' slots are arranged inside the ATDMA frame. When a collision does occur during an 'R' slot transmission, collided mobiles would select a re-transmission 'R' slot based on a random number generator (voice and video terminals have a higher priority over data during re-transmission). If the selected 'R' slot is in the same frame and is 3 slots or more away from the current 'R' slot, (this is due to acknowledgment delays) then the mobile would re-transmit in the same ATDMA frame. Otherwise, the re-transmission would be done in the next frame. The voice packet delay and loss characteristics for different input traffic levels are presented in Table 4.7. From the above results it can be seen that the arrangement shown in Fig 4.9.2 is preferred over the others because it supports slightly higher capacities. The reason why the structure in Fig 4.9.3 tends to fail at higher loads is because of packet losses due to higher contention. Higher contentions are caused by extra collisions during 'R' slot access since more re-transmission attempts are made by the mobiles in this particular structure compared to the other two structures. The structure shown in Fig. 4.9.1 causes higher delays since in most occasions, the terminals need to wait for the next frame to re-transmit their reservation requests when collisions do occur in the 'R' slots. The structure in Fig 4.9.2 is a combination of the other two structures and performs slightly better at higher loading levels. As a result, in this study the structure of Fig. 4.9.2 was used for simulation purposes.

R_1	R_2	.....	R_6	

Fig. 4.9.1 All 'R' slots at the start of the frame

R_1	R_2	R_3		R_4		R_5		R_6	

Fig. 4.9.2 Half of the 'R' slots at the start of the frame and the rest distributed in the latter half of the frame

R_1		R_2		R_3		R_4		R_5		R_6	

Fig. 4.9.3 All 'R' slots distributed throughout the frame

NOTE : In Fig 4.9.1 through to 4.9.3 unmarked areas of the frame consists of 'I slots.

More simulation results were obtained for different voice and data traffic combinations. Data traffic was modelled as a delay insensitive and a loss sensitive source of traffic. Data is generated in bursts of varying lengths between 1 kb to 40 kb. The bursts are generated with a negative exponential inter-arrival time of 15 seconds. Increasing the mean inter-arrival time would decrease the number of data packets generated per unit time. Data traffic is configured as a low priority source of traffic. Data terminals would be allocated one slot for a number of frames to transmit a data burst. The number of reserved frames a data terminal may get is fixed and it is not related to the data burst length. The reason for 'block' reservation is to give delay sensitive traffic higher priority.

If a data burst consists of 'N' data packets, the terminal would go through the normal 'I' slot allocation procedure along with other streams of traffic. When the data terminal is allocated an 'I' slot, it would only transmit say, 'M' out of the 'N' data packets in consecutive frames (in this particular case, 'M' packets are considered as a data block length). Then the 'I' slot would be released and the data terminal should re-try for another 'I' slot to transmit its remaining 'N-M' packets. This process of transmitting 'M' packets during any 'I' slot allocation interval would continue until all the 'N' packets have been transmitted. Then the terminal would wait in an inactive mode for a negative exponential time with mean of 15 seconds before generating another data burst. This process would continue for the entire simulation.

Number of mobiles	Voice Packet loss% (Fig 4.9.1)	Voice Packet loss% (Fig 4.9.2)	Voice Packet loss% (Fig 4.9.3)	Mean access delay (Fig 4.9.1)	Mean access delay (Fig 4.9.2)	Mean access delay (Fig 4.9.3)
100	.0043	.0027	.002	11.011 ms	10.995 ms	11.036 ms
110	.0074	.0036	.003	11.053 ms	11.068 ms	11.086 ms
120	.0097	.0057	.0036	11.099 ms	11.101 ms	11.127 ms
130	.0141	.0102	.0082	11.164 ms	11.175 ms	11.203 ms
140	.0185	.0122	.0091	11.178 ms	11.241 ms	11.258 ms
150	0.2448	0.2161	0.1881	13.271 ms	12.659 ms	11.62 ms
161	1.0235	0.9649	1.1676	35.063 ms	34.407 ms	38.977 ms

Table 4.7 Delay and loss characteristics for different Frame structures at transmission speed of 2006 kbps

Now, it is necessary to find out the appropriate data block length that suits higher traffic levels. The criteria used to determine the best data block length is the voice packet loss of 1%. Fig 4.10.1 shows the traffic capacity for different values of 'M'. The above simulations were carried out at a transmission rate of 1008 kbps with 45 voice terminals. This figure shows the maximum number of data terminals that can be supported before the voice packet losses reach 1% for different combinations of data block lengths. The results show that the traffic capacity varies with the value of 'M'. The results also show that M = 12 provides the best performance results. The causes for these variations in system capacity are the varying contention and allocation delays. As an example, consider a data block length of 4 (M = 4). Then for a given data burst this data terminal needs more 'I' slot allocation attempts than say, when M = 12. It is obvious that M= 4 would provide higher priorities to voice traffic than using M = 12 but, smaller values of 'M' would also increase 'R' slot access attempts and also collision. In such a situation, due to the increase in collision rates, the contention delay would increase. This invariably would increase the voice packet loss. This situation was elaborated when simulations were carried out for the M = 4 situation. M = 4 resulted in a maximum of 30 data terminals before voice terminals reached a packet loss of 1%. When 'M' is increased, voice packet loss may increase due to higher allocation delays. It is possible to conclude that, when choosing a suitable data block length, a trade-off between packet loss due contention and allocation delays have to be made. According to Fig 4.10.1, the best possible data block length was 12. Therefore, this particular value was used for rest of the simulation work.

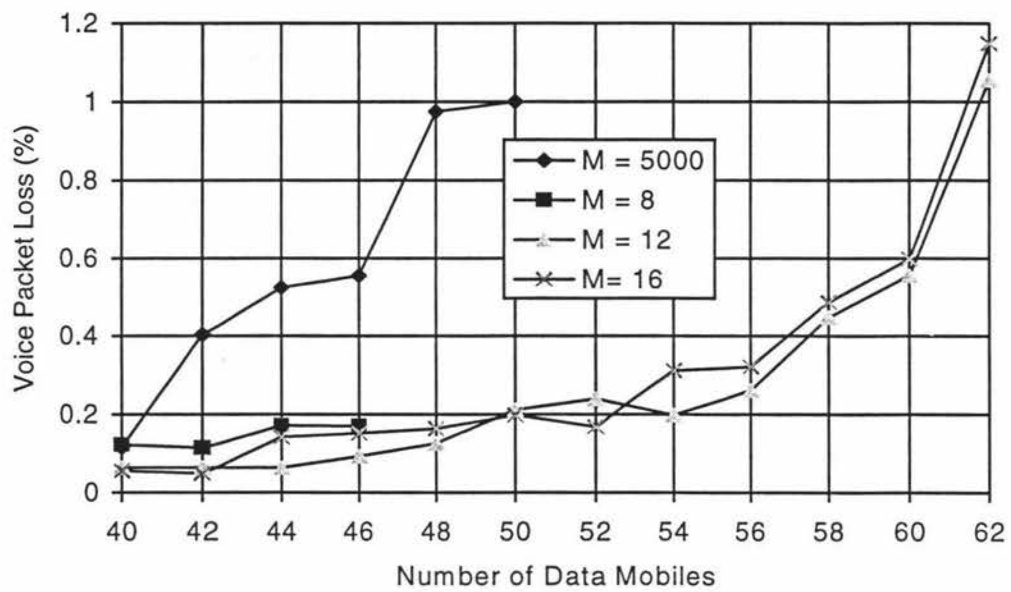


Fig. 4.10.1 Speech packet loss characteristics for varying data mobiles when different data block lengths are considered (Voice terminals are fixed at 45 and the transmission speed is taken as 1008 kbps)

Finally, mean access delay characteristics have been studied in a voice/data environment. These results are illustrated in Fig 4.10.2. As expected, mean access delay for a data terminal is slightly higher than for voice terminals. This is because when a data terminal collide during a 'R' slot access, the data terminal would have a compulsory waiting time of one frame length before re-trying for another 'R' slot. When a voice terminal or a video terminal collide, it might get another re-transmission attempt in the current frame.

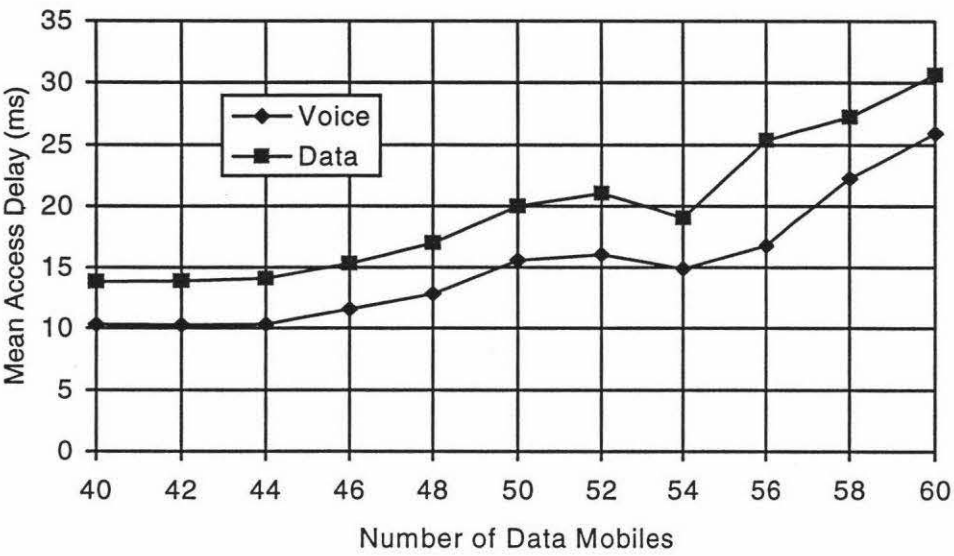


Fig. 4.10.2 Mean access delay characteristics for voice and data terminals as data terminal values are varied (Voice terminals are fixed at 45, transmission speed is taken as 1008 kbps and data block length is taken as 12)

Video traffic was added to the above model to find out the traffic performance of the ATDMA protocol for multimedia traffic. These three traffic streams have completely different characteristics and requirements but, due to the flexible nature of the ATDMA protocol, these traffic streams are accommodated through the air interface with relative ease. Table 4.8 presents all the simulation parameters that have been used during the simulations.

Performance of the protocol for mixed traffic is given in Fig 4.11.1 through to Fig 4.11.4. These characteristics were calculated at transmission speed of 1008 kbps and, keeping voice and data terminals fixed at 45 and 25, respectively. These characteristics were also studied with different number of slots allocated to each video terminal. In one of the situations, one slot per video terminal will be allocated, in another situation two slots are allocated. In the second situation if two slots are not available then one slot will be allocated. By comparing the above results, it is seen that the only advantage of adapting the two slot structure is in low average video frame transmission delays. This advantage becomes less significant as the load increases due to the unavailability of two free slots under heavy loading conditions. When two slots are not available the two slot structure would operate as same as the one slot structure, thus at high loads the video frame delays would be the same in both the situations. The one slot structure produced higher capacities due to the availability of more 'I' slots. This was expected since in the two slot structure, if two slots are allocated for video terminals then the available 'I' slots for the remaining mobiles would be lower than in the one slot arrangement. Fig 4.11.5 shows the utilisation of 'R' and 'I' slots for multimedia traffic.

Carrier bit rate	321, 704, 1008, 2006 kbps
Speech coder rate	16 kbps
ATDMA Frame length	10 ms
Down link acknowledgment delay	3 time slots
Slots per frame*	15, 32, 45, 85
Reservation slots per frame*	2, 4, 6, 11
Speech packet delay threshold	20 ms
Speech activity	43.5% - 45%
Data burst length	1 kb to 40 kb
Exponentially distributed mean data inter-arrival rate	15 Seconds
Data block length	12
Data packet delay threshold	None
Video coder rate	Variable (Max. 246 kbps)
Image size	64*64 pixels
Sub-block size	8*8 pixels
inter-arrival rate	100 ms
Video frame delay threshold	200 ms
Burst information payload	160 b
Burst overhead*	54, 60, 64, 76 b
Simulation length	300 s

(* Multiple values represents transmission rates 321, 704, 1008, and 2006 kbps respectively)

Table 4.8 Simulation parameters for ATDMA/multimedia protocol

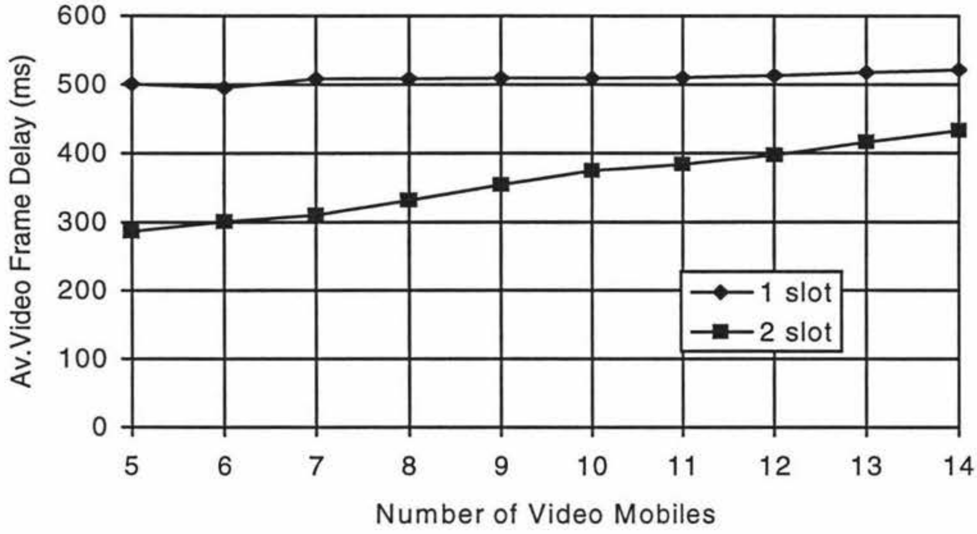


Fig. 4.11.1 Average video frame delay characteristics for 1 and 2 (combination of 1 and 2 slots) slots for video at transmission speed of 1008 kbps. Number of voice and data terminals used are 45 and 25 respectively

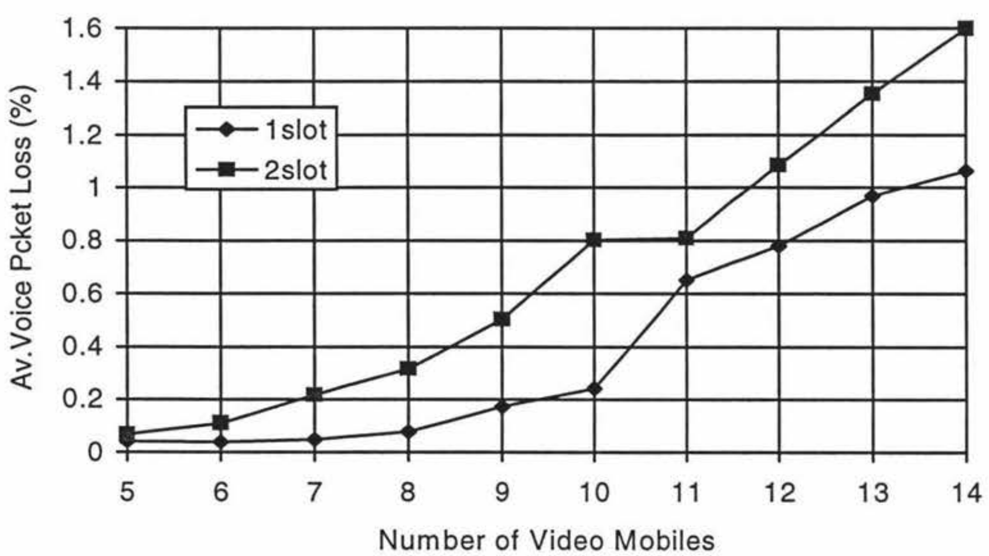


Fig. 4.11.2 Average voice packet loss characteristics for 1 and 2 (combination of 1 and 2 slots) slots for video at transmission speed of 1008 kbps. Number of voice and data terminals used are 45 and 25 respectively

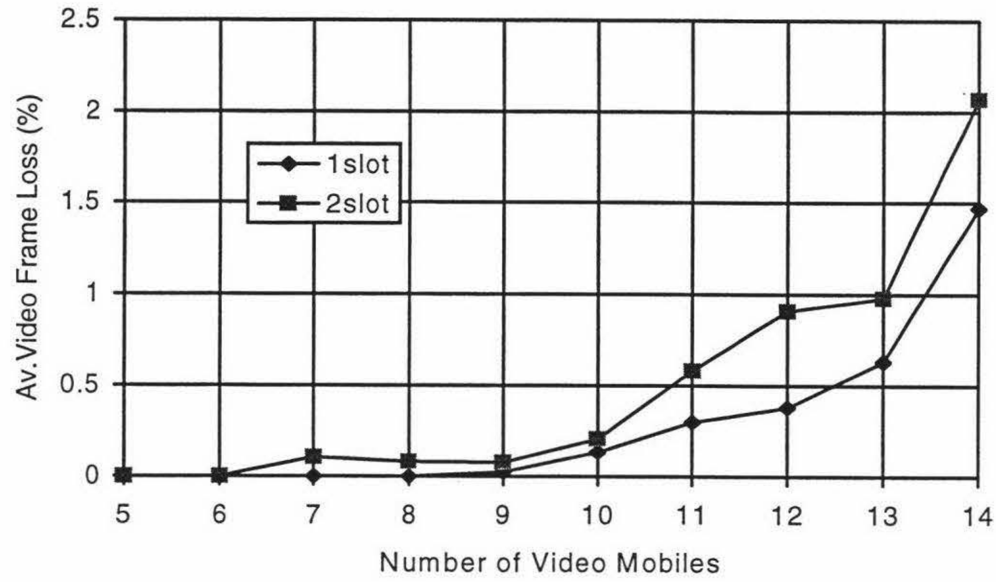


Fig. 4.11.3 Average video frame loss characteristics for 1 and 2 (combination of 1 and 2 slots) slots for video at transmission speed of 1008 kbps. Number of voice and data terminals used are 45 and 25 respectively

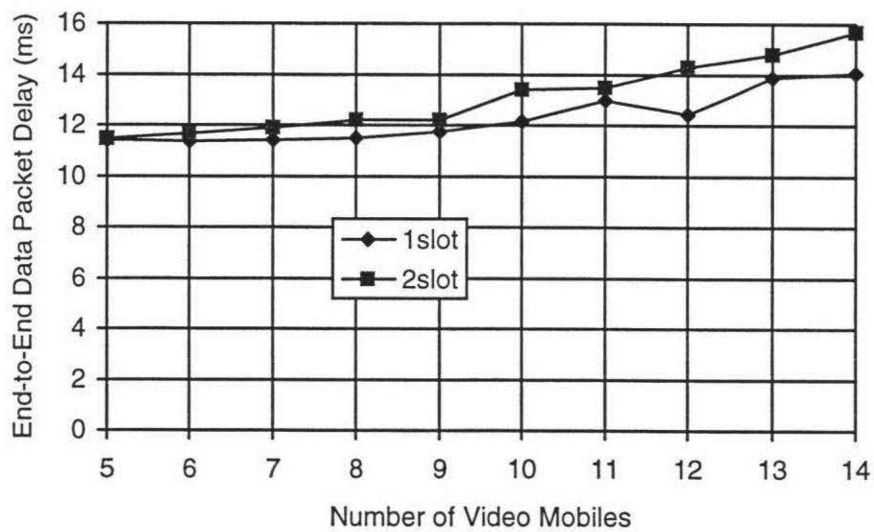


Fig. 4.11.4 End-to-end data packet delay characteristics for 1 and 2 (combination of 1 and 2 slots) slots for video at transmission speed of 1008 kbps. Number of voice and data terminals used are 45 and 25 respectively

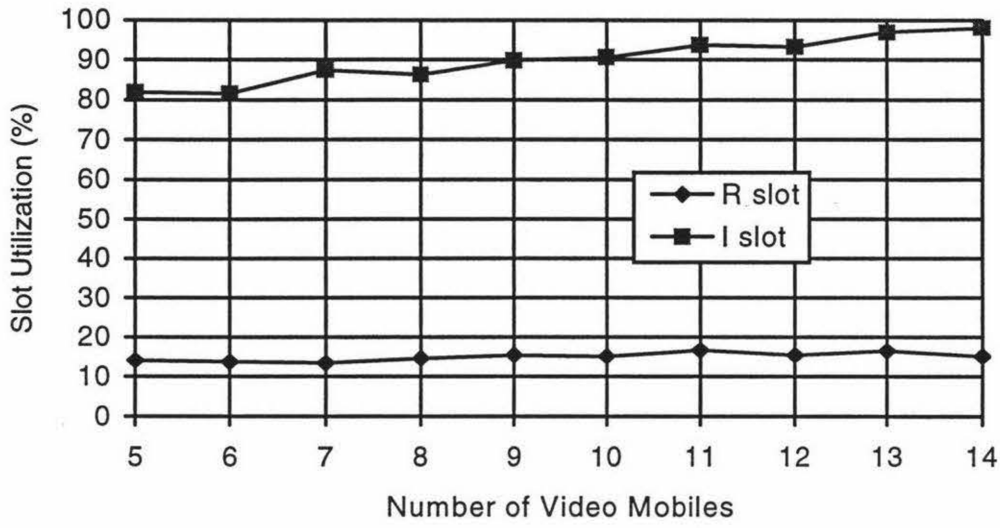


Fig. 4.11.5 Slot utilisation in a multimedia traffic scenario (1 slot/video) with transmission speed of 1008 kbps and fixed voice and data terminal values of 45 and 25 respectively

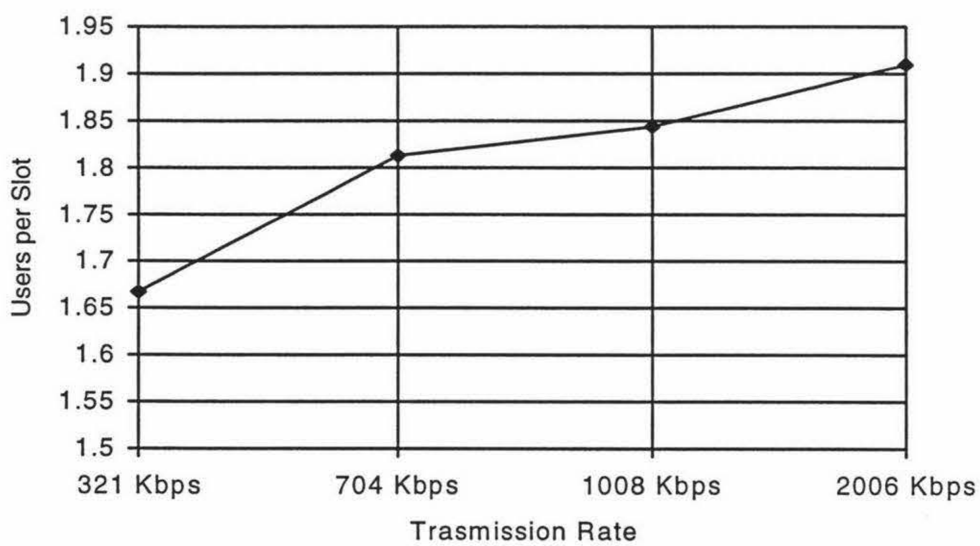


Fig. 4.12 Statistical multiplexing factor of number of users per slot in a multimedia scenario (1 'I' slot per video terminal)

Finally, Fig 4.12 shows the statistical multiplexing factor of a number of users per slot at different transmission rates. From this figure an increase is seen in the number of users per slot as the transmission speed is increased. This is because the terminals get more 'R' slot access attempts at high transmission rates. This is an extremely important situation in a micro-cellular environment.

4.6 Capacity Calculations of Up-link ATDMA Control Channels

To calculate the required control channel capacities a transmission speed of 1.008 Mbps was selected during this study. This simulation consisted of 45 voice terminals, 25 data terminals and 13 video terminals. For calculation purposes, control frames and traffic frames were assumed to be of the same length. Channel capacity calculation were based on the RACH channel assignments. This is because most of the signalling on the up-link would follow only after a successful RACH assignment. As an example, during initial access or handover, a request packet must be successfully transmitted via an 'R' slot to setup a DCCH channel. There is other signalling information exchanged in an ATDMA based system (e.g. link adaptation, power control, broadcasted information by BS). These signalling information is either carried on the ACCH or on the down-link control channels. For this reason, it is necessary to examine the behaviour of the RACH assignments. Fig. 4.13 shows the average number of 'R' slots used per frame as the number of video mobiles are changed and Fig. 4.14 shows the distribution of the maximum number of 'R' slots used per frame. The results were calculated at 20 second intervals. From the above graph, it is noticeable that during the simulation, most of the time a maximum of 5 'R' slots per frame have been utilised. Therefore, the study assumes 5 active 'R' slots per frame in the capacity calculations.

The capacity of the DCCH: A DCCH channel would be used during call access and handover. Terminals request for a DCCH via a RACH. As discussed above regardless of the mobile terminal types and the functions they want to perform (access or handover) 5 DCCH channels would be created per frame. In a steady state condition, in one ATDMA frame, 5 slots would be used for DCCH. The ATDMA packet size at 1008 kbps is 224 bits/packet (this includes the data bits, ACCH signalling bits and the overhead bits). Thus, the total bandwidth (with overhead and signalling) used for DCCH is $(5*224)/(10*10^{-3}) = 112.0$ kbps. DCCH channel capacity also consists of ACCH information. As ACCH is a logical channel it is not considered to be a dedicated capacity. ACCH signalling is passed through DCCH and TCH as part of the normal burst. In the above structure, if the 5 DCCH channels reserved for an ATDMA frame are being used, then in some situations terminals might have to wait for a DCCH once a successful 'R' slot assignment is performed.

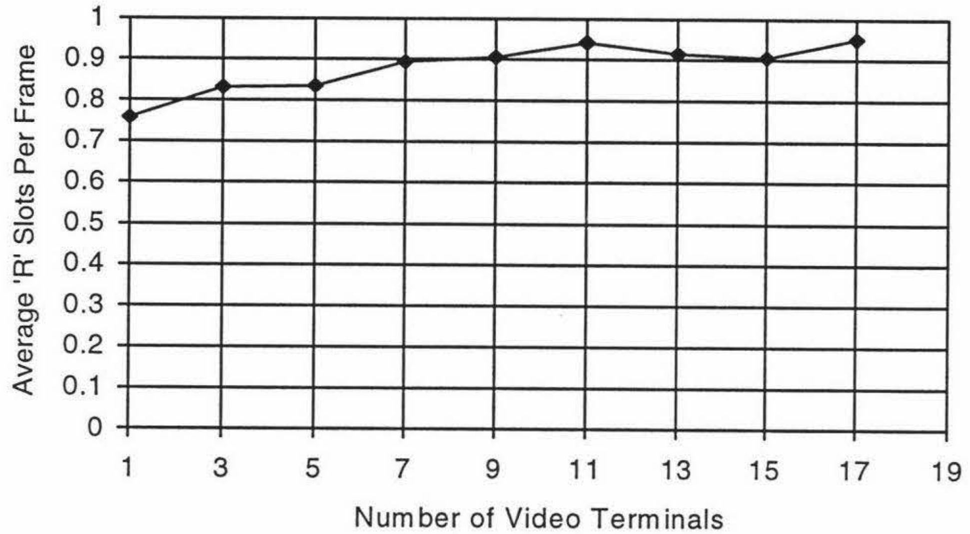


Fig. 4.13 Number of 'R' slots used per frame at transmission speed 1008 kbps

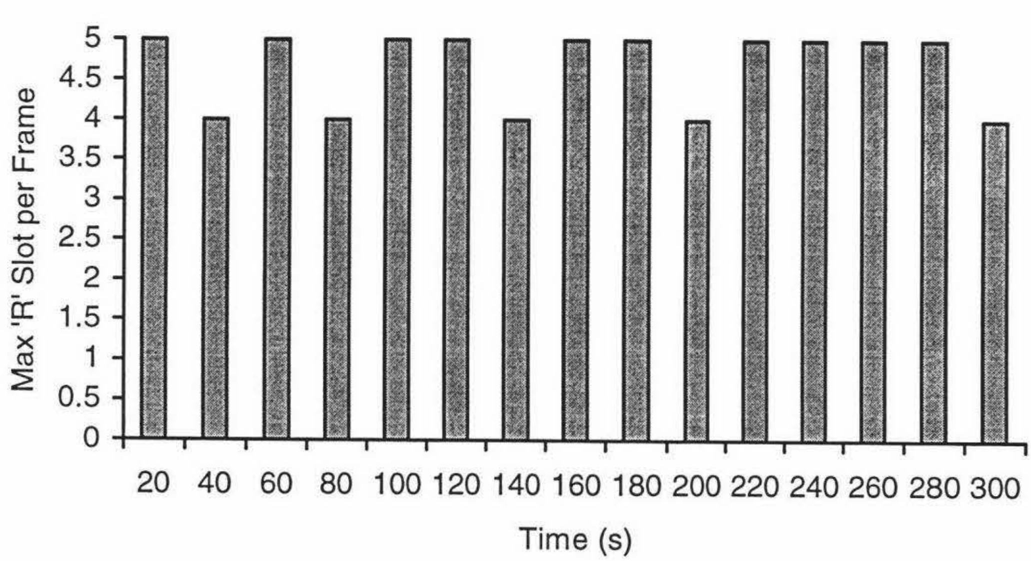


Fig. 4.14 Max. No of 'R' slots per frame characteristics at transmission speed of 1.008 Mbps (45 voice, 25 data and 13 video terminals have been used)

The capacity of RACH: RACH channels occupy the 'R' slots of an ATDMA frame. Although there are 6 'R' slots in a frame, their usage is limited to 5 slots per frame. During RACH capacity calculations it is important to assume 6 'R' slots rather than 4. Therefore, the bandwidth reserved for RACH would be $224*6/(10*10^{-3}) = 134.4$ kbps.

The capacity of LCCH: Each mobile would use a LCCH signalling channel once every 20 frames. This study assumed that each mobile would use a LCCH channel every 200 ms. LCCH signalling between MS and BS would occur for the duration of the call (during active and inactive periods). Therefore, the total number of slots used for LCCH in any given frame by the 83 mobiles would be $83/20 = 4.15 \approx 5$ slots. The resulting total bandwidth for LCCH would be $5*224/(10*10^{-3}) = 112$ kbps. The LCCH capacity would be lower than 112 kbps if no LCCH is used during the inactive periods of data terminals.

The capacity of FPak: This is an acknowledgment channel for the Fast Paging (FP) that occur on the down link. FP is a common multiplexed down-link channel used to setup down-link traffic channels. FP is used by BRA to inform MS when a traffic slot is selected for its down-link communication. Communication could begin once the MS acknowledges the received slot (channel) on the FPak. One FP slot has been used in the down-link structure [14]. This would result in one FPak on the up-link. Regardless of this slot being used or not it would reserve a bandwidth of $1*224/(10*10\text{-}3) = 22.4$ kbps.

The capacity of TCH: At 1.008 Mbps there would be 45 slots in the ATDMA frame. Six of these slots would be used for 'R' slots whereas one slot would be used for FPak. Therefore, 38 slots would be reserved for user traffic transmission. Hence, the user traffic capacity would be $(38*224)/(10*10\text{-}3) = 851.2$ kbps. This would result in a signalling volume to total traffic volume ratio of $(380.8)/(380.8 + 851.2) = 30.9$ %.

Similar calculations were carried out at transmission speed of 2.006 Mbps. The simulation used 163 mobile terminals (85 voice, 50 data and 28 video terminals). The ATDMA frame at the above transmission speed, consisted of 11 'R' slots, 72 'I' slots and 2 FPak slots. The resulting distribution of the maximum number of 'R' slots used per frame is given in Fig. 4.15. The figure shows that a maximum of 7 'R' slots per frame have been used during the simulation. The resulting capacity calculations are shown in Table 4.9 (the ATDMA packet consisted of 236 bits at 2.006 Mbps). It is important to remember that the control channel capacity calculations were done for the worst case scenario since in both the transmission speeds, the maximum number of active 'R' slots per frame were considered.

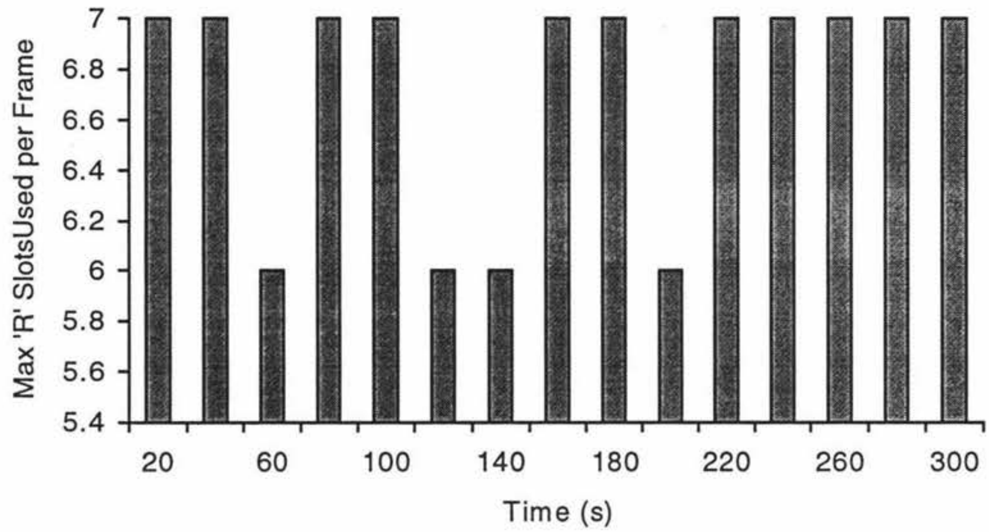


Fig. 4.15 Max. No of 'R' slots per frame characteristics at transmission speed of 2.006 Mbps (85 voice, 50 data and 28 video terminals have been used)

DCCH capacity	165.2 kbps
RACH capacity	259.6 kbps
LCCH capacity	212.4 kbps
FPak capacity	47.2 kbps
TCH capacity	1699.2 kbps
Signalling to total traffic capacity	28.7 %

Table 4.9 Channel Capacities at Transmission Speed of 2.006 Mbps

4.7 Conclusion

The main aim in simulating the versatile ATDMA protocol in a multimedia scenario was to evaluate its performance. The above discussion highlighted the necessity of using the best possible 'R' and 'I' slot structure. The results indicated that in terms of packet loss and delay, the structure presented in Fig 4.9.2 is better suited for the ATDMA protocol.

Fig 4.8.4 and Fig 4.11.5 indicated that the 'R' utilisation was low despite the random re-transmission process adopted during collisions. The low utilisation value of 'R' slots was not only due to the collisions but also due to the fact that no attempt were made on some 'R' slots to transmit reservation requests. Whatever the reason, when collisions occur during 'R' slot access (specially at low 'R' values), if a suitable 'R' slot allocation procedure could be implemented (e.g. by using the capture effect), it would result in higher 'R' slot utilisation values. Then lower R/I slot ratios would be possible thus, further increasing the capacity of the ATDMA based network.

Although the 10 ms frame structure compared to a 16 ms frame structure did not provide capacity advantages, the smaller frame size resulted in a decrease in the access delay. This is because of the shorter frame length.

When data traffic was incorporated into the simulation, its low priority, delay insensitive, loss sensitive natures were exploited. This resulted in data traffic being transmitted in blocks. The number of packets transmitted each time (or data block length) directly affected the system performance. When very small data block lengths were considered, too many 'R' slot collisions were triggered since frequent access attempts for data terminals were needed. Larger data burst can increase the delay for voice and video terminals. In this situation, voice and video information would be lost due to excessive delays. Therefore, in choosing a suitable data block length a trade-off needs to be made between video and voice packet losses due to contention and allocation. Fig 4.10.1 presents system capacities when different data block lengths were considered. These indicated that the best data block length was 12.

When video terminals were incorporated into the simulation, video terminals were allocated two slots for transmission. If two free slots were not available, a single slot was allocated. These results were presented in Fig 4.11.1 through to Fig 4.11.4. The only advantage that two 'I' slots/video-terminal have over the other structure was in terms of low video frame transmission delays. Even this advantage disappeared rapidly in heavy loaded conditions. This was due to the unavailability of two free traffic slots during 'I' slot allocation.

In this chapter some channel capacity calculations were also reported. These studies showed that the ATDMA system generated signalling to total traffic volumes of 30.9 % and 28.7 % at transmission speeds of 1008 kbps and 2006 kbps, respectively. It is important to note that the above signalling volume generated was the maximum supported on the up-link.

CHAPTER 5
Conclusions and Suggestions for Future Work

In this study several aspects of an ATDMA/ATM based PCN have been investigated. Initially, the traffic capacity of a statistically multiplexed multiple access technique known as ATDMA was extensively investigated by means of a simulation model. Simulation is a computer model based representation of a real system. Simulation is a powerful tool that can be used to study the behaviour of any system without building the system. In research, computer simulations are becoming an important tool to measure the performance of any system. Performance figures can be examined for various input conditions and operating environments. This study used a discrete event simulation language [30] to examine different characteristics of an ATDMA based mobile network for multimedia traffic. Discrete event simulation describes the model of a real system in terms of logical relationships which cause changes of states at discrete points in time. The discrete simulation model generally tends to be both stochastic and dynamic in nature. In a discrete event based simulation model, results are obtained by generating scheduled events at different points in time in a simulation model. In this case changes in the physical system are represented by a series of discrete changes or events at specific instants in time. The simulation model used in this study consists of several source models. The developed source models include voice traffic, delay insensitive data traffic and variable low bit rate video traffic. The simulation model also accurately models the ATDMA protocol.

Several parameters of the ATDMA protocol were studied using the simulation model. First the ATDMA frame structure in terms of 'R' and 'I' slot arrangements and the ratio of R/I slots was investigated. This is important because the statistical multiplexing efficiency, delay and the packet loss depend to some extent on the frame structure. The simulation results based on speech packet loss and speech packet delay indicated that the frame structure with half of the reservation slots ('R' slots) located at the beginning of the frame while rest of the 'R' slots randomly distributed throughout the latter half of the frame offers slightly better performance, as shown in Table 4.7. Simulation results also indicate that a 'R' slot to 'I' slot ratio of 13 % provides higher traffic efficiency for all transmission rates.

The Slotted-ALOHA protocol is used to access a 'R' slot, which carries the reservation request. Simulation results show that the maximum 'R' slot utilisation was 6 %, even though the slotted-ALOHA maximum R slot utilisation is 36 %. The utilisation of 'R' slots is due to a combination of the collisions in 'R' slots and fewer access attempts using 'R' slots. For example, a mobile transmitting voice packets accesses the channel on average once every 3 seconds (duration of a talkspurt + a silence period). Analytical results presented in chapter 4 show that when 160 voice terminals are operating, 0.53 'R' slots per frame would be utilised regardless of the number of 'R' slots available in the frame. However, this figure will change with the traffic type, the capacity of the system and the amount of retransmission attempts due to collisions in 'R' slots. Collisions in 'R' slots might increase the channel access delay, resulting in loss of information packets for delay sensitive traffic. The number of collisions in 'R' slots can be decreased by using the capture effect where a 'R' slot is allocated to a terminal based on the received signal strength. The capture effect will increase 'R' slot utilisation.

Receiver capture based on dominant power will recognise a particular signal with a relatively high power despite the presence of other transmitting terminals, interferes (i.e co-channel and adjacent channel interferes) and noise (mostly device noise). The probability of capture is related to the quality of reception, but only to the extent that a low capture probability corresponds to poor reception quality because of interfering terminals and other contending terminals. The receiver capture is essentially an event of the physical layer of a communication system. The main parameters which influence receiver capture are; received power level, type of modulation, robustness of the receiver synchronisation, Doppler and frequency dependent channel fading, shadowing due to variety of obstacles (i.e buildings, trees, etc.), frequency and distance dependent pathloss and interference [67]. Different capture mechanisms have been proposed for high power users to access a channel successfully [67]. Some of these are the capture ratio method [68], the classification of power level method [69], the vulnerability circle method [71] and the nonfade interval method [72]. The capture ratio mechanism is adopted in the

GSM system. In the capture ratio mechanism, a packet is regarded as correctly received at the receiver if the received power exceeds the joint interference and contender powers by at least a threshold factor (or capture ratio). Typical capture ratios for analog systems range from 3-20 dB [68], while for a GSM system, a capture ratio of 9.5 dB [72] has been described for the receiver.

Simulation results show that the ATDMA protocol supports a high number of users per slot. However, the presence of a higher number of users in a cell may degrade the quality of the overall system, due to an increase in co-channel and adjacent channel interference. The interference issues require further investigation. Interference pattern of an ATDMA based system would be different from that of a circuit switched TDMA system because of the different slot allocation procedure.

Simulation results also indicated that the statistical multiplexing efficiency of the protocol increases with the increase of transmission speed, as shown in Fig. 4.12. This is due to the increase in efficiency of the retransmission scheme. As the transmission bit rate is increased, the number of 'R' slots available in a transmission frame would increase. In this case the probability of a successful retransmission attempt is higher than that of a lower transmission speed.

Data traffic was transmitted in blocks of size 'M' packets per reservation. The 'block' transmission mechanism influences the protocol performance. A small block size (M) would increase the contention, whereas a large block size would increase the allocation delay for voice and video traffic due to the non-availability of 'I' slots. Therefore, when choosing a suitable block size effects of video and voice packet loss due to contention and allocation have to be considered. Simulations were carried out with different data block lengths. The results indicated that the best data block length is 12, as illustrated in figure 4.10.1.

In this study the optimum packet transmission strategy for a video terminal was investigated. Video terminals were allocated two 'I' slots to transmit video packets and under heavy loading conditions sometimes one 'I' slot per video terminal was allocated according to the availability of 'I' slots. Simulation results revealed that, in most situations when two 'I' slots are allocated to video terminals, the channel access delay for delay sensitive traffic would increase because more 'I' slots are occupied by the video terminals. In this case the voice packet loss and the video frame loss also increase slightly. This is due to the unavailability of 'I' slots. The only advantage of the two 'I' slot allocation is the reduced video frame transmission delay (Fig 4.11.1). This advantage becomes insignificant as the load is increased because the probability of allocating two 'I' slots for video packet transmission decreases. It is suggested, that in a multimedia environment at high traffic load, one 'I' slot per video terminal should be allocated.

Issues relating to ATDMA based PCN architecture and signalling have also been addressed in this study. Some of the important reasons why the ATM protocol has been proposed as the ideal candidate for the fixed network part of the PCN are; its packet switched nature, flexible bandwidth allocation for bursty traffic, efficient multiplexing of traffic from different sources and the ease of inter-connectivity with future networks [12]. Proper inter-working functions need to be developed in order to transport ATDMA based information through an ATM backbone network. This requires understanding the functional requirements of both the networks and then developing interworking functions based on the Open System Interconnection (OSI) standard.

A small set of bearers needs to be developed for the ATDMA based system, similar to that of ATM Adaptation Layer (AAL) bearers. These ATDMA bearers correspond to the UNL of Figure 3.4.1. However, the choice of bearers should differ from B-ISDN service classes since ATDMA bearers should specifically target the needs of the radio access system. When determining the ATDMA radio access bearer types, issues such as connection versus connectionless, speech and different classes of data bearers have to be considered. It is also important to map the UNL functions with the appropriate AAL layer functions on a fixed network in order to achieve compatibility of the networks.

When an ATM based fixed network is developed to support the ATDMA system, features such as interworking, handover, QoS management due to adverse wireless channel conditions, authentication and registration related issues have to be considered.

The interworking mechanism discussed in this study (section 3.4.1) involves the lower three layers of the OSI stack (physical, data link and network layer). The interworking functions for the user and the control planes were suggested at the base station level. This is because some of the functions of an ATDMA based system are performed at the BS. As an example, channel set-up and release functions are performed by the resource allocator located at the BS [61]. Depending on the type of information, the lower layers of the ATDMA system could also be terminated at nodes beyond the BS (i.e mobile switching center, service control point, etc). This is because some of the information transported over the radio system is intended for nodes beyond the BS. For example, interworking functions relating to handover could be located directly at a Service Control Point (SCP) [61]. The work on signalling architecture needs to be further extended when the intelligent network issues of PCN systems are considered.

Simulation results were also used to calculate the up-link channel capacities at transmission speeds of 1 Mbps and 2 Mbps. These calculations show that signalling traffic takes 30.9% and 28.7% of the total transmission bandwidth at 1 Mbps and 2 Mbps, respectively. Most of the up-link signalling is generated by the operation of RACH and DCCH channels

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APPENDIX A

```
PREAMBLE
  RESOURCES
   EVERY UNIT HAS A NO.LOSS
  DEFINE
   NO.LOSS AS AN 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 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 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 WAITINGQ AS A SET RANKED BY LOW ARRIVAL.TIME
BREAK MOBILE TIES BY LOW ARRIVAL.TIME

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, MIN.R, MAX.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
END "PREAMBLE
```

```
MAIN

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

 CALL READTALK
 CALL INITIALIZE

 START SIMULATION

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

END ''MAIN
```

```
ROUTINE BASE
 GIVEN MOBILE AND ACTION
 YIELDING TR.STATUS
" Performs A-TDMA transmission functions related to a base station

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

 SELECT CASE ACTION

 CASE .SIDTX
" If contending for an 'R' slot, figure out whether successful or not

 '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
     FILE THIS MOBILE IN TRANSMISSIONQ
     WAIT (0.5*SLOTS*1000) .MILLISECONDS  " WAIT UNTIL START OF NEXT  SLOT

IF COUNT EQ 1
" If 'R' slot allocation is successful enter the mobile in a waiting queue for 'I' allocation

          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
         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
"If more than 1 mobile contend for the 'R' slot re-assign next 'R' slot access attempt for the collided
"mobiles. Data terminals wait until next frame where as, voice and video might be contending for a 'R'
"slot in the same frame.
```

```
          ADD 1 TO TEMP.COUNT
            IF COUNT = TEMP.COUNT
              LET COUNT = 0
              LET TEMP.COUNT = 0
            ALWAYS
          REMOVE THIS MOBILE FROM TRANSMISSIONQ
          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
                  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)
                  TR.STATUS = .LOSER
          ALWAYS
ALWAYS

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

CASE .LASTPKT
" If last packet being transmitted release the 'I' slot

   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
" If tranmitting information update appropriate registers

   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
" If in the waiting queue figure out if it is possible to assign 'I' slot(s)
```

```
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
    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)
     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
'' If in the waiting queue (trying to transmit its last packet) figure out if it is possible to assign a 'I'  slot

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

 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
  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
  ELSE
      REMOVE THIS MOBILE FROM WAITINGQ
      FILE WINNER.MOB IN WAITINGQ
      SUBTRACT 1 FROM Q.COUNTER
      TR.STATUS = .LOSER
      LET RESER.MOB(MOBILE) = 0
  ALWAYS
 ALWAYS

DEFAULT
```

```
" Error situation has occurred

   WRITE AS "ERROR - UNKNOWN FLAG PASSED TO BASE",/
  ENDSELECT
'END.BASE'
  RETURN
END "BASE
```

```
ROUTINE INITIALIZE
" Performs initializing of global variables and structuring the A-TDMA frame into 'I' and  " 'R'   slots.

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

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 MAX.R             = 0
LET MIN.R             = NUM.SLOT
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

" Find the locations of 'R' slots in the A-TDMA frame

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

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
```

```
PROCESS MOBILE
" Performs some A-TDMA transmission mechanism related functions

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

 IF RESER.MOB(MOBILE) = 0
 " Checks for new calls
  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  " Assigns terminal the time slot in which the waiting
                                  "queue is examined
    LET ACTION = .INQTX           " Terminal in waiting queue
   ELSE
    IF (LAST.PACKET(STATION) EQ .SET)
     IF TYPE(MOBILE) = .DATA
      LET ACTION = .LASTPKT  " Packet about to transmit is from a data mobile and also the last
                             " packet of a "datablock" or data burst
     ALWAYS
     IF PACKET.NO(MOBILE) = 1
      LET ACTION = .LASTPKT  " Packet about to transmit is from a voice or video mobile and
                             " also the last packet of a talkspurt or a video frame
     ALWAYS
    ELSE
     LET ACTION = .DATATX      " About to transmit a voice, video or a data packet
    ALWAYS
   ALWAYS
 ELSE
  " Assigns a 'R' slot to transmit access information
  '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         " About to transmit accees information on a 'R' slot if allocated
 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              "Terminal in waiting queue to trnsmit last packet
   ELSE
       ACTION = .INQTX                        " Terminal in waiting queue
      ALWAYS
   ELSE
      SLOT.NUM = RESER.MOB(MOBILE)
      RSLOT.FLG(SLOT.NUM) = .SET
      LET ACTION = .SIDTX " About to transmit accees information on a 'R' slot if allocated
   ALWAYS
  ALWAYS
'GETTX'
  CALL SCHEDULER GIVING SLOT.NUM YIELDING TIME.DIFF
 " Timing functions are performed by scheduler

  WAIT TIME.DIFF .MILLISECONDS              "for beginning of slot

  CALL BASE GIVING MOBILE AND ACTION YIELDING TR.STATUS
" Performs A-TDMA transmission functions related to a base station

  LET STATUS(MOBILE) = TR.STATUS

 IF TYPE(MOBILE) = .VOICE
"In case of voice terminals, based on the information passed by routine BASE, find whether successful
"in transmitting access information on a 'R' slot (.WINNER), unsuccessful in transmitting access
"information (.LOSER), successful in transmitting a packet (.OK), error has occurred (.ERROR), in
"the waiting queue (.INQUE) or successful in transmitting its last packet (.RELEASE). Then update
"appropriate registers.

   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
 ELSE
  " 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
"In case of data terminals, based on the information passed by routine BASE, find whether successful in
"transmitting access information on a 'R' slot (.WINNER), unsuccessful in transmitting access
"information (.LOSER), successful in transmitting a packet (.OK), error has occurred (.ERROR) or
"successful in transmitting its last packet (.RELEASE). Then update appropriate registers

 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

 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
 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
"In case of video terminals, based on the information passed by routine BASE, find whether successful
"in transmitting access information on a 'R' slot (.WINNER), unsuccessful in transmitting access
"information (.LOSER), successful in transmitting a packet (.OK), error has occurred (.ERROR), in
```

"the waiting queue (.INQUE) or successful "in transmitting its last packet (.RELEASE). Then update "appropriate registers.

```
 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
        LET 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 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
  " Going to give it another bash...
  LET SLOT.NUM  = RESER.MOB(MOBILE)
  LET SID.TXED.FLAG(MOBILE) = .NOTSET
  GO TO 'RETRY'
 ALWAY
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
 ELSE
  " We are going to give it another bash...
   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
 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
" Pass relevant information to process STATION before re-activating it
IF RESER.MOB(MOBILE) NE (NUM.SLOT + 1)
        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
" Prints relevent ouput information

PRINT 71 LINES WITH
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 - MEANDTC),
MEANTDD,
MEANDB,
V_TNDOS,
V_TNAS,
((V_TNDOS/V_TNAS)*100),
V_SUCCESS,
(V_SUCCESS+V_NUM.LOST.PKTS),
V_NUM.LOST.PKTS,
(VIDEO.FRAME.DROPOUT/HEY)*100,
MEANVEC,
MEANVE,
(MEANVE - MEANVEC),
MEANFR,
(V_TNPKTS/HEY)
AS FOLLOWS
```

```
```````````````````````````````````````````````````````````````````````````````````````````
`        SCHEDULED ACCESS : A-TDMA                                                        `
```````````````````````````````````````````````````````````````````````````````````````````
` 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                                                       `
` ALLOCATION DELAY(DATA) ***.*** ms                                                       `
` TOTAL END-TO-END DELAY *********.**** ms                                                `
` END-TO-END DATA BURST DELAY ********.**** ms                                            `
```````````````````````````````````````````````````````````````````````````````````````````
`            VIDEO                                                                        `
```````````````````````````````````````````````````````````````````````````````````````````
` TOT VIDEO DROPOUTS   *******                                                            `
` NUM SUCCESS ASSGNTS(VIDEO)  *******                                                     `
` DROPOUTS VS SUCCESS  **.****%                                                           `
` SUCCESSFUL PACKETS(VIDEO) ******                                                        `
```
```

```
` TOT VIDEO PACKS ( VIDEO) *******                      `
` LOST PACKETS(VIDEO)     *******                       `
` AV.LOST FRAMES(VIDEO) ***.****%                       `
` CONTENTION DELAY(VIDEO) ***.*** ms                    `
` AV. ACCESS DELAY(VIDEO) ***.*** ms                    `
` ALLOCATION DELAY(VIDEO) ***.*** ms                    `
` AV.FARME DELAY(VIDEO)   *****.****** ms               `
` AV. VIDEO PACKETS PER FRAME *******.****              `
````````````````````````````````````````````````````````

SKIP 3 LINES
STOP
END"OUTPUT
```
```

```
ROUTINE READTALK
" Reads some information from the input

DEFINE METRA, AND INDEX AS INTEGER VARIABLES

OPEN UNIT 1 FOR INPUT, NAME IS "TALKDATA"
  USE UNIT 1  FOR INPUT

  RESERVE NAME1(*) AS 60
  RESERVE NAME2(*) AS 60
  RESERVE NAME3(*) 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 1

  OPEN UNIT 1 FOR INPUT, NAME IS "DATA10"
  USE UNIT 1 FOR INPUT
  LET EOF.V = 0
  FOR INDEX = 1 TO 19, READ NAME3(INDEX)
  LET EOF.V = 1

  LET INDEX = 0
  LET INDEX = INDEX + 1
  LET BIT.RATE = NAME3(INDEX) " Transmission rate
  LET INDEX = INDEX + 1
  LET CODER = NAME3(INDEX)" Channel coder rate
  LET INDEX = INDEX + 1
  LET SIMUL.LENGTH = NAME3(INDEX) " Length of simulation
  LET INDEX = INDEX + 1
  LET NUM.MOBILES = NAME3(INDEX)" Total number of mobiles
  LET INDEX = INDEX + 1
  LET FRAME.LENGTH = NAME3(INDEX) " Length of the A-TDMA frame
  LET INDEX = INDEX + 1
  LET WARM = NAME3(INDEX)" Warm-up time
  LET INDEX = INDEX + 1
  LET TALKSEED = NAME3(INDEX)" Talkseed (used for talkspurt generation)
  LET INDEX = INDEX + 1
  LET SILSEED = NAME3(INDEX)" Silseed (used for silence generation)
  LET INDEX = INDEX + 1
  LET RES.SLOTS = NAME3(INDEX)" Number of reservation slots in each frame
  LET INDEX = INDEX + 1
  LET PACK.DROP.RATE = NAME3(INDEX) " Voice packet drop threshold
  LET INDEX = INDEX + 1
  LET DATASEED = NAME3(INDEX)" Dataseed (used for data packet generation)
  LET INDEX = INDEX + 1
  LET INTERARRIVAL.TIME = NAME3(INDEX) " Data inerarrival rate
  LET INDEX = INDEX+1
  LET VOICE.MOB = NAME3(INDEX)" Number of voice mobiles
  LET INDEX = INDEX+1
  LET VIDEO.MOB = NAME3(INDEX)" Number of video mobiles
  LET INDEX = INDEX+1
  LET DATA.BLOCK = NAME3(INDEX)" Number of data packets transmitted at time
  LET INDEX = INDEX+1
  LET OVER.HEAD.BITS = NAME3(INDEX) "Number of overheads bits for the frame
  LET INDEX = INDEX+1
```

```
  LET FRAMES.PER.SEC = NAME3(INDEX) " Frames transmitted per second
  LET INDEX = INDEX+1
  LET VIDEO.SLOTS = NAME3(INDEX) " Maximum video slots allocated per terminal

  CLOSE UNIT 1
END "READTALK
```

```
ROUTINE SCHEDULER GIVEN SLOT.NO YIELDING TIME.DIFF
" Calculate the waiting time required before transmission or an attempt for
" transmission could be made.

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                    " Done for reliability
  LET ST.TIME = ST.TIME + FRAME.LENGTH
  IF ST.TIME LT TIME.V
   WRITE AS "##WARNING - 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)

RETURN
END "SCHEDULER
```

```
ROUTINE SILENCE
" Calculates silence periods

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
" Generates voice, data and video packets. Process station is also apart of the transmission "mechanism

DEFINE REFERE AS A REAL VARIABLE

WHILE TIME.V LE SIMUL.LENGTH
DO

 IF ST_TYPE(STATION) = .VOICE

"Generates silence periods (with the help of routine SILENCE) and wait for that time. It also generate
"talksputs according to the talkspurt duration passed by routine TALKSPURT.

   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

   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

" Generates data packets involved with a data burst
 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)))
        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
```

```
        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

" Generates video packets needed to be transmitted in an video frame (with the help of routine 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

   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
   " Passs relevent information to process 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)

  ACTIVATE THIS MOBILE NOW
  SUSPEND " suspends STATION

 " Update relevant information after the functions on routine BASE and process MOBILE

  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))
  LETVIDEO.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
   " If last packet, last packet indicator would be set
        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)
  " If last packet of a data block, last packet indicator would be set
    LET LAST.PACKET(STATION) = .SET
  ALWAYS
 ALWAYS

  IF TIME.V GT SIMUL.LENGTH
   CALL OUTPUT
  ALWAYS

 LOOP
LOOP

END "STATION
```

```
ROUTINE TALKSPURT
" Generate talkspurt duration

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
" Process TEMPRO is a temporary process used to print results during the simulation

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 "ATDMA10.TXT"
USE UNIT 2 FOR OUTPUT

WHILE TIME.V LT SIMUL.LENGTH
DO
 IF TIME.V GT 0
        PRINT 4 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, SUCCESS.R, MIN.R AND MAX.R THUS
        AT TIME : *****.**** Q.LENGTH *****.**** VID.PACKETS: ********    TOT.PACKET:
        ************ FRAME.NO : ****** SUCCESS.'R': **
        MIN.R : ** MAX.R : **
       LET MAX.R = 0
       LET MIN.R  = NUM.SLOT
 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
" Keeps a track of the frame number, slot number and the next 'R' slot in the A-TDMA frame

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
 IF SUCCESS.R GT MAX.R
   LET MAX.R = SUCCESS.R
 ALWAYS
 IF SUCCESS.R LT MIN.R
    LET MIN.R = SUCCESS.R
  ALWAYS

 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
```

```
ROUTINE VIDEO
" Generates video packets according to [36, 37]

   DEFINE CELLS_PER_SUBBLOCK AS INTEGER 1-DIMENSIONAL ARRAY

   DEFINE MODEINFO AS REAL 1-DIMENSIONAL ARRAY

   DEFINE TIMESTEP,
   AA,AI,DA,DI,
   U,
   TIME
   AND T AS REAL VARIABLES

   DEFINE SUBBLOCKSIZE,
   NO_OF_SUBBLOCKS,
   SECTIONNO,
   MODE,
   I,
   NO_OF_SECTIONS,
   SECTIONSIZE,
   X
   AS INTEGER VARIABLES

  DEFINE NO.OF.PACKETS AS A INTEGER VARIABLE

  "VIDEO PART

   RESERVE NAME(*) AS 20
   CELLSIZE=48
   SECTIONSIZE =8      "no of subblocks for which generator stays in same mode
   TIMESTEP = 0.0015625        "time interval in which 0 or 1 cell is produced
                               " =time interval in which one subblock is encoded

   OPEN UNIT 1 FOR INPUT, NAME IS "MODEDATA"
   USE UNIT 1 FOR INPUT
   RESERVE  MODEINFO(*)   AS  20
   FOR I=1 TO 20, READ MODEINFO(I)
   LET EOF.V = 1
   CLOSE UNIT 1
LET VID_DUR_TIME = 100.0

LET VIDEO_START_TIME = 10.0

LET SHOT_INTARR = 0.2

LET NO.OF.PACKETS = 0
LET SIGMA_LEFT = 0.5
LET SIGMA_RIGHT = 1.5
LET X = 64
IMAGESIZE= X*X                 " This project uses 64*64 pixels
SUBBLOCKSIZE = 8*8             " in pixels
NO_OF_SUBBLOCKS = IMAGESIZE/SUBBLOCKSIZE
NO_OF_SECTIONS  = NO_OF_SUBBLOCKS/SECTIONSIZE
RESERVE  CELLS_PER_SUBBLOCK(*) AS  NO_OF_SUBBLOCKS

"determine bit rate of first shot:
 MEAN_MODE= INT.F( UNIFORM.F(1.5, 4.5, 1))
 IF MEAN_MODE=6
```

```
        MEAN_MODE=5
 ALWAYS

"determine variety of first shot
SIGMA=UNIFORM.F(SIGMA_LEFT,SIGMA_RIGHT,1)

TIME= VIDEO_START_TIME

"determine point at which coming shot finishes
SHOT_SWITCH_POINT=TIME+EXPONENTIAL.F(SHOT_INTARR,1)

"determine timepoint after which no new frame is encoded
END_TIME =VIDEO_START_TIME+VID_DUR_TIME

no_of_frames=0

WHILE TIME < END_TIME DO
" NOW ENTER PROGRAM-PART FOR ONE FRAME:
"
" INPUT: mean_mode, sigma and no_of_subblocks
"
" USES GLOBAL CONSTANTS (IDENTICAL FOR EVERY IMAGE):
" sectionsize,timestep,subblocksize, cellsize and the modeinfo
" these constants are characteristics of the used coder and independent of
" frame (image) characteristics
"
" OUTPUT: array "cells_per_subblock" contains the traffic produced by the image.

   MODE=MEAN_MODE                    "start in mean_mode
   T= TIMESTEP                       "is time since last transition from other state
   CELLS_PER_SUBBLOCK(1)= INT.F( UNIFORM.F(0.0, 1.0, 1))
    "state of the first subblock
   SECTIONNO=0                  "first we consider the first section apart

   AA=MODEINFO((MODE-1)*4+1)  "retrieve information
   DA=MODEINFO((MODE-1)*4+2)  "concerning the section's
   AI=MODEINFO((MODE-1)*4+3)  "mode switching characteristics
   DI=MODEINFO((MODE-1)*4+4)  "between active and inactive

  FOR I=2 TO MIN.F(SECTIONSIZE,NO_OF_SUBBLOCKS) DO
    "now rest of the subblocks
  U=UNIFORM.F(0,1,1)       " random no. for determining next state
  IF CELLS_PER_SUBBLOCK(I-1+SECTIONNO*SECTIONSIZE)=0
    IF U < AA*T**DA              " if Pa(t) realized
        CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=1
        " then switch inact->act
         T=0                       "     and reset the time
    ELSE CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=0
        "else stay in same sta
    ALWAYS

  ELSE                      " foregoing state is active
    IF U < AI*T**DI       " if  Pi(t) realized
   CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=0
  "then switch act->inact
    T=0                      "     and reset the time
    ELSE
    " else stay in same st
```

```
        CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=1
      ALWAYS
    ALWAYS
    T = T + TIMESTEP
    LOOP            "next subblock of first section

  IF NO_OF_SECTIONS >1
  "determine mode of second section:
  MODE = MODE + INT.F(NORMAL.F(0,SIGMA,1))
  IF MODE<1
  MODE=1
  ALWAYS
  IF MODE>5
  MODE=5
  ALWAYS

  "now do the rest of the sections:
  FOR SECTIONNO=1 TO NO_OF_SECTIONS-1 DO          "loop for each new section
    AA=MODEINFO((MODE-1)*4+1)           "retrieve information
    DA=MODEINFO((MODE-1)*4+2)           "concerning the section's
    AI=MODEINFO((MODE-1)*4+3)           "mode switching characteristics
    DI=MODEINFO((MODE-1)*4+4)           "between active and inactive

  "handle each subblock of the section:
  FOR I=1 TO SECTIONSIZE DO
  U=UNIFORM.F(0,1,1)

  IF CELLS_PER_SUBBLOCK(I-1+SECTIONNO*SECTIONSIZE)=0
    IF U < AA*T**DA                 " if Pa(t) realized
    CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=1
  " then switch inact->act
    T=0                          "    and reset the time
    ELSE CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=0
  "else stay in same state
    ALWAYS

  ELSE                          " foregoing state is active
    IF U < AI*T**DI         " if  Pi(t) realized
    CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=0
  "then switch act->inact
    T=0                        "    and reset the time
    ELSE CELLS_PER_SUBBLOCK(I+SECTIONNO*SECTIONSIZE)=1
  " else stay in same state
    ALWAYS
  ALWAYS

  T = T + TIMESTEP
  LOOP  " next subblock of current section

  " determine mode of next section:
  MODE = MODE + INT.F(NORMAL.F(0,SIGMA,1))  "determine next mode
  IF MODE<1
  MODE=1
  ALWAYS
  IF MODE>5
  MODE=5
  ALWAYS
```

```
LOOP  "next section
ALWAYS
" END OF PROGRAM PART FOR EACH FRAME
"At this stage, F.PACKETS
 ALWAYS
"  time = time + timestep
LOOP        " NEXT SUBBLOCK

"NOW THE ARRAY "cells_per_subblock" HAS TO BE FILLED UP WITH THE TRAFFIC OF THE
NEXT "FRAME

'TRY_AGAIN'
IF TIME>= SHOT_SWITCH_POINT

   " 1) determine bit rate of next shot:
   MEAN_MODE= INT.F( UNIFORM.F(1.5, 4.5, 1))
   IF MEAN_MODE=6
    MEAN_MODE=5
   ALWAYS

   " 2) determine variety of next shot
   SIGMA=UNIFORM.F(SIGMA_LEFT,SIGMA_RIGHT,1)

   " 3) determine next shot termination point
   SHOT_SWITCH_POINT=TIME+EXPONENTIAL.F(SHOT_INTARR,1)

ELSE
  TIME = TIME + TIMESTEP
  GO TO 'TRY_AGAIN'
ALWAYS

NO_OF_FRAMES=NO_OF_FRAMES+1
IF NO_OF_FRAMES = 1
  GO TO 'FINISH'
ALWAYS
LOOP        " NEXT FRAME

'FINISH'

LET V.PACKETS = NO.OF.PACKETS

END"VIDEO
```

PUBLISHED ARTICLES

(1) *"Performance of the ATDMA Protocol for Personal Communication Network for Transporting Multimedia Traffic"*, Australian Telecommunication Networks and Application Conference 1996 (ATNAC 96), 3-6 December, 1996, Melbourne, Australia, pp 19-24.

(2) *"An ATM based Wireless Network for Multimedia Communications"*, Eighth IEEE Workshop on the Local and Metropolitan Area Networks, Session : Wireless Networking 11, August 25-28, 1996, Berlin/Potsdam, Germany.

(3) *"Investigation in to a high capacity personal communication network (PCN) to support multimedia applications"*, Postgraduate conference for Engineering and Technology students, Auckland, 1995, pp. 183 - 186.