EFFICIENCY IMPROVEMENTS TO THE GROUPE SPECIAL MOBILE (GSM) DIGITAL MOBILE RADIO SYSTEM

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Doctor of Philosophy

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September 1994

The University of Aston in Birmingham

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Submitted for the Degree of Doctor of Philosophy

1994

Summary

Groupe Spécial Mobile (GSM) has been developed as the pan-European second generation of digital mobile systems. GSM operates in the 900MHz frequency band and employs digital technology instead of the analogue technology of its predecessors. Digital technology enables the GSM system to operate in much smaller zones in comparison with the analogue systems. The GSM system will offer greater roaming facilities to its subscribers, extended throughout the countries that have installed the system. The GSM system could be seen as a further enhancement to European integration.

GSM has adopted a contention-based protocol for multipoint-to-point transmission. In particular, the slotted-ALOHA medium access protocol is used to coordinate the transmission of the channel request messages between the scattered mobile stations. Collision still happens when more than one mobile station having the same random reference number attempts to transmit on the same time-slot. In this research, a modified version of this protocol has been developed in order to reduce the number of collisions and hence increase the random access channel throughput compared to the existing protocol. The performance evaluation of the protocol has been carried out using simulation methods.

Due to the growing demand for mobile radio telephony as well as for data services, optimal usage of the scarce availability radio spectrum is becoming increasingly important. In this research, a protocol has been developed whereby the number of transmitted information packets over the GSM system is increased without any additional increase of the allocated radio spectrum. Simulation results are presented to show the improvements achieved by the proposed protocol.

Cellular mobile radio networks commonly respond to an increase in the service demand by using smaller coverage areas. As a result, the volume of the signalling exchanges increases. In this research, a proposal for interconnecting the various entities of the mobile radio network over the future broadband networks based on the IEEE 802.6 Metropolitan Area Network (MAN) is outlined. Simulation results are presented to show the benefits achieved by interconnecting these entities over the broadband Networks.

Key words: ATM, Slotted-ALOHA, GSM system, Medium access procedures, Metropolitan area networks (MANs)

ACKNOWLEDGEMENTS

I would like to take this opportunity to thank my supervisor Dr. R. L. Brewster for all the encouragement and constructive criticism he has given throughout the research.

I am also grateful to my wife, my daughters and my son for their patience, continuous support and encouragement during the whole period of the research. Thanks are also to all members of my family back home and all my friends here and back home for their encouragement and continuous support throughout the research.

Thanks to the Almighty ALLAH for giving me the will and the strength to complete this work.

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ABBREVIATIONS

ATM	Asynchronous Transfer Mode
BCCH	Broadcast Common Control Channel
B-ISDN	Broadband Integrated Services Digital Network
BSC	Base Station Controller
BTS	Base Transceiver Station
CCCH	Common Control Channel
DQDB	Distributed Queue Dual Bus
FDMA	Frequency Division Multiple Access
GSM	Groupe Spécial Mobile
ICMA	Idle-signal Casting Multiple Access
ICMA/CD	Idle-signal Casting Multiple Access with Collision Detection
ISDN	Integrated Services Digital Network
LAN	Local Area Network
MAN	Metropolitan Area Network
MSC	Mobile Switching Centre
PLMN	Public Land Mobile Network
RACH	Random Access Channel
SCH	Synchronisation Channel
STM	Synchronous Transfer Mode
TACS	Total Access Communication System
TDMA	Time Division Multiple Access
SCPC	Single Channel Per Carrier

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CHAPTER ONE

1.1 Introduction

First generation cellular Public Land Mobile Networks (PLMNs) have been in use for many years providing services to millions of people all over the world. In particular, the mobile radio systems provide their customers with opportunities to roam freely within the coverage area and simultaneously communicate with any phone, fax or data modem subscribers anywhere. Among these first generation systems, the two well-known nationwide systems, Advanced Mobile Phone Service (AMPS) [1], was introduced in the U.S.A. in 1978 and the Total Access Communication System (TACS) was adopted in the U.K. in 1985 [2]. The TACS system is just a modified version of the AMPS system that adopts the specifications set up by the Department of Trade and Industry (DTI). These analogue mobile networks were initially developed and designed to provide telephone services by using single-channel-per-carrier (SCPC) analogue Frequency Modulation (FM) transmission techniques. Telephone service has been and will remain the dominant service to be offered and handled by these networks. Data transmission is also achieved over these systems by fitting appropriate equipments known as modems. Data transmission is accomplished by using a modulation technique known as binary Frequency Shift Keying (FSK) [2].

There has been a rapid advance in the digital technology field since these analogue cellular systems were firstly introduced. Transmission of digitised speech instead of its analogue form over the radio path provides a lot of advantages. These advantages include integration of voice and data, encryption of speech privacy and the provision of better service quality by adopting sophisticated protocols [3].

In Europe there are currently a lot of different public land mobile radio systems in operation. Besides the TACS in the U.K., systems in operation include the C 450 in Germany, RC 2000 in France, and NMT 450 in some other European countries [4]. These systems have been designed and developed separately to fulfil each country's requirements. As a result, there is no compatibility or correlation existing between them.

In 1982, the Conference European des Postes et Telecommunications (CEPT) set up a group known as Groupe Spécial Mobile (GSM) to establish the grounds and specifications for a harmonised public mobile radio communication system [3]. The work is directed towards establishing a second generation pan-European digital cellular mobile radio communication system in the region of the 900MHz frequency band. The GSM can, therefore, be viewed as a replacement for the present uncorrelated and incompatible collection of first generation analogue systems throughout Europe towards a common cellular system. Other features the GSM system aims at are:

- efficient utilisation of the limited radio spectrum compared with its predecessor
- to bring economic prosperity to manufactures throughout Europe
- to adopt digital technology which provides better speech quality and higher data rates
- to offer greater roaming facilities nationally and internationally to its subscribers, irrespective of their location throughout all the countries that have installed the system
- compatibility and inter-working with the Integrated Services Digital

Networks (ISDN) and Public Switched Telephone Network (PSTN)

- to offer service to both vehicle and portable stations.

In the fixed telecommunication networks arena, after more than a century of reliance on analogue-based technology for telecommunication services, today there is a mixture of analogue and digital technology available and a rapid movement toward an all-digital environment. At present, specialised separate networks such as circuit and packet networks are in operation to optimally provide low delay performance for circuit-based services and low error performance for packet-based services. The integration of these traditionally separated services at a unified interface has been a long awaited user requirement. The advent of the ISDN has brought the user one step closer towards this goal. However, although the ISDN might appear to the user as an integration of services, in fact it is not integrated, since internally the two switching techniques still remain isolated and provided separately [5].

Recently, a growing interest has appeared in broadband services with the necessity to its introduction of broadband networks [6]. The anticipated broadband services will require much higher bit rates than those currently supported by the ISDN. Among business users, for example, the widespread use of Local Area Networks (LANs), operating at speeds in the range of 1-16Mb/s, depending on the transmission medium and topology involved, is generating a demand for high speed data transfer over the public networks. This demand has further increased due to the recent advances in the technology which resulted in the production of more powerful machines including personal computers and workstations. The residential subscriber, on the other hand, could also benefit from the introduction of the broadband networks. These benefits could be in the entertainment side such as high quality video broadcasting and high fidelity sound.

A basic requirement of the broadband networks is that these networks should support the existing services as well as the broad spectrum of the anticipated future services in a more flexible and efficient way. This implies that the switching and multiplexing technique (transfer mode) to be adopted for the broadband networks should lend itself very well to these required characteristics. Prior to the presentation of the transfer mode adopted for the broadband ISDN, it is appropriate to give a general introduction to the switching and multiplexing techniques used in the telecommunication field.

1.2 SWITCHING TECHNIQUES

1.2.1 Circuit Switching

Basically, three forms of switching techniques are used in communication networks. These switching techniques are: circuit switching, message switching and packet switching. In the ***circuit switching***, a dedicated route has to be established prior to any information transfer. Once the link is set up, circuit switches then transparently transfer the information in both directions for the whole period of the communication session. The link is then terminated at the end of the call. The main disadvantages of such a switching technique is that long periods of the network's valuable time is wasted due to the set up of the dedicated links. Furthermore, the channel capacity is inefficiently used when it is involved in carrying bursty traffic [7].

1.2.2 Message Switching

Information exchange between computers has directly resulted in an increased demand for effective communication means to handle them. Computer traffic is extremely bursty in nature, which means that a wastage of the network resources will result if the circuit switched based networks are used to deliver such traffic. This has led to the development of the ***message switching*** technology. Message switches handle messages which consist of user data, a header and a checksum. The header contains control information such as routing information and the checksum is used for error check purposes. Once the message is completely received, the switches examine its header and accordingly select the next appropriate link in the direction towards its destination. This process is then repeated until the message arrives at its final destination. This technique is also referred to as store-and-forward technique. Although this technique lends itself very well to bursty types of traffic, it is inefficient for interactive data such as voice [8].

1.2.3 Packet Switching

Packet switching is a modification to message switching. The packet switching technique attempts to overcome the limitations imposed by message switching. As a result, in packet switching the message is segmented into small segments known as packets. Each packet carries its own sequence number as well as other control information in its header. Limitation control may be enforced on the maximum allowable packets length. Two different packet switching protocol implementations exist: datagram and virtual circuit [9].

In the ***datagram*** protocol, each packet contains enough control information in its header to be treated as a separate entity by the switches. As a result, simultaneous transmission of packets belonging to the same message over different links is achieved. Routing decisions are made at each intermediate switch by reading a periodically updated routing table. The routing table indicates the best output trunk to each destination. At the destination, the received packets are reformatted into messages using their sequence numbers [9,10].

In the ***virtual circuit*** protocol, a logical connection (virtual circuit) across the network is established before any packet transmission is started. Packets for the same connection should carry the same virtual circuit number in their headers. This relieves the intermediate switches of making the routing decisions and packets should follow each other in a sequence. Unlike circuit switching, in virtual circuit no bandwidth is consumed after the packets have been delivered to their final destination. There are other advantages associated with packet switching technology including smaller storage capacities needed for storing these small packets and packets experience minimum delay in their progress through the network. Sharing of the network resources could be achieved and, when errors occur in a packet, only that packet is required for retransmission [9,10].

1.3 MULTIPLEXING TECHNIQUES

Multiplexing is a process which enables a number of signal sources to transmit their data simultaneously on the same circuit without interfering with each other. As a result, expensive resources could be shared among more than one user.

1.3.1 Frequency Division Multiplexing (FDM)

In the ***frequency division multiplexing*** scheme, the whole bandwidth of the medium is subdivided into smaller frequency bands. Each of these bands represents a communication channel. Guard bands are needed to ensure that the adjacent channels do not overlap with each other. Users hold the channel for the whole call period and no synchronisation is needed between the transmitter and the receiver. This multiplexing technique is traditionally used for voice transmission, but data could also be transmitted using modulators/demodulators (modems) [11].

1.3.2 Time Division Multiplexing (TDM)

Time division multiplexing is sometimes also referred to as synchronous TDM. In this scheme, the multiplexing is achieved through time sharing of the network resources. The whole transmission bandwidth is considered for a fixed time interval, known as a frame, which is occurring periodically. The frame is then subdivided into fixed time-slots in which some of these time-slots are kept for control purposes such as frame synchronisation and signalling. In this multiplexing technique, synchronisation is essential to keep both the transmitter and the receiver in synchronism. In this technique, data and digital speech services are carried in an integrated fashion (ie. without discrimination) [12].

1.3.3 Statistical Time Division Multiplexing (STDM)

Statistical time division multiplexing is also known as intelligent or asynchronous TDM. A fundamental disadvantage associated with the above mentioned TDM scheme is that when an active source enters an idle state it must keep transmitting dummy bits in its allocated time-slot to maintain its synchronisation with the receiver. To overcome this shortcoming, a more flexible type of time division multiplexing was developed known as STDM. The operation of STDM is exactly the same as TDM when all the time-slots are active. However, STDM differs from TDM when there are idle signal sources. Two different protocols could be used with this scheme. In the first protocol, a pre-defined "absence" flag has to be sent when the source is idle to maintain the synchronisation. In the second, the "absence" flag is completely suppressed and the synchronisation is accomplished by adding an explicit identification to the user data. To allow proper functioning of this protocol, user data is firstly stored in separate buffers. The assembler first scans the buffers and then constructs the next frame. The frame size is varied dynamically depending on the number of active users. However, the maximum frame size is limited to the case when all users are simultaneously active [9,10].

1.3.4 Packet Multiplexing

In the ***packet multiplexing*** scheme, user data is transferred in private packets rather than in composite packets, as in the STDM scheme. Restrictions may be placed on packet size depending on the specific implementation. Each packet is individually protected by an error detection and correction and retransmission mechanism. Packets carry sufficient information in their headers to identify the sending and receiving parties and also for routing them through the network [9,10].

Comparing all previously discussed switching and multiplexing techniques, it is clear that the packet transfer approach is more suitable where integrated services are to be carried. In the packet transfer technique, no time is wasted in constructing the frame. Further, this scheme can virtually support any transmission rate up to the maximum channel capacity. A specific packet-oriented transfer mode using the asynchronous time division multiplexing technique is now being adopted by several standardisation bodies for implementing Broadband Integrated Services Digital Networks (B-ISDN). The adopted transfer mode is known as Asynchronous Transfer Mode (ATM). An ATM based network is required to accommodate different services with different characteristics and different requirements. This technique will be outlined in more detail in the next chapter.

1.4 Research Objectives

(i) The GSM system has adopted a contention-based protocol for multipoint-to-point communication. In particular, the GSM has adopted the well-known slotted ALOHA protocol for mobile stations to transmit their channel access request messages over the random access channel. One of the objectives of this research is, therefore, to check the performance of the slotted ALOHA protocol against various traffic loads, taking into consideration the GSM specifications. The research then goes on to exploit further control parameters to control the number of contentions on the random access channel.

(ii) For the allocation of the physical channel (time-slots), the GSM system has adopted a fixed assignment scheme based on Time Division Multiple Access/Frequency

Division Multiple Access (TDMA)/(FDMA). With such a scheme, the multiplexing is achieved at the call level, which means a mobile station holds the allocated channel for the entire duration of the call period. Although such fixed assignment schemes have the advantage of a contention-free policy, they suffer poor channel utilisation when the transported information is bursty. This research attempts to increase the GSM spectrum efficiency by developing a protocol to exploit the silence intervals inherent in the speech transmission. There are two main factors which have to be taken into account when developing such a protocol. These are:

- the mobile stations that are involved in a simultaneous transmission on the same time-slot must not lose their synchronisation with the base station.

- the protocol must not add any additional delay to the interactive data transmission apart from the delay already inherited from the GSM system.

(iii) It has long been recognised that the mobile sector will benefit from the introduction of the broadband ISDN, especially in the signalling exchanges between the various entities of the mobile network. This research proposes an interconnection strategy of the GSM entities over the IEEE 802.6 MAN. In this work, the advantages of interconnecting these entities are investigated in terms of the speed of the hand-over messages exchange.

1.5 Thesis Outline

The thesis is organised as follows: chapter 2 starts with outlining how the narrowband ISDN evolved from the Integrated Digital Network (IDN). The chapter then goes on to give more technical detail on the transmission rates supported by the ISDN as well as describing the ISDN underlying transfer mode and its limitations to support the B-ISDN. The chapter ends with technical details of the two emerging broadband networks based on the IEEE 802.6 MAN and ATM.

Chapter 3 deals with the medium access procedures employed in the communication field. It begins with the description of the various communication assignment strategies and then presents the GSM random access procedure adopted on the up-link part on the common control channels. A proposed modification to the GSM access procedure is then described which achieves lower access delays by reducing the number of contentions on the random access channel.

Chapter 4 deals with a proposed protocol to increase the spectrum efficiency of the GSM system. The increase of the efficiency has been achieved through the increase in the number of simultaneous transmissions over the GSM system.

Chapter 5 presents a proposed mechanism for interconnecting the GSM entities over the B-ISDN. Fast signalling exchange is achieved which, in turn, enables the GSM to efficiently support the micro-cell structure.

Chapter 6 is devoted an in-depth description of the simulation package that was chosen to simulate the developed protocol. Events that describe the dynamic behaviour of the protocols are also presented in this chapter.

Finally, chapter 7 concludes the thesis by outlining the advantages of the developed protocols in this research. Details of further recommended work are also included in this chapter.

CHAPTER TWO

BROADBAND-ISDN AND ASYNCHRONOUS TRANSFER MODE

2.1 Narrowband ISDN

After years of standardisation and development activities, Integrated Service Digital Network (ISDN) systems have now entered the deployment stage offering end-to-end digital telecommunications services to their customers. According to CCITT, an ISDN is a network that has evolved from a telephony Integrated Digital Network (IDN) [13]. IDN is a network providing digital transmission and switching facilities of telephony services between local exchanges, based on 64kb/s channels. The evolution of these networks to ISDN networks means the conversion of the customer's access connection to the network from analogue to digital. The digital facilities are made possible by the rapid progress of technology towards a greater use of large-scale integration of circuits onto a single chip [6]. In addition, the services provided by the ISDN include voice and non-voice services, to which users have access by a limited set of standard multi-purpose user-network interfaces.

ISDN networks offer the end user a basic access structure of two B-channels (each with a transmission rate capacity of 64kb/s) and one D-channel with a capacity of 16kb/s resulting at an aggregate transmission rate of 144kb/s. B-channels are bearer channels used to convey voice and non-voice services. The transmission rate of 64kb/s was chosen because, at the time, it was the standard rate for digitised voice, and hence being introduced into the evolving IDNs. In addition, there is another standard rate interface known as primary rate interface which is provided to the user on a request basis. The primary rate interface is based on the structure nB+D (n=23 in North America, n=30 in Europe). In this primary rate interface, the D-channel has a capacity of 64kb/s. The D-channel is mainly used to carry signalling information but it could be used to carry user information as well. The signalling information does not contain any user information but is needed for control purposes. The D-channel may be divided into two equal capacity channels (8kb/s each). The 8kb/s channels are then used to carry data and signalling traffic separately [6,14].

In the current narrowband-ISDN, service-specific switches are connected by service-specific circuits accommodated in the digital transmission lines by the Synchronous Transfer Mode (STM). The STM approach is based on time division switching and multiplexing technology. The STM allocates time-slots within a recurring structure (frame) to a service for the whole duration of a call [15]. A framing slot indicates the start of each frame. Moreover, an STM channel is identified by the position of its time-slot within this synchronous structure and is hence known as position multiplexing as shown in figure 2.1. In the STM environment, slots are allocated based on the peek transfer rate of the call. As a result, the required service quality can be guaranteed even at the peak loads. Assigning time-slots on the peak rate basis results in a wastage of the bandwidth during the periods at which the information is transported below peak rates [15,16].

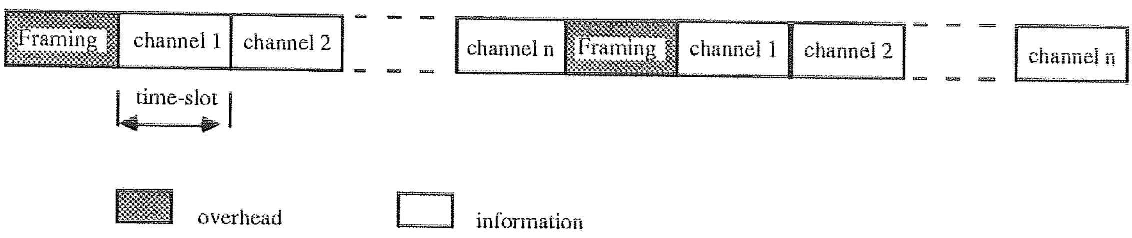


Figure 2.1 STM Multiplexing Principles

Todays customers are becoming familiar with the range of services provided by the ISDN. An ISDN socket will accept a wide variety of devices including the telephone, personal computer, a television and many more. The same socket would also provide data transmission and switching capabilities for end-to-end communications. Data transmission over the ISDN requires a set of end-to-end protocols to guarantee reliability and security. Even though these services might appear to the end user as integrated services, internally the two switching techniques remain isolated and provided separately.

With the emerging of new broadband services that require new transmission facilities, the ISDN in its current standard is unable to support them. It has already been stated that the broadband services would require more flexible user-network access interfaces and much higher bit rates than the 64kb/s already supported by the current narrowband-ISDN networks. Future broadband services will include high-speed data transfer applications, such as High Definition Television (HDTV), high resolution image transfer and retrieval, etc. [17,18].

In the data communication field, Local Area Networks (LANs) provide interconnection of terminals, computers, workstations and other intelligent systems within a building or a number of buildings that constitute a small geographical area. The maximum extent of these networks end-to-end is at most several kilometres. Communication between computers is bursty in nature. A key success of LANs in interconnecting these computers is, therefore, due to the LANs suitability in handling such bursty type of data traffic. In terms of data transmission rate facilities, LANs could offer different data transmission rates depending on the transmission medium involved [10,19].

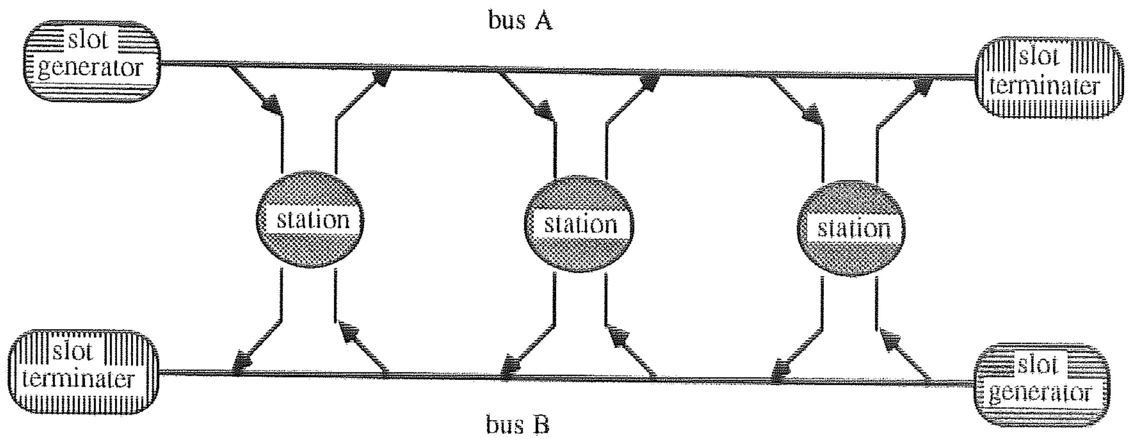
Due to the recent advances in the technology field, the processing power of personal computers, workstations, minicomputers, and laser printers is increasing rapidly. Locally, the interconnection of such powerful machines, as well as the provision of broadband services, could be satisfied by high-speed LANs such as Fibre Distributed Data Interface (FDDI). However, interconnection of homogeneous or heterogeneous existing LANs and future high-speed LANs and other applications such as desktop publishing and Computer Aided Design (CAD)/Computer Aided Manufacture (CAM) is a continuously growing need. This problem exists both in the public and the private domain, over a more and more extended area that extends beyond the local premises, across the metropolitan and wide area environment. All these factors have led to the emergence of two standards, the IEEE 802.6 Metropolitan Area Network (MAN) [20] and International Consultative Committee for Telephone and Telegraph (CCITT) study group XVIII, for broadband data networking [21,22]. These broadband networks are expected to retain many of the features of the existing networks such as LANs and ISDN but over a larger geographical scale as well as acting as backbone networks to facilitate LAN interconnections. In particular, the aim of these broadband networks is also to flexibly support a diverse range of services including voice, data, image and multimedia services using available resources. Section 2.2 outlines in detail the MAN networks, while section 2.3 is devoted for the broadband-ISDN based on the Asynchronous Transfer Mode (ATM).

2.2 Metropolitan Area Networks (MANs)

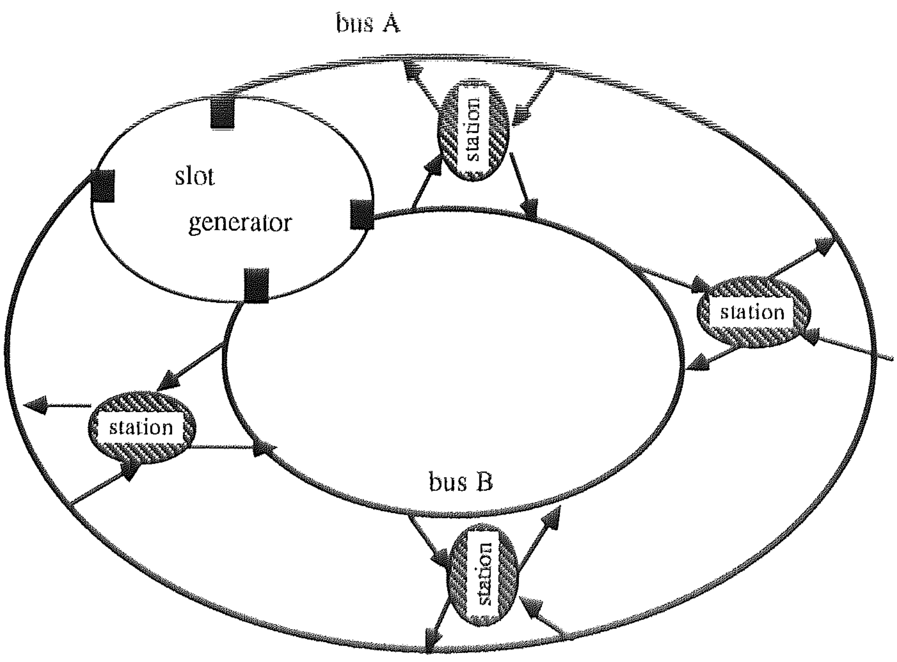
A MAN could be defined as a high-speed communication network that offers multiple services, including the connectionless services, without some of the geographical limitations of LANs. A MAN is a dual bus that could be configured in an open loop-bus topology or in a looped-bus topology [23]. Even though the buses in the latter case are looped but, they are not physically connected to form a closed loop, as shown in figure 2.2 [24]. In a MAN network, the stations are distributed along the two buses and have an access to both buses in order to enable these stations to have simultaneous read and write capabilities. Each bus has its own slot generator at its head-end. The responsibility of this slot generator is to continuously generate time-slots, each of which contains a fixed number of octets, as well as synchronising the bus. Constructing the MAN in a looped form topology, therefore, gives the advantage of co-locating the slot generators (ie. a common slot generator can be used for both buses). This type of configuration also increases the MAN reliability in the event of a link failure by closing the buses at the current head-ends and locating the new generator at a position next to the fault disconnection in the bus.

The IEEE 802.6 standard committee has adopted a reservation procedure known as a distributed queuing scheme as the medium access method for the MAN [25]. The adopted Distributed Queue Dual Bus (DQDB) medium access protocol is based on the Queue Packet and Synchronous Circuit Exchange (QPSX) medium access protocol developed by Telecom Australia [25]. The medium access protocol is particularly important for achieving both grade of service objectives, relating to call blocking probabilities for the offered traffic load, and quality of service objectives, relating to information loss and delay for the carried traffic. The DQDB protocol is intended to integrate isochronous and nonisochronous traffic on a single high bandwidth network.

As mentioned above, the responsibility of the slot generator is to continuously generate time-slots. The duration of a slot is equal to the size of a data packet plus the slot header. Two types of slot are identified: Queue Arbitrated (QA) and Pre-Arbitrated (PA). The QA slots are used to carry asynchronous segments of information, while the PA slots are used for carrying isochronous segments. The slot generator should mark any particular slot, whether the slot is PA or QA. A number of these time-slots are then grouped together to form a cycle frame. The cycle frame, therefore, consists of a frame header followed by K number of time-slots.



(a) Open loop configuration



(a) Closed loop configuration

Figure 2.2 MAN configuration

Figure 2.3 represents the cell structure for the IEEE 802.6. The figure shows that each slot contains in its header a field known as the Access Control Field (ACF). Stations attached to the DQDB system gain access to slots through a particular bit, known as the busy bit, contained in the ACF. The ACF also contains another sub-field called the request sub-field, which is used for requesting future packet transmission. The basic operation of the DQDB protocol is based on this idle/busy bit and the request sub-field. When an active station wishes to transmit information to another station, it does so by placing its request in the first available reservation free slot that passes on the "upstream" bus. The "downstream" bus refers to the bus that carries traffic towards the intended station, while the "upstream" refers to the opposite bus. By successfully placing its request, the active station has actually informed the upstream stations that an additional segment of information is queued for access to the "downstream" bus. This mechanism, therefore, acts as a global distributed queuing system. In order for a particular station to know its request position in this global queue, each station keeps an updated record of two specific counters. One counter is used for counting the number of requests and the other for the number of the empty slots that pass by. By a simple arithmetic operation, any station can determine its request position within the queue. Stations that have gained access to slots convert the busy bit to 1. The busy bits are usually set to 0 by the slot generator to indicate that these slots are idle (empty).

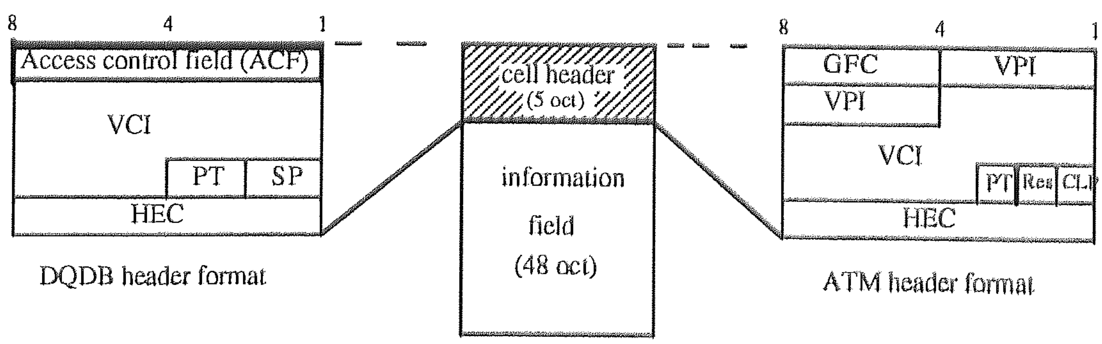


Figure 2.3 IEEE 802.6 and ATM cell structures for B-ISDN

2.3 Broadband Networks Based on ATM

The characteristics of the broad spectrum of the broadband services indicate that the transfer mode to be used in the future single-fabric broadband networks must have certain characteristics and attributes to fulfil such requirements. ATM is the most favoured transfer technique for the B-ISDN and has been chosen by several standardisation bodies for transferring of information across the User-Network Interface (UNI) [26]. The ATM is a packet-oriented transfer mode which uses the asynchronous time division multiplexing technique. It attempts to eliminate the limitations, inflexibility and inefficiency imposed by the STM-based solutions.

In an ATM environment, the multiplexed information flow is organised in fixed size blocks, called cells. An ATM cell consists of a header and an information field. In particular, an ATM cell consists of a 5-octet header field and a 48-octet information field as shown in figure 2.3 for the cell header at the UNI [26,27]. The cell length was chosen as a compromise due to the different requirements dictated by service such as voice, video and data. Although voice services can tolerate, up to some degree, loss of information, they suffer greatly from delay. On the other hand, data services can tolerate delay, but are very sensitive from information loss. The cell transmission time is equal to a slot length, and cells are assigned on a demand basis, depending on the source activity and the available resources. As a result, an ATM interface structure consists of a set of labelled, not positioned, channels. This explains why this transfer technique has been referred to as asynchronous. It is actually asynchronous due to the fact that cells, which are assigned to the same connection, may exhibit an irregular recurrence as shown in figure 2.4 [26,28].

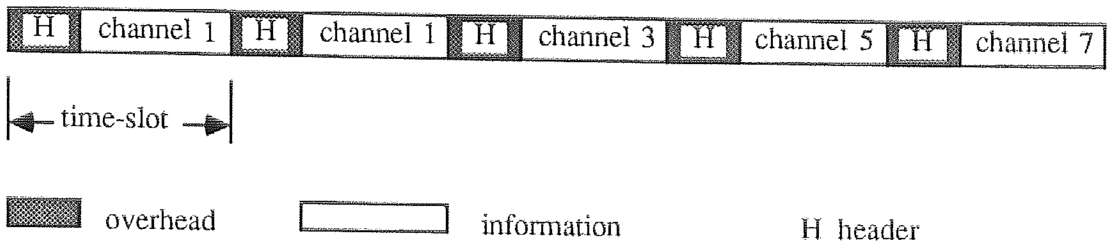


Figure 2.4 ATM multiplexing principles

By assigning cells on demand in an ATM system, no bandwidth is consumed unless information is actually being transmitted. This presents the fundamental difference between the ATM and STM transfer modes [26,29]. Also, since ATM cells are allocated on demand, an ATM network can easily accommodate variable bit rate services. Furthermore, bandwidth efficiency gains could be achieved through statistical multiplexing of bursty traffic resources.

ATM is a connection-oriented technique where header values are assigned to each section of a connection when required and released whenever no longer needed. The connections identified by the headers remain unchanged during the life-time of a call. The primary role of the header is, therefore, to identify cells belonging to the same virtual channel, virtual path as well as to provide cells with routing information. Virtual channels refers to the logical connections in the ATM network. Virtual path, on the other hand, is basically a grouping of a number of virtual circuits that have the same endpoints into a single unit. This strategy has its advantages especially in the network resource management whereby the control actions are applied to a small number of groups of connections rather than applying them to a large number of individual connections [30]. Cell sequence integrity has to be maintained for the cells belonging to the same connection. This function is accomplished by the ATM layer. The ATM layer is the next higher layer to the physical layer in the B-ISDN ATM protocol reference model. Signalling and user information are carried on separate virtual channels. Signalling information is not part of the user information but is needed for the network functionality.

ATM networks will be designed to support a wider variety of services. As a result, control procedures should be adopted to ensure a higher grade of performance. Connection Admission Control (CAC) is an example of such control procedures used to accept or reject a connection request. With such a control procedure, a call request is only accepted when sufficient resources are available to carry the call under its required quality of service. Once a call has been accepted, the required performance is guaranteed and some flow control must be operated to provide fairness among the users when congestion situations occur [31].

2.4 Summary

In this chapter a general introduction to the narrowband ISDN is given. The attention is then focused on some of the anticipated broadband services. Such services give an insight in to why the current structure of the narrowband ISDN, with its underlaying transfer mode, cannot support these services.

IEEE 802.6 MAN is an emerging broadband system. In this chapter information transfer over a MAN, as well as possible topologies, have been reviewed. There are some suggestions that the MAN will be the first step in the direction towards the ultimate broadband networks based on ATM.

ATM is a flexible and efficient transfer technique to support the existing and the future unknown services. The cell format structure of ATM, with other related information, is presented. A comparison of ATM and STM to show why the ATM has been chosen as the transfer mode for implementing the broadband ISDN is also given.

CHAPTER THREE

MEDIUM ACCESS PROCEDURES

3.1 Introduction

Efficient utilisation of the available bandwidth in communication systems is one of the challenges to communication engineers and system designers. Accordingly, the utilisation of such a communication medium needs to employ efficient methods to share this communication resource. To facilitate the sharing of this communication medium among a group of users, a suitable medium access (multiple access) control procedure must be adopted. The purpose of the medium access control procedure is to provide control, coordination and supervision of the access of these users to the shared communication facility. Moreover, medium access control protocols implemented in a specific environment might not be applicable in a different environment. This is due to the fact that different environments need different requirements to be fulfilled. Medium access control procedures adopted for the mobile radio environment, for example, differ from those medium access control protocols implemented in the fix networks. This is due to problems such as the hidden terminal problem associated with the mobile radio environment that does not exist in fixed networks [32].

The assignment of the communication medium among a group of stations can be classified into three strategies. These strategies are; fixed assignment strategy, demand assignment strategy and random access strategy. A brief description of each of these channel allocation strategies is presented in the following subsections. Two of these channel allocation strategies have actually been adopted by the pan-European digital mobile radio system, while the third strategy has been presented for comparison purposes. This chapter presents a proposed procedure which attempts to improve the performance of the medium access control protocol adopted by the pan-European digital mobile radio system to be implemented on the up-link control channels.

3.2 Fixed Channel Assignment Procedure

Under the fixed channel assignment procedure, a channel bandwidth may be dedicated to a station that it can permanently use for accessing the channel. The common access mechanisms that fall in this category are time division multiple access, frequency division multiple access and code division multiple access [33]. In time division multiple access, the time axis is divided into slots that allow the users to have an access to the whole allocated frequency band. A number of these time-slots are grouped together to form a periodic frame. A specific time-slot within each frame is exclusively allocated to a particular user for the whole duration of the call. On the other hand, in the frequency division multiple access technique, the whole allocated frequency band is subdivided into frequency subbands (channels). Each of these channels is then assigned to one and only one user for the whole duration of the communication session [33,34]. The pan-European digital mobile radio system has adopted, apart from the access technique used on the up-link part of the common control channels, a combination of time division multiple access and frequency division multiple access as an access procedure to its transmission facilities. This combined time division multiple access and frequency division multiple access technique will be outlined in more detail in later chapters when the transmission of information over the traffic channel has been dealt with.

3.3 Demand Channel Assignment Procedure

As mentioned above, in the fixed channel assignment strategy the communication resources are allocated on a fixed basis independent of the user activity. Consequently, these fixed channel assignment procedures result in a wastage of the system resources in the case where a terminal has nothing to transmit and still occupies the channel while other terminals may have some messages to transmit [35]. In contrast to these fixed channel assignment procedures, demand channel assignment procedures attempt to solve the problem and hence improve the utilisation of the communication medium. The demand channel assignment protocols allow the users to have some access control over the resource. The feature of controlling the access to the communication resources by the users does not exist with the fixed assignment method. In the demand channel assignment procedures, the access to the communication channel could be achieved through centrally or distributed controlled mechanisms. In the centrally controlled method, the access to the communication medium could be provided by two protocols; polling protocol or reservation protocol [36]. Under the polling protocol, it is the responsibility of the controller (system) to find out if the stations have messages to transmit. In comparison, under the reservation technique, it is the responsibility of the user to initiate a channel request by transmitting a channel request message to the network.

Distributed control procedures, on the other hand, give the users more control to determine access to the medium by actually allowing the users to exchange control information between themselves. Exchanging of the control information between the users yields some access coordination to the shared channel [36,37]. An example of this procedure is the local area network (LAN) with a ring topology. In this structure, a station could establish access to a shared channel by reading a specially rotated control token. The control token, which is passed from one station to another in the ring, determines whether the channel is available for access or being occupied by another terminal [36,38].

3.4 Random Access Procedure

Using basic contention random access procedures, individual stations with packets ready for transmission will simply start their packet transmission regardless of whether any other station is involved in packet transmission or not [39]. An obvious problem with such a random access technique is that unsuccessful transmission will result when more than one user attempts to transmit at the same time. There are several more sophisticated random access techniques reported in the literature. Some of these protocols such as the well-known carrier sense multiple access (CSMA) and its version with collision detection facility (CSMA/CD), attempt to improve the efficiency of the protocol by allowing the station to sense the shared channel prior to packet transmission [39,40,41]. Sensing the shared channel actually enables the stations to determine whether the shared channel is idle or busy and thus reduce the chance of simultaneous transmission. The collision detection facility will further enhance the utilisation of the shared medium by enabling the transmitting station to detect when it is involved in a collision course.

With cellular systems, such medium access control protocols cannot be used in their simple form. This is due to the fact that there are some problems that are associated with the cellular system that do not exist in the fixed network, for example, when two mobile stations are in communication or wish to communicate with each other but are out of range or separated by some physical obstacle. These problems need a controller for their solution in order to improve the performance of the adopted medium access control protocol. Busy tone multiple access (BTMA) [42], idle-signal casting multiple access (ICMA) [43] and its collision detection version (ICMA/CD) [32] are examples of modified medium access control protocols that are suitable for the cellular radio environment.

In common with other radio cellular systems, the pan-European digital mobile radio system known as Groupe Spécial Mobile (GSM) will support two distinctive types of logical channels, namely traffic channels and control channels. The traffic channels and the control channels, which are supported on the up-link as well as the down-link parts of the radio interface, will be used to carry user information and control data respectively. The GSM system has adopted a random access technique to be implemented on the up-link common control channels. These common control channels, which are used for multipoint-to-point transmission, are used by the mobile stations to transmit their access request packets over the radio link to the base station [44,45,46].

The GSM system has adopted the well-documented slotted ALOHA random access technique for implementation on the access channel. The slotted ALOHA protocol is a modified version of the pure ALOHA random access protocol. In a system where the pure ALOHA protocol is adopted, individual stations start their packet transmission as soon as the packets are ready for transmission. This pure ALOHA protocol suffers from the collision problem mentioned above if any other terminal starts its packet transmission before this terminal completes its packet transmission. In the pure ALOHA protocol, the terminal is unable to determine by itself whether it is involved in a collision with another terminal or not. However, the terminal assumes an unsuccessful transmission if it does not receive an acknowledgment message from the base station within a prescribed period of time. In the event of collision, i.e. no acknowledgement message is received after waiting a certain time, the terminal schedules a retransmission of the same packet at a later time [39,47]. Retransmission of the packets might be achieved through the execution of back-off algorithms. There are various types of back-off algorithm, such as binary exponential, used to randomise the retransmission times of the packets and thus reduce the occurrence of collisions. Reducing the occurrence of collisions will have a positive impact on the performance of these medium access control protocols. It has been reported [39,19] that the channel throughput of ALOHA medium access control protocols is 1/(2e) where e is the exponential constant. Thus pure ALOHA protocol achieves a maximum throughput of 0.184 or 18.4%.

The ALOHA protocol suffers greatly from the non-synchronisation between the users. It is, however, possible to modify the protocol to operate with synchronised users. Roberts [48] suggested that the channel may be slotted by requiring each user to start his packet transmission at certain fixed instants of time (i.e. at the beginning of the slots). The slot sizes are chosen to be equal to a packet transmission time. This modified protocol is referred to as " slotted ALOHA random access protocol " or " slotted ALOHA ". In the slotted ALOHA, collisions will happen only when more than one user attempts to transmit at the beginning of a specific time-slot. By adopting this protocol, the pure ALOHA channel throughput has been doubled from 1/(2e) to 1/(e) (i.e. from 18.4% to 36.8%).

3.4.1 GSM Access Protocol

This section outlines the random access procedure adopted by the GSM system. Prior to establishing a radio resource connection, a mobile station has to transmit a channel request message on the Random Access Channel (RACH). The radio resource connection enables the mobile station to establish a point-to-point communication. In the GSM system, the scattered mobile stations will attempt to send their channel request messages on the RACH by following the slotted ALOHA random access protocol mentioned above. In the GSM system it is required that the channel request messages sent in one time-slot must reach the base station at the exact specified time and must not overlap with the messages sent in the next time-slot. The base station, therefore, provides a reference signal which, by definition, arrives at the mobile stations along a time-base depending on their location. Thus they signal back at different times synchronous with the same time-base.

The GSM system has been designed to serve radio cells that have radii up to 35 kilometres [44]. This implies that a transmitted signal requires 0.252 ms to reach the cell periphery and return. The access burst which is sent on the RACH is characterised by this extended guard period and a graphical representation of the access burst has been presented in figure 3.1.

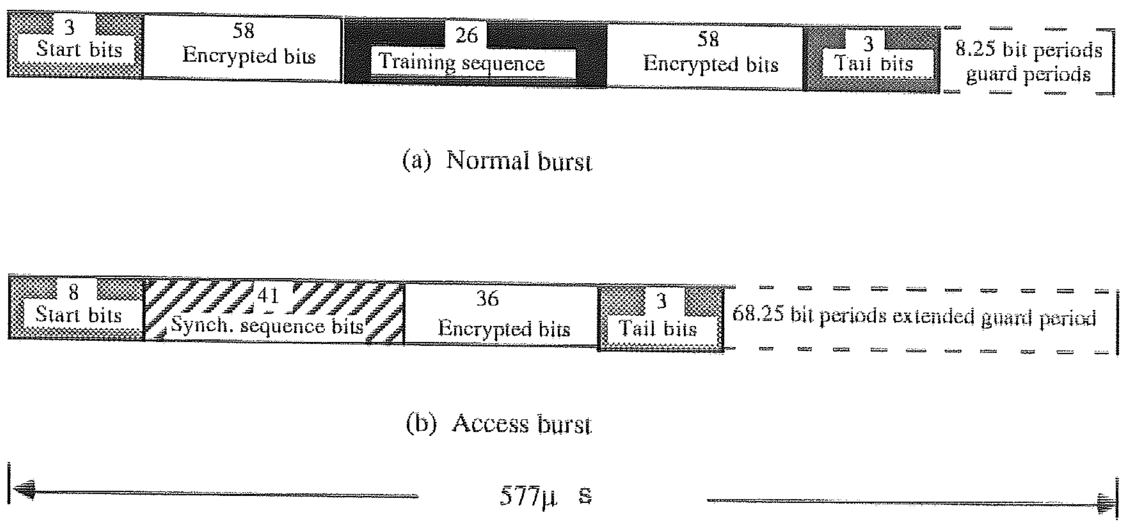


Figure 3.1 GSM burst structure (a) normal burst (b) access burst

The sending of the channel request message on the RACH is scheduled by using a timer known as T3120. The timer is started and at its expiry the channel request message is dispatched. The value of the timer is drawn randomly according to a specific statistical law. All the mobile stations will receive a RACH control parameter information element included in every broadcast system information message. These control parameters will be used in the calculation of the timer value. GSM recommendation [04.08] [49] presents in more detail how the value of the timer T3120 can be determined.

The channel request message will carry two important pieces of information, one indicating the reason why a radio resource connection is needed and the other is a random reference number. The former will assist the network to give priority to services such as emergency calls, whilst the latter is used by the mobile station as a temporary identification.

The mobile station starts the timer T3120 with a new value as soon as the sending of the initial channel request message is completed. While the T3120 is running, the mobile station will keep listening to the full down-link common control channel (ie. be ready to receive an access grant message which might arrive immediately). However, if the T3120 expires and no answer from the network is received, the mobile station repeats the sending of the channel request message if the "maximum re-transmission" allowed in the radio cell is not exceeded. The information concerning the "maximum re-transmission" allowed in the radio cell is obtained from the system information broadcast on the broadcast control channel. It is worth mentioning here that the mobile station draws a new random reference value prior to every re-attempt of re-transmission of the channel request message. In the case when both T3120 expires and the "maximum re-transmission" allowed has been reached, the mobile station waits for some time to allow the network to answer. A cell re-selection will then be performed prior to any future request attempt [49].

If it has succeeded in reading a channel request message, the network side responds by sending an immediate assignment message on the down-link part of the common control channel. The ability of the network to read a channel request message depends on the probability that the received signal strength of this message will be higher by a certain value that the summation of the other received signal strengths received in the same slot (a phenomenon known as " capture effect ").

The information transmitted in the information field of the immediate assignment message will include information on the dedicated control channel, information field of the received channel request message, the frame number in which the request message is received, description of the new configuration and the initial time advance. The time advance is used by the mobile stations to ensure their bursts arrive at the base station at the correct time (ie. not overlapping with other bursts from other mobile stations).

As soon as an immediate assignment message is received, the mobile station switches to the dedicated control channel and establishes a signalling link on this channel by sending a set asynchronous balanced mode frame. The set asynchronous balanced mode (SABM) frame will include the actual mobile identification number and a duplicate of the transmitted SABM frame will be stored in the mobile station. The network responds to a SABM frame by returning the same SABM frame back towards the mobile stations. The mobile station will compare the received SABM received on the down-link with the one it has already stored. The mobile station maintains the dedicated control channel if the two SABM frames match, otherwise it leaves the channel. By adopting this procedure, the mobile station resolves the contentions if several mobile stations have accessed the same random access slot with the same temporary identification random number [49].

3.5 Description of the Proposed Procedure

As mentioned above, the GSM system has adopted a random access channel protocol to achieve a multipoint-to-point transmission. In this procedure, a mobile station (MS) wishing to transmit on the RACH will use a seven bit random number as a temporary identification sent in its access burst.

Upon successful reception of an access burst, the network side will respond by sending an access grant message containing, among other information, the same information received in the access burst and information about the assigned dedicated control channel. The access grant messages are actually sent to allocate a stand-alone dedicated control channel or, in some situations, a traffic channel. The mobile station will use the traffic channel as a dedicated control channel if a traffic channel has been allocated to it in the access grant message [40,49].

Due to the capture effect, a possible contention on the same dedicated control channel will result if several mobile stations have accessed the same random access slot with the same temporary identification number. This inevitably leads to a lower utilisation of the dedicated control channel as well as an increase in the access delay times. This research, however, presents an algorithm which exploits the difference in the distance separation between the mobile stations and the base station as an added control dimension to reduce the probability of contentions on the dedicated control channel. Reducing the number of possible contentions on the dedicated control channel will improve the efficiency of the dedicated control channel as well as the random access channel.

In the mobile radio environment, the base station which is centred at the radio cell (radio zone) will provide service to the subscribers in that cell. The operating principle of the protocol is, however, to virtually subdivide the radio zone into a number of mini-zones. The size of the mini-zones is determined by the running speed of the new timer. By adopting this procedure, a contention on the dedicated control channel will only take place when more than one mobile station existing in the same mini-zone attempts an access in the same time-slot and having the same temporary identification random number.

In this protocol, each MS will be fitted with this new timer known as the contention avoidance timer (T_{ca}). The new timer runs on a one quarter of a bit period basis (ie. the timer is incremented 4 times per bit duration). The starting of the timer will be triggered by the expiry of T3120 when an access burst is transmitted. The timer will be kept running until either an answer to the channel request message is received or the T3120 has expired after being restarted upon completion of the burst access transmission.

Upon receiving an answer to a channel request message, the contention avoidance timer at the mobile station would have been running for a period of time (T) according to the following equation :-

$$T = T_{pg} + T_{fx} + T_{pr} \qquad \text{..............................} \qquad (3.1)$$

where

T_{pg} = propagation delay time (base station > mobile station > base station)

T_{fx} = the time needed to transmit the channel request message

T_{pr} = processing time. The processing time is actually the time it takes the base station to answer back upon successfully receiving a channel request message. This time is calculated from the time that the last bit in the access burst is received to the time the immediate assignment message is scheduled to be transmitted on the down-link part of the common control channel known as the access grant channel. The processing time includes the data analysis time by the base station, which is equal for all the mobile stations.

In this procedure the base station is required to include the processing time in the information conveyed by the access grant message. Upon receiving this information, the mobile station will be able to calculate the propagation delay time. The calculated propagation delay time will be compared with the time advance transmitted in the access grant message. The advance time usually represents the distance separating the mobile station from the base station calculated by the base station. The advance time, which is updated according to the movement of the mobile station, is used by the mobile stations to transmit their normal bursts in advance so that the burst reaches the base station at its specified time. The advance time will be explained in more detail in a later chapter. The proposed procedure therefore compares the mobile station to base station distance separation signalled by the base station with the one that has been calculated by the mobile station. This new feature assists the mobile station to decide whether or not to contend on the dedicated control channel. In this case, the mobile station will now only contend on the allocated control channel if the calculated propagation delay time matches the time advance and the same reference number as transmitted in the access burst is received. The new timer will be reinitiated prior to every transmission or retransmission on the RACH.

3.6 Simulation of the Proposed Procedure

In order to explore the capabilities, the efficiency and to perform a comparison between the access protocol adopted by the GSM system and its modified version considered in this research, each of these access protocols was simulated on the computer using a special oriented simulation language known as simulation language for alternative modelling (SLAM). The reader should refer to chapter (6) for more detail of the reasons behind the choice of this particular simulation language as well as how the SLAM language is used to code both access protocols.

In simulating the access protocols, there are some parameters which have to be assumed. The assumptions of these parameters are needed in order to simplify the simulation task. The assumed parameters are:-

(i) one simulation time is normalised to 3.33μs real time

(ii) the system has been simulated for a period of 5 seconds real time which is equivalent to $1.5 x 10^6$ simulation time periods.

(iii) the mobile station chooses its temporary identification random reference number by selecting an integer number from a uniform distribution between 0 and 127. The temporary identification random reference number is included in the channel request message sent on the RACH and the random numbers selected from the uniform distribution correspond to the 128 bit patterns of the binary seven bits random reference number as mentioned earlier.

(iv) the spatial distribution of the mobile stations is assumed to be uniformly distributed over the radio cell areas.

(v) during the whole simulation period, the system has been assumed to be working with a perfect capture. As a result of the perfect capture assumption, the base station will always successfully decode one channel request message even if there is an overlapping of several channel request messages.

(vi) In this simulation, the access protocols have been tested against the worst case scenario. The worse case scenario considered in this research is defined as in the event of simultaneously receiving more than one access burst having the same random reference number, the base station captures one of the access bursts that have a repeated random reference number rather than capturing any other access burst that contains a unique random reference number.

The above assumptions have been considered for both access protocols. However, there are some more parameters which have to be assumed for the proposed protocol only. These assumptions are:-

(i) each mobile station is assumed to be fitted with the new timer termed in this research the contention avoidance timer.

(ii) the protocol has been tested against three cell radii. These cell radii are 1km, 3km and 5km.

3.7 Discussion of the Simulation Results

In the GSM access protocol, the number of retransmission attempts of the channel request messages on the RACH is limited. The network broadcasts the maximum number of the retransmission attempts allowed in a particular radio cell on the broadcast control channel. It is, therefore, required for each mobile station to check whether or not it has reached the maximum retransmission attempts number allowed in the cell prior to any reattempt of transmission. In the event when the maximum retransmission attempts number has been reached, the mobile station should wait for some time, allowing the network to answer back before it restarts the whole process again, which might involve a cell reselection procedure. In order to explore the effect of controlling the number of retransmission attempts, the GSM protocol, together with its modified version, have been simulated for the case when there is no control placed on the number of retransmission attempts of the channel request message and for the case where there is a control limiting the number of retransmission attempts is applied. Figure 3.2 represents the case for the GSM access protocol when there is no limit applied to the protocol. In the event of unsuccessful transmission of the access burst, the mobile stations always reschedule their retransmission of the access burst by selecting a random time period from a uniform distribution between one to four access burst periods. Since there are several parameters that are randomly chosen, three simulation runs were performed and all the figures in this research represent plots of the average value of the three simulation runs.

Figure 3.2 presents the results of different offered loads against the channel throughput for the GSM access protocol. The channel throughput is defined as the number of access bursts that have been successfully delivered to the base station multiplied by the time duration of the access burst divided by the whole simulation period. The figure shows that the channel throughput starts with a low value and then starts to increase as the traffic load increases up to a maximum point before it starts to decrease again. The low channel throughput experienced at the beginning is due to the low volume of the offered traffic loads to the channel which, in turn, results in a situation where the channel is idle for some time. As the offered traffic volume increases the channel utilisation increases to a peak region where the retransmission attempts number reaches a level at which it becomes a predominant factor. The retransmission attempts number is directly affected by the probability of simultaneous transmission of more than one mobile station having the same reference number. The probability of more than one mobile station having the same reference number is small with low traffic loads and this probability increases with increasing the offered load.

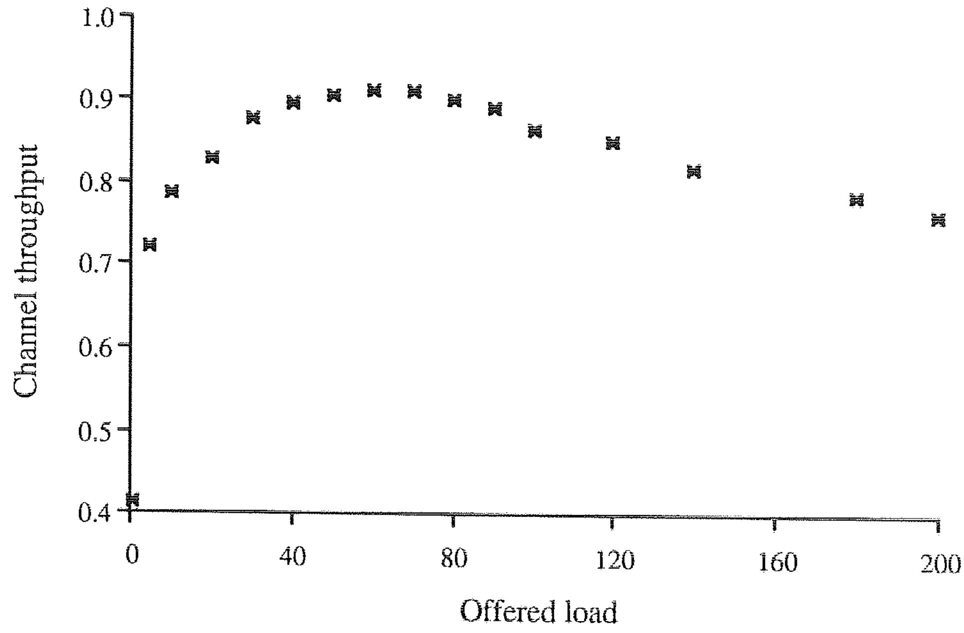


Figure 3.2 Channel throughput versus offered traffic loads for the GSM access protocol without limit control

Figure 3.3 represents the relationship between the access delay and the offered traffic loads for the GSM access protocol that has no limit on the number of retransmission attempts. The access delay is defined as the time between the initial attempt of channel request message transmission and the time the channel request message has been successfully transmitted. As expected, the relationship between the access delay and the offered traffic loads is almost a linear relationship. This is due to the fact that at higher traffic loads the network becomes congested due to the increase in the number of retransmission as well as the newly created messages.

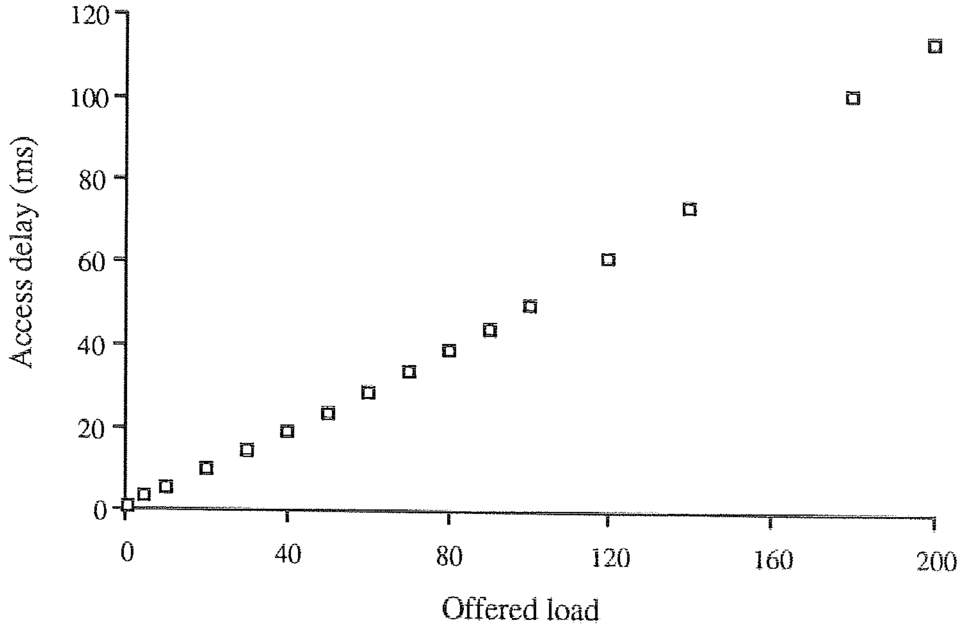


Figure 3.3 Access delay versus offered traffic loads for the GSM access protocol without limit control

Figure 3.4 shows the channel throughput versus the offered traffic loads for the GSM access protocol when a limit control is applied to the maximum number of retransmission attempts. Figure 3.4 shows that this protocol offers a very similar performance to the case when no control is applied as far as the channel throughput is concerned. This may be explained by the fact that there is now a number of mobile stations which have dropped out of the system because they have reached their upper limit of retransmission attempts whilst the remaining number is still sufficient to allow a channel throughput performance similar to the case when there is no control applied.

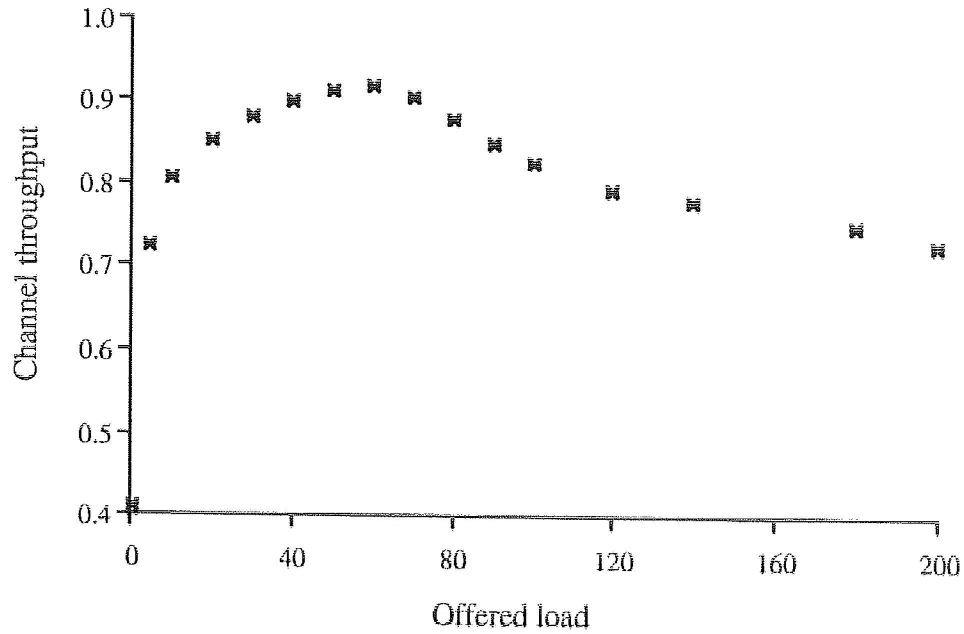


Figure 3.4 Channel throughput versus offered traffic loads for the GSM access protocol with limit control

Figure 3.5 shows the relationship between the access delay and the volume of the offered traffic to the channel. The figure shows that, although the protocol still exhibits the access delay versus offered traffic loads relationship as for the protocol without retransmission control, the access protocol with control limit offers much lower access delays compared to the access protocol without limit control. The lower access delays are achieved since the mobile stations which have reached their upper limit of retransmission attempts no longer have permission to attempt retransmission and are dropped out of the system.

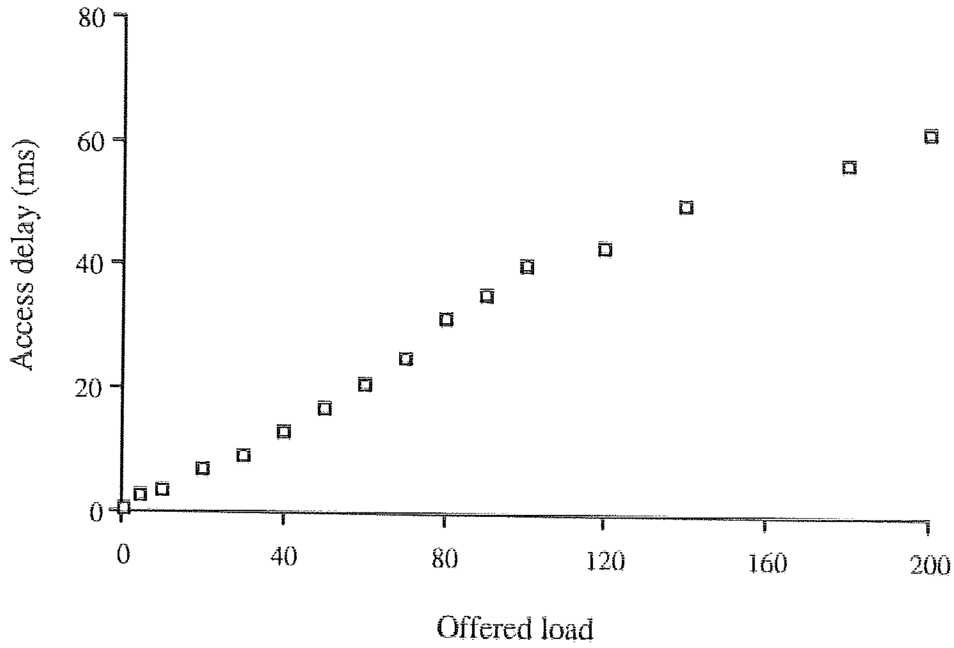


Figure 3.5 Access delay versus offered traffic loads for the GSM access protocol with limit control

Figure 3.6 illustrates the relationship between the access delay and the channel throughput for the GSM access protocol with and without control limit on the maximum retransmission attempts. This figure of access delay versus the channel throughput shows that, as the offered traffic loads increases, the channel throughput increases but at the expense of the access delay. The figure also shows that the channel throughput increases with increasing offered loads to a maximum point before it starts to decrease, whilst the access delay maintains a linear relationship with the offered load.

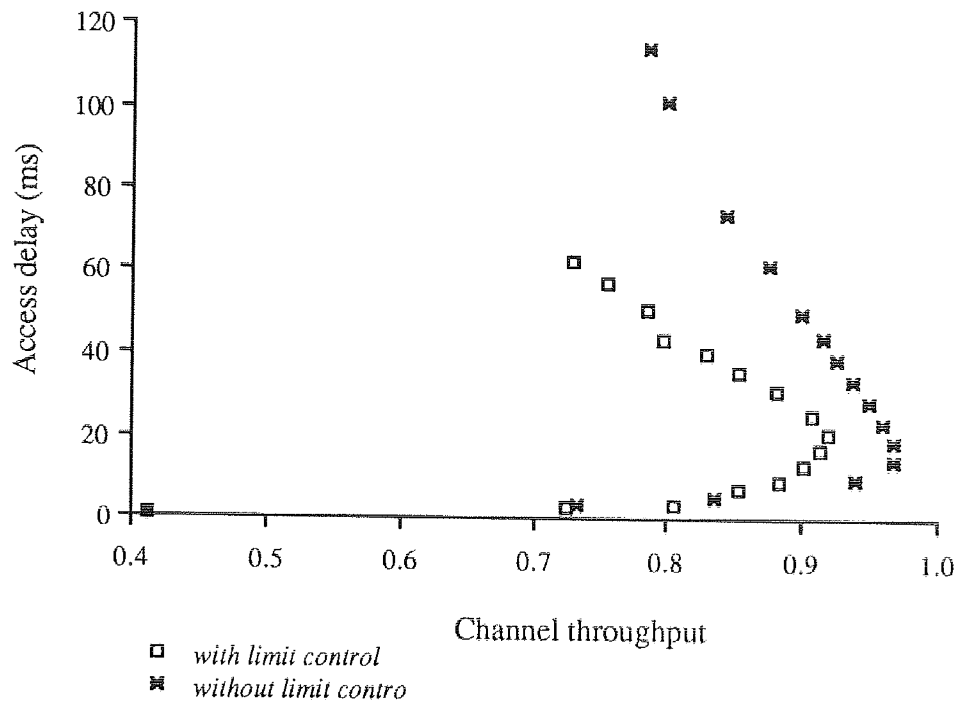


Figure 3.6 Channel throughput versus access delay for the GSM access protocol with and without limit control

The channel throughput against the offered load plots of the modified version of the GSM access protocol without limiting the retransmission attempts are illustrated in figure 3.7. The figure presents the channel throughput versus the offered traffic loads for various radio cell radii, together with the plot of figure 3.2, which is re-plotted on the figure 3.7 for comparison purposes.

Figure 3.7 demonstrates the advantage of applying the distance separation as a control parameter to the GSM access in the case where no control limit is applied to the number of reattempts. Applying the distance control parameter in this case has improved the channel throughput performance, especially at higher offered traffic loads. The increase in the channel throughput performance is achieved since in this case the mobile stations will only contend on the dedicated control channel when they exist in the same mini-zone area and have the same random reference number. The modified protocol exhibits a slight increase in the channel throughput as the cell coverage area increases. This is due to the fact that the probability of more than one mobile station having the same random reference number existing in the same mini-zone decreases with increasing cell coverage area.

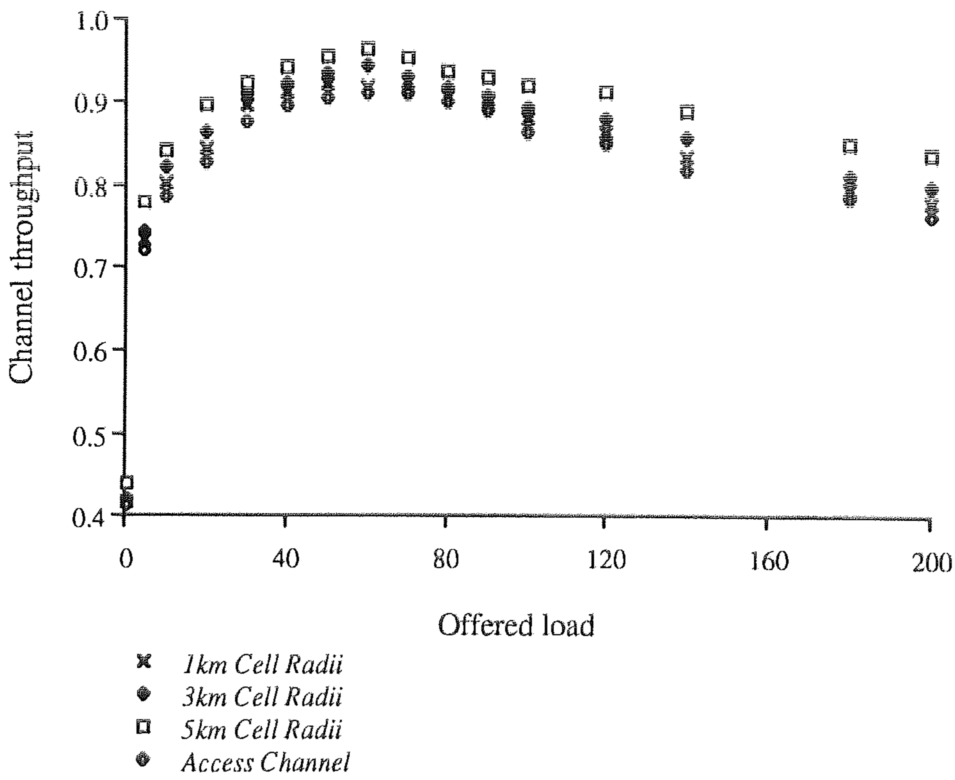


Figure 3.7 Channel throughput versus offered traffic loads for the modified protocol where distance parameter is applied and no control limit

Figure 3.8 presents the access delay versus the offered traffic loads for the above protocol. The figure clearly shows that the modified version of the access protocol offers a better access delay performance. This is due to more access bursts being successfully transmitted as a result of less contention on the dedicated control channel. The modified protocol also offers a slight improvement in the access delay performance as the coverage area increases. This is as a result of further improvement in the possible number of mobile stations that can contend on the same control channel.

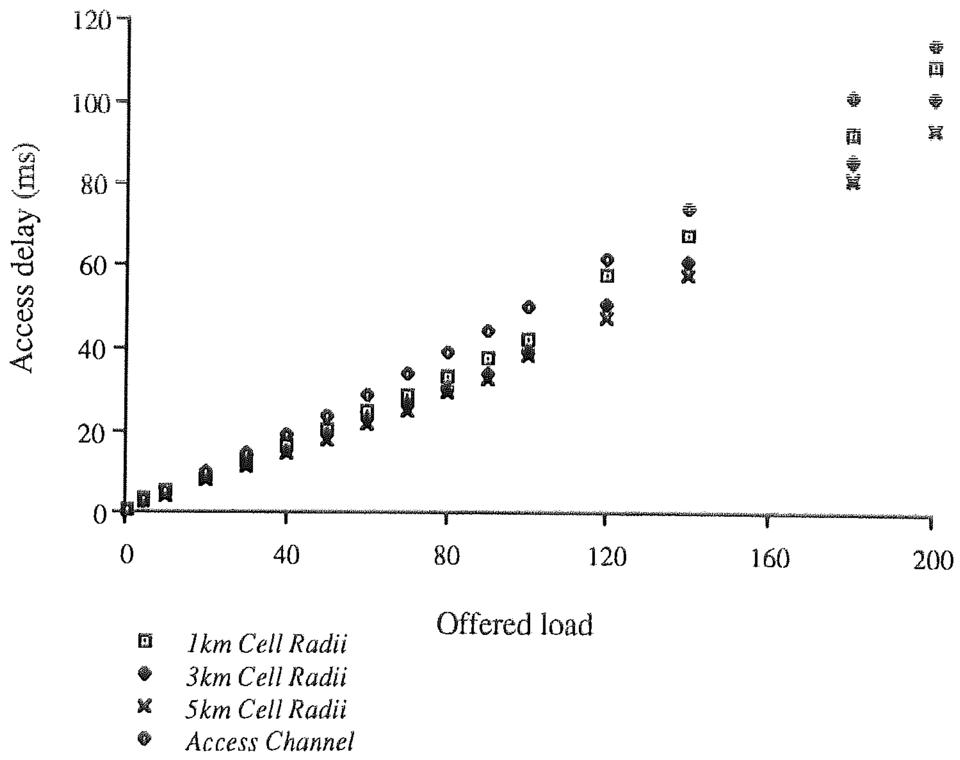


Figure 3.8 Access delay versus offered traffic loads for the modified protocol where distance parameter is applied and no control limit

Figure 3.9 represents the channel throughput versus the offered traffic loads for the access protocol when the distance separation control parameter is applied to the GSM access protocol and with a control limit on the number of retransmission attempts. The figure again presents the plots for different radio cell radii together with the plot of figure 3.4 for the comparison purpose. The figure shows that the modified protocol offers a good channel throughput performance against the offered traffic loads. This is due to the high rate of successful delivery of the access bursts.

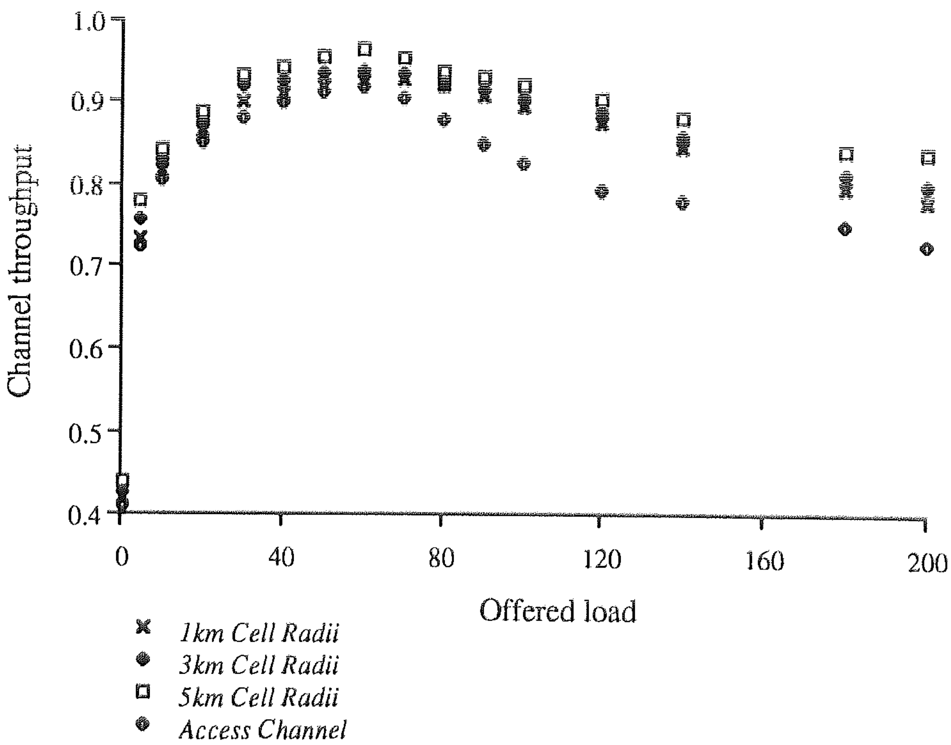


Figure 3.9 Channel throughput versus offered traffic loads for the modified protocol where distance parameter as well as the control limit are both applied

Figure 3.10 illustrates the access delay relationship with the offered traffic loads of the above protocol. The figure clearly demonstrates the advantage of applying the distance separation as well as limiting of the number of retransmission attempts as combined control parameters for the possible contention on the dedicated control channel. This improvement of the access delay performance is a result of dropping mobile stations out of the system when they reach the maximum allowable number of retransmission attempts permitted in the radio cell, coupled with the exploitation of the advantages of the distance separation control parameter.

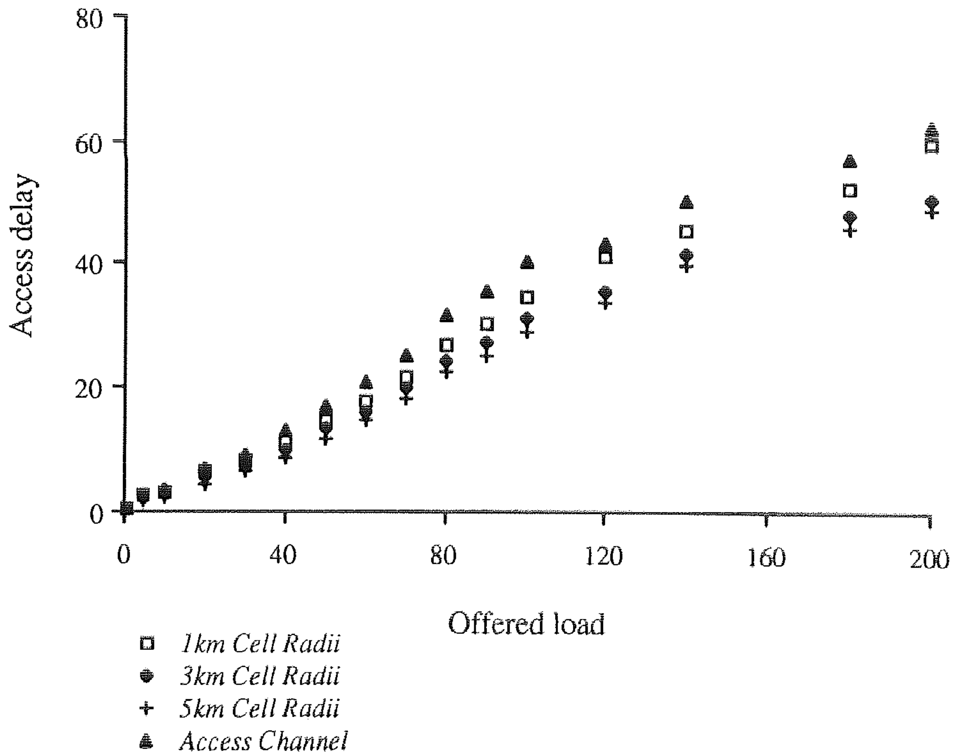


Figure 3.10 Access delay versus offered traffic loads for the modified protocol where distance parameter as well as control limit are both applied

Figures 3.11 and 3.12 are presented to show the relationship between the access delay and the channel throughput for the modified protocol without and with limiting the number of retransmission attempts, respectively. Both figures show that the channel throughput increases with increasing offered traffic load up to a peak point before it slowly starts to decrease. On the other hand, the access delay increases with increasing channel throughput for both cases, until the peak point, after which the access delay deteriorates very rapidly for the case when there is no limit applied. This deterioration of the access delay is due to the congestion in the system. However, the plots show that the modified protocol is superior as far as the access delay is concerned when it is applied to the actual GSM access protocol which applies a control on the maximum number of retransmission attempts.

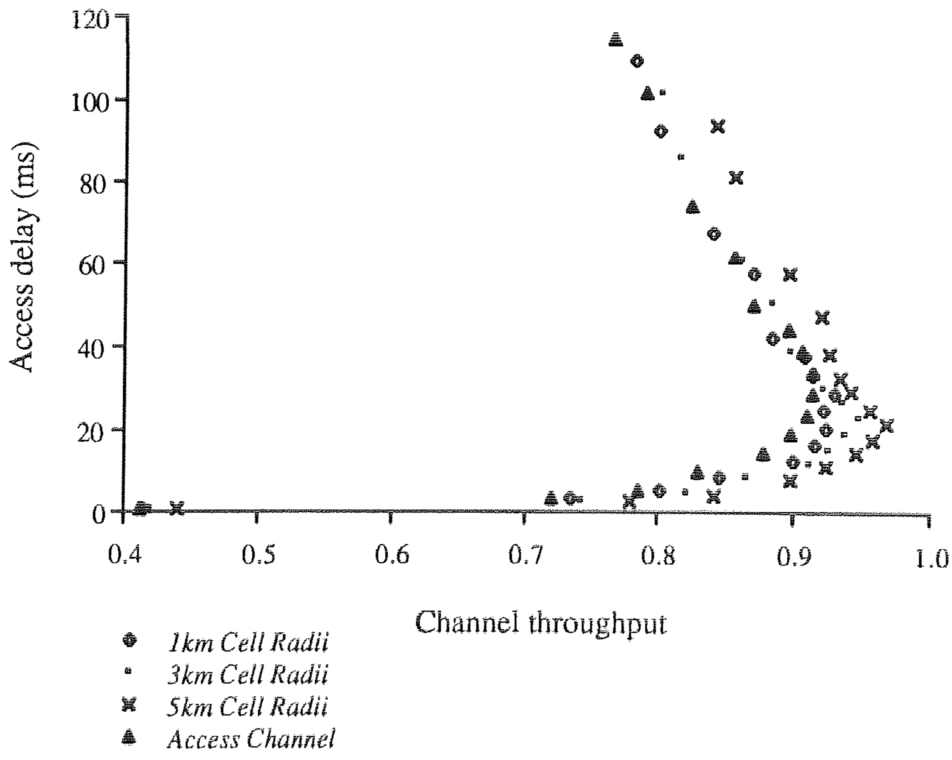


Figure 3.11 Channel throughput versus access delay for the modified protocol where distance parameter is applied and no limit control

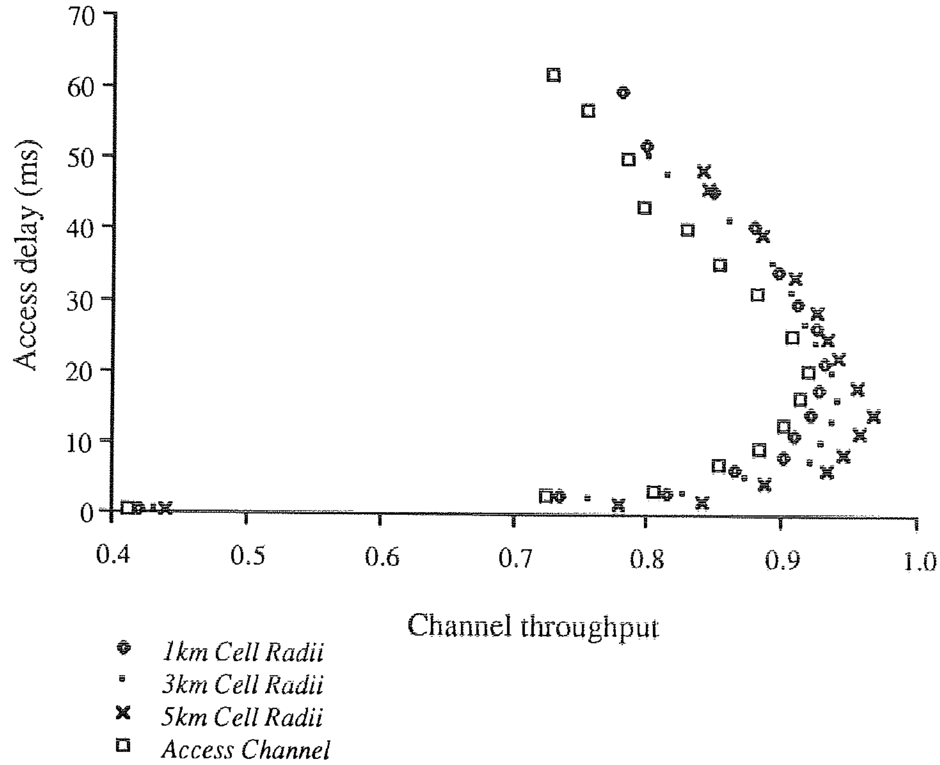


Figure 3.12 Channel throughput versus access delay for the modified protocol where distance parameter as well as the control limit are both applied

3.8 Summary

An algorithm to improve the efficiency of the up-link control part of the GSM system has been presented. The algorithm relies on a new timer to reduce the number of mobile stations contending on the dedicated control channel. The improvement has been achieved in terms of reduction in the access delay which will have a positive impact on the efficiency of the GSM system as whole.

CHAPTER FOUR

VOICE AND DATA MAC PROTOCOLS FOR THE GSM SYSTEM

4.1 Introduction

Since the bandwidth assigned to the mobile radio services is limited, the transmission and the network techniques to be used in these mobile radio communication systems have to be very efficient in terms of utilising this precious resource. In the early conventional mobile radio systems, the mobile services are provided by assigning one base station equipped with a transmitter radiating at a high power at a high elevation to cover as large an area as possible. Each base station is assigned a group of disjoint radio channels in order to prevent adjacent channel interference and the system configuration does not change for the life-time of the system [50]. This approach has led to congestion and limited radio channels, since the channel frequencies cannot be reused in the same geographical area. As the demand for the mobile services is increasingly growing, the mobile radio systems have to find ways of increasing their efficiency without any increase in the spectrum allocation [51,52]. As a result, the mobile radio systems have to adopt and implement the cellular mobile radio concept, as a parameter among others, to increase their efficiency. In the cellular systems, the entire coverage area is subdivided into a number of smaller cells [1]. Each one of these cells is equipped with an individual base station operating on a dedicated set of radio channels and a low power transmitter at an elevation just high enough to achieve the desired coverage. Subsequently, frequency reuse technology can be used whereby radio channels on the same carrier frequency are used to cover different areas. A basic requirement with this technique is that the distance separating two radio cells using the same radio carrier frequency should be sufficient so that co-channel interference is not objectionable [51,1,2].

The GSM mobile radio system has been defined to operate in two 25MHz frequency bands, one for the mobile station to the base station and the other for the base station to the mobile station transmissions. The two 25MHz frequency bands are in the region of 900MHz which is currently used by the existing Total Access Communication System (TACS) [53,54]. With each radio frequency carrier occupying a bandwidth of 200KHz and supporting eight time-slots, the GSM offers the same total number of physical channels that is offered by the TACS system using the single-channel-per-carrier (SCPC) technique at a carrier spacing of 25KHz [2]. In contrast to the TACS system, the GSM system has adopted digital technology. It is the adoption of the digital technology that makes the GSM system more spectrum efficient compared with its predecessors [55]. Digital technology actually makes the GSM system more robust to interference than the analogue systems and hence it enables the system to operate at a much lower channel-to-interfence (C/I) ratio which, in turn, contributes to an increase in the system efficiency by allowing the system to operate at much smaller radio cell size arrangements [55,3].

As can be expected, the subscriber density has a direct relationship with the radio cell sizes to be considered. This implies that in the heavily populated areas the radio cell sizes will be much smaller that in the less populated areas. However, despite the fact that reducing radio cell sizes increases the channel reuse rate, there are some practical limitations to how small the radio cell size should be. Limiting factors such as the number of times a mobile terminal crosses from one cell boundary to another cell boundary (hand-over rate) have been taken into consideration [1,56,57].

As the demand for mobile speech and data services is increasing far beyond the expectation, a lot of research activities are now directed towards the development of protocols that enable an integrated services over the wireless medium. A modified version of the Packet Reservation Multiple Access (PRMA) was proposed [115], whereby the unreserved slot, which is used by the mobile stations to send their packets, is subdivided into a number of sub-slots. GSM parameters were used to show the improvement of such protocol over the GSM [116]. In this research a medium access protocol has been developed in order to increase the spectrum efficiency of the GSM system. The proposed medium access protocol explores the fact that speech can be represented as short bursts of vocal energy known as talkspurts separated by intervals of no vocal energy known as silence gaps. The talkspurt is actually considered when the vocal energy activity exceeds a given threshold while the silence interval is defined when the vocal energy activity sinks below this reference level [58]. The increase in the spectrum efficiency of the proposed medium access protocol is achieved by inserting data information in these silence gaps inherent in the speech. The performance of the mobile radio systems in delivering speech and data services is highly influenced by the characteristics of the mobile radio channel. In order to present an investigation into the performance of the proposed medium access protocol, perform a comparison with the GSM medium access protocol and outline how the proposed protocol can fit into this environment, it is appropriate at this time to briefly discuss the mobile radio channel characteristics and the measures adopted to reduce the effect of these channel characteristics on the transmitted information.

4.2 Mobile Radio Channel Characteristics

As opposed to fixed point-to-point communications, in the mobile radio communications environment the mobile stations are most frequently in motion. Thus the mobile stations will move at different speeds as well as travel in various directions [51,59]. As a result, the mobile stations pass many types of scatterers as they traverse their routes.

Obstacles such as high buildings in an urban environment and hills in rural areas frequently prevent the transmitter and the receiver lying on a direct "line-of-sight" path [60]. As a result, the radio propagation in the mobile radio environment is by way of signal reflection and diffraction from these scatterers surrounding the mobile receiver [51,59,61]. Thus the received signal is actually the sum of the scattered replicas of the transmitted signal due to multi-path propagation. Receiving the signal through different paths results in signal fading along with other path loss factors. It is possible to divide these scatterers and reflectors into two types: some large natural or man-made reflectors relatively at a distant from the mobile receiver called remote scatterers and nearby clusters of buildings and other objects known as local scatterers [62]. As mentioned above, the presence of the scatterers and reflectors along the radio path produces a dynamic change of the multi-path geometry with time as the mobile station traverses its route [59,60]. As a result, two classes of signal variations exist, short-term variations, also known as short-term fading or multi-path fading, and long term variations, known as long-term fading or shadowing. The short-term fading is due to the multi-path propagation of the signal as a result of scattering, reflection and refraction of the signal [62]. In this case the received signal envelope is characterised by its rapid fluctuations and the statistics of the received signal envelope has been shown to be a Rayleigh distribution. The phase of the received signal is uniformly distributed. Shadowing, on the other hand, is due to terrain topography such as hills, buildings and other obstacles. The mean value of the received signal caused by this type of fading obeys a log-normal distribution [51,60,61].

In the mobile radio communications environment, if the transmitter and the receiver are stationary then the mobile radio channel could be considered as a stationary channel resulting in a "static multi-path" situation as the obstacles which cause the scattering and diffraction along the path are fixed. This is in contrast to the case when the transmitter or the receiver is in motion. In this case, the fading rate of the received signal is directly proportional to the speed of the vehicle. For example, a vehicle cruising at 40 mile per hour (mph) and operating at a carrier frequency of 900MHz will suffer a fading rate of approximately 54 fades per second. The duration and depth of these fades depends on the vehicle speed as well as the signal-to-noise ratio (S/N) [50,63]. Moreover, the components of the received signal will also have a Doppler frequency shift associated with each path due to the movements of the transmitter or the receiver. This Doppler frequency shift sometimes introduces a small drift in the operating carrier frequency, causing an instability of the received signal frequency [50].

The multi-path propagation phenomenon introduce time delays to the scattered components of the transmitted signal. The magnitude of these delays is directly proportional to the length of the path that the signal has followed. In Lee [51] it has been shown that when the base station transmits an impulse to the mobile station, the received impulse at the mobile station's receiver is significantly lengthened (spread out in time) due to the multi-path propagation. It has also been shown that as the number of scatterers and reflectors surrounding the mobile station increases, the transmitted impulse becomes a continuous signal pulse with a pulse length Δ known as delay spread.

The delay spread imposes a central problem in the digital mobile communications field. This is due to the fact that individual transmitted pulses tend to spread out and overlap in the multi-path fading channel resulting in a phenomenon known as "inter symbol interference" (ISI) [51]. In digital transmission systems, it is this time delay spread which imposes an upper limit on the transmission bit rate. The ISI actually becomes a predominant data speed transmission restricting factor when the delay spread becomes comparable to or greater than the bit period [51,3].

The ISI degrades the bit error rate (BER) which cannot be improved simply by increasing the transmission power. However, there are several countermeasure techniques adopted in the digital mobile communications environment in order to improve the signal transmission performance in terms of BER. Equalisation is an effective technique used by mobile radio systems to enhance their digital information transmission capabilities. The equalisation process actually allows for the wanted signal to be recovered from a severely corrupted received signal due to multi-path propagation effects [64,65,66,68]. The operational mechanism of the equalisers can be simply explained as the equaliser making an estimate of the channel impulse response and then adjusting its coefficients accordingly to construct an inverse filter [67,73]. The received signal is then passed through the inverse filter which compensates for the heavy distortion due to the multi-path propagation.

In principle, it is always possible to construct an equaliser that will reduce the effect of the ISI if the channel characteristics are precisely known. However, in practice individual channels will have different channel characteristics. A channel being used for a particular connection will therefore be a random channel in the sense that this channel is one of a set of possible channels [65,66]. Consequently, constructing a fixed equaliser on the basis of an average channel characteristic may not be adequately effective against the ISI. An adaptive equaliser is then required which updates its channel impulse response according to the characteristics of the channel being used. As explained earlier, in the cellular environment it is expected that a mobile will be in motion, which causes a dramatic change in the channel impulse response. In order to assist the equaliser to operate adequately in such environment, a suitable training sequence is transmitted through the channel as a guidance for the equaliser to adjust its parameters. In the GSM system, eight different training sequences have been defined. Each training sequence is 26-bits long and every transmitted burst includes a training sequence placed in the middle of the burst. This allows the receiver to obtain synchronisation and at the same time obtains a new estimation of the channel impulse response at each received burst by correlating the received signal with the training sequence which is known at the receiver [64,65,66].

4.3 Speech Coding Scheme For The GSM System

All subscriber information in the GSM system is conveyed via a digital channel and this requires that the analogue speech needs to be coded into digital information prior to its transmission over the radio path. For this purpose the European Postal and Telecommunication Committee (CEPT) Groupe Spécial Mobile (GSM) has carried out intensive research and experimental tests based on different proposals to select a suitable speech coding scheme to be implemented for the GSM system [69,72,80]. The selection of the speech coding/decoding (codec) criterion was based, among other criteria, on the design which minimises the amount of the bandwidth required for the transmission of the speech at a quality at least comparable to the first generation of cellular systems. Initially, more than 20 different codec proposals from 9 European countries were under consideration. As a result of these subjective tests it was finally recommended the acceptance of the Regular Pulse Excited Linear Predictive Coder (RPE-LPC) with Long-Term (pitch) Prediction [70,71,79]. This speech coder uses techniques of spectral analysis and prediction in order to convert human speech into a bit stream very efficiently.

The RPE-LPC-LTP is a block based coder with a net bit-rate of 13kb/s [71]. Comparing the RPE-LPC-LTP with straightforward analogue/digital conversion and Pulse Code Modulation (PCM) coding, in which the latter requires 64kb/s without error correction, indicates the level of data reduction already achieved. However, the RPE-LPC-LTP still requires protection to the important bits in the information from the disturbance in the radio link as will be seen in the speech coding mechanism described below. A more detailed description of the GSM RPE-LPC-LTP algorithm has been given by Vary [70].

Figure 4.1 shows the primary elements of a speech link for a call established over the GSM system. The incoming speech patterns are analysed and processed in blocks of 20ms duration. Operating at its net bit-rate, the speech encoder produces an output of 260 bits corresponding to every 20ms speech sample. The 260 output bits of the speech encoder constitute a frame of speech. In contrast to waveform encoders such as PCM, the 260 output bits of the GSM speech encoder are not of equal importance as far as the effect on the speech quality is concerned [71,82]. The above mentioned dynamic behaviour of the mobile radio channel characteristics can sometimes result in a loss of entire speech frames. To minimise the influence of the radio propagation and thus enabling the transmission systems to offer a reliable level of performance, protective measures have to be employed to any information transmitted over the radio link. Error-control protocols can then be employed whereby errors in the transmitted information can be detected and can possibly be corrected [74]. This can be simply described as the addition of redundancy bits to the transmitted information bits over the radio link. Two different error-control mechanisms can be used. For error detection an

automatic repeat request (ARQ) protocol can be used. In the ARQ systems, a few check bits are added to each block of transmitted data so that transmission errors can be detected. Once errors are detected, an erroneous block can then be requested for retransmission until it is successfully received. Different strategies are possible for this error detection and retransmission protocol such as ***stop-and-wait*** ARQ, ***selective-repeat*** ARQ and ***go-back-n*** ARQ [74,75]. The GSM system, on the other hand, has adopted a forward error correction (channel coding decoding) mechanism to protect its data transmission against the effects of the mobile radio channel. With a forward error correction (FEC) protocol there will be no repetition of the erroneous blocks, instead the receiver attempts to correct the transmission errors. This mechanism also requires the addition of some redundancy bits to the transmitted bits prior to their transmission over the radio channel as can be seen in the channel coding algorithm adopted by the GSM system for speech transmission over the radio link.

The GSM speech coding scheme is ordered to produce speech data bits that are ranked according to their subjective importance [4]. The resulting output of the speech data bits is divided into two classes; class one consists of the first 182 data bits and class two consists of the remaining 78 data bits [55,71,82]. It has been found by subjective assessment that bits in class one are the most critical bits as far as the speech quality is concerned. By exploiting the fact that some bits within the encoded speech frame have a greater influence on the speech quality than the others, it is then possible to employ unequal error protection and thus produce a more robust channel coding scheme by offering a higher level of protection to the most important bits. This implies that protective measures have to be applied to protect the bits of class one. This process is illustrated in figure 4.2.

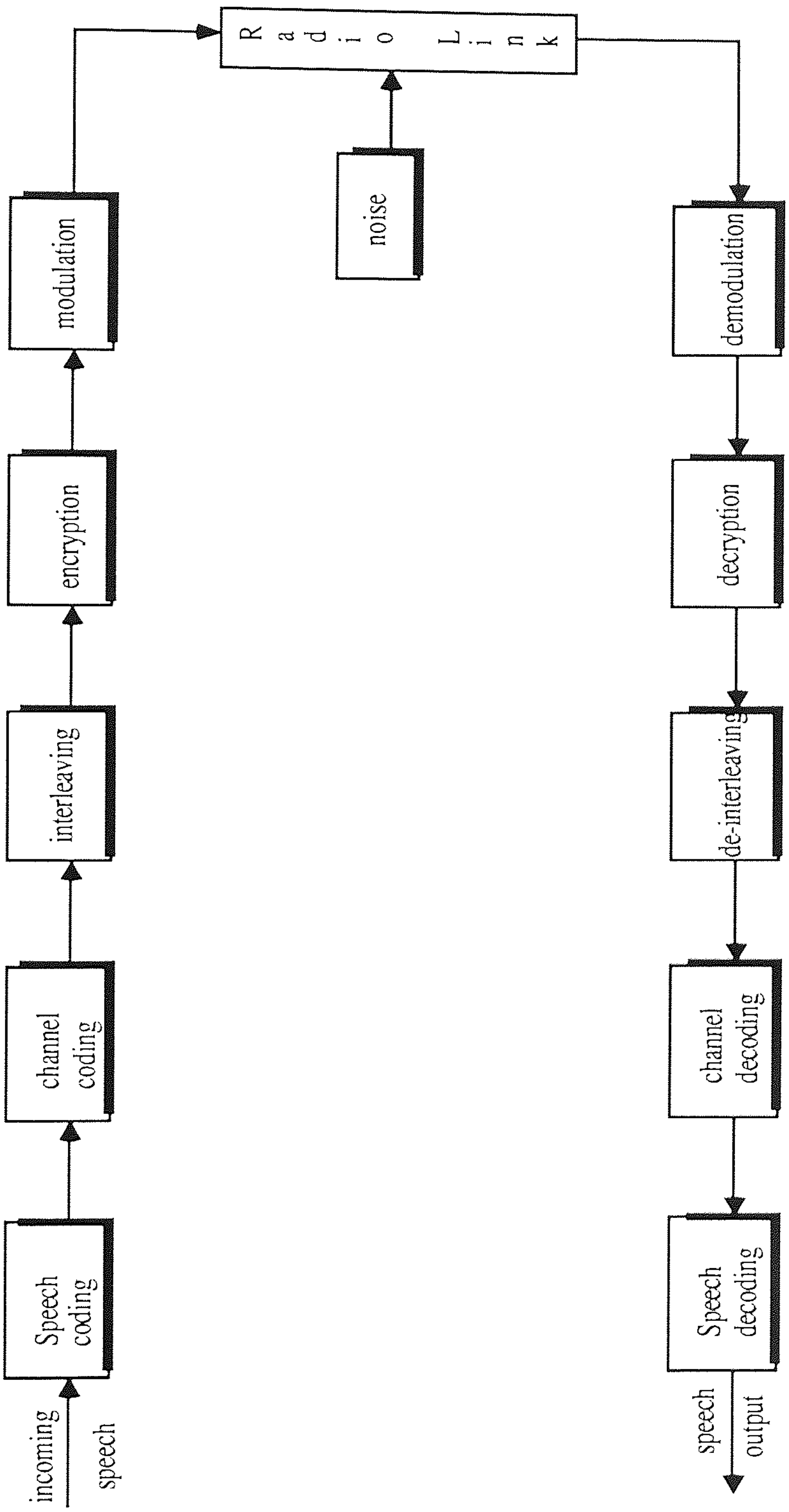


Figure 4.1 Speech link over the GSM system

The data bits of class one are further subdivided into class 1a and class 1b. Class 1a includes the leading 50 data bits and class 1b includes the remaining 132 data bits. Class 1a, which includes the most sensitive bits, has three parity added bits generated by a cyclic redundancy check (CRC) code. The addition of the three CRC parity bits to the most significant bits of the speech coder is in order to perform error detection. If the detector detects uncorrected errors in the class 1a bits, the complete speech frame is discarded. This is due to the fact that errors in the most significant bits has been found to result in serious speech degradation. When corrupted speech frames are discarded, the GSM system might use the speech frame substitution (SFS) strategy in order to improve the speech quality services. In the SFS strategy a corrupted speech frame is substituted by a previous uncorrupted speech frame [76].

As a result of the addition of the parity bits, the total number of class one data bits has now risen to 185 bits. A flag of four bits, all equal to zero, is then appended to the class one data bits. The purpose of this flag is to assist the decoder in making the right decisions and thus reducing the decoded error rate. The flag is actually used to inform the decoder of that it has received the final data bits. All of the 189 data bits of class one are now ready to go through a convolutional encoder having 1/2 (input bits to output bits) ratio. The convolutional encoder produces a measure of error correction in addition to error detection. Class two are low level (relatively unimportant) data bits and hence no coding is applied to them. The convolutional encoder output (a total of 378 data bits) is then added to the unprotected 78 data bits of class two to give a final speech frame ready for transmission consisting of 456 data bits. With this speech coding scheme adopted by the GSM system, every 20ms of analogue speech is finally encoded to 456 bits resulting in a gross bit-rate of 22.8 kb/s [55,71,77].

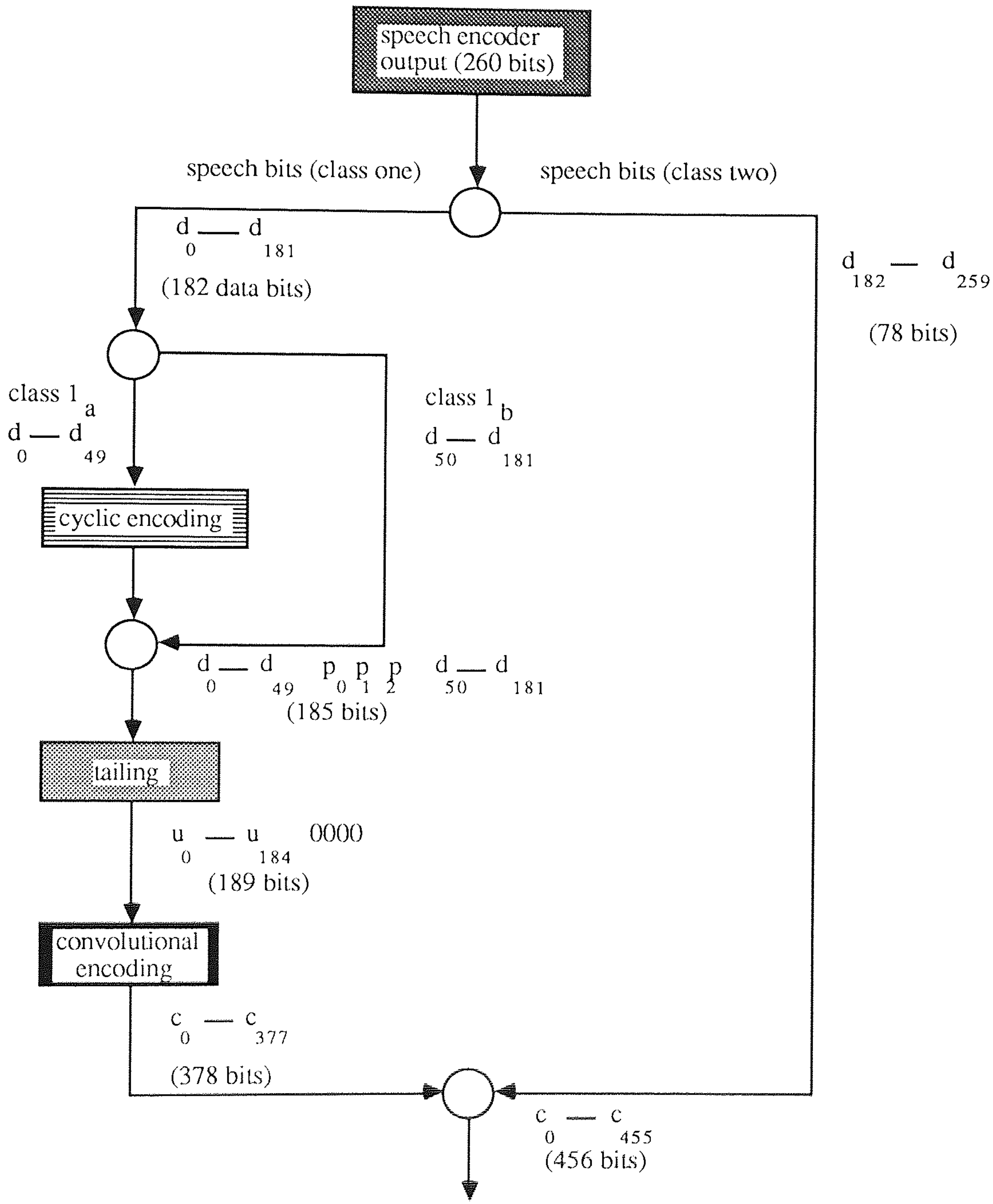


Figure 4.2 Speech channel encode for the GSM system

As mentioned earlier, fading due to multi-path propagation is the main source of correlated errors occurring over the propagation paths. With the GSM system transmitting at a rate of approximately 271kb/s, errors tend to occur in bursts since the fades occur at a much slower rate than the GSM transmission rate. To increase the efficiency of the error correcting code, measures have to be considered whereby the errors are distributed in time. The GSM system has adopted interleaving as well as reordering as a means to achieve this purpose [78]. Prior to the interleaving process, every block of 456 data bits has to be reordered and organised into sub-blocks of even and odd bits. Refer to the GSM recommendation 05.03 [78] for further details on the actual reordering process.

Interleaving is an effective technique in randomising errors when the channel exhibits error bursts [35]. With this technique, the burst-error channel is transformed into an independent-error channel for which many coding techniques are effective. Interleaving is actually a process in which successive bits transmitted over the communication channel are widely separated in the data sequence. There are two types of interleaving in existence in the GSM system known as intra-interleaving and inter-interleaving. For further details on these types of interleaving, the reader should refer to GSM recommendation 05.03 [78].

In the interleaving mechanism for the speech transmission adopted by the GSM system, each time-slot includes two sub-blocks belonging to two successive speech frames. As expected, increasing the interleaving depth will certainly improve the speech quality at the expense of increasing the transmission delay [77,35] It is therefore necessary to reach a compromise between the depth of the interleaving and the transmission delay. In the GSM system for speech transmission an interleaving depth of 8 has been adopted, which introduces an interleaving delay of approximately 37ms. A 'block diagonal' type of interleaving has been chosen for the speech transmission where a new data block starts every 4th block as illustrated in figure 4.3. On the other hand, different interleaving types and interleaving depth values have been considered for data transmission, where different transmission requirements are needed. The GSM interleaving process is illustrated in figure 4.3.

4.4 GSM Speech Transmission Over The Radio Link

The GSM system has adopted a burst mode transmission based on a slot and frame structure to achieve communication between the mobile stations and the base station. A burst is defined as the period that a data stream modulates a radio frequency (RF) carrier [53]. Two classes of burst exist in the GSM system depending on the information being conveyed, one contains user information, whether it is encoded speech or data, and the other contains control information, which includes the control signalling data needed for the system operation [53,54,4]. It is the transmission of these bursts in the consecutive TDMA frames that accounts for the speech and data transmission. The recurrence of one particular time-slot in each TDMA frame makes up one physical channel. Eight consecutive time-slots are grouped together to form a time division multiple access (TDMA) frame as shown in figure 4.4. The resulting TDMA frame has a duration of 4.616ms and thus there are about 217 TDMA frames per second. This primary TDMA frame is then used as a basis for constructing multi-frames. The multi-frames then form a basis for building up super frames which, in turn, are used to build up the hyper-frame as shown in figure 4.5.

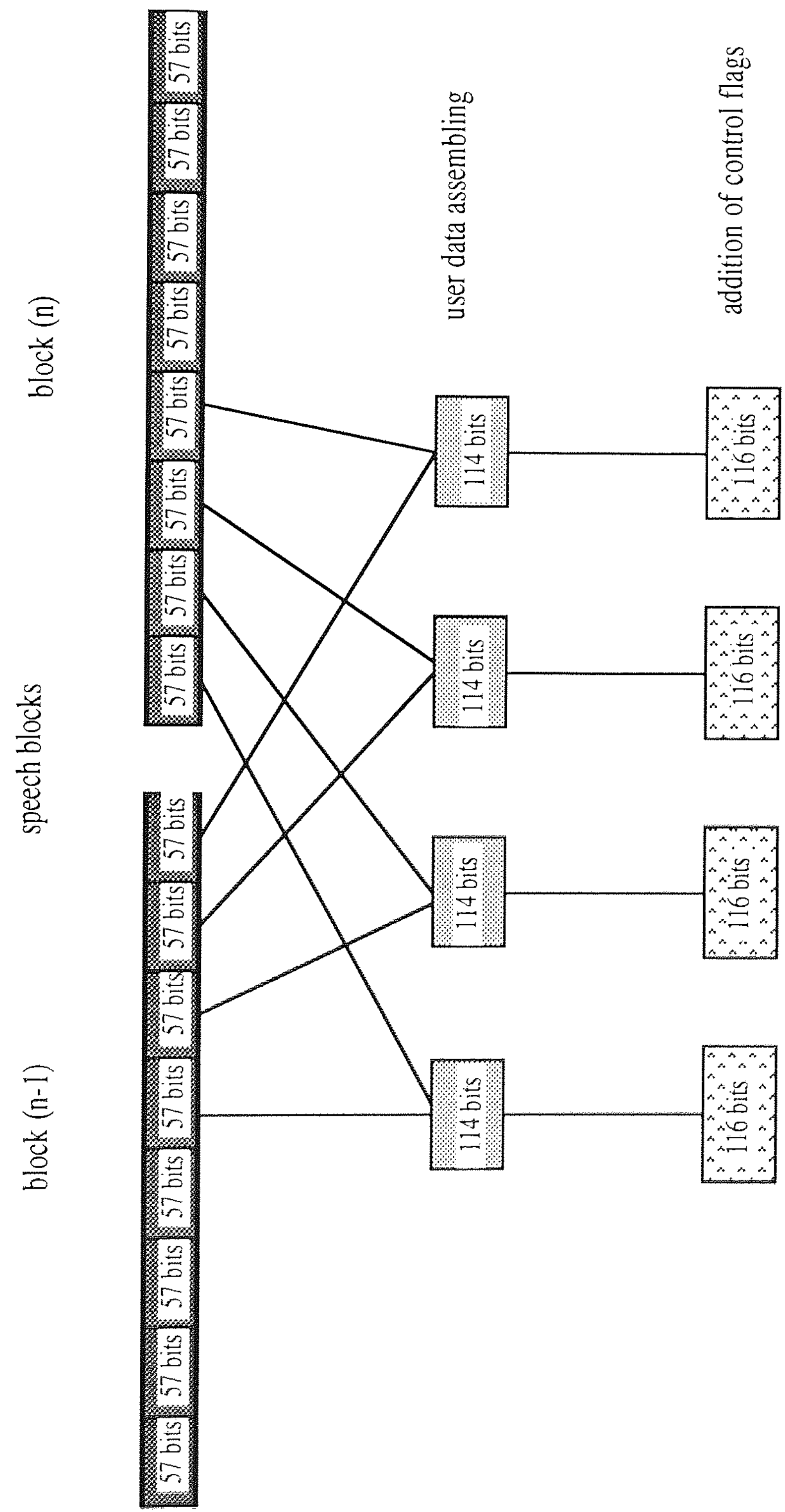


Figure 4.3 GSM speech interleaving process

Despite the fact that all the bursts have the same duration, the type of the burst depends on the nature of the data in contains. Each burst has a useful part where meaningful information is transmitted and a guard period where nothing is transmitted. In speech transmission, for example, the burst is built with two sub-blocks belonging to two successive speech frames as shown in figure 4.3. As mentioned earlier, in the mobile radio environment, subscribers are most likely to be in motion during a call session. Because of the cellular concept, the mobile station movement during a call period could result in a situation when the mobile station crosses from one radio cell territory to another radio cell territory. If this happens then the mobile station is transferred to a new radio link supported by the new base station which is serving the new radio cell. The change of the radio link during a communication phase is known as a hand-over process [56,87]. This requires a prompt delivery of the hand-over messages. The GSM system achieves this purpose by using the fast associated control channel (FACCH) which gains access to the physical link by stealing frames from the traffic channel it is associated with [46]. Two single bit flags corresponding to the two sub-blocks of the speech frames are then added to indicate to the decoder whether odd or even numbered bits haves been stolen for this signalling purposes. Stealing bits for signalling purposes will introduce gaps in the speech frames. The proposed solution is that the encoder attempts to bridge the gaps by using a prediction based on the previous speech frames in the cases when the gaps are short or to mute the output in the cases when the gaps are longer. The number of bits so far included in the speech burst has now risen to 116 bits. A 26 bit-long training sequence is now inserted in the middle of the burst to enable the equaliser to adjust to compensate for the multi-path propagation effects. Finally, a total of 6 bits are added to the 142 bits for the purpose of the start and stop flags to yield a complete burst of 148 bits. One fundamental requirement with the TDMA technology adopted by the GSM system is that the bursts transmitted on a single RF carrier must reach the base station at the exact specified time. If the bursts do

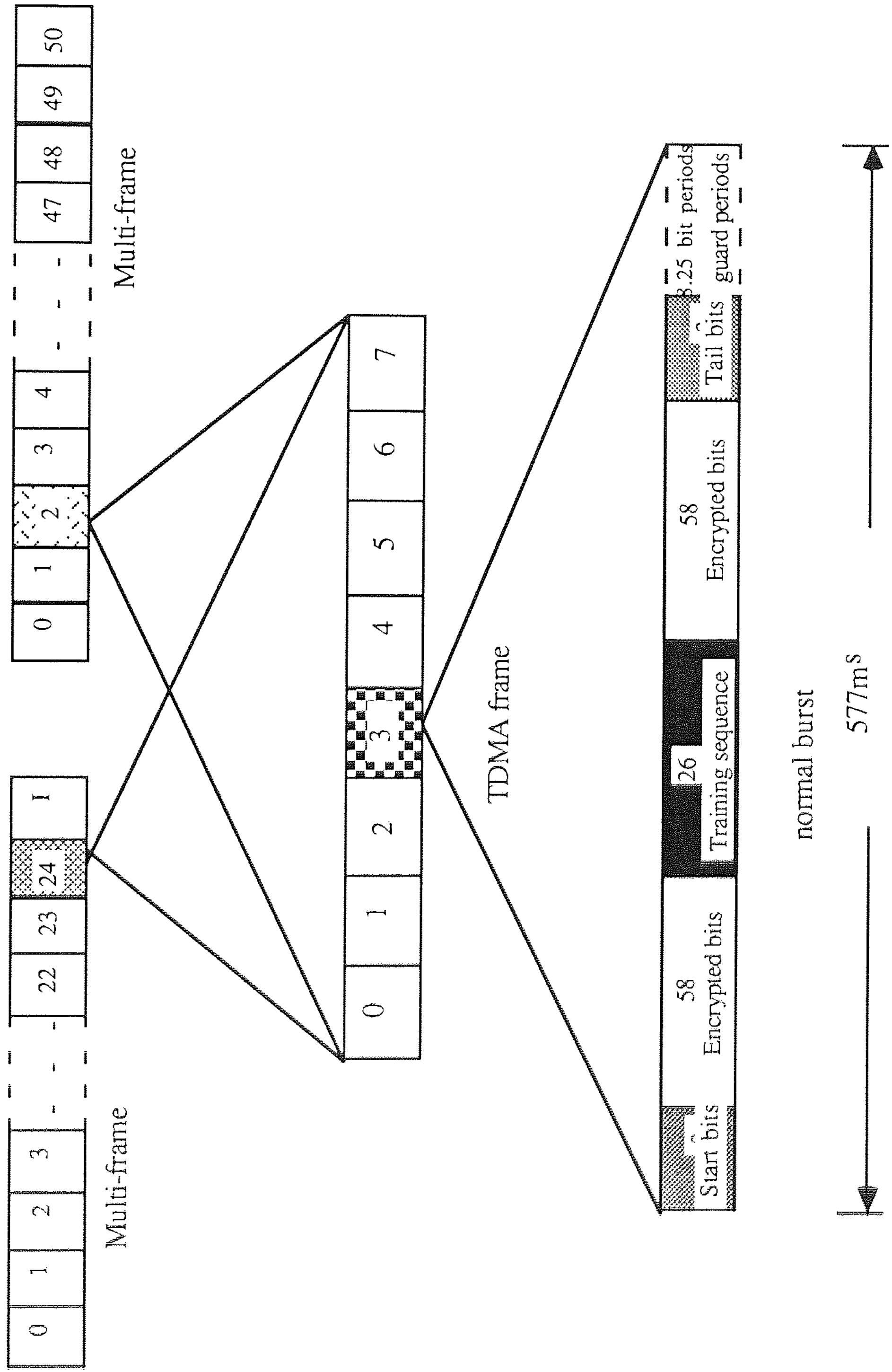
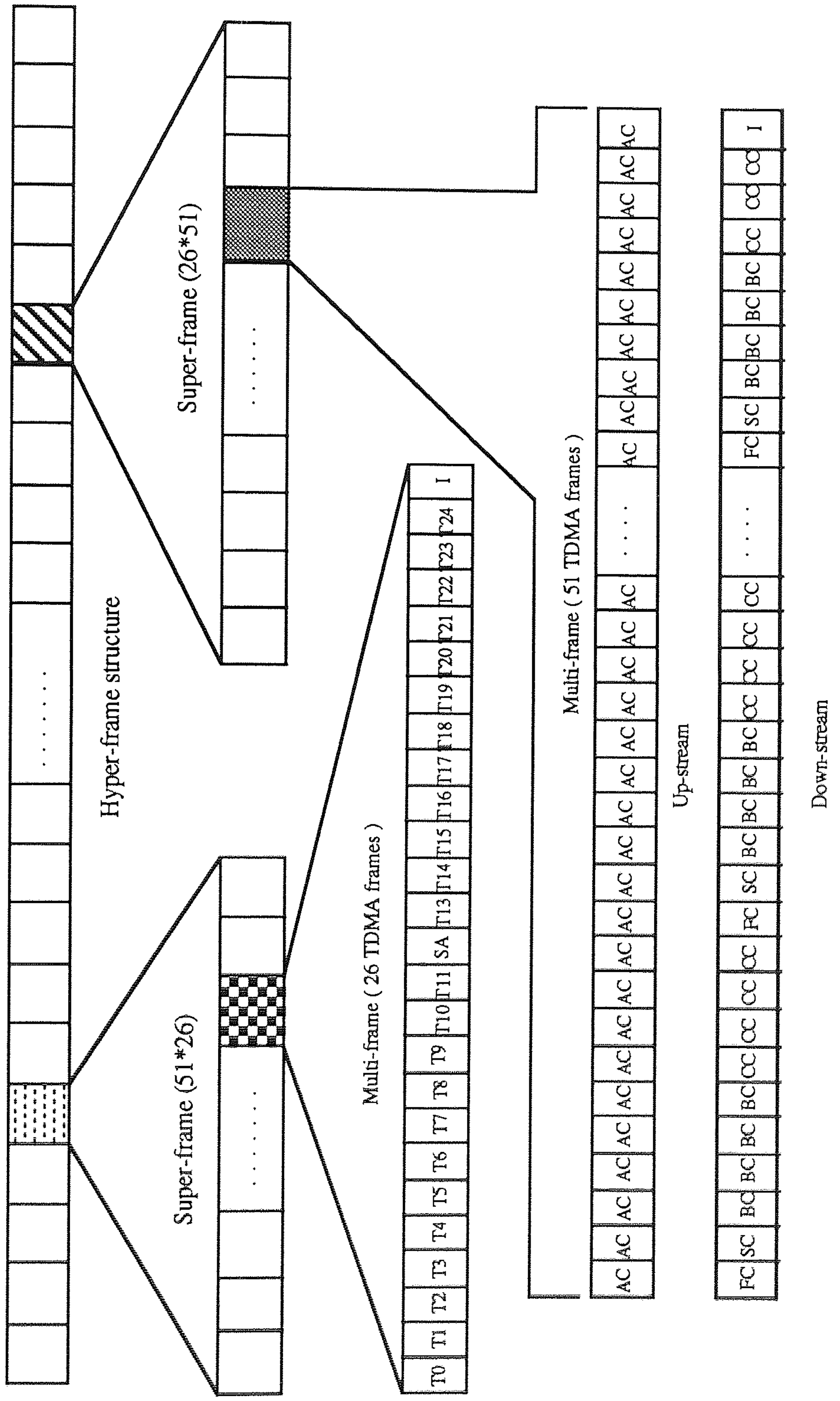


Figure 4.4 Time slot & TDMA frame structure



T = terminal   AC = access channel   FC = freq. correction ch.   SC= synch. ch.   BC= broadcast ch.   SA = associated control ch.   I = idle frame

Figure 4.5 Hyper- frame structure

not reach at the specified time then adjacent bursts in time will overlap with each other. A period of 8.25 bit duration has been used in the GSM system as a guard period between consecutive time-slots. With the GSM system transmitting at a rate of approximately 271kb/s this implies that a burst of 156.25 bit duration requires a period of 577μs to be transmitted over the radio link which fits exactly into a time-slot period. As a result of this transmission rate, each bit has a duration of 3.69μs.

4.5 Time Advance Strategy

The GSM system has been defined to operate in radio cell sizes of 70km. Thus transmitting signals in the free space means that every burst should have a guard period of 0.233ms for the round trip delay in order to prevent adjacent received bursts having their useful parts overlapped. Comparing this guard period with the 8.25 bit duration guard period assigned by the GSM system for the normal bursts means that the transmission over the GSM radio link will be useless unless some measures are taken. The GSM system has approached this problem by instructing the mobile station to start transmission in advance in order for the bursts to arrive at the base station at the specified time [53,54]. The time advance is calculated by the base station after a successful reception of the access burst that was sent by a mobile station. The initial calculated time advance is sent to the mobile station prior to its start of transmission and thereafter whenever it is required. The time advance is thus a distance dependent parameter. A new time advance is always signalled to the mobile station whenever there is a change in the transmission delay as a result of the change in the distance separating the mobile station from the base station of more than one bit period. The delay is detected by the base station which continuously monitors the signal from the mobile station. The new time advance forces the mobile station to either advance or delay its transmission. Access burst in the GSM system needs a very long guard period because the mobile station does not known the time advance when it starts transmission of its access burst as illustrated in figure (3.1).

4.6 Associated Signalling Transmission over The GSM System

Once the mobile station has successfully accessed the base station, the base station assigns a time-slot which is exclusively used by the mobile station for transmission and reception of information during the whole duration of the call. After equal intervals, the mobile station also transmits signalling information over the slow associated control channel (SACCH) mapped onto the same physical channel. It is the transmission of the SACCH over the radio path that provides the GSM system with the out-of-band signalling capability that is unavailable with it predecessor [81]. To achieve this purpose, the GSM has adopted the 26 TDMA frames based multi-frame as shown in figure 4.5. This type of multi-frame is structured by grouping together 26 TDMA frames. The resulting multi-frame has a duration of 120ms and is used to support traffic channels and their associated control channel. In the GSM system, the mobile station uses the same time-slot number on the up-link TDMA frame and down-link TDMA frame for transmission and reception of information respectively. In order to assist the mobile station implementing this procedure and at the same time avoiding the need for parallel processing at the transceiver stage, the start of the up-link frame is delayed by a fixed period of 3 time-slots from the start of the down-link frame [53,54]. The TDMA frame structure adopted, coupled with this frame transmission and reception algorithm means that a mobile station will be active in two time-slots per

TDMA frame. The TDMA frame is a module of 8 time-slots and thus the two time-slots period corresponds only to 25% of the frame time in which the mobile station is actually engaged in transmission or reception. In the free time-slots the mobile station enters a different mode known as the monitoring mode. Figure 4.6 shows how the mobile station switches between the transmission, reception and monitor modes.

During the free time-slots, the mobile station switches on the BCCHs of the base stations in its vicinity, including the BCCHs of the serving base station, to have their received signal strength measured. In addition to the received signal strength, the mobile station also measures the signal quality of its serving base station. In order for the mobile station to carry out such measurements effectively, the BCCH carriers should be continuously switched on. Furthermore, the measurements are sequentially carried out and then averaged and transmitted once in 480ms over the SACCH to the serving base station. A 480ms period corresponds to 104 TDMA frames duration. During this interval the mobile station would have been active in 100 TDMA frames and idle in the remaining TDMA frames. It is in these active TDMA frames that the mobile station carries out the signal measurements. Since the mobile station makes the measurements, it is therefore necessary for the mobile station to identify which of the base stations is being measured. Basically, a mobile station would recognise an individual base station within a cluster by its unique BCCH carrier frequency. With small cluster sizes, the reuse distance is small and consequently it is possible that two surrounding cells being measured are using the same frequency. The GSM system has approached this problem by adding a new form of base station identification known as base station identify code (BSIC). The BSIC is broadcast by the base station on the BCCH carrier.

As can be seen from the figure 4.5, only 24 out of 26 TDMA frames are used to support the traffic channel. Of the remaining two frames, frame number 12 (start numbering from 0) is used to carry the slow associated control channel (SACCH) information while frame number 25 is kept idle. There will be four idle TDMA frames in the 480ms period. During these four idle TDMA frames, the mobile station synchronises to the synchronisation channel of the BCCH carriers in its vicinity. Once the mobile station has synchronised to the synchronisation channel, it is then be able to read the BSIC belonging to that particular base station. Once it has determined the identity of the surrounding base stations being measured, the mobile station then reports the averaged values of the signal strength along with the identification information over the SACCH.

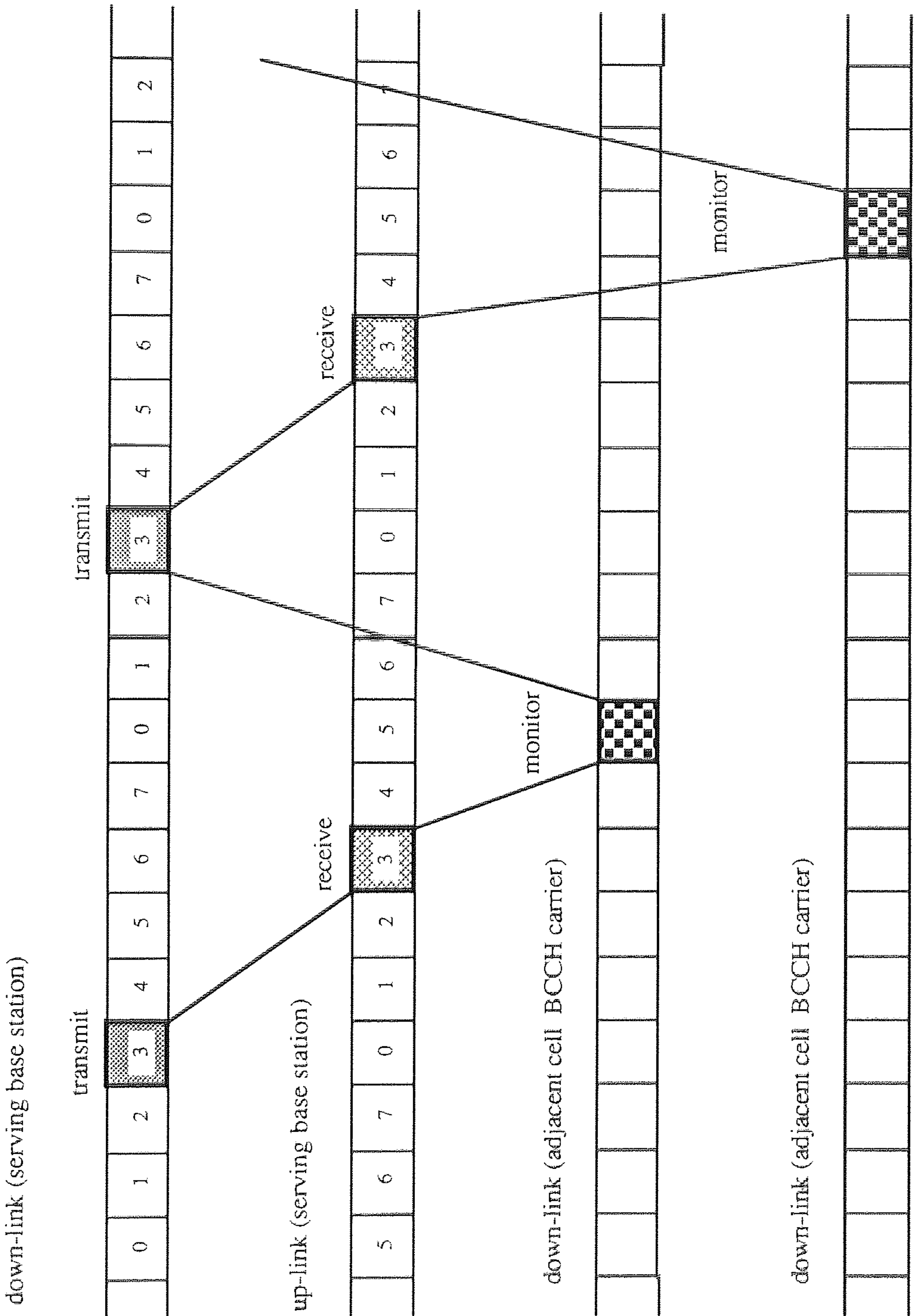


Figure 4.6 Mobile station switching between the modes

4.7 Simulation Of The GSM MAC Protocol

The GSM MAC protocol, which defines the way how several mobile stations share common transmission facilities provided by the network based on the TDMA mechanism, was simulated on the computer using the special oriented package known as SLAM. Prior to the presentation of the GSM MAC simulation results and discussions, it is worthwhile outlining some of the parameters and their definition that were used during the simulation of the all protocols in this research. The parameters are as follows:

- ***access protocol*** during the idle state, the mobile station keeps monitoring the paging channel to be ready to receive any paging messages that are addressed to it. As soon as it wishes to gain an access to the wireless medium (time-slot) to transmit its information, whether it is speech or data, it does so by following the GSM access protocol based on the slotted ALOHA (S-ALOHA) access protocol. The GSM access protocol is presented in more detail in chapter three.

- ***burst and frame structure*s** the information is transmitted on the same burst structure defined by the GSM system. As explained above, the GSM burst contains 156.25 bit duration and has a length of 577μs. By normalising one simulation time to 3.33μs real time, every burst is equivalent to approximately 173.3 simulation time periods. In addition, every eight time-slots are then grouped together to constitute a TDMA frame which, in turn, has a duration of 1386.4 simulation time periods.

- ***transmission rate*** the bit transmission rate adopted by this simulation is the 270.88kb/s transmission rate that is defined by the GSM system.

- ***number of traffic channels*** in the GSM system, every base station is typically allocated a number of two to three RF carriers. With every RF carrier supporting eight traffic channels, there are therefore between sixteen to twenty four traffic channels available to every base station. In this simulation twenty four traffic channels are considered available to individual base stations for information transmission.

- ***overall simulation time*** the system has been simulated for a period of 5 seconds real time which is equivalent to 1.5×10^6 simulation time periods.

- ***Channel throughput*** channel throughput (channel capacity) is defined as the channel time that it has been used for good transmission. In this simulation, throughput (S) is given by

$$S = (P \times K)/H$$

  where P is the packet period time, K is the number of successfully delivered packets and H is the total simulation time.

- ***mobile stations spatial distribution*** all the mobile stations are assumed to be uniformly distributed all over the radio cell area.

Two versions of the GSM MAC protocol were simulated and examined:

(i) One GSM MAC protocol was simulated by assuming the ***maximum multiplexing case.*** The maximum multiplexing case considered in this simulation means that once the mobile station succeeds in transmitting its channel request message, by following the GSM access scheme, the mobile station continuously transmits bursts of information in every time-slot that has been assigned to it for the whole duration of the simulation period. With this strategy there will be no silence gaps considered during which virtually nothing is transmitted and the only time wasted for no burst transmission is the period of the time from the time of initiation of the call to the time when the first speech burst of the first talkspurt in the call is ready for transmission. This elapsed time of no transmission taking place includes ***access delay time*** defined as the period of time between the initial attempt of transmitting the channel request message and the time of successful reception of the channel assignment message. It also includes the processing time of the speech encoder. Thus this ideal multiplexing mechanism represents the upper spectrum efficiency bound of the GSM system that the proposed protocol is aiming to achieve. It also allows for the comparison with the actual GSM MAC protocol in order to evaluate, for example, the wastage in time for the spectrum efficiency during the silence gaps in the speech transmission.

(ii) The actual GSM MAC has also been simulated. The mobile station once again performs the GSM access procedure to gain access to the physical channels. In this case it has been assumed that the mobile stations will only be involved in transmission of speech information. As a result, speech bursts will only be transmitted during the talkspurts and nothing is transmitted during the silence periods.

Figure 4.7 illustrates the ideal channel throughput for the individual physical channels (time-slots). The term ideal used here represents the case where no time delay is introduced apart from the contention delay used by the mobile stations to access the traffic channels. It represents the channel throughput when the channel is used for transmitting both information and control packets. Figure 4.8, on the other hand, shows the channel throughput when the channel is used for transmitting the information packets only and shows clearly the channel time that has been used for transmitting control packets.

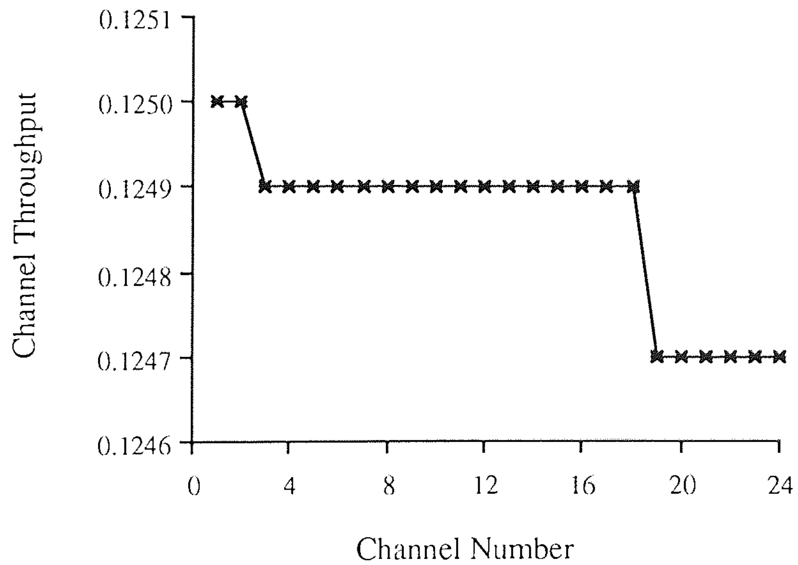


Figure 4.7 Ideal channel throughput

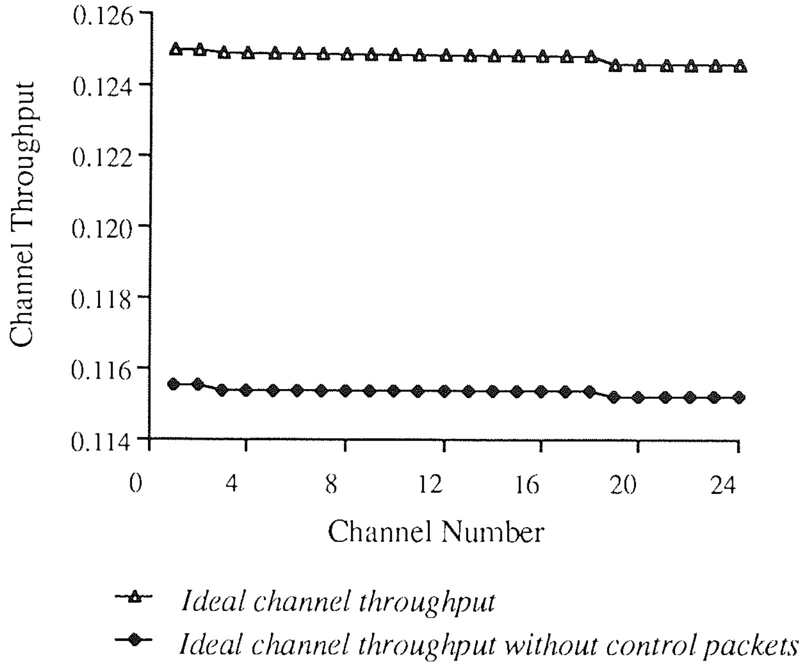


Figure 4.8 Ideal channel throughput with and without control packets

Figure 4.9 illustrates the channel throughput for the maximum multiplexing case described in paragraph (i) above along with a re-plotting of the ideal channel throughput case for comparison purposes. It represents the case where the channel is used for transmitting information as well as control packets. On the other hand, figure 4.10 represents the channel throughput when the channel is used for information transmission only. Both figures represent the ideal and maximum multiplexing cases.

The difference in the channel throughput is due to the delay introduced by the GSM speech coding scheme. Both figures show that in the maximum multiplexing case, all the channels support an equal number of packets. This is explained by the fact that the time delay due to contention is very small compared with the time introduced by the speech coding scheme. This time is useful for the portable stations where the off-air call set-up (OACSU) procedure is implemented [86]. Furthermore, the maximum multiplexing case in figure 4.10 actually represents the maximum channel throughput that the proposed protocol is aiming to achieve.

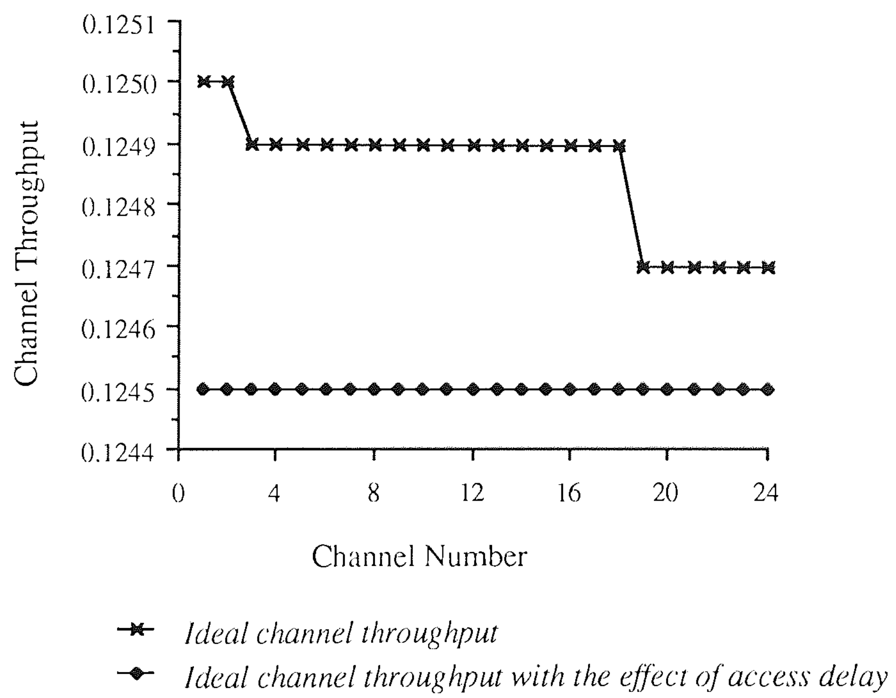


Figure 4.9 Ideal channel throughput with control packets and ideal channel throughput with control packets and with the effect of the GSM access delay

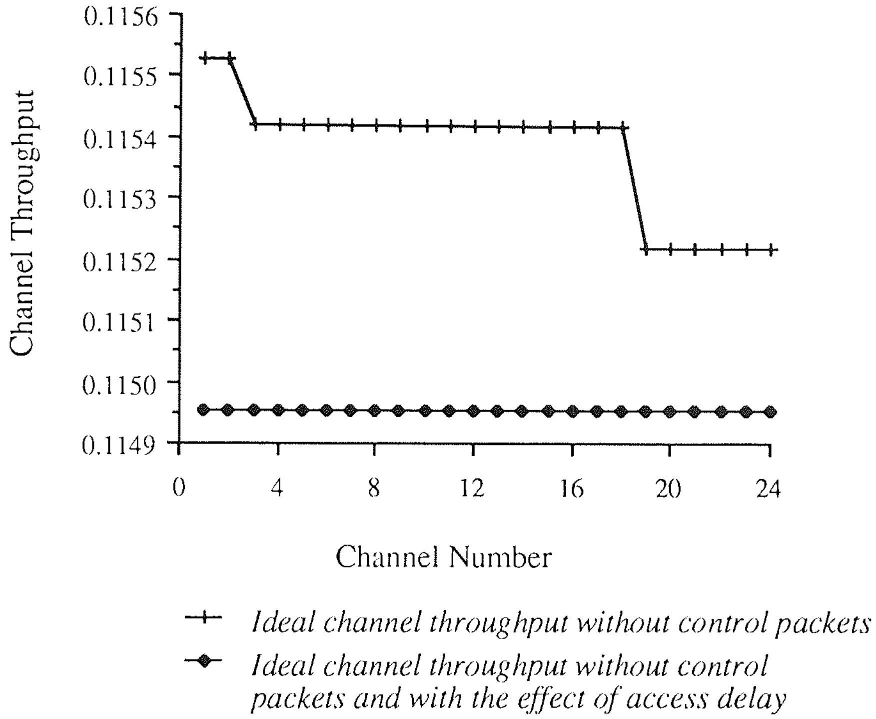


Figure 4.10 Ideal channel throughput without control packets and ideal channel throughput without control packets and with the effect of the access delay

Figure 4.11 shows the channel throughput for the individual time-slots when they are used for speech packet transmission only. The figure clearly shows the random number of speech packets which result from the random number of the talkspurts supported by the individual traffic channels.

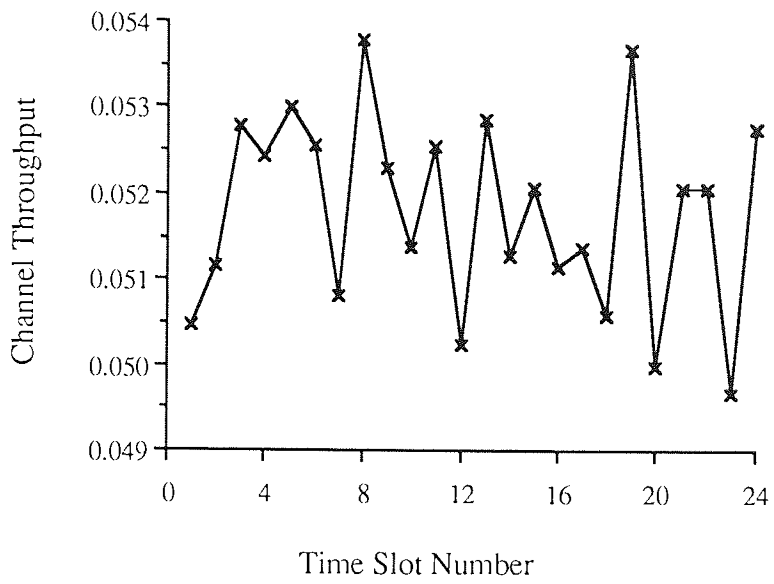


Figure 4.11 Channel throughput for speech transmission

4.8 Description Of The Proposed MAC Protocol

The GSM system has adopted a fixed assignment scheme based on TDMA as a procedure for the wireless multiple access. In this classical TDMA, the multiplexing is achieved at the call level, which means a physical channel (time-slot) is devoted to a source for the entire duration of the connection. The proposed protocol, however, attempts to increase the GSM spectrum efficiency by allocating the time-slots to the speech terminals only during their activity, which means multiplexing at the talkspurts level. It is, therefore, offering an increase of the GSM system spectrum efficiency by inserting data packets during the silence periods. The proposed protocol achieves the multiplexing of the voice terminals and the data terminals on the same time-slot by using a ***controlled collision mechanism.*** The controlled collision mechanism is made possible by the fact that the voice terminal and the data terminals both exist within the coverage area of the same base station.

Transmission of speech packets and data packets have different transmission requirements. Digitised speech require effectively prompt delivery but can tolerate relatively high bit-rates. On the other hand, data services require accurate transmission but can tolerate relatively lengthy delays. This implies that the developed protocol should acknowledge these transmission requirements when accommodating both types of data on the same time-slot. The developed protocol fulfils these requirements by granting the speech terminals a preemptive priority over that data terminals which results in no additional delay being introduced into the speech service apart from the delay which has already been defined by the GSM system. On the other hand, the data packets will encounter a very small probability of collision due to the jamming packets of the speech terminals.

The GSM system has been defined to support data transmission for data terminals having data rates from less than or equal to 2.4kb/s up to data rates of 9.6kb/s. All these data rate will end up with a data frame and a burst structure similar to a speech frame and speech burst structure consisting of 456 bits and 156.25 bit duration respectively.

In this research it has been assumed that data terminals that are to be multiplexed onto the same physical channel with a speech terminal should have their data packets buffered prior to transmission and be prepared to give up the channel as soon as the speech terminal needs it. These type of data terminals are referred to as data terminals group B. Data terminals which require to have a control over the time-slot are referred to as data terminals group A and will be treated as speech terminals. Accordingly, the data terminals group B will be offered a time-slot every time a speech terminal terminates its call or enters a silence period.

In the case when a voice terminal simultaneously shares the time-slot with a data terminal group B and the voice terminates the call, the base station broadcasts in an explicit form the availability of a free time-slot intended for speech terminals or data terminals group A. Furthermore, the base station allows the data terminal B which is already using the channel to continue using it as long as it has data packets to transmit and still exit within the cell boundaries under the control of the same base station. When the time-slot is assigned to a new voice terminal or data terminal A, the present occupier of data terminal group B will be paged of the new assignment in order to reschedule or abort its transmission and to request a new channel by performing a random access procedure when there is a time-slot available as broadcast by the base station.

Due to the interleaving process, the speech terminals will store the speech packets that are ready for transmission. Speech and data multiplexing can then be achieved in the following way. The speech terminal should inform the base station of the last speech packet in the current talkspurt. If the last speech packet in the current talkspurt happened, for example, in the (L) TDMA frame number, then the speech terminal sends a special packet on the (L+1) TDMA frame number. Optionally, the special packet may contain parameters that can be used by the receiver to generate "comfort noise". As soon as the base station detects the special packet, which is an indication of the speech terminal entering the silence period, the base station signals to the data terminals of group B the availability of a free time-slot for data transmission. In this case, a choice of two disciplines can be implemented by the base station to inform the data terminal of group B of the availability of the free time-slot:

(i) Channel request messages from data terminals group B, which are accurately detected by the base station, are queued at the base station in a buffer according to their arrival times. A particular data terminal will then be notified that its channel request message has been accepted and that it has been queued for the future available free time-slots. Once the data terminal receives this paging message it keeps monitoring the paging channel for any paging messages that are addressed to it. Although this strategy gives the first come first served (FCFS) discipline it suffers from the fact that the base station does not know whether the first data terminal in the queue still exists in the service area that is under its control or not (i.e. the mobile station has not crossed to another radio cell boundary) at the time of the availability of the free time-slot. The base station can solve this problem by periodically checking the first M number of data terminals in the queue. The checking is achieved by paging the M number of data terminals and deletes any data terminal that does not respond to the paging message. Alternatively, the queued data terminals send a duplicate of the channel request message at certain periods of time to update their request. The base station will preserve the initial channel request message for any duplicate message that has been sent to maintain the FCFS discipline and deletes any channel request message that has not been updated after a certain time has expired. The period of time that the data terminals should update their channel request messages is a system parameter which is broadcast to the data terminals on the down-link.

(ii) The base station broadcasts in an explicit form the availability of a free time-slot for the data terminals group B every time a speech terminal enters a silence period. The data terminals group B then contend for this free time-slot by sending their channel request messages on the random access channel (RACH). The simulation has shown that the access delay needed by the terminals to access the assigned time-slots is in the order of few micro-seconds. This is due to the fact that the random access channel is used for accessing a very limited number of traffic channels. As a result, there is sufficient time for the data terminal that has successfully delivered its channel request message and the base station to exchange signalling and other control information, such as the time-slot number, time advance parameter and transmitting power level, before the transmission is due in the (L+2) TDMA frame. Optionally, the data terminal might be required to transmit a control packet in the time-slot of the first TDMA frame that has been assigned to it (i.e. (L+1) TDMA frame) and in every time-slot after the speech terminal enters a silence period thereafter. This enables the base station to estimate the radio link quality in terms of bit error rate (BER) as well as confirming other parameters such as the identity of the data terminal and time advance. It is the transmission of the training sequence which allows for the estimation of the radio link quality. In this research the strategy has been adopted in which the base station broadcasts on the BCCH the necessary information to inform the data terminals group B of the availability of the free time-slots as a result of speech terminals entering the silence periods.

During the silence periods, there must be some signalling information exchange between the active mobile station transmitting speech packets and the base station. In the multi-frame structure of the GSM system, TDMA frame 12 starting from TDMA frame 0 has been allocated for sending the SACCH messages and frame number 25 is kept idle. The proposed protocol has to coordinate between the speech terminal that has entered the silence period and the data terminal which is currently occupying the time-slot. The proposed protocol approaches this problem by informing the data terminal to stop transmission in two TDMA frames in every multi-frame as long as the speech terminal stays in the silence period. The TDMA frame number 11 will be used by the speech terminal to send a control packet, having a length similar to the length of the GSM access burst, while TDMA frame number 12 will be used for the normal SACCH message. The purpose of the control packet is for frame re-alignment as well as the other purposes explained above. On the other hand, the data terminal will use the TDMA frame number 25 to transmit its SACCH message and enters idle periods in the TDMA frame number 12. Moreover, in the present structure of the GSM system, the SACCH on the down-link is used to convey useful control information such as changes in the power control and changes in the time advance. In the proposed protocol, however, it is assumed that the function of the SACCH on the down-link remains unchanged as specified by the GSM system for the mobile station that currently occupies the channel. If the speech terminal is currently involved in speech transmission, then the information conveyed by the down-link SACCH will be useful to the speech terminal and ignored by the data terminal and vice versa.

When the speech terminal changes into a talking mode from a silence mode (i.e. starts the next talkspurt which is detected by the speech activity detector), the speech terminal re-acquires the time-slot by firstly sending a jamming packet, for example, in the TDMA frame number (N-2) prior to the readiness of its speech packet in the TDMA frame number (N). The jamming packet should have a length similar to the length of the GSM access burst and carry no useful information. This is because the speech terminal does not known the new time advance which it must use for its normal burst transmission. The sending of this jamming packet is termed the ***controlled collision mechanism*** . The collision will be detected by the base station which then informs the data terminal to abort its transmission. The data terminal then reschedules the transmission of the same data packet that has been involved in the collision at a later stage. In the (N-1) TDMA frame, the speech terminal sends a control packet that has a similar length to the GSM access burst. The transmission of the control packet in the (N-1) TDMA frame enables the base station to calculate the new time advance, assess the radio link quality and calculate the required transmitting power level. This information is then signalled to the speech terminal before it starts its speech transmission in the (N) TDMA frame. Since the speech transmission commences in the (N) TDMA frame, the proposed MAC protocol with the controlled collision mechanism introduces no additional delay as far as the speech transmission is concerned. Furthermore, the only delay suffered by the speech transmission is the delay that has already been defined by the GSM system, which includes the delay due to the speech coding scheme and the delay due to the interleaving process. No speech degradation occurs as a result of inserting a data packet during the silence periods due to the fact that the speech packet is not involved in any collision.

4.9 Simulation Of The Proposed MAC Protocol

The proposed protocol has been simulated using the same computer package and applying the same parameters that are described in section 4.7.

Figure 4.12 represents a re-plotting of the relationships of figures 4.11 and the maximum multiplexing case of figure 4.10 to show the amount of the channel time that is wasted due to the silence periods occurring in the speech.

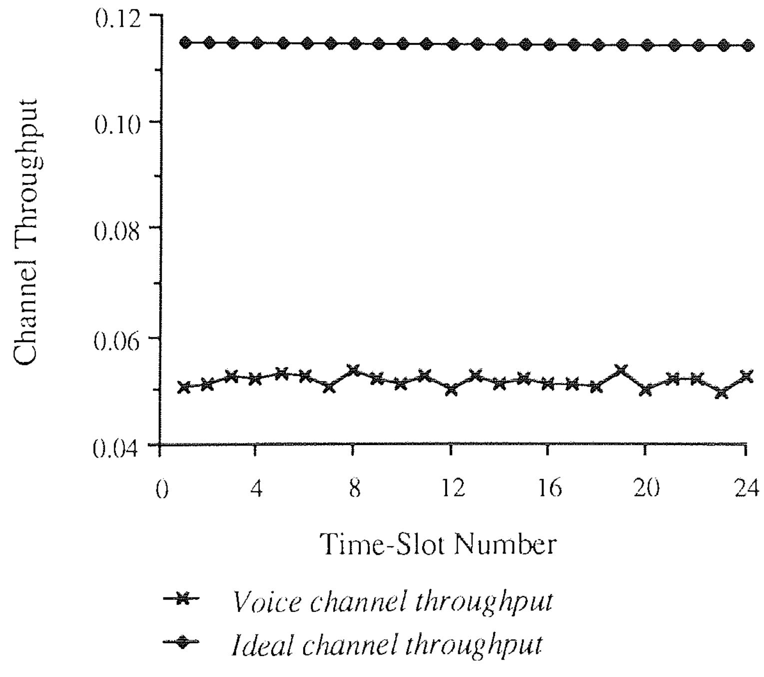


Figure 4.12 Channel throughput for speech packets and ideal channel throughput without control packets

Figure 4.13 illustrates the channel throughput for multiplexing speech and data packets on the same physical channels along with plots in figure 4.12 for the comparison purposes. Figure 4.13 clearly shows the gain made by the proposed protocol. The gain of the proposed protocol is made possible by the insertion of the data packets during the silence periods of the speech.

It is common for MAC protocols that use contention as a procedure for transmitting data packets to drop speech packets that suffer excessive access delay [83]. Dropping speech packets at the start of the talkspurt is known as 'front-speech clipping'. Front-speech clipping has been found to be less objectionable than loosing packets during talkspurts [84].

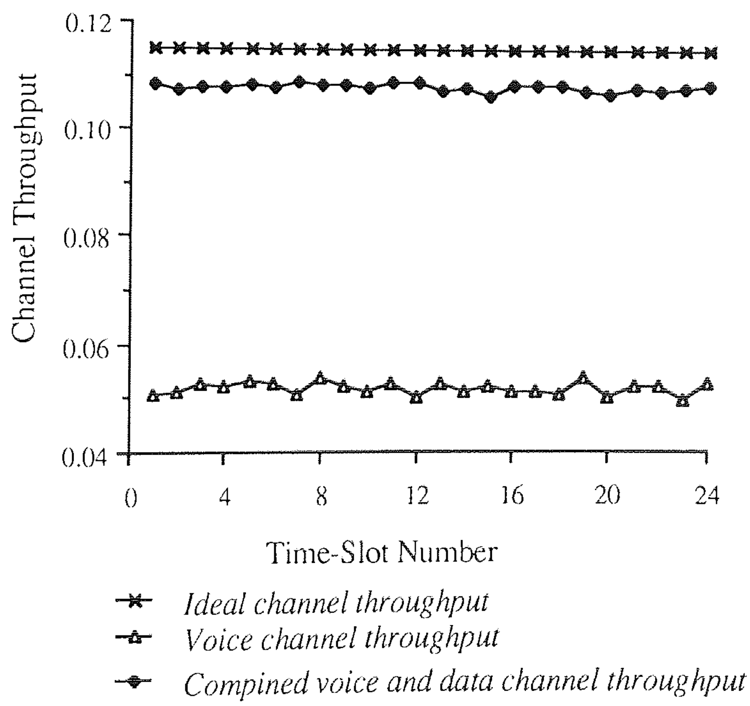


Figure 4.13 Channel throughput for speech and data transmission

In the GSM MAC protocol, the advantage of the interleaving process is applicable from the fourth packet at the start of any talkspurt and upwards. Dropping the packets that are not included in the interleaving process (i.e. at the beginning of talkspurts), will certainly improve data transmission. Dropping speech packets at the beginning of the talkspurts account for 10ms of speech loss from every talkspurt.

Figure 4.14 illustrates the advantage in channel throughput for transmitting data when the dropping of the speech packets at the start of talkspurts is considered.

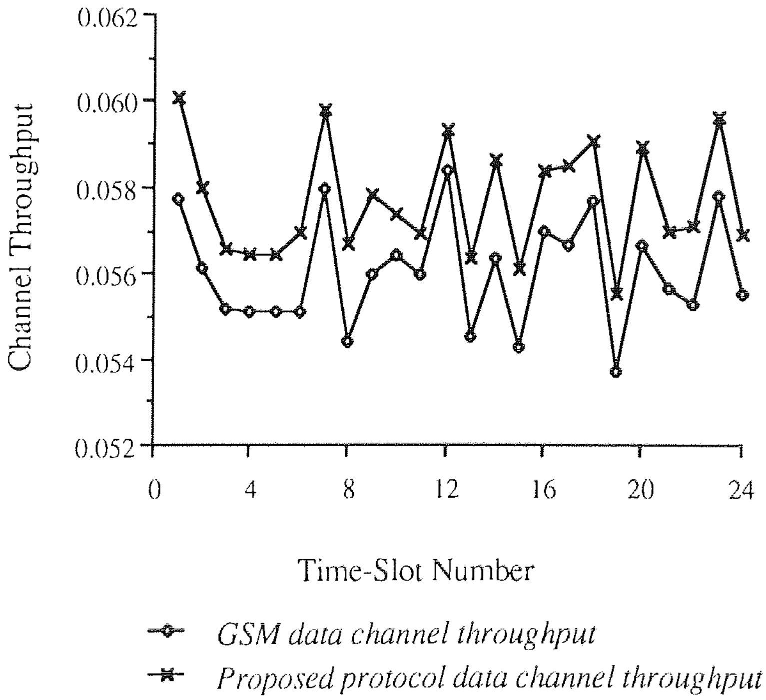


Figure 4.14 Channel throughput for the GSM data transmission and the channel throughput of the proposed protocol as a result of the front speech clipping

During this simulation, it has been found that the total loss of the speech packets due to dropping of speech packets is given by:

$$\begin{aligned}\text{total loss of speech packet} &= (\text{total dropped packets})/(\text{total transmitted packets}) \\ &= (348)/(10765) = 3.2\%\end{aligned}$$

the 3.2% packet loss is less than the 5% speech loss which has been found acceptable without significantly affecting the quality [85]. Moreover, the transmission of the jamming packets, in order for the speech terminal to reacquire the channel, introduces a collision with the data packets that have to be retransmitted at a later stage. In this simulation, the value of the data collision probability is calculated by dividing the number of data packets which collided with the jamming packets by the total number of data packets that have been successfully delivered.

$$\text{Data collision probability} = (87)/(11647) = 0.75\%$$

4.10 Summary

The GSM system is more spectrum efficient compared to its predecessor. This is due to the incorporation of digital technology which, in turn, allows for the deployment of smaller radio cell sizes. Further increase in the GSM spectrum efficiency is achieved through the proposed algorithm. The algorithm relies on the insertion of data packets during the silence periods inherent in the speech transmission.

CHAPTER FIVE

INTERWORKING OF THE GSM SYSTEM WITH B-ISDN

5.1 Introduction

The GSM system, which is a cellular radio system, relies on frequency reuse technology to increase its capacity. By adopting digital technology, the GSM system is able to operate with a much lower carrier-to-interface (C/I) ratio compared with its predecessor, which relies on analogue technology. Operating at lower C/I means the system is more robust to co-channel interference. This feature, in turn, allows the GSM system to use smaller cell sizes in the heavily populated areas where the demand on the mobile service is at its peak. In urban areas, such as London, radio service areas of around 1-2 kilometres have already been adopted [88,89]. Such small radio service areas leads to more frequent boundary crossing by the roaming mobile stations. Consequently, the volume of signalling exchanges increases and hence the signal processing on the Mobile Switching Centres (MSCs) increases. Despite the fact that the GSM system has partially relieved the MSC of some signalling tasks compared with its predecessor, where the MSC takes all the decisions related to signalling control, it is still a centralised system as the MSCs still control most of the channel selection and hand-over decisions.

In parallel with the deployment of the second generation of PLMNs, fixed networks are also evolving towards a Broadband Integrated Service Digital Networks (B-ISDN). This evolution has been made possible and feasible by the realisation of single-fabric networks which accommodate the broadband services as a result of recent advances in computing, micro-electronics, optical fibre and other technologies. Asynchronous Transfer Mode (ATM) has already been defined by the International Telegraph and Telephone Consultative Committee (CCITT) as the multiplexing and routing protocol for these B-ISDNs [90,91,92]. The evolution of the fixed networks towards these ultimate B-ISDNs might consist of several evolutionary stages driven by parameters such as current service demands and progress in standards development [93]. However, by the time the B-ISDNs enter the deployment phase, which is expected by the late nineties, the second generation of PLMNs would have been in a widespread use.

Metropolitan Area Networks (MANs) seem to be the most likely solution prior to implementation of the full ATM based networks [94]. Distributed Queue Dual Bus (DQDB) has already been standardised as the IEEE 802.6 transporting mechanism for use in MANs [23]. DQDB permits a mixture of isochronous (fixed bandwidth for voice and video traffic) and nonisochronous (data traffic) services. The cell format and the cell size of the MAN has been chosen, apart from a small difference in the header format, to coincide with the ATM cell format. The compatibility of the cell formats of the 802.6 and ATM simplifies the interworking between these systems.

A key success in the introduction of the broadband services, whether these services are provided by ATM-based networks or MANs, is the rapid and widespread offering of these new services to the users. This implies that interface units, among other functions such as signalling, should be evolved and developed to interface B-ISDNs with the existing networks whether these networks are public or private, mobile or fixed, as well as with the future anticipated Universal Mobile Telecommunication System (UMTS) and Future Public Land Mobile Telecommunication Systems (FPLMTS). Gallagher at el [95] have suggested that there will be some advantages to the mobile radio networks if these networks are interconnected with the B-ISDN. Interworking PLMNs with B-ISDNs offers the PLMNs an access to the heterogeneous services provided by the B-ISDNs. Fast routing and fast exchanging of information is another area where the PLMNs could benefit from interworking with the B-ISDNs. As a result, it is essential that the MAN/ATM based networks are planned to interwork with the PLMNs.

Logically, MANs will firstly be introduced to areas with a high density of broadband communications demand. GSM, on the other hand, has responded to demand for mobile service in such areas by adopting very small cell sizes. Reducing cell sizes increases the number of required base stations and subsequently the number of MSCs to handle such high volumes of mobile traffic. Currently the GSM system, like any other existing PLMN, relies upon leased private circuits with fixed capacity for interconnection, as shown in figure 5.1. Interconnecting the GSM entities over public MAN(s) will change this by allowing the GSM to dynamically share the communication resources offered by MAN(s) with other networks. Dynamic sharing of the communication resources by more than one user over a MAN will be offered by exploiting virtual path and virtual circuit technology. By careful management of the high speed MAN link, the congestion of the fixed links that currently link the GSM entities, which result from the deployment of reduced cell sizes, might be prevented. Furthermore, interconnecting the GSM entities over the MAN offers the GSM fast routing and exchange of information between different MSCs and between MSCs and Home Location Register(s) (HLR).

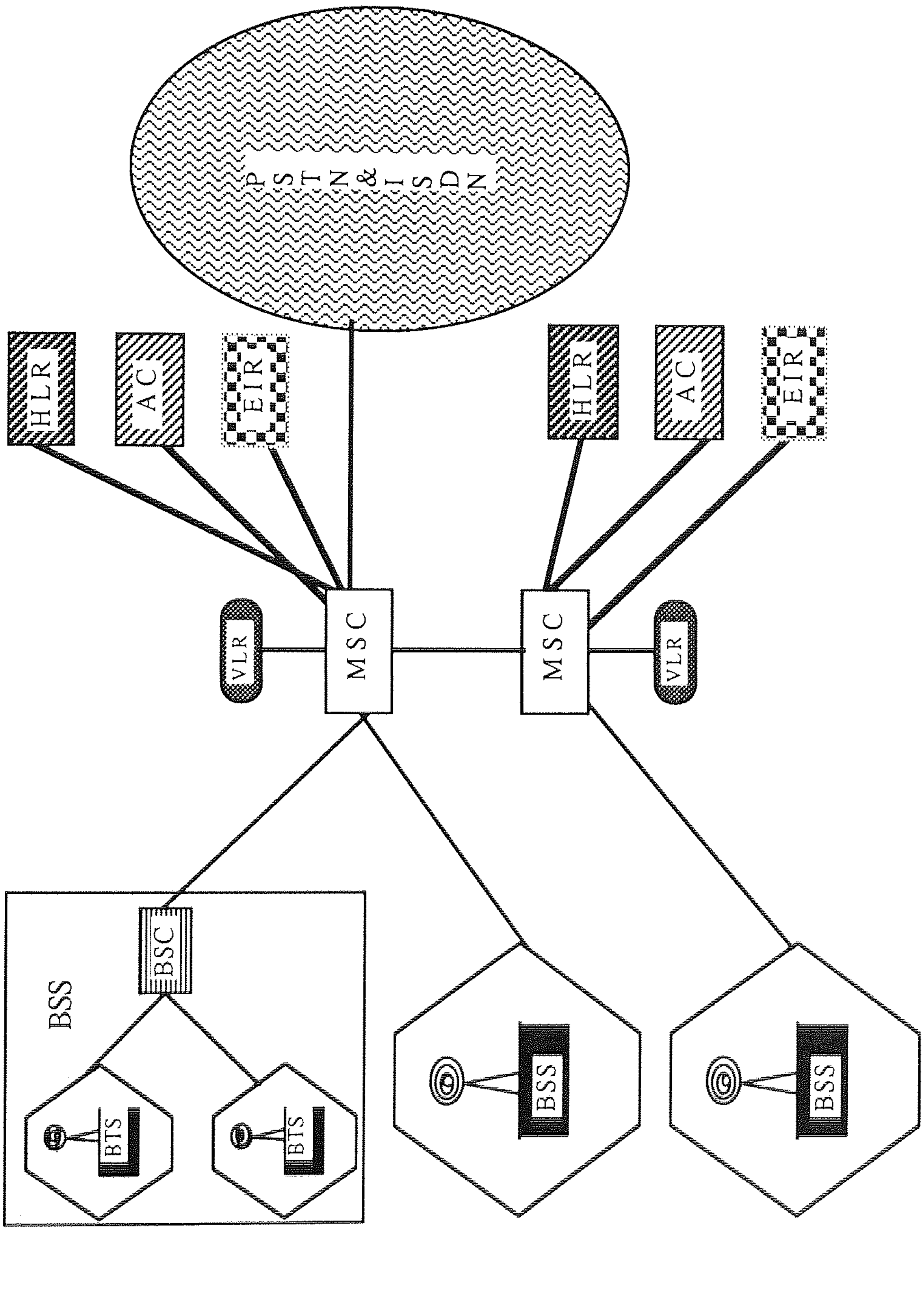


Figure 5.1 GSM Configuration

5.2 Interconnecting The Local GSM MSCs Over Public MAN

In this research a possible two-phase interconnection of GSM entities over a MAN is suggested. The first phase deals with the possibility of interconnecting local MSCs and the second phase deals with the interconnection of the Base Station Subsystems (BSSs) of the GSM system by public MANs based on the proposed IEEE 802.6 standard. The BSSs comprise Base Transceiver Stations (BTSs) and Base Station Controllers (BSCs). Figure 5.2 illustrates the new proposed MAN architecture where the GSM system interfaces with the MAN at the MSC level. The GSM gains access to the MAN by using new MSC interface units. The GSM already includes several interface units which interfaces it with the fixed networks such as the Integrated Service Digital Network (ISDN), the Public Switching Telephone Networks (PSTN) and Packet and Circuit Switched Public Data Networks (PSPDN & CSPDN). The new interface units, which might logically be separated from the MSC but physically integrated with it, terminate the GSM interface (ie. signalling and user channels) and provide the interworking functionality needed to interface with MANs. Current GSM signalling protocols, which are based on the Signalling System Number 7 (SSN7), might be further developed and then used to initiate the establishment of the connection over the MAN links. In particular, the interface unit acts upon the signalling requests to establish the virtual paths and virtual circuits. The interworking functionality also include translation of information between the two formats involved. For routing and switching purposes, it is required that each interface unit connected to the MAN should have a unique identification number which might be based on the Integrated Broadband Communication Network (IBCN) numbering plan [96]. Packets destined for a particular interface unit should carry the unique identification number of that particular interface unit in their headers or, alternatively, should carry an equivalent virtual circuit connection number which is assigned to that interface unit at the call set up.

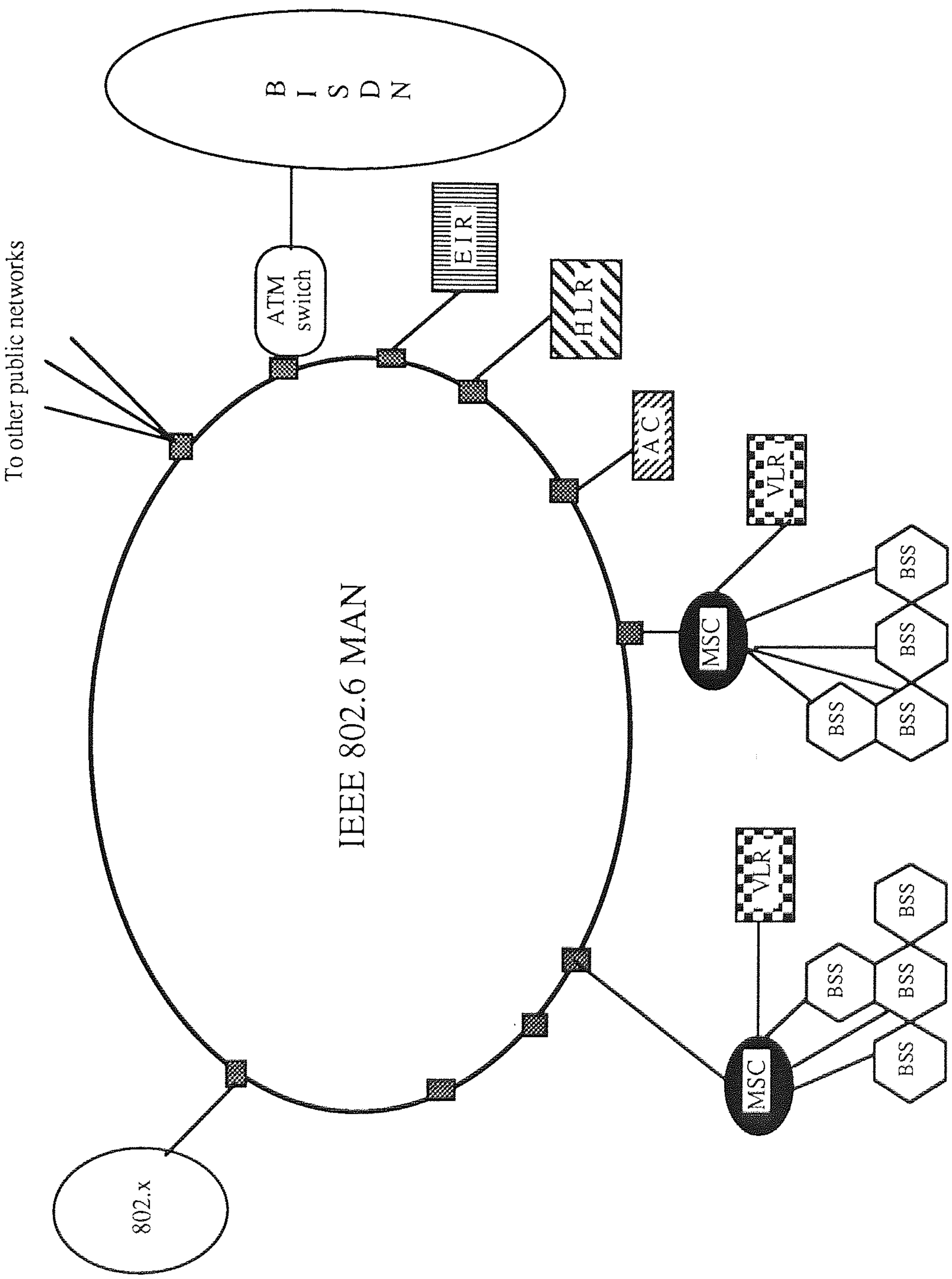


Figure 5.2 Interconnecting MSCs over public MAN

Signalling packets should always be given a priority over other packets. In the MAN environment based on the DQDB this task could be achieved by assigning different queues for different priorities [97,98]. Placing the signalling packets in the queue with the highest priority will ensure that these packets gain access to the MAN before any other packets with lower priorities. This feature is very useful, especially in the hand-over signalling exchanges. Sharing the MAN transmission facilities with other networks, such as private local area networks (LANs) and high speed terminals that access the MAN directly, might result in a situation where the volume of the traffic presented to the MAN is too high to handle. To ensure the delivery of the signalling as well as the isochronous connection packets in such a congestion situation, a number of virtual paths and virtual circuits could permanently be allocated to handle these services.

The provision of the personal mobile communication services anticipated for the future means working with very small radio cell sizes. Such services will have a significant impact on the B-ISDN. This is due to the number of signalling exchanges required for control functions such as location finding, location updating and hand-over. Databases such as the Home Location Register (HLR) are used to store static data related to all the mobile stations that are registered with it. The static data includes access capabilities, subscribed services and supplementary services. The HLR also holds dynamic information as far as the location areas are concerned. The Visitor Location Register (VLR) is another database that is used for storing user's related information. The VLR is, actually, a database that holds all the information about the mobile stations that are currently roaming in a geographical area under its control [99,100]. The information related to a particular mobile station will be kept as long as the mobile station stays in the area and deleted as soon as the mobile station leaves the area. The VLR is therefore considered as a dynamic subscribers database. In this research it has been assumed that one HLR will be used for each individual MAN. The HLR will have its unique interface unit number, which is used in any packets that are forwarded to it. To facilitate MANs with global connection, it is required to interwork these MANs with B-ISDNs based on ATM. The interworking of MANs with B-ISDNs based on ATM will be via ATM switches. Other interface units will be used to interwork MAN with other current networks such as the ISDN. In addition to being used as a means to gain access to the B-ISDNs, the ATM switch can be used to manage the bandwidth between MAN users by giving the ATM switch the task of generating the frames on the MAN. The management of the MAN bandwidth could be achieved through the allocation of virtual path and virtual circuit numbers as well as the assignment of Queue-Arbitrated (QA) and Pre-arbitrated (PA) time-slots. Each ATM switch will again have a unique interface number and all the packets destined to it will use this number in their headers. In this architecture, the MAN is not only used for interconnecting users that are attached to it but is also used as a concentrator of traffic to the B-ISDN node. In the following paragraphs, a location updating procedure and a mobile station call termination procedure are presented to illustrate the volume of signalling exchanges that take place in the GSM system to achieve such procedures.

5.3 GSM Mobile Station Location Registration (Tracking Procedure)

One of the main requirements to enable a mobile station to have greater roaming capabilities is the knowledge of the mobile station's location within the PLMN at any given time. The requirement for information on the mobile stations whereabouts is obvious in a situation where a fixed network subscriber initiates a call intended for a mobile network subscriber. In this case it is necessary for the PLMN to know in which of the base station zones the mobile station actually exists so that the call can be successfully routed to the mobile station. Alternatively, a call which is intended for a mobile subscriber has to be transmitted on every base station existing in the PLMN. The latter procedure is impractical and results in an overloading of the limited number of signalling channels and, consequently, the capacity of the system [101].

The GSM system deals with the mobile station location registration or mobile station tracking process by subdividing the geographical coverage area into a number of location areas. Each location area carries an unique identity code which is broadcast on every Broadcast Control Channel (BCCH) that exists in the location area [102]. It is the mobile station's responsibility to inform the fixed side of the PLMN of any change in the location area. The process of informing the system of any change in the location area is known as "location registration". The location registration is accomplished by triggering a location updating procedure. The location updating procedure is always activated while the mobile station is in an idle mode (ie. not involved in transmission or reception of voice or data services) [100,49]. The location updating procedure is used for normal location updating, period updating and International Mobile Subscriber Identification (IMSI) attach.

The normal location updating procedure is used whenever there is a mismatch between the location area identity code broadcast by the system and the one already stored in the mobile station memory. The periodic updating procedure, on the other hand, is invoked at regular intervals after the mobile station has switched on and the mobile station is still roaming in the same location area as the one stored in its memory. The purpose of periodic updating is to inform the system of the mobile station availability to the network. Finally, the location updating is also used when an IMSI attach is to be performed. The IMSI attach procedure is initiated if the IMSI was deactivated while the mobile station was in "idle updated" state and no change in the location area identity code had been detected. In all cases the mobile station should inform the system of why the updating procedure is invoked by using the location updating type information element in the location updating request message.

The mobile station decodes the location area identity code whenever it switches on and at regular periods thereafter. Once the mobile station has determined the location area identity code, it then compares the decoded identity code with the one already stored in its memory. If the two codes match each other, then there will be no action taken, otherwise the mobile station has to update the VLR by sending a layer 2 set asynchronous balanced mode (SABM) frame containing a location updating request message through the base station controller (BSC) [49].

In response to the layer 2 frame, the VLR records the received information and checks whether the mobile station already exists in a location area under its control. If the mobile station was already under its control, the VLR updates the relevant information and no further action is taken. If this is a new VLR, then the VLR assigns a new Mobile Station Roaming Number (MSRN). The MSRN will be used for forwarding the calls intended for this mobile station initiated by a fixed network subscriber. When the mobile station enters a new location area, it is necessary for the VLR database to update the HLR database with the new MSRN and with any other related information. In addition, when required, the VLR issues another number known as the Temporary Mobile Station Identity (TMSI) used for secret identification of mobile stations over the radio link. The replacement of an IMSI by a TMSI is to protect the user against being identified and located by an unauthorised person [49].

The HLR responds to the new MSRN by updating itself with the new information and then by sending two messages, one intended for the new VLR and the other for the previous VLR. The purpose of the message sent to the old VLR is to instruct the old

VLR to cancel the old MSRN. The message sent to the new VLR includes all the mobile subscriber parameters necessary for call control and operation of supplementary services. The old VLR responds to the cancellation message by sending a cancellation acknowledgement to the HLR. The reception of the subscriber related parameters by the new VLR is taken as an updating acknowledgement from the HLR. The new VLR then sends an updating acknowledgement to the MSC which passes it to the mobile station over the radio link [49].

5.4 Call Termination

Routing calls to a mobile station involves several connection stages. In this case the fixed network subscriber initiates the call by dialling a unique number. The unique number is a PSTN/ISDN number assigned by the fixed network as a permanent identification of the mobile station in the fixed network. There are several routing possibilities available. In one of the possibilities, it is assumed that the local exchange (or a transit exchange) of the PSTN/ISDN has the capability to interrogate the HLR. In the second possibility, the HLR will be interrogated by a gateway interconnecting the PSTN/ISDN with the GSM system.

As soon as the PSTN/ISDN number is received, the local exchange or the gateway uses this number to obtain the address for the mobile station's HLR. With this procedure, the advantage is that the fixed network subscriber always initiates the call to the mobile radio subscriber by dialling the PSTN/ISDN number regardless of the whereabouts of the mobile station at the time of the call initiation. After the mobile station's HLR address is known, the local exchange or the gateway interrogates the HLR to get the MSRN. Once the MSRN is obtained, it is used to route the call to the appropriate

MSC. Upon receiving the call, the MSC exchanges signalling with its VLR to get the mobile subscriber's data. The MSC now pages the mobile station through the base station that is serving it. Since there are several base stations controlled by the MSC and the exact location of the mobile station is not known, the MSC has no alternative but to page the mobile station through all the base stations that it is controlling. The call is then directed to the intended mobile station as soon as a response is received. If the mobile station is busy at the time of receiving the call, then the MSC has to return the busy signal [100].

5.5 BSS Interconnection over MAN

According to the present configuration of the GSM system, a base station subsystem could have a number of base transceiver stations that are connected to a base station controller, which controls and manages them. Alternatively, the base station subsystem could include an integrated base transceiver station and a base station controller. The type of configuration to be adopted will be influenced by the type of area to be covered, that is, whether the area is urban, suburban or rural.

Prior to the presentation of a proposed possible ultimate configuration for an integrated GSM system with B-ISDNs, it is worth mentioning that the future personal communication networks concept has also been taken into consideration. There are suggestions that future personal communication networks, which will provide their customers with ubiquitous communication coverage, will be based on the coexistence and cooperation of the different networks such as cordless telephones, telepoint, cellular systems, etc. [103,104,105]. This requires that the future personal communication networks will have flexible interfaces suitable for working with each system. A comprehensive study of such flexible interfaces has been given in the literature [103,104,105]. In Europe, ETSI has defined the Digital Cellular System (1800MHz) (DCS1800) as a standard for the future personal communication networks. The DCS1800 is based on the GSM900 standards. Before proceeding further, it is important to note that for this section only, the term GSM system will always refer to the GSM/DCS system.

As discussed earlier, one of the main problems of the second generation PLMNs, such as the GSM, is their inability to handle all the anticipated vast volume of signalling exchanges. This is due to the centralisation of the decision-making by the MSCs. In this research, a proposed solution to this problem has been considered through the implementation of the MAN concept. By interconnecting the GSM system entities, the processing and decision tasks can then be distributed among several decision making controllers.

Figure 5.3 shows the possible ultimate configuration, which might take several implementation stages to achieve. These implementation stages might be driven by factors such as cost of implementation and service demands. In this configuration, the mixed cell strategy has been adopted [106]. The mixed cell configuration means a hybrid configuration made up of microcells and macrocells. The microcells are used to serve slow-moving pedestrians; whilst the macrocells are intended for the fast-moving vehicle-mounted customers. This technique has its advantage in providing users with high capacity in densely populated areas. As can be seen from the configuration, LANs could be used to interconnect the individual BTSs with a number of the future microcell stations. Microcell stations should provide the coverage into buildings such as railway stations, streets and shopping malls. Interworking units will again be used to interconnect LANs that belong to the same BSC. Private LANs could also access these public LANs. Level 0 MANs (MANs-0) are used to interconnect a number of BTSs that belong to the same BSC as their controlling BSC. BTSs interface MAN-0 through the interface units (interworking units). In the future, it is anticipated that the majority of the traffic that is originated by the mobile stations will be directed towards the fixed networks or towards another PLMN, or vice-versa. Additionally, the traffic originated by mobile stations that is intended for other mobile stations within the same BSS will remain low. This implies that most of the signalling exchanges between BTSs, between BTSs and BSCs and between BSCs themselves that belong to the same MSC will be related to hand-over signalling. This is, in turn, created by the movement of the users between the cell boundaries.

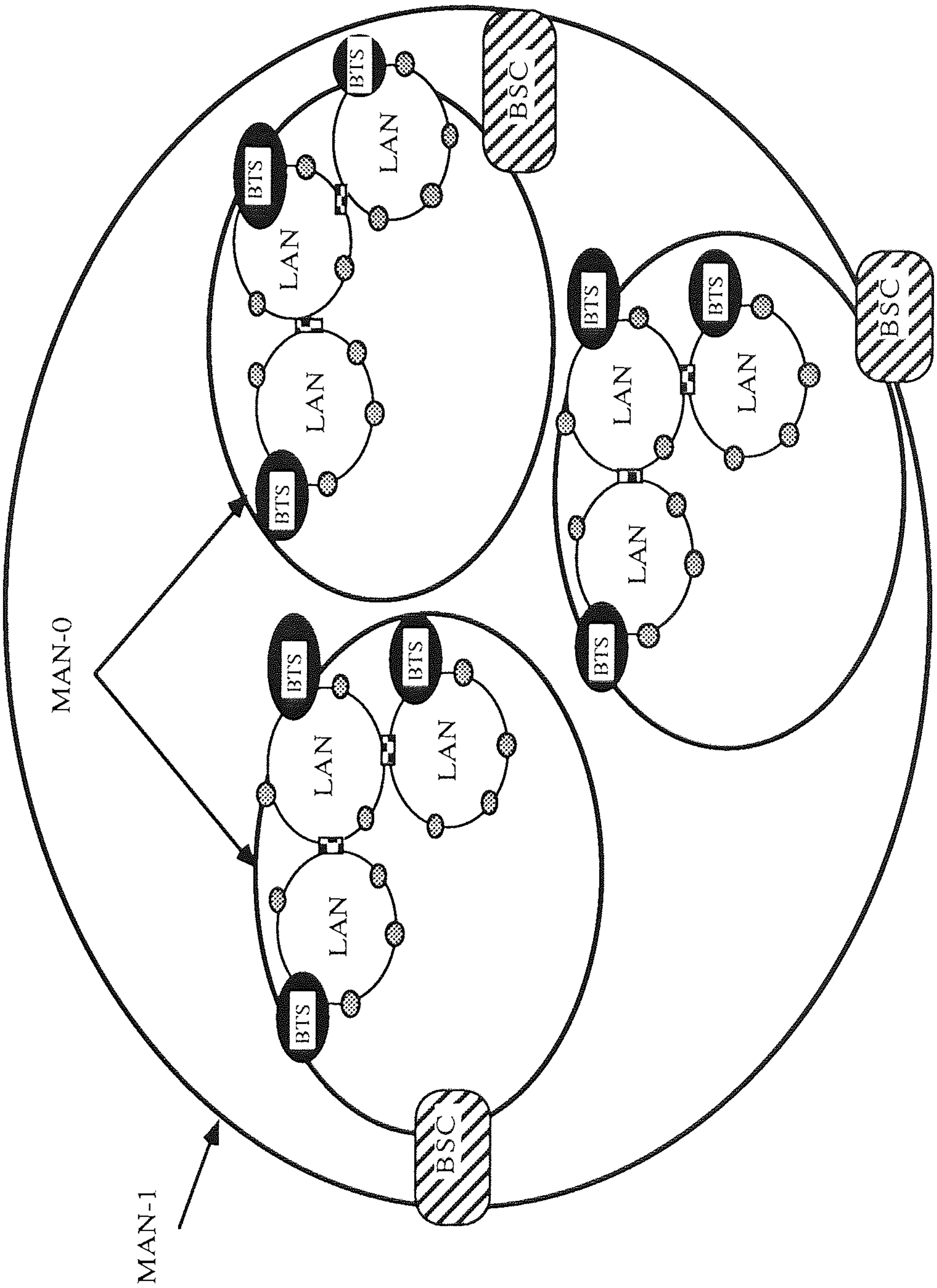


Figure 5.3 Ultimate configuration of interconnecting GSM entities over MANs

In order to relieve MAN-0 from such signalling traffic, interface units interconnecting the LANs of the same BSC are suggested. As a result, most of the traffic through these interface units will be related to the hand-over signalling exchanges between the BTSs (ie. bypassing MAN-0); whilst most of the traffic through the interface units at MAN-0 is intended for fixed or another PLMNs. As explained earlier, every interface unit, whether this unit is on LAN or MAN, will have its unique identification number for routing and switching the packets to it. MANs-1 are used to interconnect BSCs that belong to the same MSC as well as interconnecting them with the MSC and VLR that these BSCs belong to. The MSC also interfaces MAN-2 which interconnects it with other MSCs and the HLR. At this stage, it is worth mentioning that networks such as private LANs/MANs can interface these MANs at any level (levels 0-2) in order to share the MANs transmission facilities. Since the proposed structure interconnects the GSM entities that exist on the fixed side of the PLMN, this structure is independent of the radio interfaces. This advantage will enable the structure to support the systems with the future anticipated flexible interfaces. Prior to the presentation of the benefits of this MAN hierarchy in terms of decentralisation of the decision making task of the GSM system, it is appropriate to illustrate how the hand-over decisions are taken and executed under the current structure of the GSM system.

5.6 GSM hand-over Procedure

Because of the cellular concept, the mobile station's movement during a call period could result in a situation where the mobile station has crossed to another cell area or travelled too far away from the serving base station. In both cases the radio link performance falls below a prescribed threshold level such that an enhancement to the radio link could be achieved if the mobile station is transferred to a new radio channel supported by a neighbouring base station. The movement of the mobile station also implies that the radio link characteristics between the two parties involved (ie. the mobile station and the base station) might change so that the grade of service is no longer provided by the existing radio link. In some cases the radio link could possibly be improved by transferring the mobile station to another radio channel. In addition to these circumstances, the MSC sometimes has its own reasons for transferring a mobile station served by one base station to another base station. The reasons for the latter case could include parameters such as instantaneous traffic loads supported by individual radio cells. This, in turn, assists in fair distribution of traffic loads among various radio cells controlled by the system and prevents overload in the system.

The changing of the radio channels during a communication phase is known as the hand-over process. Sometimes the hand-over happens on the same base station, a phenomenon known as "intra-cell hand-over". When the hand-over involves different base stations then it is known as "inter-cell hand-over". It is this hand-over process which actually enables the system to provide continuation of calls despite the fact that the users are roaming between areas supported by different base stations [56].

When it is decided to implement a transfer, the mobile station will be instructed to tune to the new radio channel. The mobile station will be informed on the radio path of all the necessary information related to the new radio channel as explained in later sections. In addition, the GSM system has a new added dimension whereby the mobile station is also transferred to a new radio channel if the new radio channel can provide a better or equivalent link quality at the existing or a reduced power level. This new feature is known as the "power budget" method [49,3]. The adoption of the power budget method minimises the chance of co-channel interference, which has a direct impact on the system capacity as a whole [3].

The first generation of analogue cellular mobile radio systems use signal parameters measured on the up-link part (ie. mobile station-to-base station direction) by the base station for a hand-over procedure to be triggered. The signal parameters to be measured include the signal power, the estimated signal-to-noise ratio (S/N), or the estimated bit error rate (BER) in the signalling blocks. The first generation of cellular systems give a passive role to the mobile stations as far as the signal parameter measurements are concerned. In contrast, the GSM mobile radio system assigns a very active role to the mobile stations in order to improve the hand-over decision making.

5.6.1 Measurements made by the base station

Among other information broadcast by individual base stations on the down-link is the information related to synchronisation. The synchronisation information is transmitted on the synchronisation channel (SCH). As no synchronisation exists between the base stations, it is necessary for the mobile stations to re-acquire synchronisation once they switch to a different base station. This re-acquisition of synchronisation must also be applied during the hand-over process.

Due to the non-synchronisation between the base stations, in the GSM system the base stations cannot measure the signal strength and signal quality of an up-link used by a particular mobile station that they are not serving. This is a fundamental difference between the GSM system and its predecessors as far as hand-over is concerned. Instead, the mobile stations measure the signal strength of the down-link of the neighbouring base stations. The mobile stations actually use the signals broadcast on the BCCH carrier to measure the received signal strength of the neighbouring base stations as well as the serving base station. By adopting this protocol, there is no longer a need for the base stations to contain additional radio frequency equipment, which is used in the existing analogue cellular systems for measuring the up-link signal strength of the mobile stations that are candidate for the hand-over.

There are several measurements taken by the base station in the GSM system which include the continuous measurements of the received signal strength as well as the signal quality of the up-link of each active mobile station in its coverage area. The base station uses the outcome of these measurements and the measurements reported by the mobile station to trigger the hand-over procedure. As soon as the base station realises that there is a need for a hand-over, it transmits a hand-over required message to the MSC. The hand-over required message should include information, such as the reason for the hand-over initiation, along with a list representing the preferred target cells. It is the duty of the MSC to choose the target cell from the list of preferred cells supplied by the base station.

5.6.2 Measurements made by the mobile stations

A significant improvement in the GSM system regarding the hand-over decision is brought about by the inclusion of the measurements on the down-link part taken by the mobile station. The mobile station measures the received signal strength of the down-link of the surrounding base stations and then reports the outcome of these measurements to its serving base station. The mobile station not only measures and reports the received signal strength of the down-link of its serving base station but also the signal quality. The time-slot structure provides for a fixed training sequence to be included in the middle of every time-slot. It is this training sequence which allows the signal quality to be estimated. The signal quality is based upon the estimated BER.

Once the mobile station has successfully accessed the base station, the base station assigns a time-slot which is exclusively used by the mobile station for transmission and reception of information during the whole duration of the call. The mobile station actually uses the same time-slot number on the up-link TDMA frame and down-link TDMA frame for transmission and reception of information respectively. In order to assist the mobile station implementing this procedure, and at the same time avoiding the need for parallel processing at the transceiver stage, the start of the up-link frame is delayed by a fixed period of 3 time-slots from the start of the down-link frame [45].

The TDMA frame structure adopted, coupled with the frame transmission and reception algorithm explained above, means that a mobile station will be active in two time-slots per TDMA frame. The TDMA frame is a module of 8 time-slots. The two time-slots period corresponds to 25% of the frame time in which the mobile station is actually engaged in transmission or reception. In the free time-slots the mobile station enters a different mode known as a monitoring mode.

During the free time-slots, the mobile station switches onto the BCCHs of the base stations in its vicinity, including the BCCHs of the serving base station, in order to measure their received signal strength as well as the signal quality of its serving base station. In order for the mobile station to carry out such measurements effectively, the BCCH carriers should be continuously switched on. Furthermore, the measurements are sequentially carried out and then averaged and transmitted once in 480ms over the slow associated control channel (SACCH) to the serving base station. A 480ms period corresponds to 104 TDMA frames duration. During this interval the mobile station would have been active in 100 TDMA frames and idle in the remaining TDMA frames. It is in these active TDMA frames that the mobile station carries out the signal measurements. Since the mobile station makes the measurements, it is therefore necessary for the mobile station to identify which of the base stations is being measured. Basically, the mobile station would recognise the individual base station within a cluster by its unique BCCH carrier frequency. With small cluster sizes the reuse distance is small and it is therefore possible that two cells in close proximity and using the same frequency are being measured. The GSM system approaches this problem by adding a new form of base station identification known as the Base Station Identity Code (BSIC). The BSIC is broadcast by the base station on the BCCH carrier. During the 4 idle frames mentioned above, the mobile station synchronises to the synchronisation channel of the BCCH carriers in its vicinity. Once the mobile station establishes a synchronisation with a particular base station, it is then able to decode the BSIC.

Once the mobile station determines the identity of the surrounding base stations being measured, it reports the averaged values of the six strongest received signal strengths to its serving base station on the SACCH. The base station keeps monitoring the data reported by the mobile station as well as the measurements carried out by itself. As soon as it realises there is a need for a hand-over, the base station informs the MSC by sending a hand-over request message. Additionally situations arise when, for example, signalling data related to hand-over is urgently required to be transmitted. In this case the signalling data transfer is achieved by stealing bits from speech frames.

The GSM system has been designed to operate with four different types of hand-over. Some of these hand-over decisions are autonomously taken by the BSS; whilst the others have to be considered by the MSC. These four types of hand-over are as follows:

(i) the mobile station is instructed to tune to a different radio channel that is supported by the same BTS. This type of hand-over is known as " ***intra-BTS hand-over*** ". This hand-over can autonomously be taken by the BTS.

(ii) the mobile station is instructed to tune from one radio channel that is supported by one BTS to another radio channel that is supported by a different BTS that is being controlled by the same BSC. This hand-over is known as "***inter-BTS, intra-BSC hand-over*** ". This hand-over decision can autonomously be considered by the BSC.

(iii) the mobile station is instructed to tune from one radio channel that is served by a BSS to another radio channel that is served by another BSS that is being controlled by the same MSC. This hand-over is known as "***inter-BSS, intra-MSC hand-over*** ". The decision on such hand-over has to be taken by the MSC.

(iv) the mobile station is instructed to tune from a radio channel that is supported by a BSS belonging to a MSC to another radio channel supported by another BSS that is being controlled by another MSC. This type of hand-over is known as "***inter-MSC hand-over*** " and it has to be taken by the MSC.

From the above, the types of hand-over can be broadly divided into two types as far as the BSS is concerned, namely internal and external hand-overs. The internal hand-over refers to the hand-overs that the BSS can autonomously take the decision on, while the external hand-over refers to the hand-overs where the MSC has to take the decision on. The GSM system uses SSN7 to achieve network signalling between its various functional entities such as BTS, BSC, MSC, etc.. The GSM has added a number of extensions to the SSN7 signalling features to include those needed to support mobile communications. Furthermore, BSS terminates the Radio Resource Management (RM), which is the lowest sublayer of the signalling layer 3. In particular, the BSS interprets and processes the RM messages to decide whether a hand-over decision is needed and then maps them into messages of the Base Station System Management Application Part (BSSMAP) for transportation to the MSC. For the other signalling messages of the higher sublayers of layer 3, the BSS uses Direct Transfer Application Part (DTAP) for transportation to the MSC [107].

With the present structure of the GSM system, no direct link is available between the BSSs. However, with the proposed structure, direct links are available between the BSSs under the control of the same MSC. This allows for the "***intra-MSC hand-overs***" to be taken autonomously by the BSSs. To achieve the purpose, the following procedure could be followed:

(i) the BSS checks the identity code of the first targeted BSS in the list supplied by the mobile station as well as the list prepared by itself as explained earlier. This requires that each BSS should keep a record of all the BSSs that are under the control of the same MSC. If the first targeted BSS is in its list, then it proceeds with the following steps. Otherwise the BSS transfers the message to the MSC using the BSSMAP.

(ii) the serving BSS sends a packet addressed to the targeted BSS. The packet contains a request for allocation of radio channel to the mobile station. The packet should contain any additional information that might be needed such as the type of the requested channel (ie. voice, data or control).

(iii) if the targeted BSS has a free channel to offer, then it responses by sending a packet to the serving BSS that contains all the necessary information related to the new channel. The information is related to the new channel, such as the frequency of the channel and the number of the time-slot within the TDMA frame.

(iv) upon receiving a positive response, the serving BSS passes all the information regarding the new radio channel to the mobile station over the radio link. The mobile station switches to the new channel and the old channel (ie. the channel that has just been released) becomes free.

(v) the targeted BSS has to respond by sending a request rejection packet if it has no free channel to offer at the time of receiving the channel request packet.

(vi) if case (v) occurs, then the serving BSS tries the next BSS down on the list, as long as the BSS is under the control of the same MSC, by starting the procedure from step (ii). Step number (vi) will be repeated as long as the BSS receives a negative response to its request.

Since it is now a part of the BSS functionality to execute "***intra-MSC hand-overs***", the base station should inform the MSC of any resulting assignment. By adopting this procedure, it is expected that there will be some benefits in terms of partially lifting some of the signal processing load regarding the hand-over of the MSC. To illustrate these advantages, the GSM hand-over algorithm has been simulated using the SLAM package. In this simulation, it has been assumed that the BSSs belonging to the same MSC are interconnected by a MAN as explained earlier. This allows for the BSSs to communicate directly with each other and hence intra-MSC hand-over decisions can take place without the intervention of the MSC. The MSC will be notified of any hand-over execution which then passes it to its VLR. Such information might be needed for keeping a fairness of traffic load distribution among the BSSs. Since the BSSs involved exist in the same location area, which means the MSRN is still valid, no further signalling exchanges will take place as far as this hand-over execution is concerned. A few simulation assumptions have been assumed as follows:

- a 7-cell cluster configuration has been assumed. This configuration gives situations when intra-BSS and inter-MSC hand-over executions are needed as the mobile station travels along its route.

- the BTSs are located at the centre of the service areas.

- three radii cell areas were considered. These radii are 1km, 2km, and 3km.

- the initial distance separates the mobile stations from the BTSs is chosen from a uniform distribution having values between (0 and 1) times R where R is the cell radius. In particular, 0 means that the initial starting point of the mobile station is at the centre of the cell and 1 means the point is at the edge of the cell.

- the mobile station will have three speed limits chosen from a uniform distribution. These speed limits are: 20-30km/h, 30-40km/h and 40-50km/h.

- the mobile stations choose their call duration periods from an exponential distribution between 2-7 minutes. The call duration periods will result in a situation where some calls will end before any hand-over is needed as well as call periods which allow for several hand-overs to be performed.

- the system has been simulated for 7.5 minutes of real time.

Figure 5.4 shows the mean number of intra-BSS and inter-MSC hand-overs per user. This figure represents the data for the case when the vehicle is cruising at a speed of 20-30km/h and the cell has a 1km radius. It can be seen that the number of intra-BSS hand-overs exceeds the number of inter-MSC hand-overs. This might be due to the initial location of most of the mobile stations (ie. at which the mobile stations start their routes), and the direction in which the mobile stations are travelling (either towards another BTS or another BSS).

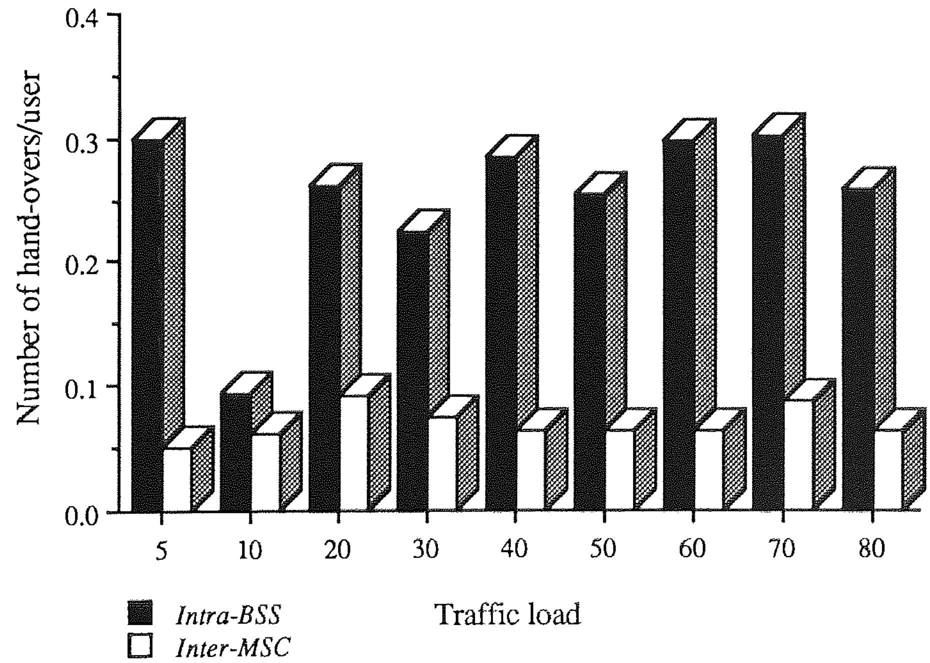


Figure 5.4 Comparison of the intra-BSS hand-overs with the inter-MSC hand-overs for the 20-30km/h vehicle speed and 1km cell radius.

Figures 5.5, 5.6, and 5.7 show the number of intra-BSS hand-overs for 1km, 2km and 3km cell radii with speed limits of 20-30km/h, 30-40km/h and 40-50km/h respectively. All the figures show that the number of intra-BSS hand-overs increases with increasing cell size. This is true because, with large cells, the calls tend to end either within the same cell or have just crossed to a neighbouring cell.

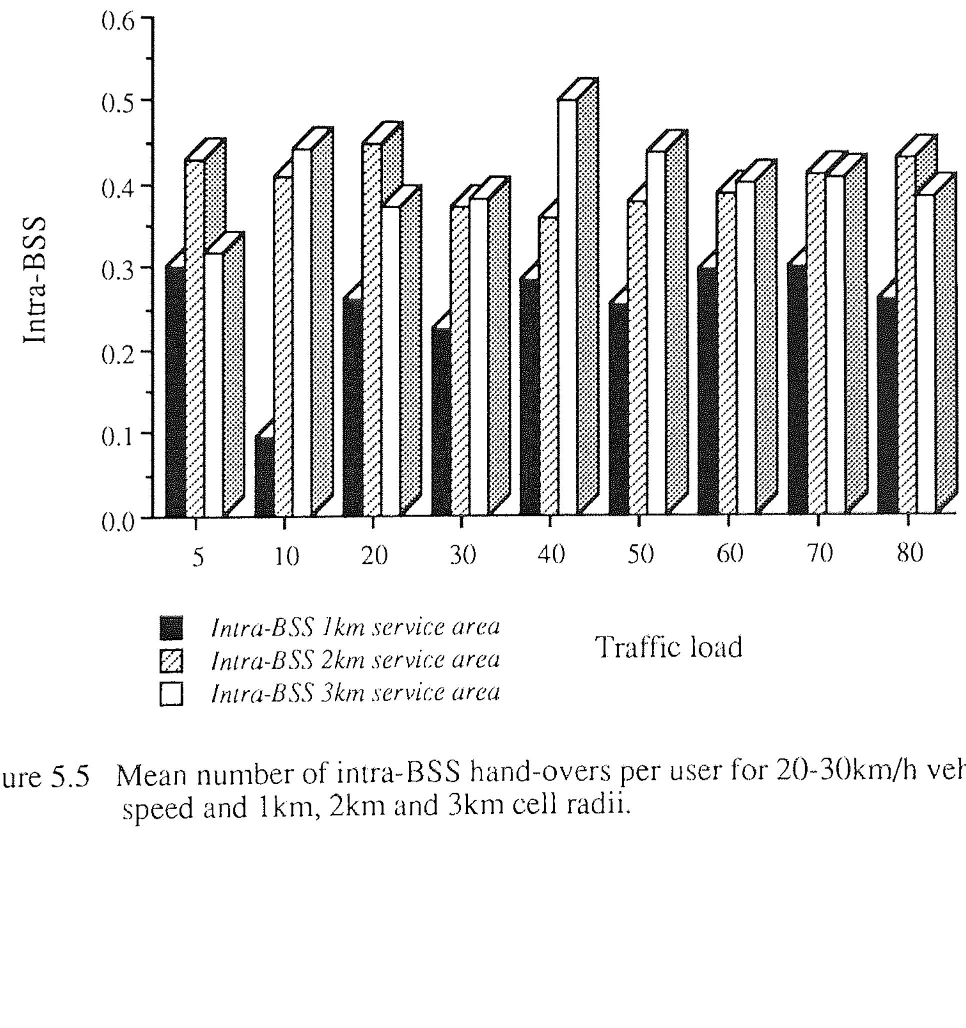


Figure 5.5 Mean number of intra-BSS hand-overs per user for 20-30km/h vehicle speed and 1km, 2km and 3km cell radii.

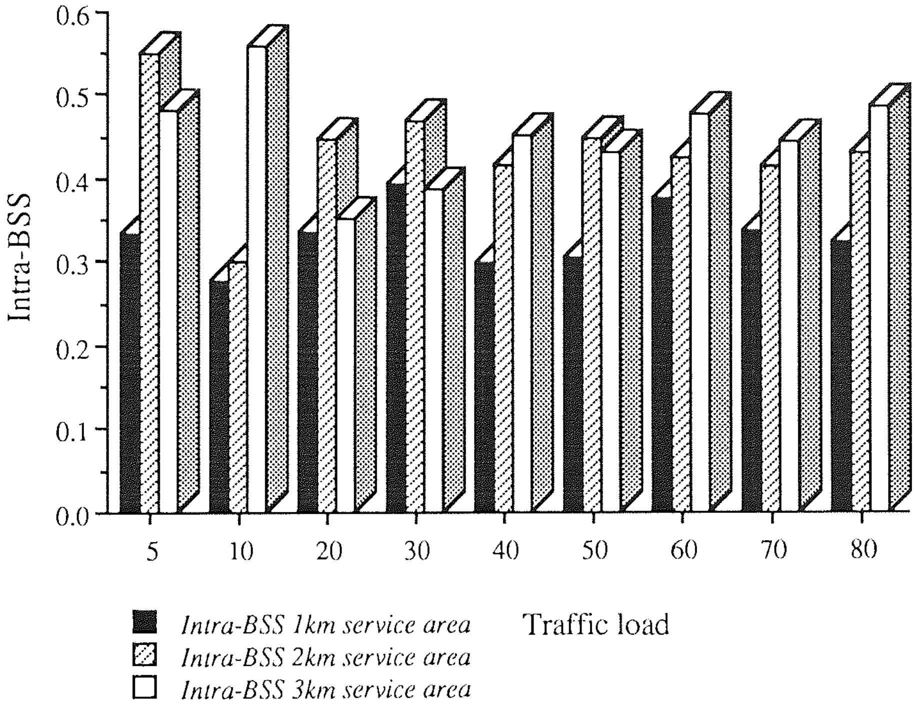


Figure 5.6 Mean number of intra-BSS hand-overs per user for 30-40km/h vehicle speed and 1km, 2km and 3km cell radii.

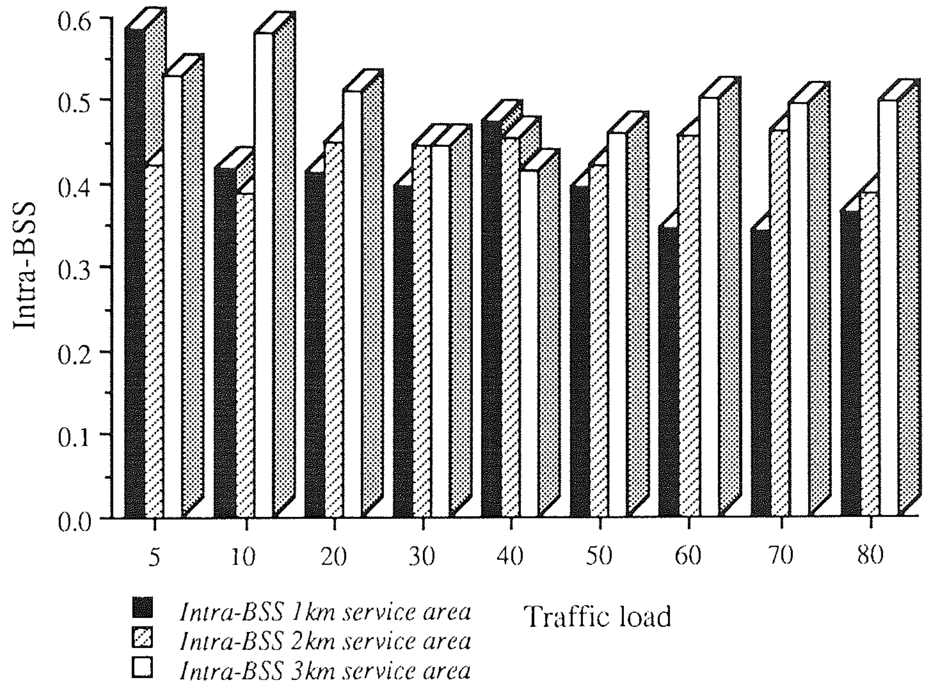


Figure 5.7 Mean number of intra-BSS hand-overs per user for 40-50km/h vehicle speed and 1km, 2km and 3km cell radii.

Figures 5.8, 5.9 and 5.10 represent the number of inter-MSC hand-overs per user for 1km, 2km and 3km cell radii with speed limits of 20-30km/h, 30-40km/h and 40-50km/h respectively. All the figures show that the number of inter-MSC hand-overs per user decreases as the cell size increases. This is true as this represents exactly the opposite situation to the above case.

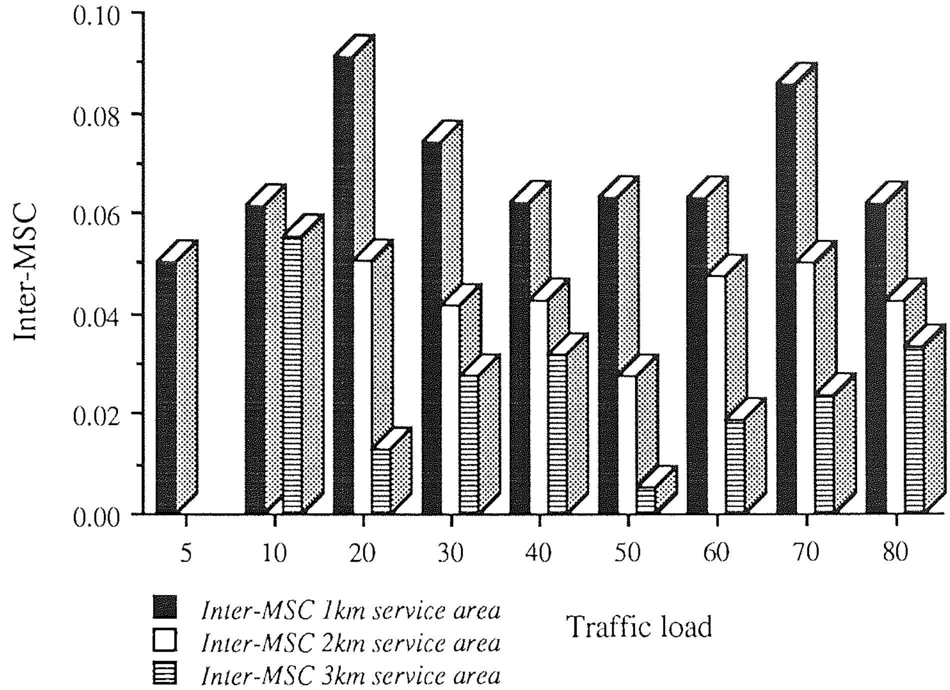


Figure 5.8 Mean number of inter-MSC hand-overs per user for 20-30km/h vehicle speed and 1km, 2km and 3km cell radii.

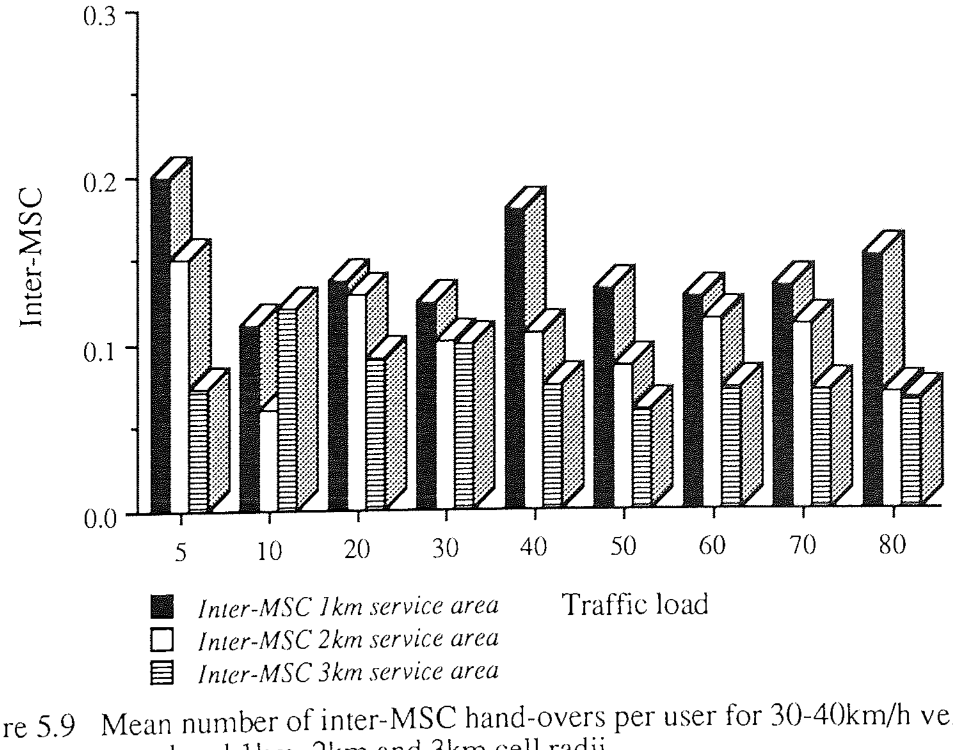


Figure 5.9 Mean number of inter-MSC hand-overs per user for 30-40km/h vehicle speed and 1km, 2km and 3km cell radii.

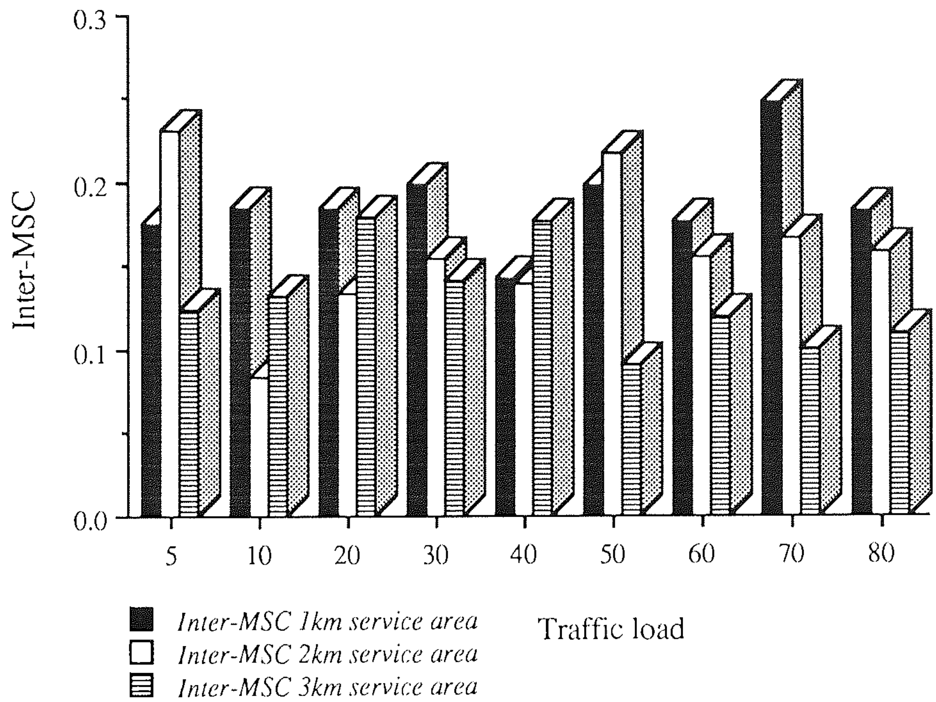


Figure 5.10 Mean number of inter-MSC hand-overs per user for 40-50km/h vehicle speed and 1km, 2km and 3km cell radii.

Figures 5.11, 5.12 and 5.13 represent the percentage of inter-BSSs, intra-MSC hand-over loads on the system for the various cell radii and speed limits mentioned earlier. These figures actually represent one of the benefits that the GSM system will achieve from the proposed interconnection structure. In the present GSM arrangement these inter-BSSs, intra-MSC loads will be dealt with by the MSC, whilst in the proposed arrangement such loads will be distributed among the BSSs.

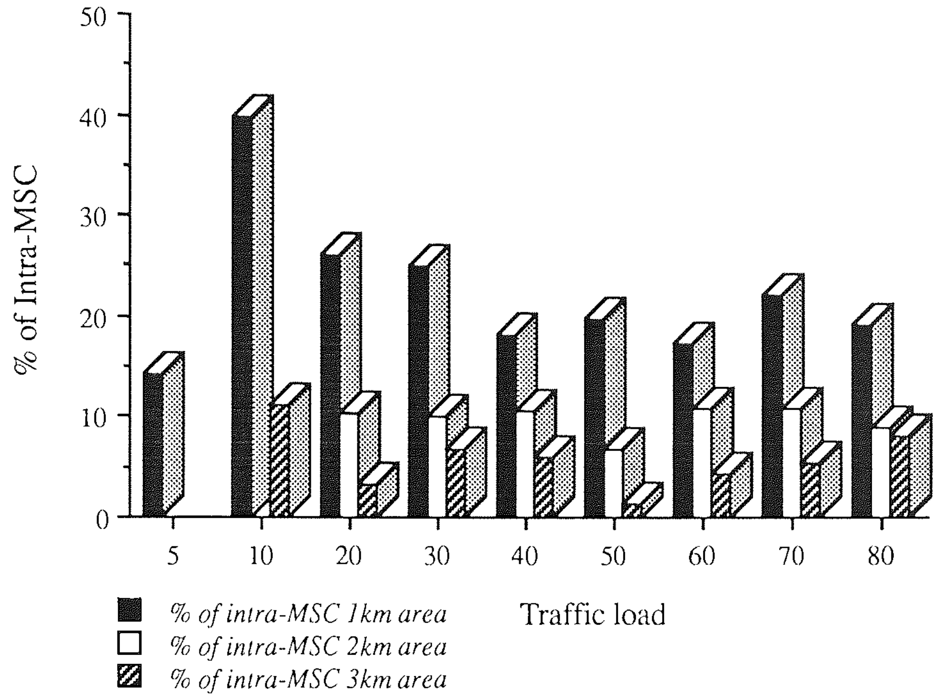


Figure 5.11 Percentage of the inter-BSSs, intra-MSC hand-overs for the 20-30km/h vehicle speed and 1km, 2km and 3km cell radii.

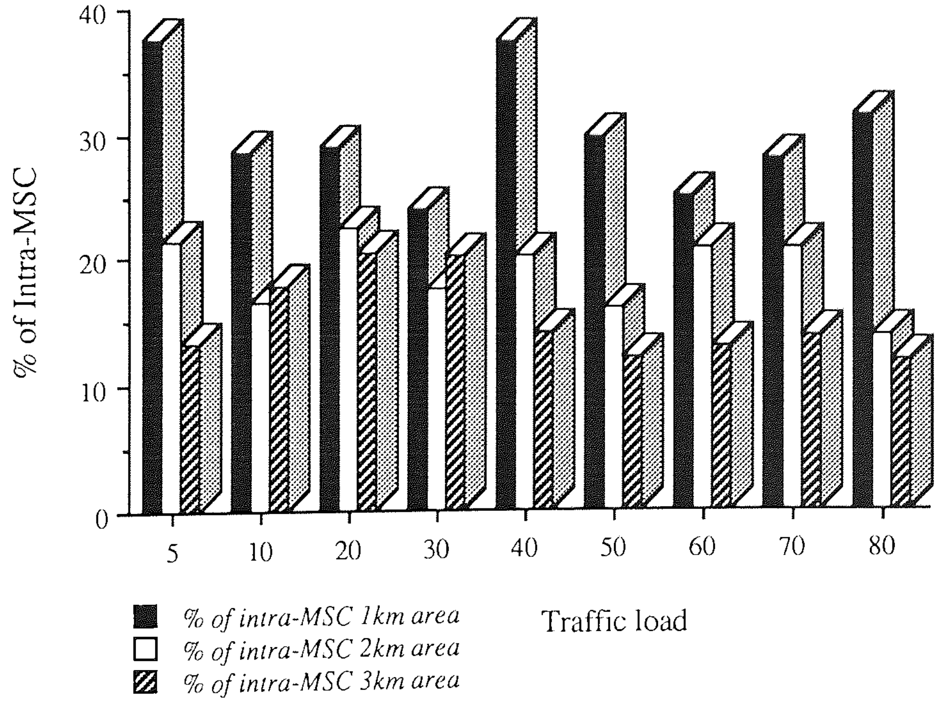


Figure 5.12 Percentage of the inter-BSSs, intra-MSC hand-overs for the 30-40km/h vehicle speed and 1km, 2km and 3km cell radii.

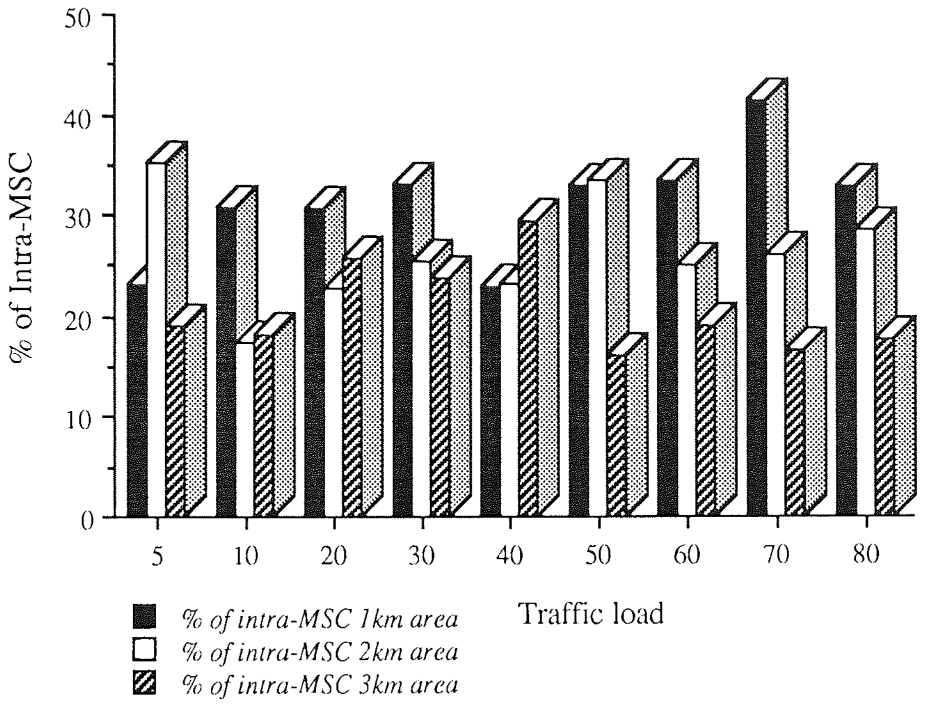


Figure 5.13 Percentage of the inter-BSSs, intra-MSC hand-overs for the 40-50km/h vehicle speed and 1km, 2km and 3km cell radii.

Figures 5.14 represents the effect of speed limits on the inter-BSS, inter-MSC hand-overs for the 1km cell radius, while figures 5.15 and 5.16 represent the same relationship for the 2km and 3km cell radii respectively. Furthermore, figure 5.17, 5.18 and 5.19 represent the effect of speed limits on the number of inter-MSC hand-overs for the mentioned cell radii. All the figures show that the rate of either type of hand-overs increases with the increase of the vehicle speed limits.

This simulation output results have shown the benefits of interconnecting BSSs of the same MSC. These results will also be useful for determining the size of the areas to be covered by individual BTSs as well as by individual BSSs.

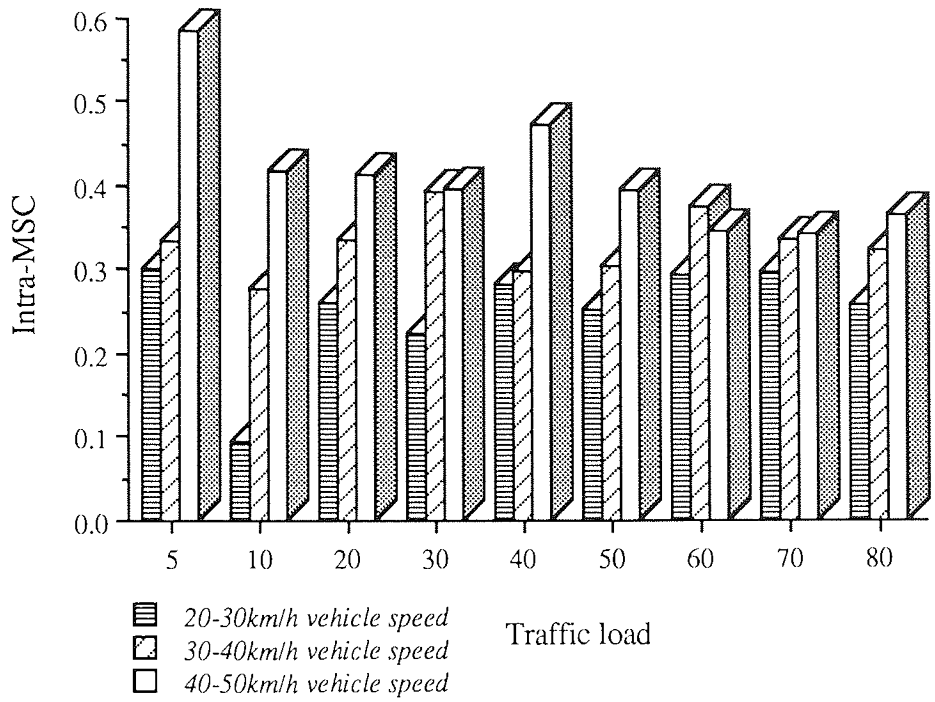


Figure 5.14 The effect of speed limits on the inter-BSS, intra-MSC hand-overs on the 1km cell radius.

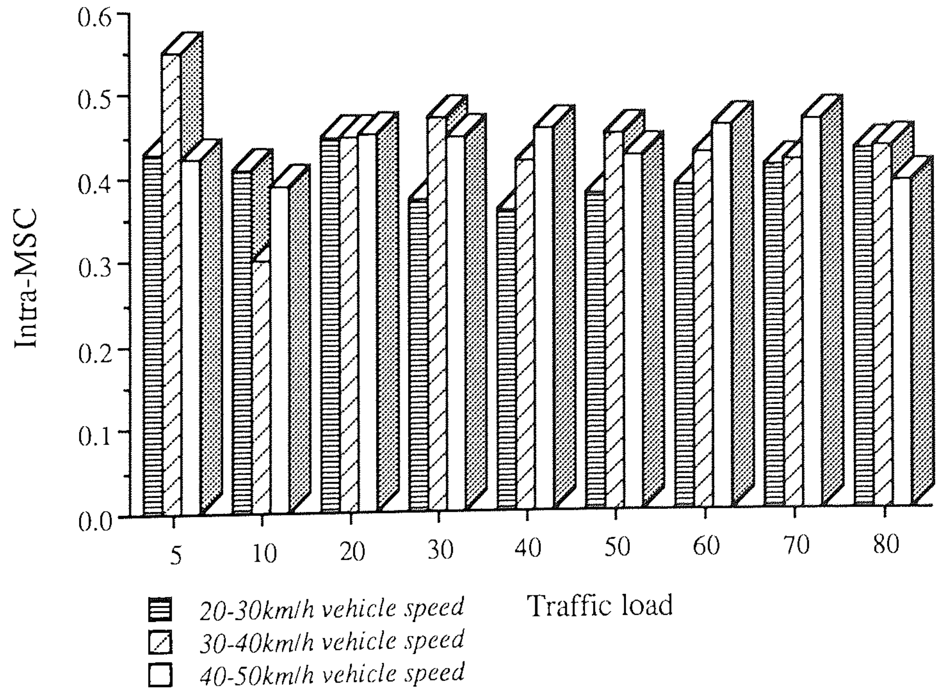


Figure 5.15 The effect of speed limits on the inter-BSS, intra-MSC hand-overs on the 2km cell radius.

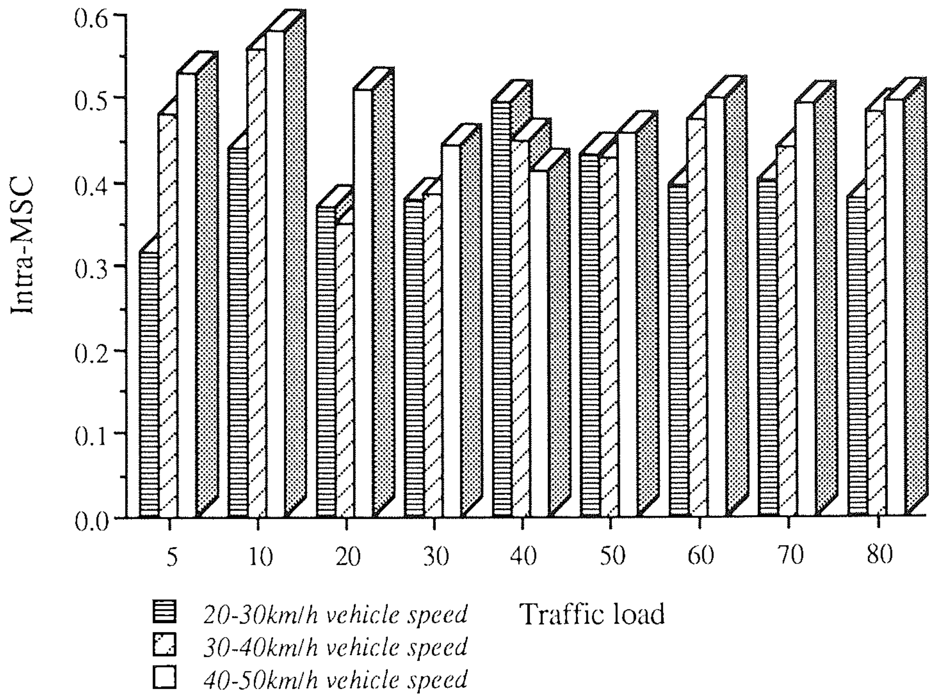


Figure 5.16 The effect of speed limits on the inter-BSS, intra-MSC hand-overs on the 3km cell radius.

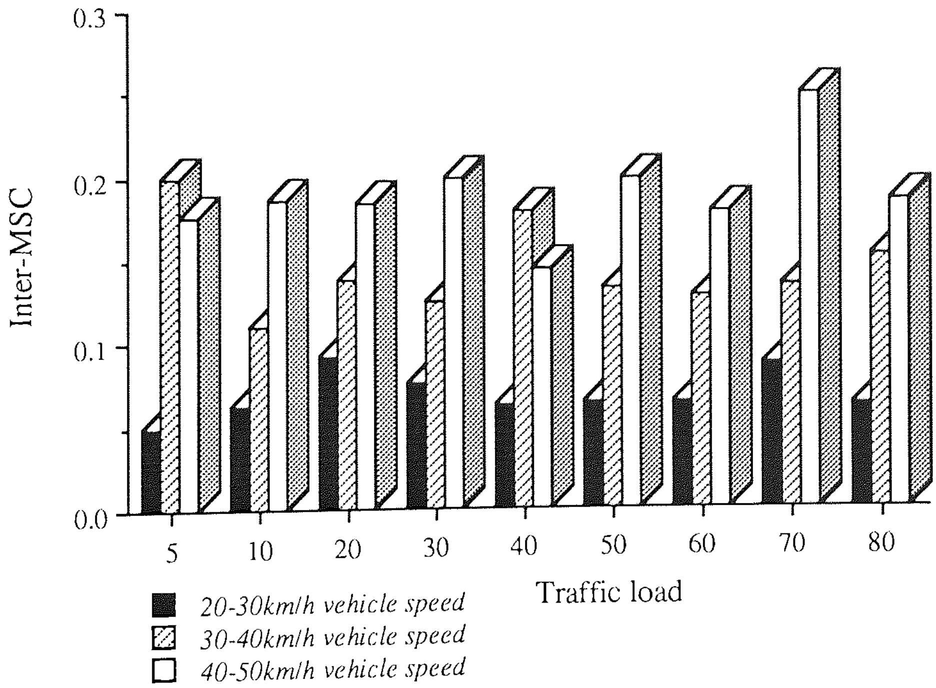


Figure 5.17 The effect of speed limits on the inter-MSC hand-overs on the 1km cell radius

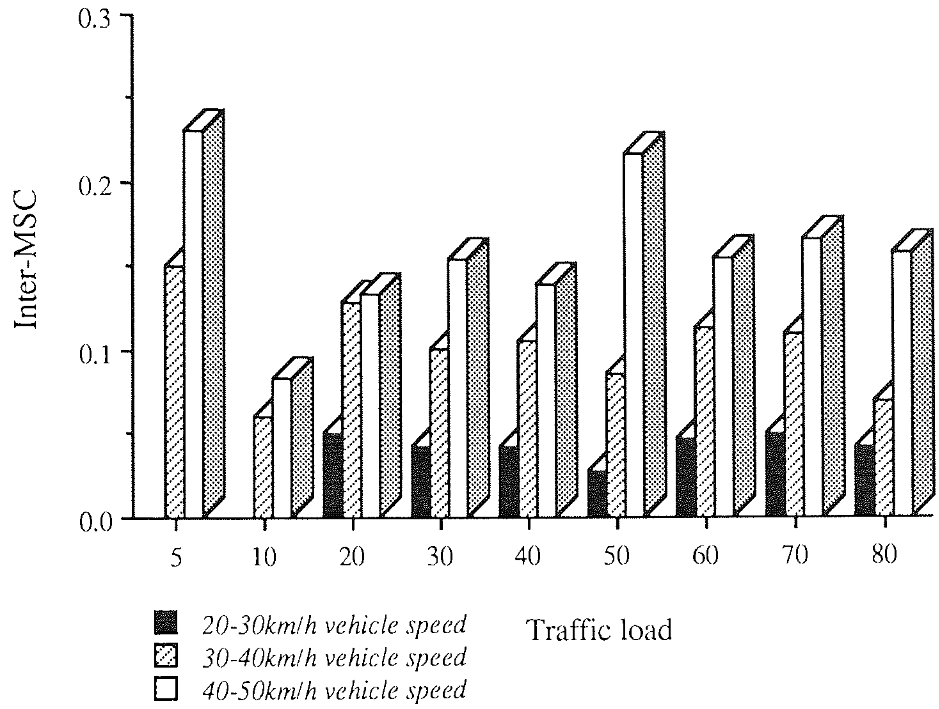


Figure 5.18 The effect of speed limits on the inter-MSC hand-overs on the 2km cell radius.

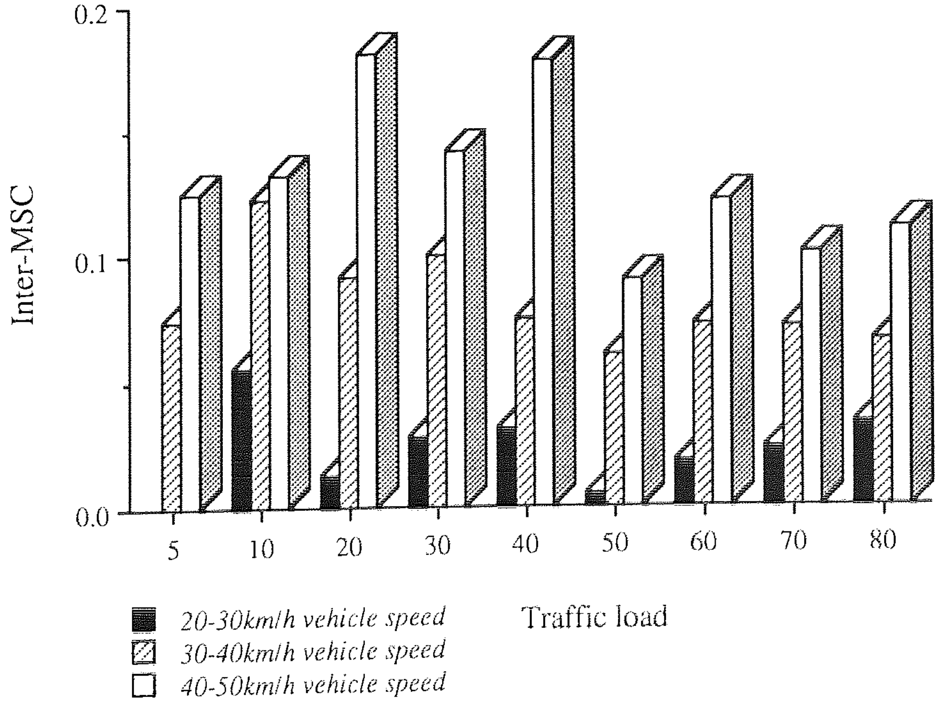


Figure 5.19 The effect of speed limits on the inter-MSC hand-overs on the 3km cell radius.

With the present GSM system's structure, the intra-BTSs and intra-BSCs (ie. intra-BSSs) hand-overs are executed without the intervention of the MSC. The intra-BSSs hand-overs are dependent on the size of the coverage area as well as the speed of the vehicle. Figures 5.4, 5.5, 5.6 and 5.7 are presented to show the effect of the speed and area size on the intra-BSSs hand-overs. Figures 5.8, 5.9 and 5.10 present the mean number of the inter-MSC hand-overs per customer. In this case the speed and the cell size factors are also taken into account. Figures 5.11, 5.12 and 5.13 represent the percentage of the volume of the inter-BSSs, intra-MSC hand-overs against the speed limits and cell radii. On the other hand, figures 5.14, 5.15 and 5.16 represent the effect of various speed limits against the cell size for the inter-BSSs, intra-MSC hand-overs. Finally, figures 5.17, 5.18 and 5.19 represent the relationship between various speed limits versus coverage area for the inter-MSC hand-overs.

Under the current structure of the GSM, inter-BSSs, intra-MSC hand-overs are dealt with by the MSC. However, under the proposed layout, where a link is established between the BSSs, the inter-BSSs, intra-MSC hand-overs will be executed by the BSSs without the intervention of the MSC. Furthermore, information on the effect of speed limits and cell radii on the rate of the hand-overs is a useful information when dealing with network aspects such as the consideration of the coverage area's size.

5.7 Summary

This chapter begins with an outlining of the current trends in the communication systems. A two-phase scenario of interconnecting the GSM entities over the broadband networks based on the IEEE 802.6 MAN is presented. The initial phase deals with interconnecting the mobile switching centres. The second phase presents the strategy of interconnecting the base station controllers and base transceiver stations. The advantages of such interconnection is evaluated in terms of the speed of exchanging of the hand-over control information between these entities.

CHAPTER SIX

DEVELOPED PROTOCOLS MODELLING BY SIMULATION

6.1 Introduction

In the past few decades, communication and computer systems have evolved very rapidly and become increasingly complex. As a result, it may be virtually impossible to reliably evaluate and measure their performance by exact mathematical modelling methods. Simulation is another attractive and powerful tool that can be used for evaluating the performance of any type of network [111]. At present, there are several software packages that have been created to analyse and evaluate the behaviour of these systems. The advantages of using simulation as a tool for evaluating the performance of these networks may include the following:

* It enables the system designer to model the system and hence to understand, evaluate and improve the performance of the system prior to the expense of its implementation.

* It aids the system analyst to evaluate the performance of a real system in the case when it is impossible to obtain direct measurements, or when it is economically infeasible to carry out direct measurements such as the case in this research for evaluating the GSM system.

* It is a time saving method in the case where it may take a very long period of time to produce results on an ongoing system.

6.2 Description of the Simulation Modelling

Computer simulation modelling is a process whereby a real system is presented by a mathetmatical-logic model and experiments on this model are carried out using a computer. A set of variables is then used to characterise the system, with each combination of variable values representing a unique state or condition of the system. Movement of the system from state to state is simulated by the manipulation of the variable values. Changes in the state of the system can occur continuously over the simulation time or at discrete points in the simulation time. In communication systems, the manipulation of the variable values is done in accordance with well-defined communication rules and regulations known as protocols. Simulation is therefore regarded as the representation of the dynamic behaviour of the system from state to state governed by these protocols.

6.3 Package Choice

A number of software packages are now widely used for simulating communication systems. In the literature, references [112,113] provide a comprehensive review of these software packages. There are several factors which have to be considered when choosing a particular simulation package, such as the type of the system to be modelled. Simulation Language for Alternative Modelling (SLAM II) package has been selected to model the developed protocols in this research. The selection of the SLAM

II simulation package was based on factors which include its suitability for modelling the developed protocols, its flexibility in allowing the modeller to write his own subroutines in normal Fortran computer language and its availability at Aston University.

6.4 Simulation by SLAM II

SLAM II is a Fortran-based simulation language that allows an alternative approach to modelling [114]. Two simulation techniques could be used in simulating a system. The selection of a particular simulation technique is directly dependent on the nature of the changes of the system parameters or variables:

Continuous simulation. In this simulation approach, dependent variables are used to represent the model. These dependent variables change continuously over the simulation time and their values are available at any point in the simulation time. Construction of a continuous simulation model involves the definition of a set of differential or difference equations. The set of equations describe the dynamic behaviour of the state variables. The dynamic behaviour of the state variables simulates the real system.

Discrete simulation. In this approach to simulation, the changes in the state of the system occur discretely at specified points in the simulation time. These specified points in the simulation time are referred to as event times. This implies that the system remains constant between the event times. Advancing the simulation time from one event to the next is referred to as the event-oriented approach. The complete dynamic behaviour of the system is then obtained by sequentially processing events and recording status values at event times. In the event-oriented approach, the modeller defines the events and the potential changes which they may cause in the system. The modeller also provides Fortran subroutines for the mathematical and logical relationships describing the changes that each type of event will produce.

In this research, discrete event simulation has been used to simulate the developed protocols. With this simulation technique, SLAM II offers the modeller a set of standard subprograms to be used for performing discrete event functions, including statistic collection, event scheduling, file manipulation and entity tracing through the system. The SLAM II processor also completely relieves the modeller of the responsibility for chronologically ordering the events. Figure 6.1 depicts the discrete event simulation using SLAM II.

Common to all the simulation runs, the SLAM II processor starts by reading the SLAM II input statements and initialising the SLAM II variables. These control variables include variables such as the maximum number of attributes per entry, dimensioning of the array NSET/QSET for storing the entries and the maximum number of entries that can exist in the system at any time. This is followed by a call to the INTLC subroutine which specifies any additional initial conditions and variables for the simulation runs. The processor then begins execution of the simulation by removing any particular event that has been specified by the modeller in the subroutine INTLC and then starts by removing the first event in the event calender. Events are queued in the event calender according to their sequence values of event times (ie. first event will be the event that has the earliest event time and so on).

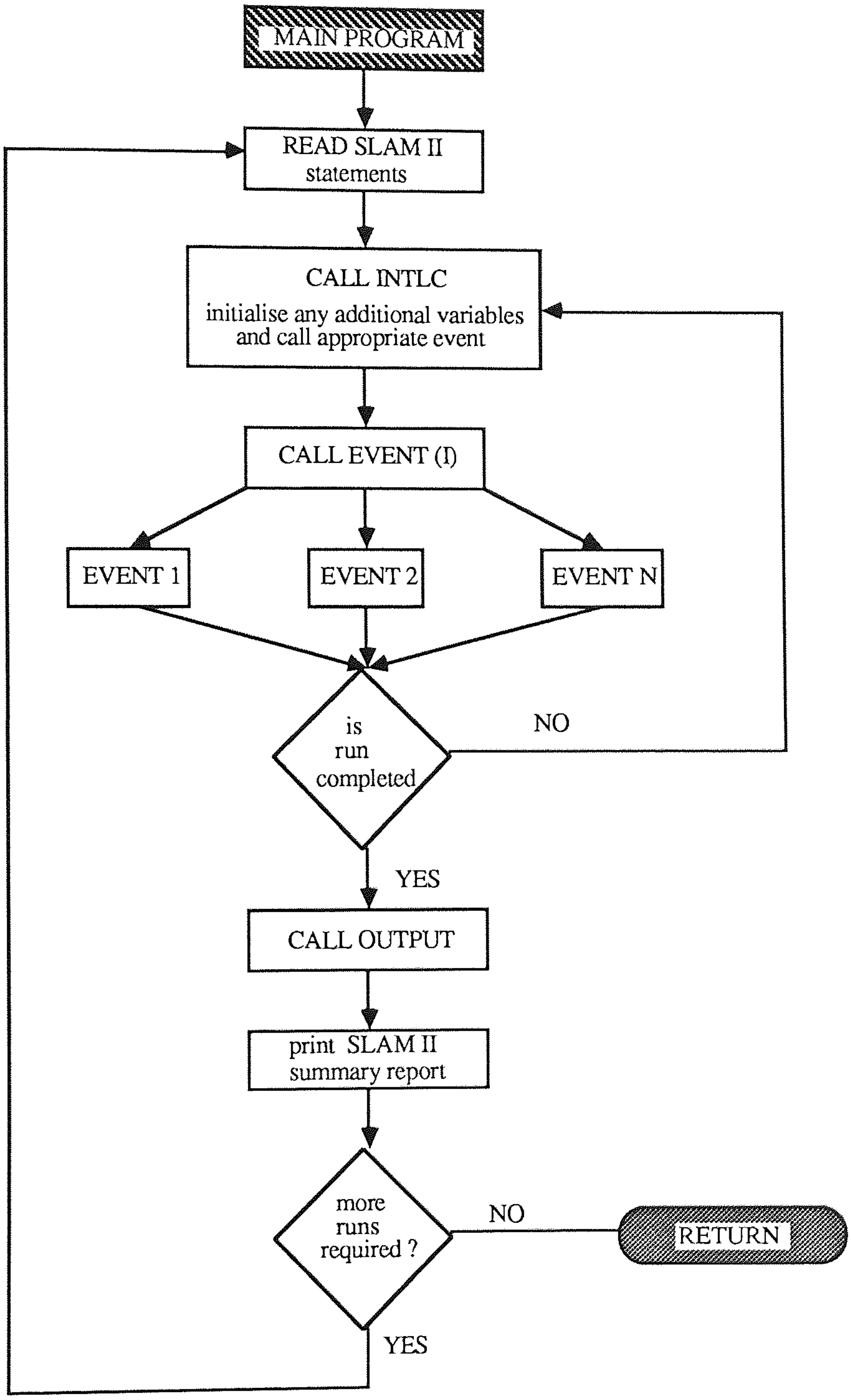


Figure 6.1 Flowchart of the discrete event simulation using SLAM II

6.4.1 Definition of the Events

In discrete event modelling, all the variables which define the status of the system are firstly determined. The number of the events, which characterise the dynamic behaviour of any protocol, will be different from one protocol to another. Program 1 in the appendices represents the coded program of the developed access protocol for the GSM system. Four events were needed to model the developed access protocol of the GSM system. These events are briefly described below:

ARRIVAL EVENT. This event is used to generate mobile users. Every mobile user has his associated attributes such as arrival time, maximum number of retransmission attempts specified by the system at the time of the initial access attempt, initial transmission time on the random access channel and the selected random reference number. All the necessary statistics, which will be needed later in the simulation, are collected and kept in their specific associated attributes.

SENSING EVENT. This event is used by the mobile users to check whether or not they have successfully transmitted their channel request messages. All the attempts of channel request transmission on the same time-slot will be queued in a buffer at the controller. Since the worst case scenario has been adopted in this research, all the mobile users who have the same random reference number need further contention on the dedicated control channel.

REATTEMPT EVENT. This event is used by the mobile users who unsuccessfully transmit their channel request messages and want to reschedule retransmission of their channel request messages. Prior to any retransmission attempt, every mobile user has to check whether or not he has reached the maximum number of retransmission attempts broadcast by the system. If the maximum number of retransmission attempts has not yet been reached, then the user schedules the next attempt and increments the attributes containing the number of retransmission attempts by 1; otherwise the attempt is abandon.

SUCCESS EVENT. This event is used by all the mobile users who have successfully transmitted their channel request packets. In particular, this event is used for the collection of statistics such as the access delay time and the access channel throughput.

Program 2 in the appendices represents the coded program for voice transmission over the GSM system. Four additional events to the events presented for the access protocol are needed with the voice transmission protocol. These additional events are:

TRANSMISSION EVENT. This event is needed for the synchronisation of the mobile users. In particular, every mobile user will be assigned both the time-slot number and the radio carrier number to transmit his voice packets. The time-slots are sequentially allocated to the users.

Additionally, every mobile user will be assigned an initial time for transmission (ie. initial time advance).

ALLOCATION EVENT. A call to this event is made from the transmission event, after the initial time advance has elapsed, to mark the start of the speech packets transmission. All the necessary attributes are updated.

END SERVICE EVENT. A call to this event is made from the allocation event at the end of each packet transmission. This event is therefore used for statistics collection, such as the number of transmitted speech packets per user and the throughput of the allocated channels. A check is then made on each user's buffer. If the buffer contains more packets for transmission, a call to the allocation event is then made after a frame time period, otherwise the user is presumed to have entered the silence period.

SUBS-TRANSMISSION EVENT. A call to this event is made when a mobile user has entered a subsequent speech transmission after being in a silence period.

In modelling the developed protocol for inserting data packets during the silence periods of the speech transmission, two events, in addition to the events already used for speech transmission protocol described above, were needed. Program 3 in the appendices represents the coded program for this protocol. The two additional events are briefly described below:

DATA START EVENT. This event is called every time an associated speech terminal enters silence periods and the time is due for data transmission. In this simulation, it has been assumed that data terminals firstly gain access and then keep listening to any paging message for commencing their data transmission.

DATA END EVENT. This event is called at the end of each data packet transmission or as soon as the associated speech terminal reacquires the channel. The main purpose of this event is, therefore, for statistics collection on the data transmission.

Hand-over is a control procedure applied in the mobile communication environment when a mobile terminal crosses from one cell boundary to another cell boundary during a communication session. Five events were required to simulate the developed hand-over procedure. Program 4 represents the coded program for this procedure. Some of these events perform similar functions to those functions performed by the events described above. In particular, these events are used to perform functions including estimations of call duration, initial mobile location, selection of the vehicle speeds and checking when and whether an intra or inter hand-over procedure is needed as explained in chapter 5.

6.5 Summary

Since their introduction, computer simulation techniques have contributed on a very large-scale in every engineering field. In this research, SLAM II has been used to model the developed protocols due the complexity associated with exact mathematical analysis. In particular, the discrete event approach was used in simulating these developed protocols. All the events, which are used to describe the dynamic behaviour of the different protocols, were presented and described.

CHAPTER SEVEN

CONCLUSION AND RECOMMENDATION FOR FURTHER WORK

7.1 Conclusion

The pan-European Groupe Spécial Mobile (GSM) has entered the operational phase in some of the European countries. Furthermore, more than 20 non-European countries have also shown interest in the GSM system. Some of these countries have already adopted the system, while in the others the system is still under consideration. This makes the GSM system not only a pan-European system but a global system [108].

In the GSM system, mobile stations, which are scattered over the coverage area, use the slotted-ALOHA medium access protocol to contend on the random access channel to send their channel request messages. The channel request messages contain a random reference number as well as the reason for requesting the channel. The random reference number is used as a temporary identification for the accessing mobile stations. Moreover, the GSM system is designed to operate in radio cells as small as 1km and as large as 35km. This implies that the channel request messages should have very large guard periods of time in order to compensate for the round trip propagation delay and hence prevent the overlapping of the adjacent received messages at the base station. Due to the capture effect, it is likely that one of the simultaneously transmitted channel request messages will be correctly received by the base station. As a result, all the mobile stations that have accessed at the same time-slot and having the same random reference number will contend on the same dedicated control channel to check which one of them had its channel request message successfully received by the base station. Mobile stations use the layer 3 service request message, which contains the actual identification of the mobile station, to resolve this contention. In this work, simulation results have shown that the GSM access procedure suffers long access delays especially at higher offered traffic loads. This inefficiency in the GSM access protocol is related to the number of mobile stations that contend on the dedicated control channel. Long access delay is an undesirable phenomenon particularly in the small radio zones where the mobile stations are expected to exist in each one for a short time before crossing to another radio cell. By fitting the mobile stations with a timer, the distance separating the mobile stations from the base station could be used as an added control parameter to increase the efficiency of the GSM access protocol. In this work, simulation results have shown that shorter access delays are achieved when the mobile stations are fitted with the timer. This can be explained since, as a result of the fitted timer, a smaller number of mobile stations actually contend on the dedicated control channel. This proposed modification of the GSM access protocol has other attractive advantages, which including the following:

(1) Fitting the mobile stations with the new timer may be achieved with a minimum disruption to the existing software and hardware of the mobile stations.

(2) Improvement in the efficiency of the access delay will result in a positive impact on the efficiency of the system as a whole.

The current structure of the GSM is based on the realisation that voice services are assumed to be dominant. However, some market analysts [109] predict that by the end of the century the volume of the data traffic will outweigh that of the voice traffic by a large margin.

The GSM system has adopted a fixed assignment method based on Time Division Multiple Access (TDMA) as a protocol for the wireless transmission of information over the traffic channels. The operation of the TDMA protocol resembles the operation of the circuit-switched based technology whereby the multiplexing is achieved at the call level. Multiplexing at the call level means that the source occupies the assigned channel for the whole duration of the call period. Due to the fact that speech is bursty, although its burstiness is not as high as data, a considerable amount of the channel's time will be wasted during the periods when there is no information to be transmitted. The burstiness of the speech arises due to the fact that the speech is present for approximately 40% of the duration of the call [110], the rest of it being silence between speech and listening. In this research, a protocol to increase the spectrum efficiency has been developed. The increase in the spectrum efficiency is achieved through the insertion of data packets during the silence periods of the speech transmissions. This protocol is suitable for data transmission that has a stringent information loss but can tolerate delay. The other advantages of the developed protocol including the following:

(1) The protocol enables the GSM system to achieve higher spectrum efficiency without any need for additional radio spectrum allocations.

(2) The protocol allows for the mobile stations that are involved in the speech transmission to have the upper hand in the reacquisition of the channel. The ability of the speech mobile stations to reacquire the channel at any time they need the channel is a feature needed in order to prevent the speech transmission suffering any additional delay as a result of the data transmission during the silence periods.

(3) The implementation of the developed protocol requires no additional hardware to be fitted to the mobile stations.

Currently there are a lot of research activities toward standardisation of Metropolitan Area Networks (MANs). The most important achieved agreement so far was the standardisation of the packet size and format. The agreed cell format for the MANs, which highly matches the ATM cell format, will enable easy interworking as well as assisting in the provision of a common set of interfaces. MANs are designed to provide connection-oriented and connectionless services based on the DQDB medium access protocol. MANs are considered to be the first step towards an ultimate single fabric based on the IEEE 802.6 and B-ISDN protocols to provide an integrated voice and non-voice services. Public broadband networks are expected to be mature by the mid nineties. However, this is subject to some factors such as the progress in the standardisation activity, the maturity of the customer for the broadband services and the advance in the customer premises equipment.

The GSM system, on the other hand, is now in the implementation phase throughout Europe. This implies that the GSM system will be in service well beyond the turn of the century and after the implementation of broadband networks. While the transition towards the broadband network may take a longer period of time, urban areas will see the first implementation of these broadband networks. The demand on the mobile service is also rapidly increasing in these urban areas. In the present structure, the fixed side of the GSM system relays on private leased lines that have a capacity of about 2Mb/s for interconnection of its entities.

In urban areas the mobile networks operate in small coverage areas, which has a direct impact on the volume of signalling exchanges between the mobile network entities. In this work, a configuration has been presented whereby the GSM system local entities could be interconnected over different levels of public MANs. The advantages of this interconnection strategy include:

(1) Fast signalling exchanges could be achieved between the GSM system entities.

(2) Sharing of the connection links with other networks could offer the mobile services at reduced prices.

(3) Fast signalling exchanges lead to a shorter connection set up times.

(4) Mobile radio customers could benefit from the services offered by the MANs such as the connectionless services.

(5) The proposed strategy is independent on the technology adopted over the air interface.

7.2 Recommendation for Further Work

* Practically, the GSM system operates with a limited number of traffic channels per radio cell. Simulation results have shown that the improvement in the access procedure, due to the developed protocol, has resulted in situations where the random access channel is used for smaller periods of time in transmitting the channel request messages. As a result, further improvement in the usage of the random channel can, therefore, be achieved if the channel is used for other purposes, such as short message transmission, during its idle periods. Utilisation of the random channel during its idle periods requires the knowledge of the status of channel, whether the channel is in the

idle state or busy state, by all the mobile stations that exist in the cell.

Controllers should continuously broadcast, on their common broadcast channels, the state of the random access channel. The idle signal is broadcast when all the traffic channels are busy. This idle signal then inhibits those who wish to transmit their channel request message and enables those wish to transmit their short messages.

* The disadvantage of the current MANs structure is that the stations which exist further away from the slot generator will suffer longer delay periods due to the fact that the stations that exist nearer to the slot generator are most likely to capture and reserve the free slots. This condition is acute in heavy traffic situations. In the final draft of the DQDB standard, bandwidth balancing was added to improve fairness. Bandwidth balancing places a restriction on the allowed number of reservations per station.

The disadvantage of this restriction scheme is that there will be situations when the upstream stations may have too many packets to transmit but are offered a limited number of time-slots and, at the same time, the downstream stations may be offered too many time-slots but have a smaller number of packets. This situation could be improved by a careful planing strategy of positioning the mobile radio entities over the MANs. If the mobile radio entities are allowed a number of reserved time-slots in each frame, then this protocol will not only guarantee the grade of services to the mobile users but also improve the efficiency of MAN by allowing the mobile entities to release those unneeded time-slots for use by those stations that exist further downstream to the mobile entities.

7.3 Summary

This chapter gives the performance of the developed protocols. Since the developed protocols are intended for the GSM system, which is already a standardised system and has entered the operation phase, the developed protocol for improving the access protocol and the protocol for inserting the data packets during the silence periods in the speech have been developed in such a way that causes a minimum disruption to the existing hardware and software of the GSM system.

In the future, the technology, standardisations and economic benefits will allow for the interfacing of mobile radio systems with broadband networks. The simulation results show some of the benefits that mobile radio systems, such as GSM, could achieve from such interfacing. Decentralisation of the control decisions could lead to less complexity of the MSCs and also to faster responses.

REFERENCES

(1) MacDoland, V. H.
AMPS: The cellular concept,
Bell System Tech. J., vol.58, no.1, pp 15-41, Jan. 1979.

(2) A guid to the Total Access Communications System (TACS),
Department of Trade & Industry , U.K. 1985.

(3) Macario, R. C. V.
Personal & mobile radio system,
Peter Peregrinus , U.K. 1991.

(4) Balston, D.M.
Pan-European cellular radio 1991 and all that,
Electronics & Communication Engineering Journal, pp 7-13, Feb., 1989.

(5) Special issue on ISDN,
Proc. IEEE, vol.79, no.2, Feb. 1991.

(6) Stallings, W.
ISDN and broadband ISDN,
Macmillan Publishing Company, 2nd ed., 1992.

(7) Special issue on circuit switching,
Proc. IEEE, vol.65, Sept. 1977.

(8) Tanenbaum, A. S.
Computer networks,
Prentice-Hall Inc. Englewood Cliffs, N. J. 1981.

(9) Special issue on packet communication,
Proc. IEEE, vol.66, Nov. 1978.

(10) Stallings, W.
Data and computer communications,
Macmillan Publishing Company, New York, 2nd ed., 1988.

(11) Gregg, W. D.
Analog and digital communications,
Wiley, New York, 1977.

(12) Carlson, A. B.
Communication system: an introduction to signals and noise in electrical communication,
McGraw-Hill Inc., 3rd ed., 1986.

(13) International Telecommunication Union
Integrated Service Digital Network (ISDN): general structure and service capabilities,
Blue Book, Geneva, 1989.

(14) Helgert, H.
Integrated Services Digital Networks: architecture, protocols and standards,
Addison-Wesley, Reading, MA, 1991.

(15) Pauwels, B.
A flexible customer premises network concept based on ATM principles,
IEEE GLOBCOM'88, pp 246-250, U.S.A. 1989.

(16) Handel, R.
Evoluation of ISDN towards B-ISDN,
IEEE Network, pp 7-13, Jan. 1989.

(17) Byrne, W. R., Kilm, T. A., Nelson, B. L. and Soneru, M. D.
Broadband ISDN technology and architecture,
IEEE Network, pp 23-28, Jan. 1989.

(18) Beau, O. and Pual, Jean-Lin.
Technical concepts in broadband ATM,
International J. of Digital and Analog Cabled Systems, vol.1, pp 207-211, 1988.

(19) Keiser, G. E.
Local area networks,
McGraw-Hill, Computer Science Series, 1989.

(20) Batorsky, D. V., Spears, D. R. and Tedesco, A.R.
The evolution of broadband network architectures,
IEEE GLOBCOM'88, pp367-373, 1988.

(21) Quinquis, JP., Lespagnol, A. François, J. and Gastaud, G.
Data services and LANs interconnection using ATM technique,
XIII International Switching Symposium, Proc. vol.II, pp 9-14, May 1990.

(22) Minzer, S. E.
Broadband ISDN and asynchronous transfer mode (ATM),
IEEE Commun. MAgazine, pp 17-24, Sept. 1989.

(23) Yokotani, T., Sato, H. and Nakatsuka, S.
A study on a performance improvement algorithm in DQDB MAN,
Computer Networks and ISDN, pp 1107-1117, Sept. 1993.

(24) Yap. Ma-Tit and Hutchison, D.
An emulator for evaluating DQDB performance,
Computer Networks and ISDN, pp 1177-1204, Sept. 1993.

(25) Newman, R. M., Budrikis, Z. L. and Hullet, J. L.
The QPSX MAN,
IEEE Commun. Magazine, vol.26, no.4, pp 20-28, April 1988.

(26) Special issue on Asynchronous Transfer Mode,
International J. of Digital and Analog Cabled Systems., vol.1, no.4, 1988.

(27) CCITT Recommendation I.311
B-ISDN general network aspects,
1990.

(28) Handel, R. and Huber, M. N.
Integrated broadband networks- an introduction to ATM-Based networks,
Addison-Wasley, 1991.

(29) Kawarasaki, M and Jabbari, B.
B-ISDN architecture and protocol,
IEEE J. on Select. Areas in Commun., vol.9, no.9, pp 1405-1415, Dec. 1991.

(30) Adams, J. L.
The virtual path identifier and its application for routing and priority of connectionless and connection oriented services,
International J. of Digital and Analog Cabled Systems., vol.1,no.1, pp 257-262, 1988.

(31) Murase, T., Suzuki, H., Sato, S. and Takeuchi, T.
A call admission control scheme for ATM networks using a simple quality estimate,
IEEE J. on Select. Areas in Commun., pp 1461-1470, Dec. 1991.

(32) Murase, A. and Imamura, K.
Idle-signal casting multiple access with collision detection (ICMA/CD) for land mobile radio,
IEEE Trans. Veh. Tech., vol.VT-36, pp 34-50, May 1987.

(33) Nuspl, P, Brown, K. E., Steenaart, W. and Ghlcopoulos, B.
Synchronization methods for TDMA,
Proc. IEEE, vol.65, no 3, pp 434-444, Mar. 1977.

(34) Rubin, I.
Message delays in FDMA and TDMA communication channels,
IEEE Trans. on Commun., vol.Com-27, no 5, pp 769-777, May 1979.

(35) Taub, H. and Schilling, D. L.
Principles of communication system,
McGraw-Hill Inc., 2nd ed., 1987.

(36) Carter, G. [Ed].
Local area networks, the principle of data communications,
ICL. Publication, London 1984.

(37) Clark, D. D., Pogran, K. T. and Read, D. P.
An introduction to local area networks,
Proc. IEEE, vol.66, pp 1497-1577, Nov. 1978.

(38) IEEE Network
Special issue on token-ring local area networks,
IEEE Network, vol.1, no.1, Jan. 1987.

(39) Kleinrock, L. and Tobagi, F. A.
Packet switching in radio channels: Part I - carrier sense multiple-access modes and their throughput-delay characteristics,
IEEE Trans. on Commun., vol.COM-23, no. 12, pp 1400-1416, Dec. 1975.

(40) Schwartz, M
Telecommunication networks: protocols, modelling and analysis,
Addison-Wesley Publishing company, 1987.

(41) Tobagi, F. A. and Hunt, V. B.
Performance analysis of carrier sense multiple access with collision detection,
North-Holland Publishing company, Computer Networks 4, pp 245-259, 1980.

(42) Tobagi, F. A. and Kleinrock, L.
Packet switching in radio channels: Part II - the hidden terminal problem in carrier sense multiple access and the busy tone solution,
IEEE Trans. on Commun., vol.COM-23, no. 12, pp 1417-1433, Dec. 1975.

(43) Murase, A. and Imamura, K.
Idle-signal casting multiple access with data slot reservation (ICMA/DR) packet radio communication,
IEEE Trans.Veh. Tech., vol.VT-38, no.2, pp 50-54, May 1989.

(44) GSM Recommondation 05.01

(45) GSM Recommondation 05.02

(46) Maloberti, A.
Radio transmission interface of the digital pan-European mobile system,
39th. IEEE Veh. Tech. Conf., pp 712-717, 1989.

(47) Abramson, N.
The ALOHA system-Another alternative for computer communications,
AFIPS Conf. Proc., Fall Joint Computer Conf., vol.37, pp 281-285 1970.

(48) Roberts, L. G.
ALOHA packet system with and without slots and capture,
Comput. Commun. Rev., vol.5, pp 28-42, April 1975.

(49) GSM Recommondation 04.08

(50) Jakes, W. C.
Microwave mobile communications,
John Wiley & Sons, New York, 1974.

(51) Lee, W. C. Y
Mobile cellular telecommunications systems,
McGraw-Hill, New York 1989.

(52) Oetting, J.
Cellular mobile radio: an emerging technology,
IEEE Commun. Magazine, pp10-15 1983.

(53) GSM Recommendation 05 Series.

(54) Hodges, M. R. L.
The GSM interface,
British Telecom. Tech. J. , vol. 8. no. 1 pp 31-43 Jan. 1990.

(55) Gardiner, J. G.
Second generation cordless (CT-2) telephony in the UK: telepoint service and the common air-interface,
Electronics and Commun. Eng. J., pp 71-78 April 1990.

(56) Mende, W. R.
Evaluation of a proposed handover algorithm for the GSM cellular system,
40th IEEE Veh. Tech. Conf., pp 264-269, 1990.

(57) Munoz-Rodriguez, D. and Cattermole, K. W.
Multiple criteria for hand-off in cellular mobile radio,
IEE Proc., vol. 134, Pt. F, no. 1, pp 85-88, Feb. 1987.

(58) Jayant, N. S., and Noll, P.
Digital coding of waveforms: principles and applications to speech and video,
Prentice-Hall International Inc., London UK 1984.

(59) Persons, J. D. and Gardiner, J. G.
Mobile communication systems,
Blake & Son, Glasgow 1989.

(60) Gürdenli, E., and Huish, P. W.
Propagation measurement and modelling for digital cellular radio systems,
British Telecom. Tech. J., vol. 8. no. 1 pp 44-56 Jan. 1990.

(61) Hodges, M. R. L., Jensen, S. A., and Tattersall, P. R.
Laboratory testing of digital cellular radio systems,
British Telecom. Tech. J., vol. 8. no. 1 pp 57-66 Jan. 1990.

(62) Lee, W. C. Y.
Mobile communications engineering,
McGraw-Hill, New York, 1982.

(63) Stein, S.
Fading channel issues in system engineering,
IEEE J. on Select. Areas in Commun., vol. SAC-5, no. 2 pp 68- , Feb. 1987.

(64) Raith, K. and Uddenfeldt, J.
Capacity of digital cellular TDMA systems,
IEEE Transactions On Veh. Tech., vol. 40. no. 2 pp 323-332 May 1991.

(65) D'Aria, G., Muratore, F. and Palestini, V.
Simulation and performance of the pan-European land mobile radio system,
IEEE Transactions On Veh. Tech., vol. 41. no. 2 pp 177-189 May 1992.

(66) D'Aria, G., Piermarini, R. and Zingarelli, V.
Fast adaptive equalizers for narrow-band TDMA mobile radio,
IEEE Transactions on Veh. Tech., vol. 40, no. 2, pp 392-404, May 1991.

(67) Proakis, J. G.
Digital communications,
McGraw-Hill, New York, 1983.

(68) Muratore, F. and Palestini, V.
Performance of a simulated narrow-band TDMA mobile radio system,
CSELT Technical Reports, vol. XVII, no. 2, pp 139-145, April 1989.

(69) Natvig, J. E.
Speech coding in the pan-European digital mobile radio system,
Speech Commun., Special Issue no.1 1988.

(70) Vary, P., Hellwig, K., Hofmann, R., Sluyter, R. J., Galand, C. and Rosso, M.
Speech codec for the European mobile radio system,
International Conf. on Acoustics Speech and Signal Processing, pp 227-230 1988.

(71) GSM Recommendation 06.10
Full Rate Speech Transcoding.

(72) Coleman, A., Gleiss, N., Sotscheck, J., Usai, P. and Scheuermann, H.
Subjective performance evaluation of the RPE-LTP codec for the pan-European cellular digital mobile radio system,
IEEE GLOBCOM'89, pp 1075-1079, Nov. 1989.

(73) Haykin, S.
Communication systems,
Wiley, New York, 2nd ed., 1983.

(74) Blahut, R.
Theory and practice of error control codes,
Addison-Wesley, 1983.

(75) Lin, S.and Costello, D.
Error control coding,
Prentice-Hall, 1982.

(76) Southcott, C. B., Freeman, D., Cosier, G., Sereno, D., Van der Krogt, A., Gilloire, A. and Braun, J. H.
Voice control of the pan-European digital mobile radio system,
IEEE Globcom'89, pp 1070-1074, Nov. 1989.

(77) Natvig, J. E., Hansen, S. and Jorge de Brito.
Speech processing in the pan-European digital mobile radio system (GSM) - system overview,
IEEE Globcom'89, pp 1060-1064, Nov. 1989.

(78) GSM Recommendation 05.03
Channel coding.

(79) Hellwig, K., Vary, P., Massaloux, D., Petit, J. P., Galand, C. and Rosso, H.
Speech codec for the European mobile radio system,
IEEE GLOBCOM'89, pp 1065-1069, Nov. 1989.

(80) Natvig, J. E.
Evaluation of six medium bit-rate coders for the pan-European digital mobile radio system,
IEEE J. on Select. Area in Commun., vol.6, pp 324-331, Feb. 1988.

(81) Goodman, D. J.
Second generation wireless information networks,
IEEE Trans. on Veh. Tech., vol.40, pp 366-374, May 1991.

(82) Lee, L. H. C. and Farrell, P. G.
Concatenated coding and interleaving for half-rate GSM digital mobile radio system,
Electronics Letters, vol. 27, no. 14, pp 1274-1276, July, 1991.

(83) Goodman, D. J. and Wei, S. X.
Efficiency of packet reservation multiple access,
IEEE Trans. on Veh. Tech., vol.40, pp 170-176, Feb. 1991.

(84) Gruber, J. G. and Strawczynski, L.
Subjective effects of variable delay and speech clipping in dynamically managed voice system,
IEEE Trans. on Commun., vol.Com-33, no.8, pp 801-808, Aug. 1985.

(85) Goviello, G. J.
Comparative discussion of circuit vs. packet switched voice,
IEEE Trans. on Coommun., vol.Com-27, no.8, pp 1153-1159, Aug. 1979.

(86) Thomas, R., Mouly, M. and Gilbert, H.
Performance evaluation of the channel organization of the European digital mobile communication system,
38th IEEE Veh. Tech. Conf., pp 38-43, 1988.

(87) Tekinay, S. and Jabbari, B.
Handover and channel assignment in mobile cellular networks,
IEEE Commun. Magazine, pp 42-46, Nov. 1991.

(88) Groves, I.S.
Personal mobile communications - a vision of the future,
British Telecom. Tech. J., vol.8, no.1, pp 7-11, Jan. 1990.

(89) McFarlane, D.A. and Chia, S.T.S.
Microcellular mobile radio systems,
British Telecom. Tech. J., vol.8, no.1, pp 79-84, Jan. 1990.

(90) Burgin, J. and Dorman, D.
Broadband ISDN resource management: the role of virtual paths,
IEEE Commun. Magazine, vol.29, no.9, pp 44-48, Sept. 1991.

(91) Aoyama, T., Tokizawa, I. and Sato, Ken-Ichi
Introduction strategy and technologies for ATM VP-based broadband networks,
IEEE J. on Select. Areas in Commun.., vol.10, no.9, pp 1434-1447, Dec. 1992.

(92) Bonatti, M., Gobbi, R., Dammicco, G., Mocci, U., Lippi, M., Montagna, M., Polese, P. and Roso, G.
B-ISDN economical evaluation in metropolitan areas,
IEEE J. on Select. Areas in Commun., vol.10, no.9, pp 1489-1502, Dec. 1992.

(93) Mollenauer, J.F.
Metropolitan area networks and ATM technology,
International J. of Digital and Analog Cabled Systems, pp 223-228, 1988.

(94) Byrne, W. R., Luderer, G. W. R., Clapp, G, Nelson, B. L. and Kafka, H. J.
Evolution of metropolitan area networks to broadband ISDN,
IEEE Commun. Magazine, pp 69-82, Jan. 1991.

(95) Gallagher, I., Ballance, J. and Adams, J.
The application of ATM techniques to the local network,
British Telecom. Tech. J., vol.7, no.2, pp 151-160, April 1989.

(96) Sykas, E. and Theologou, M.E.
Numbering and addressing in IBCN for mobile communications,
Proc. IEEE, vol.79, no.2, pp 230-241, Feb. 1991.

(97) Bisdikian, C.
A queueing model with applications to bridges and the DQDB (IEEE 802.6) MAN,
Computer Networks and ISDN Systems, pp 1279-1289, 1993.

(98) Davidson, R.P. and Muller, N.J.
Internetworking LANs: operation, design and management,
Artech House Inc., Norwood, MA. 1992.

(99) Feldmann, M. and Rissen, J.P.
GSM network system and overall system integration,
Electrical Commun., 2nd Quarter, pp 141-153, 1993.

(100) Wizgall, M., Weiss, W. and Stier, W.
The ISDN approach for mobile radio,
XIII international Switching Symposium Proc., vol.VI, pp 161-167, 1990.

(101) Appleby, M.S. and Garrett, J.
The cellnet cellular radio network: part 1 - general system overview,
British Telecom. Eng., vol.4, pp 62-69, July 1985.

(102) Ballard, M. and Issenmann, E.
Cellular mobile radio as an intelligent network application,
Electrical Commun., vol.63, no.4, pp 389-399, 1989.

(103) Grillo, D. and MacNamee, G.
European Perspectives on third generation personal communication systems,
40th IEEE Veh. Tech. Conf., pp 135-140, 1990.

(104) Rashidzadeh and Mabey, P.
Prospects for a common air interface for cellular and cordless systems,
5th International Conf. on Mobile Radio and Personal Commun., pp 162-166, Dec. 1989.

(105) Grillo, D., MacNamee, R. J. G. and Rashidzadeh, B.
Towards third generation mobile radio systems: a European possible transition path,
Computer Networks and ISDN Systems, pp 947-961, 1993.

(106) Grillo, D. and Chia, S.T.S.
UMTS-mobile communication beyond the year 2000: requirements, architecture and options,
Electronics & Commun. Eng. J., pp 331-340, Oct. 1992.

(107) Rahnema, M.
Overview of the GSM system and personal architecture,
IEEE Commun. Magazine, pp 92-100, April 1993.

(108) Dechaux, C. and Scheller, R.
What are GSM and DCS,
Electrical Commun., 2nd Quarter, pp 118-127, 1993.

(109) Torstensson, A.
Technology for trunked mobile data services,
Communication International, pp 75-76, Nov. 1989.

(110) Campanella, S. J.
Digital speech interpolation,
COMSAT Tech. Rev., no.6, pp 127-158, 1976.

(111) Ilyas, M. and Mouftah, H. T.
Performance evaluation of computer communications networks,
IEEE Commun. Magazine, vol.23, pp 18-29, April 1985.

(112) Shanmugan, K. S.
An update on software packages for simulation of computer systems {links),
IEEE J. on Select. Areas in Commum., vol.6, no.1, pp 5-12, Jan. 1988.

(113) Kurose, J. F. and Mouftah, H. T.
Computer aided modelling, analysis, and design of communication networks,
IEEE J. on Select. Areas in Commum., vol.6, no.1, pp 130-145, Jan. 1988.

(114) Pritsker, A. A. B.
Introduction to simulation and SLAM II,
John Wiley & Sons, 3rd ed., New York, 1986.

(115) Tafazolli, R., Richharia, M. and Evens, B. G.
Multiple access sheme for future land mobile communication systems,
IEE, 5th. International Conf. on Mobile Radio and Personal Communi., Warwick University, Coventry, U.K., Dec. 1989.

(116) Tafazolli, R., Richharia, M. and Evens, B. G.
A Flexible multiple access scheme for mobile communication,
IEE Colloquium, London, U.K., 1992.

APPENDICES

```
****************************************************************
THIS PROGRAM SIMULATES THE DEVELOPED ACCESS
PROTOCOL FOR THE GSM SYSTEM
****************************************************************

 DIMENSION  NSET (2E+06)
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5;KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER  (MXMSG=250)
 PARAMETER  (MXPOUT=6)
 INCLUDE  ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON QSET(2E + O6)
 EQUIVALENCE (NSET(1), QSET(1))
 NNSET = 2E + O6
 NCRDR = 5
 NPRNT = 6
 NTAPE = 7
 CALL SLAM
 STOP
 END
```

```
*************************************************************
*************************************************************

SUBROUTINE EVENT (I)
        GO TO (1,2,3,4), I
 1      CALL ARRIVAL
        RETURN
 2      CALL SENSING
        RETURN
 3      CALL REATTEMPT
        RETURN
 4      CALL SUCCESS
        RETURN
        END

*************************************************************
*************************************************************

SUBROUTINE INTLC

 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER (MXMSG=250)
 PARAMETER (MXPOUT=6)
 INCLUDE ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
 COMMON/UCOM1/CIRRAD,VULPRD,APKTTIM,NUMGA,OFERD,PKTTIM,MMPAK,
1 VLO,VHI

          READ(NCRDR,*)CIRRAD,OFERD
```

```
CALL COLCT(CIRRAD,1)
CALL COLCT(OFERD,1)
PKTTIM=173.26
VULPRD=35.
APKTTIM=138.26
CALL GO5CCF
XX(1)=0.
NUMGA =0
NARCU =0
MMPAK=0
CALL SCHDL(1,0.,ATRIB)
RETURN
END
```

****

****

SUBROUTINE ARRIVAL

```
SOL=PKTTIM/OFERD
NTH=GO5DAF(1.,9.)
TIMB=NPSSN(SOL,NTH)
CALL SCHDL(1,TIMB,ATRIB)
IF((NARCU-NUMGA).GE.INT(OFERD)) RETURN
ATRIB(1)=TNOW
CARVT=TNOW
NARCU=NARCU+1
ACC = FLOAT(NARCU)
CALL COLCT(ACC,3)
IF (CARVT.EQ.0.) THEN
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(1,ATRIB)
SVULPRD =2. *VULPRD
CALL SCHDL(2,SVULPRD,ATRIB)
ELSE
SUBART=MOD(CARVT,(2.*VULPRD))
IF (SUBART.GT.VULPRD) THEN
UTML=PKTTIM-SUBART
CALL SCHDL(3,UTML,ATRIB)
ELSE
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(1,ATRIB)
IF(SUBART.EQ.0.) THEN
CVULPRD=2.*VULPRD
CALL SCHDL(2,VULPRD,ATRIB)
ELSE
BOSLOT=(2.*VULPRD) - SUBART
```

```
        CALL SCHDL(2,BOSLOT,ATRIB)
        ENDIF
        ENDIF
        ENDIF
        RETURN
        END
```

```
**************************************************************
**************************************************************
```

SUBROUTINE SENSING

```
        IF (NNQ(1).EQ.O.OR.XX(1).EQ.1.) RETURN
        MMPAK=0
        L=1
60      CALL COPY(L,1,ATRIB)
        J=NNQ(1)
        LL=ATRIB(2)
        NEXT=MMLE(1)
40      IF(NEXT.EQ.0) GOTO 90
        IF(J.EQ.L) GOTO 35
        CALL COPY(-NEXT,1,ATRIB)
        IF(ATRIB(2).EQ.LL) THEN
        MMPAK=MMPAK+1
        PAK=FLOAT(MMPAK)
        CALL COLCT(PAK,4)
        ENDIF
35      NEXT=NPRED(NEXT)
        J=J-1
        GOTO 40
90      L=L+1
        IF(L.GT.NNQ(1)) GOTO 100
100     IF(MMPAK.GE.2) THEN
        GOTO 105
        ELSE
        GOTO 150
        ENDIF
        RETURN
105     LM=NNQ(1)
        DO 120 I=1,LM
        CALL RMOVE(1,1,ATRIB)
        AHI=(GO5DAF(1.,4.))*PKTTIM
        CALL SCHDL(3,AHI,ATRIB)
120     CONTINUE
        RETURN
150     LJ=NNQ(1)
        DO 170 K=1,LJ
```

```
          CALL RMOVE(1,1,ATRIB)
          IF(K.EQ.1) THEN
          XX(1) =1.
          CALL SCHDL(4,APKTTIM,ATRIB)
          ELSE
          BHI = (GO5DAF(1.,4.))*PKTTIM
          CALL SCHDL(3,BHI,ATRIB)
          ENDIF
170       CONTINUE
          RETURN
          END

*****************************************************************
*****************************************************************
```

SUBROUTINE REATTEMPT

```
          SART =TNOW
          DVULPRD= (2.*VULPRD)
          BART =MOD(SART,DVULPRD)
          IF(BART.GT.VULPRD) THEN
          UTML = PKTTIM -SUBART
          CALL SCHDL(3,UTML,ATRIB)
          ELSE
          ATRIB(2)=INT(GO5DAF(1.,30.))
          CALL FILEM(1,ATRIB)
          IF(BART.EQ.O.) THEN
          BVULPRD =2.*VULPRD
          CALL SCHDL(2,BVULPRD,ATRIB)
          ELSE
          DOSLOT = (2.*VULPRD) - BART
          CALL SCHDL(2,DOSLOT,ATRIB)
          ENDIF
          ENDIF
          RETURN
          END
*****************************************************************
*****************************************************************
```

SUBROUTINE SUCCESS

```
          XX(1)=0.
          TTGA= TNOW-ATRIB(1)
          CALL COLCT(TTGA,5)
          NUMGA = NUMGA + 1
          RETURN
          END
```

```
******************************************************************
THIS PROGRAM SIMULATES THE VOICE TRANSMISSION
OVER THE GSM SYSTEM
******************************************************************

 DIMENSION  NSET (2E+06)
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER  (MXMSG=250)
 PARAMETER  (MXPOUT=6)
 INCLUDE  ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON QSET(2E + O6)
 EQUIVALENCE (NSET(1), QSET(1))
 NNSET = 2E + O6
 NCRDR = 5
 NPRNT = 6
 NTAPE = 7
 CALL SLAM
 STOP
 END


******************************************************************
******************************************************************

SUBROUTINE  EVENT  (I)
        GO TO (1,2,3,4,5,6,7,8), I
1       CALL ARRIVAL
        RETURN
```

```
2         CALL SENSING
          RETURN
3         CALL REATTEMPT
          RETURN
4         CALL SUCCESS
          RETURN
5         CALL TRANSMISSION
          RETURN
6         CALL ALLOCATION
          RETURN
7         CALL END SERVICE
          RETURN
8         CALL SUBS-TRANSMISSION
          RETURN
          END
```

****

****

SUBROUTINE INTLC

```
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER (MXMSG=250)
 PARAMETER (MXPOUT=6)
 INCLUDE ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON/UCOM1/CIRRAD,VULPRD,APKTTIM,NUMGA,OFERD,NOSACC,
1NTSLTS,NRFS,NTOTAL,MSTRF,MNU,NOSUBF,FRAME,PKTTIM,MTEST,KSFIL,M
2MPAK,NPAK,NPAK2,NPAK3.NPAK4,SSAMPLE
```

```
      READ(NCRDR,*)CIRRAD,OFERD
      CALL COLCT(CIRRAD,1)
      CALL COLCT(OFERD,2)
      PKTTIM=173.26
      FRAME=1386.
      VULPRD=35.
      APKTTIM=138.26
      SSAMPLE=6006.
      CALL GO5CCF
      NTSLTS=8
      NRFS=3
      NTOTAL=NTSLTS*NRFS
      MSTRF=NTOTAL+1
      NRFF=MSTRF+1
      DO 10 K=1,NRFS
      CALL FILEM(MSTRF,ATRIB)
10    CONTINUE
      NN=1
      DO 20 I=1,NRFS
      NN1=NN*8 - 8
      DO 30 J=1,8
      NN2=NN1+J
      XX(NN2)=0.
      CALL FILEM(NRFF,ATRIB)
30    CONTINUE
      NN=NN+1
      NRFF=NRFF+1
20    CONTINUE
      XX(MSTRF) = 0.
      NUMGA =0
      NOSACC=0
      NARCU =0
      MTEST=0
      MNU=0
      NOSUBF=0
      KSFIL = MSTRF+NRFS+1
      NPAK1=0
      NPAK2=0
      NPAK3=0
      NPAK4=0
      DO 25 NMK=30,53
      DO 36 KK=1,900
      CALL FILEM(NMK,ATRIB)
36    CONTINUE
25    CONTINUE
      CALL SCHDL(1,0.,ATRIB)
      RETURN
```

```
END
```

```
************************************************************
************************************************************
```

SUBROUTINE ARRIVAL

```
SOL=PKTTIM/OFERD
NTH=GO5DAF(1.,9.)
TIMB=NPSSN(SOL,NTH)
CALL SCHDL(1,TIMB,ATRIB)
IF((NARCU-NUMGA).GE.INT(OFERD)) RETURN
ATRIB(1)=TNOW
CARVT=TNOW
NARCU=NARCU+1
ACC=FLOAT(NARCU)
CALL COLCT(ACC,3)
IF(CARVT.EQ.O.) THEN
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(KSFIL,ATRIB)
SVULPRD=2.*VULPRD
CALL SCHDL(2,SVULPRD,ATRIB)
ELSE
SUBART =MOD(CARVT,VULPRD
IF(SUBART.GT.VULPRD) THEN
UTML = PKTTIM -SUBART
CALL SCHDL(3,UTML,ATRIB)
ELSE
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(KSFIL,ATRIB)
IF (SUBART.EQ.O.)THEN
CVULPRD=2.*VULPRD
CALL SCHDL(2,CVULPRD,ATRIB)
ELSE
BOSLOT = (2.*VULPRD) - SUBART
CALL SCHDL(2,BOSLOT,ATRIB)
ENDIF
ENDIF
ENDIF
RETURN
END
```

```
************************************************************
************************************************************
```

SUBROUTINE SENSING

```
        IF(NNQ(KSFIL).EQ.O.OR.XX(MSTRF).EQ.1.) RETURN
        MMPAK=0
        L=1
60      CALL COPY(L,KSFIL,ATRIB)
        J=NNQ(KSFIL)
        LL=ATRIB(2)
        NEXT=MMLE(KSFIL)
40      IF(NEXT.EQ.O.) GOTO 90
        IF(J.EQ.L) GOTO 35
        CALL COPY(-NEXT,KSFIL,ATRIB)
        MMPAK=MMPAK+1
        PAK=FLOAT(MMPAK)
        CALL COLCT(PAK,4)
        END IF
35      NEXT=NPRED(NEXT)
        J=J-1
        GOTO 40
90      L=L+1
        IF(L.GT.NNQ(KSFIL)) GOTO 100
100      IF(MMPAK.GE.2) THEN
        GOTO 105
        ELSE
        GOTO 150
        ENDIF
        RETURN
105     LM=NNQ(KSFIL)
        DO 120 I=1,LM
        CALL RMOVE(1.KSFIL,ATRIB)
        AHI=(GO5DAF(1.,4,))*PKTTIM
        CALL SCHDL(3,AHI,ATRIB)
120     CONTINUE
        RETURN
150     LJ=NNQ(KSFIL)
        DO 170 K=1,LJ
        CALL RMOVE(1,KSFIL,ATRIB)
        IF(K.EQ.1) THEN
        XX(MSTRF)=1.
        CALL SCHDL(4,APKTTIM,ATRIB)
        ELSE
        BHI=(GO5DAF(1.,4.))*PKTTIM
        CALL SCHDL(3,BHI,ATRIB)
        ENDIF
```

```
170     CONTINUE
        RETURN
        END
```

****
****

SUBROUTINE REATTEMPT

```
        SART=TNOW
        DVULPRD=2.*VULPRD
        BART=MOD(SART,DVULPRD)
        IF(BART.GT.VULPRD) THEN
        UTML = PKTTIM-SUBART
        CALL SCHDL(3,UTML,ATRIB)
        ELSE
        ATRIB(2)=INT(GO5DAF(1.,30.))
        CALL FILEM(KSFIL,ATRIB)
        IF(BART.EQ.O.) THEN
        BVULPRD=2.*VULPRD
        CALL SCHDL(2,BVULPRD,ATRIB)
        ELSE
        DOSLOT=(2.*VULPRD)-BART
        CALL SCHDL(DOSLOT,ATRIB)
        END IF
        ENDIF
        RETURN
        END
```

****
****

SUBROUTINE SUCCESS

```
        NOSUBF=NOSUBF+1
        XX(MSTRF)=0.
        IF(NOSUBF.GT.NTOTAL) RETURN
        TTGA=TNOW - ATRIB(1)
        CALL COLCT(TTGA,5)
        NUMGA=NUMGA+1
        SCHTIM = (SSAMPLE -TTGA)
        IF(SCHTIM.LE.O.) THEN
        CALL SCHDL(6,O.,ATRIB)
        ELSE
        CALL SCHDL(6,SCHTIM,ATRIB)
        END IF
```

```
      RETURN
      END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE TRANSMISSION

```
      LL=INT(ATRIB(3))
      XX(LL)=1.
      CALL SCHDL(7,PKTTIM,ATRIB)
      RETURN
      END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE ALLOCATION

```
      K=1
      NORFF=MSTRF+1
10    IF(NNQ(NORFF).EQ.O) GOTO 30
      MTEST = MTEST+1
      IF(MTEST.GT.8) MTEST=1
      CALL RMOVE(1,NORFF,ATRIB)
      ATRIB(4)=TNOW
      ATRIB(5)=0
      NN=((K*8) - 8+MTEST)
      KRAM=INT(GO5DAF(1.,9.))
      TSPURTL=EXPON(402402.,KRAM)
      SPPTN=(TSPURTL/SSAMPLE)
      CALL COLCT(SPPTN,11)
      NUPATS=INT(SPPTN*4.)
      IF(NUPATS.LT.2) NUPATS=2
      DO 35 J=1,NUPATS
      CALL FILEM(NN,ATRIB)
35    CONTINUE
      ATRIB(3)=FLOAT((K*8)-8 + MTEST)
      TOARV=MOD(TNOW,FRAME)
      YSLOT=FLOAT(MTEST)*PKTTIM
      YTIMDEF=YSLOT - TOARV
      IF(YTIMDEF.LE.O.) THEN
      NPAK1=NPAK1+1
      PAK1=FLOAT(NPAK1)
      CALL COLCT(PAK1,7)
      YREQTIM=FRAME - PKTTIM - ABS(YTIMDEF)
```

```
      CALL SCHDL(5,YREQTIM,ATRIB)
      ELSE IF (YTIMDEF.LT.PKTTIM) THEN
      NPAK2=NPAK2+1
      PAK2=FLOAT(NPAK2)
      CALL COLCT(PAK2,8)
      YREQTIM1=(YSLOT-PKTTIM)+(FRAME - TOARV)
      CALL SCHDL(5,YREQTIM1,ATRIB)
      ELSE IF(YTIMDEF.EQ.PKTTIM) THEN
      NPAK3=NPAK3+1
      PAK3=FLOAT(NPAK3)
      CALL COLCT(PAK3,9)
      CALL SCHDL(5,0.,ATRIB)
      ELSE
      NPAK4=NPAK4+1
      PAK4=FLOAT(NPAK4)
      CALL COLCT(PAK4,10)
      YREQTIM2=(YSLOT-PKTTIM-TOARV)
      CALL SCHDL(5,YREQTIM2,ATRIB)
      END IF
      RETURN
30    K=K+1
      NORFF=NORFF+1
      IF(K.GT.NNQ(MSTRF)) GOTO 100
      GOTO 10
100   RETURN
      END
```

****
****

SUBROUTINE END SERVICE

```
      KK=INT(ATRIB(3))
      XX(KK)=0.
      IF(NNQ(KK).EQ.O) RETURN
      IF(NNQ(KK).GT.1) THEN
      ATRIB(5)=ATRIB(5) +1
      CVB=ATRIB(5)
      CALL COLCT(CVB,12)
      CALL RMOVE(1,KK,ATRIB)
      REFRAME=FRAME-PKTTIM
      ATRIB(3)=FLOAT(KK)
      CALL SCHDL(5,REFRAME,ATRIB)
      ELSE
      ENTIME=TNOW-ATRIB(4)
      CALL COLCT(ENTIME,6)
      CALL RMOVE(1,KK,ATRIB)
```

```
      ATRIB(5)=ATRIB(5)+1
      DDV=ATRIB(5)
      CALL COLCT(DDV,12)
      LRAM=INT(GO5DAF(1.,9.))
      SPERD=EXPON(67567.LRAM)
      SETRM=(SPERD + SSAMPLE)
      ATRIB(3)=FLOAT(KK)
      CALL SCHDL(8,SETRM,ATRIB)
      END IF
      RETURN
      END
```

****

****

SUBROUTINE SUBS-TRANSMISSION

```
      MM=INT(ATRIB(3))
      IF(MM.LE.8) THEN
      LL=MM
      J=1
      ELSE IF(MM.LE.16.) THEN
      LL=MOD(MM,16)
      J=2
      IF(LL.EQ.O) LL=16
      LL=LL-8
      ELSE
      LL=MOD(MM,24)
      J=3
      IF(LL.EQ.0) LL=24
      LL=LL-16
      END IF
      ATRIB(4)=TNOW
      JJ=((J*8) - 8 +LL)
      JRAM =INT(GO5DAF(1.,9.))
      VTSPURTL=EXPON(402402.,JRAM)
      CSPPTN=(VTSPURTL/SSAMPLE)
      CALL COLCT(CSPPTN,11)
      MUPATS=INT(CSPPTN*4.)
      IF(MUPATS.LT.2) MUPATS=2
      DO 45 MJ=1,MUPATS
      CALL FILEM(JJ,ATRIB)
45    CONTINUE
      ATRIB(3)=FLOAT((J*8)-8 +LL)
      BTOARV=MOD(TNOW,FRAME)
      BYSLOT=FLOAT(LL)*PKTTIM
      BYTIMDEF=BYSLOT-BTOARV
```

```
IF(BYTIMDEF.LE.O.) THEN
NPAK1=NPAK1+1
PAK1=FLOAT(NPAK1)
CALL COLCT(PAK1,7)
BYREQTIM=FRAME-PKTTIM-ABS(BYTIMDEF)
CALL SCHDL(5,BYREQTIM,ATRIB)
ELSE IF(BYTIMDEF.LT.PKTTIM) THEN
NPAK2=NPAK2+1
PAK2=FLOAT(NPAK2)
CALL COLCT(PAK2,8)
BYREQTIM1=(BYSLOT-PKTTIM)+(FRAME-BTOARV)
CALL SCHDL(5,BYREQTIM1,ATRIB)
ELSE IF (BYTIMDEF.EQ.PKTTIM) THEN
NPAK3=NPAK3+1
PAK3=FLOAT(NPAK3)
CALL COLCT(PAK3,9)
CALL SCHDL(5,0.,ATRIB)
ELSE
NPAK4=NPAK4+1
PAK4=FLOAT(NPAK4)
CALL COLCT(PAK4,10)
BYREQTIM2 =(BYSLOT-PKTTIM-BTOARV)
CALL SCHDL(5,BYREQTIM2,ATRIB)
END IF
RETURN
END
```

```
*****************************************************************
        THIS PROGRAM SIMULATES THE  COMBINED
            TRANSMISSION OF SPEECH AND DATA
*****************************************************************

 DIMENSION  NSET (2E+06)
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$  KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$  KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,
   KVCVCO=12,
$  KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
   KVSLEN=5,
$   KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$   KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$  KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$  KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
   KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$  KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$  KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$  KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER  (MXMSG=250)
 PARAMETER  (MXPOUT=6)
 INCLUDE  ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON QSET(2E + O6)
 EQUIVALENCE (NSET(1), QSET(1))
 NNSET = 2E + O6
 NCRDR = 5
 NPRNT = 6
 NTAPE = 7
 CALL SLAM
 STOP
 END

*****************************************************************
*****************************************************************
SUBROUTINE  EVENT  (I)
         GO TO (1,2,3,4,5,6,7,8,9,10), I
 1       CALL ARRIVAL
         RETURN
```

```
2          CALL SENSING
           RETURN
3          CALL REATTEMPT
           RETURN
4          CALL SUCCESS
           RETURN
5          CALL TRANSMISSION
           RETURN
6          CALL ALLOCATION
           RETURN
7          CALL END SERVICE
           RETURN
8          CALL SUBS-TRANSMISSION
           RETURN
9          CALL DATA START
           RETURN
10         CALL DATA END
           RETURN
           END
```

****
****

SUBROUTINE INTLC

```
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER (MXMSG=250)
 PARAMETER (MXPOUT=6)
 INCLUDE ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
 COMMON/UCOM1/CIRRAD,VULPRD,APKTTIM,NUMGA,OFERD,NOSACC,
```

```
1NTSLTS,NRFS,NTOTAL,MSTRF,MNU,NOSUBF,FRAME,PKTTIM,MTEST,KSFIL,M
2MPAK,NPAK1,NPAK2,NPAK3.NPAK4,SSAMPLE

        READ(NCRDR,*)CIRRAD,OFERD
        CALL COLCT(CIRRAD,1)
        CALL COLCT(OFERD,2)
        PKTTIM=173.26
        FRAME=1386.
        VULPRD=35.
        APKTTIM=138.26
        CALL GO5CCF
        NTSLTS=8
        NRFS=3
        NTOTAL=NTSLTS*NRFS
        MSTRF=NTOTAL+1
        NRFF=MSTRF+1
        DO 10 K=1,NRFS
        CALL FILEM(MSTRF,ATRIB)
10      CONTINUE
        NN=1
        DO 20 I=1,NRFS
        NN1=NN*8 - 8
        DO 30 J=1,8
        NN2=NN1+J
        XX(NN2)=0.
        CALL FILEM(NRFF,ATRIB)
30      CONTINUE
        NN=NN+1
        NRFF=NRFF+1
20      CONTINUE
        XX(MSTRF) = 0.
        NUMGA =0
        NARCU =0
        MTEST=0
        NOSUBF=0
        KSFIL = MSTRF+MSTRF+NRFS+1
        NPAK1=0
        NPAK2=0
        NPAK3=0
        NPAK4=0
        DO 25 NMK=30,53
        DO 36 KK=1,900
        CALL FILEM(NMK,ATRIB)
36      CONTINUE
25      CONTINUE
        CALL SCHDL(1,0.,ATRIB)
        RETURN
        END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE ARRIVAL

```
SOL=PKTTIM/OFERD
NTH=GO5DAF(1.,9.)
TIMB=NPSSN(SOL,NTH)
CALL SCHDL(1,TIMB,ATRIB)
IF((NARCU-NUMGA).GE.INT(OFERD)) RETURN
ATRIB(1)=TNOW
CARVT=TNOW
NARCU=NARCU+1
ACC=FLOAT(NARCU)
CALL COLCT(ACC,3)
IF(CARVT.EQ.O.) THEN
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(KSFIL,ATRIB)
SVULPRD=2.*VULPRD
CALL SCHDL(2,SVULPRD,ATRIB)
ELSE
SUBART =MOD(CARVT,VULPRD)
IF(SUBART.GT.VULPRD) THEN
UTML = PKTTIM -SUBART
CALL SCHDL(3,UTML,ATRIB)
ELSE
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(KSFIL,ATRIB)
IF (SUBART.EQ.O.)THEN
CVULPRD=2.*VULPRD
CALL SCHDL(2,CVULPRD,ATRIB)
ELSE
BOSLOT = (2.*VULPRD) - SUBART
CALL SCHDL(2,BOSLOT,ATRIB)
ENDIF
ENDIF
ENDIF
RETURN
END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE SENSING

```
IF(NNQ(KSFIL).EQ.O.OR.XX(MSTRF).EQ.1.) RETURN
MMPAK=0
```

```
          L=1
60        CALL COPY(L,KSFIL,ATRIB)
          J=NNQ(KSFIL)
          LL=ATRIB(2)
          NEXT=MMLE(KSFIL)
40        IF(NEXT.EQ.O.) GOTO 90
          IF(J.EQ.L) GOTO 35
          CALL COPY(-NEXT,KSFIL,ATRIB)
          MMPAK=MMPAK+1
          PAK=FLOAT(MMPAK)
          CALL COLCT(PAK,4)
          END IF
35        NEXT=NPRED(NEXT)
          J=J-1
          GOTO 40
90        L=L+1
          IF(L.GT.NNQ(KSFIL)) GOTO 100
100        IF(MMPAK.GE.2) THEN
          GOTO 105
          ELSE
          GOTO 150
          ENDIF
          RETURN
105       LM=NNQ(KSFIL)
          DO 120 I=1,LM
          CALL RMOVE(1.KSFIL,ATRIB)
          AHI=(GO5DAF(1.,4,))*PKTTIM
          CALL SCHDL(3,AHI,ATRIB)
120       CONTINUE
          RETURN
150       LJ=NNQ(KSFIL)
          DO 170 K=1,LJ
          CALL RMOVE(1,KSFIL,ATRIB)
          IF(K.EQ.1) THEN
          XX(MSTRF)=1.
          CALL SCHDL(4,APKTTIM,ATRIB)
          ELSE
          BHI=(GO5DAF(1.,4.))*PKTTIM
          CALL SCHDL(3,BHI,ATRIB)
          ENDIF
170       CONTINUE
          RETURN
          END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE REATTEMPT

```
SART=TNOW
DVULPRD=2.*VULPRD
BART=MOD(SART,DVULPRD)
IF(BART.GT.VULPRD) THEN
UTML = PKTTIM-SUBART
CALL SCHDL(3,UTML,ATRIB)
ELSE
ATRIB(2)=INT(GO5DAF(1.,30.))
CALL FILEM(KSFIL,ATRIB)
IF(BART.EQ.O.) THEN
BVULPRD=2.*VULPRD
CALL SCHDL(2,BVULPRD,ATRIB)
ELSE
DOSLOT=(2.*VULPRD)-BART
CALL SCHDL(DOSLOT,ATRIB)
END IF
ENDIF
RETURN
END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE SUCCESS

```
NOSUBF=NOSUBF+1
XX(MSTRF)=0.
IF(NOSUBF.GT.NTOTAL) RETURN
TTGA=TNOW - ATRIB(1)
CALL COLCT(TTGA,5)
NUMGA=NUMGA+1
SCHTIM = (SSAMPLE -TTGA)
IF(SCHTIM.LE.O.) THEN
CALL SCHDL(6,O.,ATRIB)
ELSE
CALL SCHDL(6,SCHTIM,ATRIB)
END IF
RETURN
END
```

```
****************************************************************
****************************************************************
```

SUBROUTINE TRANSMISSION

```
        LL=INT(ATRIB(3))
        XX(LL)=1.
        CALL SCHDL(7,PKTTIM,ATRIB)
        RETURN
        END
```

```
****************************************************************
****************************************************************
```

SUBROUTINE ALLOCATION

```
        K=1
        NORFF=MSTRF+1
10      IF(NNQ(NORFF).EQ.O) GOTO 30
        MTEST = MTEST+1
        IF(MTEST.GT.8) MTEST=1
        CALL RMOVE(1,NORFF,ATRIB)
        ATRIB(4)=TNOW
        ATRIB(5)=0
        NN=((K*8) - 8+MTEST)
        KRAM=INT(GO5DAF(1.,9.))
        TSPURTL=EXPON(402402.,KRAM)
        SPPTN=(TSPURTL/SSAMPLE)
        CALL COLCT(SPPTN,11)
        NUPATS=INT(SPPTN*4.)
        IF(NUPATS.LT.2) NUPATS=2
        DO 35 J=1,NUPATS
        CALL FILEM(NN,ATRIB)
35      CONTINUE
        ATRIB(3)=FLOAT((K*8)-8 + MTEST)
        TOARV=MOD(TNOW,FRAME)
        YSLOT=FLOAT(MTEST)*PKTTIM
        YTIMDEF=YSLOT - TOARV
        IF(YTIMDEF.LE.O.) THEN
        NPAK1=NPAK1+1
        PAK1=FLOAT(NPAK1)
        CALL COLCT(PAK1,7)
        YREQTIM=FRAME - PKTTIM - ABS(YTIMDEF)
        CALL SCHDL(5,YREQTIM,ATRIB)
        ELSE IF (YTIMDEF.LT.PKTTIM) THEN
        NPAK2=NPAK2+1
        PAK2=FLOAT(NPAK2)
```

```
          CALL COLCT(PAK2,8)
          YREQTIM1=(YSLOT-PKTTIM)+(FRAME - TOARV)
          CALL SCHDL(5,YREQTIM1,ATRIB)
          ELSE IF(YTIMDEF.EQ.PKTTIM) THEN
          NPAK3=NPAK3+1
          PAK3=FLOAT(NPAK3)
          CALL COLCT(PAK3,9)
          CALL SCHDL(5,0.,ATRIB)
          ELSE
          NPAK4=NPAK4+1
          PAK4=FLOAT(NPAK4)
          CALL COLCT(PAK4,10)
          YREQTIM2=(YSLOT-PKTTIM-TOARV)
          CALL SCHDL(5,YREQTIM2,ATRIB)
          END IF
          RETURN
30        K=K+1
          NORFF=NORFF+1
          IF(K.GT.NNQ(MSTRF)) GOTO 100
          GOTO 10
100       RETURN
          END
```

****

****

SUBROUTINE END SERVICE

```
          KK=INT(ATRIB(3))
          XX(KK)=0.
          IF(NNQ(KK).EQ.O) RETURN
          IF(NNQ(KK).GT.1) THEN
          ATRIB(5)=ATRIB(5) +1
          CVB=ATRIB(5)
          CALL COLCT(CVB,12)
          CALL RMOVE(1,KK,ATRIB)
          REFRAME=FRAME-PKTTIM
          ATRIB(3)=FLOAT(KK)
          CALL SCHDL(5,REFRAME,ATRIB)
          ELSE
          ENTIME=TNOW-ATRIB(4)
          CALL COLCT(ENTIME,6)
          CALL RMOVE(1,KK,ATRIB)
          ATRIB(5)=ATRIB(5)+1
          DDV=ATRIB(5)
          CALL COLCT(DDV,12)
          LRAM=INT(GO5DAF(1.,9.))
          SPERD=EXPON(67567.LRAM)
```

```
      SETRM=(SPERD + SSAMPLE)
      ATRIB(3)=FLOAT(KK)
      CALL SCHDL(8,SETRM,ATRIB)
      ATRIB(5)=FLOAT(KK+29)
      ATRIB(6)=SETRM
      DTRTIM = FRAME - PKTTIM+(5.0*FRAME)
      CALL SCHDL(9,DTRTIM,ATRIB)
      END IF
      RETURN
      END
```

****
****

SUBROUTINE SUBS-TRANSMISSION

```
      MM=INT(ATRIB(3))
      IF(MM.LE.8) THEN
      LL=MM
      J=1
      ELSE IF(MM.LE.16.) THEN
      LL=MOD(MM,16)
      J=2
      IF(LL.EQ.O) LL=16
      LL=LL-8
      ELSE
      LL=MOD(MM,24)
      J=3
      IF(LL.EQ.0) LL=24
      LL=LL-16
      END IF
      ATRIB(4)=TNOW
      JJ=((J*8) - 8 +LL)
      JRAM =INT(GO5DAF(1.,9.))
      VTSPURTL=EXPON(402402.,JRAM)
      CSPPTN=(VTSPURTL/SSAMPLE)
      CALL COLCT(CSPPTN,11)
      MUPATS=INT(CSPPTN*4.)
      IF(MUPATS.LT.2) MUPATS=2
      NJJ=JJ+53
      CALL FILEM(NJJ,ATRIB)
      DO 45 MJ=1,MUPATS
      CALL FILEM(JJ,ATRIB)
45    CONTINUE
      ATRIB(3)=FLOAT((J*8)-8 +LL)
      BTOARV=MOD(TNOW,FRAME)
      BYSLOT=FLOAT(LL)*PKTTIM
      BYTIMDEF=BYSLOT-BTOARV
```

```
IF(BYTIMDEF.LE.O.) THEN
PAK1=NPAK1+1
PAK1=FLOAT(NPAK1)
CALL COLCT(PAK1,7)
BYREQTIM=FRAME-PKTTIM-ABS(BYTIMDEF)
CALL SCHDL(5,BYREQTIM,ATRIB)
ELSE IF(BYTIMDEF.LT.PKTTIM) THEN
NPAK2=NPAK2+1
PAK2=FLOAT(NPAK2)
CALL COLCT(PAK2,8)
BYREQTIM1=(BYSLOT-PKTTIM)+(FRAME-BTOARV)
CALL SCHDL(5,BYREQTIM1,ATRIB)
ELSE IF (BYTIMDEF.EQ.PKTTIM) THEN
NPAK3=NPAK3+1
PAK3=FLOAT(NPAK3)
CALL COLCT(PAK3,9)
CALL SCHDL(5,0.,ATRIB)
ELSE
NPAK4=NPAK4+1
PAK4=FLOAT(NPAK4)
CALL COLCT(PAK4,10)
BYREQTIM2 =(BYSLOT-PKTTIM-BTOARV)
CALL SCHDL(5,BYREQTIM2,ATRIB)
END IF
RETURN
END
```

****

****

SUBROUTINE DATA START

```
VTIME=TNOW-ATRIB(6)
NM=INT(ATRIB(5)-29.)
IF(VTIME.LE.FRAME.OR.NNQ(NM).GT.O) RETURN
XX(NM)=1
CALL SCHDL(10,PKTTIM,ATRIB)
RETURN
END
```

```
*************************************************************
*************************************************************
```

SUBROUTINE DATA END

```
LK=INT(ATRIB(5))
MJ=LK-29
XX(MJ)=0
CALL RMOVE(1,LK,ATRIM)
RTIM=TNOW-ATRIB(6)
IF(RTIM.LE.FRAME.OR.NNQ(MJ).GT.O) RETURN
SECTRA = FRAME-PKTTIM
ATRIB(5)=FLOAT(LK)
CALL SCHDL(9,SECTRA,ATRIB)
RETURN
END
```

```
*************************************************************
   THIS PROGRAM SIMULATE THE PROPOSED HANDOVER
*************************************************************

DIMENSION NSET (2E+06)
 PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$  KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$  KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$  KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
   KVSLEN=5,
$  KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$  KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$  KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$  KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
   KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$  KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$  KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$  KVSNWF=33,KVUNWF=15,KVWNWF=8)
 PARAMETER  (MXMSG=250)
 PARAMETER  (MXPOUT=6)
 INCLUDE ' SLAM$DIR:PARAM.INC '
 COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON QSET(2E + O6)
 EQUIVALENCE (NSET(1), QSET(1))
 NNSET = 2E + O6
 NCRDR = 5
 NPRNT = 6
 NTAPE = 7
 CALL SLAM
 STOP
 END

*************************************************************
*************************************************************

SUBROUTINE EVENT (I)
         GO TO (1,2,3,4,5), I
1        CALL ARRIVAL
         RETURN
```

```
2          CALL CHECK
           RETURN
3          CALL COLLECT
           RETURN
4          CALL TERMINATE
           RETURN
5          CALL END SERVICE
           RETURN
           END
```

****
****

SUBROUTINE INTLC

```
PARAMETER
(KVACP1=1,KVACP2=2,KVALEN=3,KVACAP=4,KVADIR=5,KVANTX=6,
$ KVANCO=7,KVASTO=8,KVAMNO=9,KVATLU=10,KVAPFE=11,KVAPLE=12,
$ KVCSPR=1,KVCNWL=2,KVCPFE=3,KVCPLE=4,KVCMXL=5,KVCPPL=6,
$ KVCCRC=7,KVCNTE=8,KVCCST=9,KVCSTB=10,KVCTLU=11,KVCVCO=12,
$ KVCRRC=13,KVCCPI=14,KVSESP=1,KVSLSP=2,KVSACC=3,KVSDEC=4,
  KVSLEN=5,
$ KVSBUF=6,KVSCKZ=7,KVSIFL=8,KVSJRQ=9,KVSINI=10,KVSREP=11,
$ KVSNTL=12,KVSNTU=13,KVSNUL=14,KVSNUU=16,KVSNUE=18,KVSNUF=20,
$ KVSPLE=30,KVSPAL=31,KVSNOV=32,KVUPVS=1,KVUCSG=2,KVUCCP=3,
$ KVUICP=4,KVUDCP=5,KVUPCL=6,KVUVMO=7,KVUCSI=8,KVUSPD=9,
  KVUCBF=10,
$ KVUCBT=11,KVUTLU=12,KVUSCP=13,KVUSHT=14,KVUNTL=15,KVUSPT=16,
$ KVFVSI=4,KVWIFL=4,KVWVSI=5,KVWVRQ=6,KVWVRE=7,
$ KVWCPI=8,KVMCPI=4,KVANWF=13,KVCNWF=15,KVFNWF=5,KVMNWF=4,
$ KVSNWF=33,KVUNWF=15,KVWNWF=8)
PARAMETER (MXMSG=250)
PARAMETER (MXPOUT=6)
INCLUDE ' SLAM$DIR:PARAM.INC '
COMMON/SCOM1/ATRIB(MATRB), DD(MEQT), DDL(MEQT), DTNOW, II, MFA,
1MSTOP, NCLNR, NCRDR, NPRNT, NNRUN, NNSET, NTAPE, SS(MEQT),
2SSL(MEQT), TNEXT, TNOW, XX(MMXXV)
COMMON/UCOM1/CIRRAD,NARCU,NUMGA,OFERD,MMPAK,NPAK,KAL,SLO,SHI

      READ(NCRDR,*)CIRRAD,OFERD,SLO,SHI
      CALL COLCT(CIRRAD,1)
      CALL COLCT(OFERD,2)
      CALL COLCT(SLO,7)
      CALL COLCT(SHI,8)
      DLO=3.
      NARCU=0
      MMPAK=0
```

```
NPAK=0
NUMGA=0
KAL=0
CALL GO5CCF
CALL SCHDL(1,0.,ATRIB)
RETURN
END
```

****
****

SUBROUTINE ARRIVAL

```
SOL=50./OFERD
NTH=GO5DAF(1.,9.)
TIMB=NPSSN(SOL,NTH)
CALL SCHDL(1,TIMB,ATRIB)
IF((NARCU-NUMGA).GE.INT(OFERD+1)) RETURN
ATRIB(1)=TNOW
NARCU=NARCU+1
ACC=FLOAT(NARCU)
CALL COLCT(ACC,3)
TRLOCT=(GO5DAF(DLO,CIRRAD)*100.)
CALDUR=((GO5DAF(0.1,5))*300000.)
SPED=(GO5DAF(SLO,SHI)*1000.)
MDIR=GO5DAF(1.,3.)
IF(MDIR.EQ.1) THEN
TDIS=1000.- TRLOCT
TREQ=(((TDIS*3600.)/SPED)*5000.)
IF(TREQ.GT.CALDUR) THEN
ATRIB(4)=0
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(2,TREQ,ATRIB)
END IF
ELSE IF(MDIR.EQ.2) THEN
SDIS=1000.+TRLOCT
STRQ=(((SDIS*3600.)/SPED)*5000.)
IF(STRQ.GT.CALDUR) THEN
ATRIB(4)0
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(3,STRQ,ATRIB)
```

```
        END IF
        ELSE
        DDIS=1000.-TRLOCT
        XDIS=2000.+TRLOCT
        DTRQ=(((DDIS*3600.)/SPED)*5000.)
        XTRQ=(((XDIS*3600.)/SPED)*5000.)
        VDIS=1000.+TRLOCT
        VTRQ=(((VDIS*3600.)/SPED)*5000.)
        IF(DTRQ.GT.CALDUR) THEN
        ATRIB(4)=0
        CALL SCHDL(4,CALDUR,ATRIB)
        ELSE IF (XTRQ.GT.CALDUR) THEN
        ATRIB(2)=SPED
        ATRIB(3)=CALDUR
        CALL SCHDL(2,VTRQ,ATRIB)
        ELSE
        ATRIB(2)=SPED
        ATRIB(3)=CALDUR
        CALL SCHDL(5,XTRQ,ATRIB)
        END IF
        END IF
        RETURN
        END
```

****
****

SUBROUTINE CHECK

```
        MMPAK=MMPAK+1
        CPAK=FLOAT(MMPAK)
        CALL CLOCT(CPAK,4)
        MDIR=GO5DAF(1.,3.)
        CALDUR=ATRIB(3)-TNOW
        SPED=ATRIB(2)
        IF(MDIR.EQ.1) THEN
        TDIS=2000.
        TREQ=(((TDIS*3600.)/SPED)*5000.)
        IF (TREQ.GT.CALDUR) THEN
        CALL SCHDL(4,CALDUR,ATRIB)
        ELSE
        ATRIB(2)=SPED
        ATRIB(3)=CALDUR
        CALL SCHDL(2,TREQ,ATRIB)
        END IF
        ELSE IF(MDIR.EQ.2) THEN
        SDIS=2000.
```

```
STRQ=(((SDIS*3600.)/SPED)*5000.)
IF(STRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(3,STRQ,ATRIB)
END IF
ELSE
DDIS =1000.
XDIS=2000.+TRLOCT
DTRQ=(((DDIS*3600.)/SPED)*5000.)
XTRQ=(((XDIS*3600.)/SPED)*5000.)
VDIS=1000.+TRLOCT
VTRQ=(((VDIS*3600.)/SPED)*5000.)
IF(DTRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE IF(XTRQ.GT.CALDUR) THEN
ATRIB(2)=SPED
ATRIB(3=CALDUR
CALL SCHDL(2,VTRQ,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(5,XTRQ,ATRIB)
END IF
END IF
RETURN
END
```

```
*****************************************************************
*****************************************************************
```

SUBROUTINE COLLECT

```
NPKT=NPKT +1
VPAK=FLOAT(NPKT)
CALL COLCT(VPAK,5)
CALDUR=ATRIB(3)-TNOW
SPED=ATRIB(2)
MDIR=GO5DAF(1.,3.)
IF(MDIR.EQ.1) THEN
TDIS=2000.
TREQ=(((TDIS*3600.)/SPED)*5000.)
IF(TREQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
```

```
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(2,TREQ,ATRIB)
END IF
ELSE IF (MDIR.EQ.2) THEN
SDIS=2000.
STRQ=(((SDIS*3600.)*SPED)*5000.)
IF(STRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(3,STRQ,ATRIB)
END IF
ELSE
DDIS=1000.
XDIS=2000.+TRLOCT
DTRQ=(((DDIS*3600.)/SPED)*5000.)
XTRQ=(((XDIS*3600.)/SPED)*5000.)
VDIS=1000.+TRLOCT
VTRQ=(((VDIS*3600.)/SPED)*5000.)
IF(DTRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE IF(XTRQ.GT.CALDUR) THEN
ATRIB(2)=SPED
ATRIB(3)CALDUR
CALL SCHDL(2,VTRQ,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(5,XTRQ,ATRIB)
END IF
END IF
RETURN
END
```

****

****

SUBROUTINE TERMINATE

```
IF(ATRIB(4).EQ.0) THEN
KAL=KAL+1
XPAK=FLOAT(KAL)
CALL COLCT(XPAK,6)
END IF
```

```
NUMGA = NUMGA +1
RETURN
END
```

****
****

SUBROUTINE END SERVICE

```
MMPAK=MMPAK+1
CPAK=FLOAT(MMPAK)
CALL COLCT(CPAK,4)
CALDUR=ATRIB(3)-TNOW
SPED=ATRIB(2)
MDIR=GO5DAF(1.,3.)
IF(MDIR.EQ.1) THEN
TDIS=2000.
TREQ=(((TDIS*3600.)/SPED)*5000.)
IF(TREQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(2,TREQ,ATRIB)
END IF
ELSE IF (MDIR.EQ.2) THEN
SDIS=2000.
STRQ=(((SDIS*3600.)/SPED)*5000.)
IF(STRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(3,STRQ,ATRIB)
END IF
ELSE
DDIS=1000.
XDIS=2000.+TRLOCT
DTRQ=(((DDIS*3600.)/SPED)*5000.)
XTRQ=(((XDIS*3600.)/SPED*5000.)
VDIS=1000.+TRLOCT
VTRQ+(((VDIS*3600.)/SPED)*5000.)
IF (DTRQ.GT.CALDUR) THEN
CALL SCHDL(4,CALDUR,ATRIB)
ELSE IF(XTRQ.GT.CALDUR) THEN
ATRIB(2)=SPED
ATRIB(3)=CALDUR
```

```
CALL SCHDL(2,VTRQ,ATRIB)
ELSE
ATRIB(2)=SPED
ATRIB(3)=CALDUR
CALL SCHDL(5,XTRQ,ATRIB)
END IF
END IF
RETURN
END
```