THE ENHANCEMENT OF GQ-MAC PROTOCOL ON GSM/GPRS NETWORK

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A THESIS SUBMITTED IN PARTIAL
FULFILLMENT OF THE REQUIREMENTS FOR
THE DEGREE OF MASTER OF SCIENCE
(COMPUTER SCIENCE)
FACULTY OF GRADUATE STUDIES
MAHIDOL UNIVERSITY
2007

Thesis
Entitled

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Was submitted to the Faculty of Graduate Studies, Mahidol University
For the degree of Master of Science (Computer Science)
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ACKNOWLEDGMENT

I would like to express my sincere gratitude and deep appreciation to my advisor Dr. Sudsanguan Ngamsuriyaroj for giving helpful comments, guidance, and encouragement throughout. He never lacks his support and responsibility. I am equally grateful to Dr. Panomporn Suvannapattana. This thesis would not have been possible without the guidance, technical supports of my two advisors, research committee, friends, and colleagues.

I would like to thank all the lecturers of the Computer Science Department at Mahidol University for their valuable advices and all the teachers who have taught me since my childhood.

I thank my friends who are graduate students of Computer Science at Mahidol University, for their high spirit and friendship. In addition, my thankful acknowledgement goes to all office personnel of Mahidol University Computing Center who were contact points during this thesis. My appreciation also goes to the Department of Computer Science and the Computing Center of Mahidol University for giving me valuable knowledge throughout my study.

Finally, I special thank my family for their encouragement and taking good care of me during the entire period of my study.

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ABSTRACT

The GSM network is the most popular network and many researchers are interested to develop applications on it. GPRS is one of the technologies used to transmit data packets over the GSM network that allow mobile communication with the Internet.

One important problem is how to guarantee the QoS of data transmission in the GPRS/GSM network. Several techniques have been proposed to control QoS in the GSM network. One technique called the GQ-MAC protocol is able to guarantee the QoS of four types of traffic: streaming, conversation, interactive and background. Another technique is called Link Adaptation which helps improve the performance of packet transmission in the GPRS network by adjusting the coding scheme in changing environment. There are four coding schemes: CS1, CS2, CS3, and CS4. Each coding scheme can transfer traffic at different data rate from lowest to highest. The Link Adaptive technique uses BLER (Block Error Rate) calculation to adjust the coding scheme used.

This research demonstrates an improvement to the performance of the GQ-MAC protocol using the Link Adaptation technique. The proposed model increases the performance of packet transmission for all traffic. Via CSIM simulation tool, four experiments were conducted. The results show better performance of each traffic transmission. Streaming traffic has higher data bit rate. Conversation traffic has a greater number of users, and interactive traffic can guarantee the throughput.

KEY WORDS: GSM, GPRS / GQ-MAC PROTOCOL / QUALITY OF SERVICES / LINK ADAPTIVE

72 pp.

การปรับปรุงโปรโตคอล จีคิว-แม็ก บนเครือข่าย GSM/GPRS

(THE ENHANCEMENT OF GQ-MAC PROTOCOL ON GSM/GPRS NETWORK)

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บทคัดย่อ

เครือข่าย GSM เป็นเครือข่ายที่มีผู้ใช้งานมากที่สุดเครือข่ายหนึ่งและได้มีการนำเทคโนโลยี GPRS เข้ามาช่วยเพิ่มประสิทธิภาพในการรับ-ส่งข้อมูลแบบ Packet ซึ่งทำให้เครือข่าย GSM สามารถใช้ข้อมูลร่วมกับเครือข่ายแบบ เครือข่าย internet เป็นต้น

ปัญหาที่สำคัญในการส่งข้อมูลแบบ Packet บนเครือข่าย GSM/GPRS คือ การรับประกันคุณภาพบริการ (QoS) ซึ่งก็มีงานวิจัยมากมายที่เกี่ยวกับ QoS บนเครือข่ายนี้ งานหนึ่งที่น่าสนใจคือ GQ-MAC Protocol ซึ่งสามารถช่วยให้รับประกัน QoS ของการส่งข้อมูล Packet บนเครือข่าย GPRS ได้ ข้อมูล Packet แบ่งเป็น 4 แบบ ได้แก่ Stream Traffic, Conversation Traffic, Interactive Traffic, Background Traffic ซึ่ง GQ-MAC Protocol ได้นำเสนอวิธีการรับประกัน QoS ของข้อมูลแบบต่างๆ ดังกล่าวได้อย่างเหมาะสม

นอกจากนี้ เทคนิค Link Adaptation สามารถปรับประสิทธิภาพของการส่งข้อมูลบนเครือข่าย GPRS ให้เหมาะสมได้ โดยการปรับ Coding Scheme ในการรับ-ส่งข้อมูลให้เหมาะสมกับสภาพแวดล้อม ซึ่งมี Coding Scheme อยู่ 4 แบบ ได้แก่ CS1, CS2, CS3, CS4 แต่ละแบบสามารถส่งข้อมูลในอัตรา bit rate ได้น้อยไปมากตามลำดับ การทำงานของ Link Adaptation จะใช้ค่า BLER (Block Error Rate) ซึ่งเป็นค่าที่วัดอัตรา Packet Loss ถ้ามีค่าสูงขึ้น ก็จะปรับ Coding Scheme ลง เพื่อให้อัตรา Packet Loss เหมาะสมกับสภาพแวดล้อม ณ เวลานั้น

งานวิจัยนี้เสนอ Proposed Model ที่นำวิธีการใช้ Link Adaptation เพิ่มประสิทธิภาพการทำงานของ GQ-MAC Protocol ซึ่งทำให้การส่ง Packet แต่ละประเภทข้อมูลเพิ่มประสิทธิภาพมากขึ้นโดยยังรับประกัน QoS ได้ Link Adaptation สามารถเพิ่มประสิทธิภาพการส่งข้อมูลได้ทุกประเภท โดยการใช้โปรแกรมจำลองชื่อว่า CSIM มาใช้ทำการทดลอง 4 การทดลองและทำการวัดประสิทธิภาพการส่งข้อมูลแบบต่างๆ ซึ่งผลการทดลองแสดงให้เห็นถึงประสิทธิภาพการส่งข้อมูลที่เพิ่มขึ้น คือ การส่งข้อมูล Stream traffic ที่ Bit Rate สูงขึ้น, การรองรับจำนวนผู้ใช้งานข้อมูลแบบ Conversation ได้มากขึ้นและการรองรับการส่งข้อมูลแบบ Interactive ได้ดีขึ้น

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CHAPTER I
INTRODUCTION

General Packet Radio Service (GPRS) technology is originated from the demand to send data of Internet access via the cellular GSM network. The GSM network is one of the vital mechanism that allows the seamless connection between two distinct networks: mobile and the Internet.

GPRS technology has caused new and diverse applications. It permits the connection between voice data in GSM network and non-voice data in most of computers and Internet world. Hence, talking with our colleagues through a mobile phone connecting to the Internet and a computer connecting to the GPRS technology is possible. In addition, sending an e-mail to our colleagues would be easy since we can do it anywhere. More and more applications are now within our reach because of the connectivity between these two technologies.

Nonetheless, adopting GPRS technology would not be instantly or easy. The GPRS is just fundamental technology that allows sharing data between the GSM and the Internet. We will encounter several obstacles such as the efficiency of data transmission, the stability of the two networks when connecting and the security of data sent between these networks.Those topics has still been on-going developed to meet the continuous demands of using GPRS.

One interesting research topic is to guarantee the quality and the efficiency of data transmission over the GPRS which is the so called Quality of Service or QoS. For example, when having the voice communication on the GSM/GPRS network, how we can ensure that such voice would not delay or get lost since the original version of the GSM was not designed to support packet data transmission. Thus, adopting the GPRS technology will cause an efficiency problem during data transmission.

There are several studies that develop a QoS over the GPRS. Those studies focus on providing sufficient and efficient data transmission with a variety of network characteristics. Even though the GPRS technology would help connect between the

GSM and the Internet, the frequency resources are still limited. Theoretically, we can send data with the maximum speed of 170 kbps; however, such speed cannot accommodate multiple users of the GPRS simultaneously.

Thus, in summary, the study of QoS on the GPRS would be one of the most interesting topics since it would not only help transmit data reliably between two networks, but also it would be a key factor in adopting the GPRS over GSM in an effective manner. This would lead to the development of novel applications in the future.

1.1 Problem statement

On the GSM network, data are transmitted using the circuit switching which users hold a communication channel from the start to the end of the communication. In contrast, the Internet uses the packet switching which enables multiple communication to each other simultaneously. The GPRS technology allows the GSM network to transmit data via packet switching which has one crucial problem in quality assurance or so called the Quality of Service (QoS) guarantee.

The GSM/GPRS has different techniques to control QoS since the GSM network is a cellular and wireless network. One of the popular protocols used on the GSM is RLC/MAC which has no QoS guarantee. After doing a literature survey, we found one interesting work proposed a QoS protocol called GQ-MAC Protocol[1]. It divides the data traffic into four categories: Streaming, Conversation, Interactive and Background. The GQ-MAC protocol support such four types of traffic efficiently.

The GQ-MAC protocol works at the MAC layer and guarantee the service at the bounded delay of each class's traffic. It also gives the better optimization of the frequency resources. However, the experimental results implied that the network capacity did not maximize. In addition, the GQ-MAC Protocol used only one coding scheme, CS-2.

Coding scheme is to encode the traffic data into a specific format which can be categorized into four levels: CS-1, CS-2, CS-3, and CS-4. Each level determines the amount of data sent to a receiver. In addition, most studies use only CS-2 due to limited hardware capability, and putting complications into the GSM network would add extra delay time which highly affects the network stability for voice data. Thus, no one actually applies the four coding scheme levels.

More data can be sent out if the coding scheme level is properly adjusted. Thus, if the data are sent over the GSM/GPRS using GQ-MAC Protocol and the appropriate level of coding scheme is adjusted, the sufficient quality and maximum efficiency can be achieved. In this work, we will use the four coding scheme levels and having them adjusted according to the applications and network environment.

1.2 Objective of the Thesis

1. To propose a queueing model that simulates the GQ-MAC protocol that would adapt to changing coding scheme and to different traffic types.
2. To improve the efficiency of data transmission by applying the adaptive GQ-MAC protocol that works with the adjustment of coding scheme under various network environment.
3. To simulate the performance of the proposed protocol under different traffic types and individual and mixed coding schemes.

1.3 Scope of the thesis

The scope of the thesis is described as follows.

1. This thesis will study data transmission results which has been the main objective indicated in Proposed Model.
2. This thesis will study only data transmission process and will not include other processes such as new call or hand-off.
3. Csim19 will be used in the testing simulation by using only variable indicated in proposed model and the results will not be directly compared with the previous thesis [1].
4. Our experiment will not consider the delay of a mobile station, a base station or other variables incurred due to hardware limitations.

1.4 Organization of the Thesis

The thesis is organized into seven chapters described as follows.

- **Chapter 1: Introduction** describes the motivation, the problem statement, the objectives and the scope of this thesis.
- **Chapter 2: Background** gives some background knowledge such as GSM/GPRS Network Overview, TDMA Frame structure and Coding Scheme.
- **Chapter 3: Literature Survey** explains research work related to our thesis. Many approaches proposed to guarantee the quality-of-service of data transmission on the GSM/GPRS network.
- **Chapter 4: Proposed Model** presents the details of the proposed model which include the model process and the model parameters.
- **Chapter 5: Implementation** describes the implementation details including the tools used for simulating the proposed model. The flow of all processes and their corresponding algorithms for the model is implemented.
- **Chapter 6: Experiment Results** explains the experiements and the results. We also discuss the analysis of the results.
- **Chapter 7: Discussion and Conclusion** discusses the performance of the proposed model, and the limitations. We also conclude our work, and give some suggestions for future work.

CHAPTER II
BACKGROUND

In the past few years, fixed networks have witnessed a tremendous growth in data traffic due in good part to the increasing popularity of the Internet. Consequently, new data applications are emerging and are reaching the general public. At the same time, the market is witnessing a remarkable explosion of cellular and mobile technologies leading to demand that data applications become available to mobile users.

GSM (Global System for Mobile communications) is the European standard for cellular communications developed by ETSI (European Telecommunications Standards Institute). Throughout Europe and the rest of the world (including North America), GSM has been widely adopted. It has already been implemented in over 100 countries. the most important service in GSM is voice telephony. Voice is digitally encoded and carried by the GSM network as a digital stream in a circuit-switched mode.

2.1 GPRS/GSM network overview

GPRS (General Package Radio Service) stands out as one major development in the GSM standard that benefits from packet switched techniques to provide mobile subscribers with the much needed high bit rates for busty data transmissions. It is possible theoretically for GPRS subscribers to use several time slots (packet data channels) simultaneously reaching a bit rate of about 170kbit/s. Volume-based charging is possible because channels are allocated to users only when packets are to be sent or received. Bursty data applications make it possible to balance more efficiently the network resources between users because the provider can use transmission gaps for other subscriber activities.

2.2 Basic of the GSM/GPRS Network

In order to understand the GPRS network architecture, some fundamental GSM terminology are necessary. This section describes some of the main components of the GSM network (as show in Figure2.1). The GSM PLMN is divided into two major subsystems: the BSS (Base Station Subsystem), and the NSS (Network Switching Subsystem). A GSM subscriber requires a terminal called MS (Mobile Station) to connect to the network using the radio interface (Um).

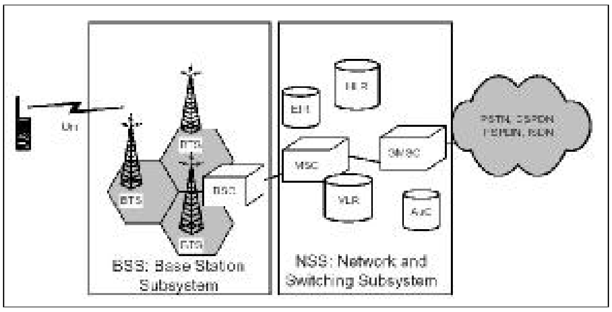


Figure 2.1 GSM Network Architecture

- **HLR (Home Location Register)**

The HLR is a database used to store and manage permanent data of subscribers such as service profiles, location information, and activity status.

- **MSC (Mobile Switching Center)**

The MSC is responsible for telephony switching functions of the network. It also performs authentication to verify the user's identity and to ensure the confidentiality of the calls. The Authentication Center (AuC) provides the necessary parameters to the MSC to perform the authentication procedure. The AuC is shown as a separate logical entity but is generally integrated with the HLR. The Equipment Identity Register (EIR) is on the other hand a database that contains information about the identity of the mobile equipment. It prevents calls from unauthorized, or stolen MSs.

- **VLR (Visitor Location Register)**

The VLR is a database used to store temporary information about the subscribers and is needed by the MSC in order to service visiting subscribers. The MSC and VLR are commonly integrated into one single physical node and the term

MSC/VLR is used instead. When a subscriber enters a new MSC area, a copy of all the necessary information is downloaded from the HLR into the VLR. The VLR keeps this information so that calls of the subscriber can be processed without having to interrogate the HLR (which can be in another PLMN) each time. The temporary information is cleared when the mobile station roams out of the service area.

- **GMSC (Gateway Mobile Switching Center)**

A GMSC is an MSC that serves as a gateway node to external networks, such as ISDN or wireline networks.

- **The Base Station Subsystem**

The BSS performs radio-related functions. It consists of BTSs (Base Transceiver Stations) and BSCs (Base Station Controllers). The BTS handles the radio interface to the MS(Mobile Station). It consists of radio equipment (transceivers and antennas) required to service each cell in the network. The BSC provides the control functions and physical links between the MSC and the BTS. A number of BSCs are served by one MSC while several BTSs can be controlled by one BSC.

2.3 GPRS Architecture

GPRS is considered as a service or feature of GSM. It was designed by ETSI to be implemented over the existing infrastructure of GSM without interfering with the already existing services. The aim is quick GPRS deployment with minor impact on existing GSM PLMN components. Figure 2.2 illustrates the logical architecture of a GSM network supporting GPRS. The components of GPRS are described as follows.

- **GPRS Terminals**

GPRS and GSM systems provide inter-working and sharing of resources dynamically between users. For this reason, three types of terminals have been defined:

A **class-A** MS can carry a circuit-switched and a packet switched connection simultaneously enabling the subscriber to initiate or receive a voice call without interrupting a data transmission or reception activity. This type of terminal probably will not be available when GPRS is initially deployed due to its complexity and high cost.

A **class-B** MS is able to connect to both GSM and GPRS at the same time but an incoming voice call requires GPRS data transactions in progress to be suspended for the duration of the call. GPRS data transactions can then resume at the end of the voice call.

A **class-C** MS allows subscribers to access one service type only at a given time in an exclusive manner.

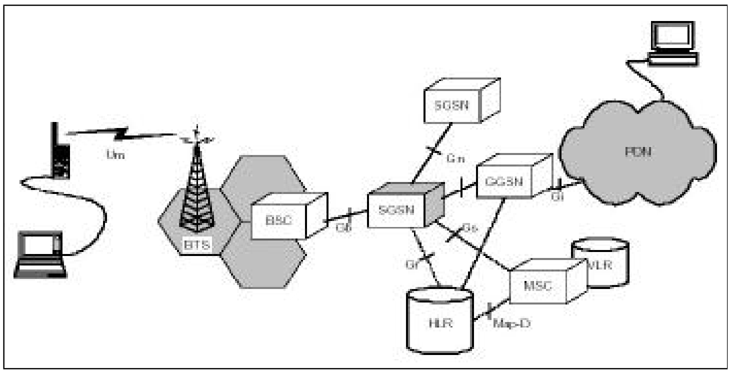


Figure 2.2 Logical architecture of GPRS Network

The GPRS Mobile Station has two components: a MT (Mobile Terminal) which is typically a handset used to access the radio interface as a radio modem, and a TE (Terminal Equipment) which is typically a laptop or a Personal Digital Assistant (PDA). GPRS Mobile Station will also come as one unit combining the functionalities of a MT and a TE.

- **GPRS BSS**

GPRS has minor impact on the existing GSM BSS ant it is easy to reuse existing component and links without major modifications. This is possible because GPRS uses the same frequency bands and hopping techniques, the same TDMA frame structure, the same radio modulation and the same burst structure as GSM.

A new functional component, called PCU, (Packet Control Unit) was added to the BSS in the GPRS standard to support the handling of data packets. The PCU is placed logically between the BSS and the GPRS NSS. Unlike the voice circuit connections, connections in GPRS have to be established and released between the BSS and the MS only when data needs to be transported over the air interface. Therefore, ETSI has defined new procedures to adapt such connections.

- **GPRS NSS**

The GPRS NSS can be viewed as an overlay network ensuring the link between mobile users and data networks. GPRS introduces a new functional element to the GSM infrastructure (Figure 2.3): GSN (GPRS Support Node) which can be either a SGSN (Serving- GSN) or a GGSN (Gateway-GSN). This addition is necessary for the GSM network in order to support packet data services. The network is generally divided into several service areas controlled by separate SGSNs. Only one SGSN serves a MS at a given time provided it is located in its service area.

The SGSN is primarily responsible for keeping track of the MSs it serves, and for access control to data services. The GGSN on the other hand provides the interface to external PDNs (Packet Data Networks). The SGSN is connected to the BSS by Frame Relay and to possibly several GGSNs via a GPRS backbone network. The HLR database is updated to contain GPRS subscriber information. Adaptations to an existing MSC/VLR are not required but the GPRS standard suggests some enhancements to coordinate between the SGSN and the MSC/VLR if the optional interface between the two is to be supported.

Several interfaces have been introduced in GPRS to define entity-to-entity interactions. For instance, the Gb interface is required between the BSC and the SGSN. Two GSNs communicate through a Gn interface, and the SGSN sends queries and receives subscriber information to/from the HLR through the Gr interface. The Gs interface between the SGSN and the MSC/VLR was left optional while the Gi interface which connects a GGSN to a PDN was not specified in the standard to allow implementation preferences.

As mentioned, GPRS standard activities focused mainly on PTP connections to IP PDNs at the Gi interface. An example of such IP PDN can be a corporate

Intranet where access is restricted to authenticated corporate employees allowing them to access for instance the corporate web and mail servers. Another example is connectivity to an Internet Service Provider (ISP) offering Internet access and related services.

- **Transmission / Signaling Planes in GPRS**

A layered protocol structure is adopted for the transmission and signaling planes in GPRS (Figure). The SNDCP (SubNetwork Dependent Convergence Protocol) serves as a mapping of the characteristics of the underlying network such as IP. Mobility management functionality is supported by the GMM (GPRS Mobility Management) and SM (Session Management) layers.

The LLC (Logical Link Control) layer provides a logical link between the MS and the SGSN and manages reliable transmission while at the same time supporting point-to-point and point-to-multipoint addressing. The RLC (Radio Link Control), MAC (Medium Access Control), and GSM RF (Radio Frequency) layers control the radio link, the allocation of physical channels and radio frequency. LLC PDUs (Packet Data Units) between the MS and the SGSN are relayed at the BSS.

The BSSGP (Base Station System GPRS Protocol) layer handles routing and QoS between the BSS (Base Station System) and the SGSN. The GTP (GPRS Tunneling Protocol) is the basis for tunnel signaling and user PDUS between the SGSN and GGSN. We do not describe the rest of the layers of Figure because they are already well known.

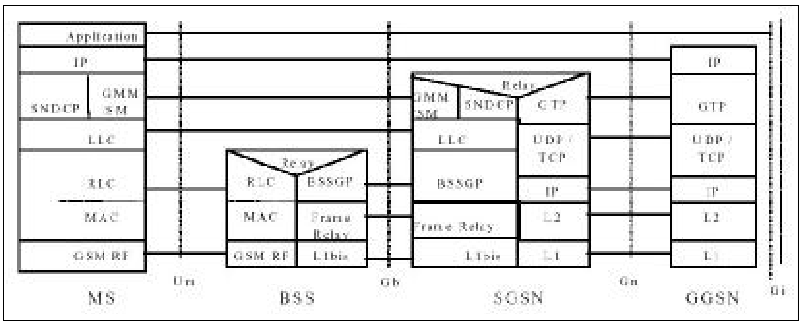


Figure 2.3 GPRS Transmission/Signaling Planes From MS to GGSN

2.4 GPRS Interfaces

This section describes the GPRS interfaces shown in Figure 2.4: Um, Gb, Gn, Gp, Gs and Gi. In this thesis will describe only the Um interface because other interfaces do not related to our work.

Um is the radio interface between MS and BTS. GPRS radio technology is based on GSM radio architecture, which introduces new logical channel structure to control signaling and traffic flow in the Um radio interface. In this subsection, we elaborate on the Um channel structure, Um protocol layers and the enhanced Um for GPRS.

The physical channel dedicated to packet data traffic is called a Packet Data Channel (PDCH). Different packet data logical channels can occur on the same PDCH. The logical channels are described below.

GPRS utilizes Packet Data Traffic Channel(PDTCH) for data transfer. High spectral efficiency is achieved through timeslot sharing where multiple users may share one PDTCH. Furthermore, a user may simultaneously occupy multiple PDTCHs.

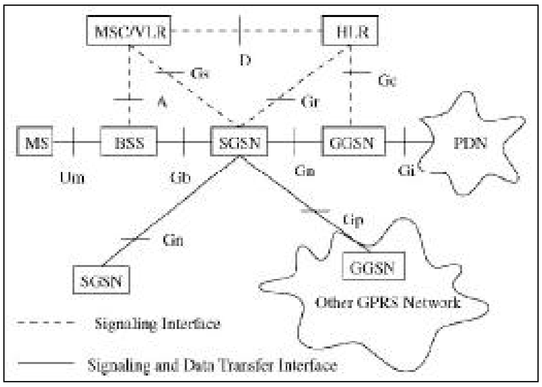


Figure 2.4 GPRS Signaling Interface

Several Packet Common Control Channels (PCCCHs) are introduced in GPRS. The Packet Random Access Channel is the only uplink PCCCH, which is sent from the MS to the BTS to initiate uplink transfer for data or signaling. The following downlink PCCCHs are sent from the BTS to the MS.

- Packet Paging Channel pages an MS for both circuit-switched and packet data services.
- Packet Access Grant Channel is used in the packet transfer establishment phase for resource assignment.
- Packet Notification Channel is used to send a Point-To-Multipoint Multicast (PTM-M) notification to a group of MSs prior to a PTM-M packet transfer.
- Packet Broadcast Control Channel, GPRS (PBCCH) broadcasts system information specific for packet data. If PBCCH is not allocated, the packet data specific system information is broadcast on the existing GSM BCCH channel.

Several Packet Dedicated Control Channels are defined in GPRS:

- Packet Associated Control Channel (PACCH) conveys signaling information such as power control, resource assignment and reassignment information. The PACCH shares resources with PDTCHs. An MS currently involved in packet transfer can be paged for circuit-switched services on PACCH.
- Packet Timing Advance Control Channel in the Uplink direction (PTCCH/U) is used by an MS to transmit a random access burst. With this information, the BSS estimates timing advance. In the downlink, the BSS uses PTCCH/D to transmit timing advance information updates to several MSs.

The GPRS common control signaling is conveyed on PCCCH. If PCCCH is not allocated, the existing GSM Common Control Channel is used. Two concepts are employed for GPRS channel management. In the master–slave concept, a master PDCH accommodates PCCCHs to carry all necessary control signaling for initiating packet transfer. Other PDCHs serve as slaves for user data transfer (PDTCH) and for dedicated signaling. In the capacity-on-demand concept, PDCHs are dynamically allocated based on the actual amount of packet transfers. Also, the number of allocated PDCHs in a cell can be increased or decreased according to traffic change.

GPRS performs fast release of PDCH to share the pool of radio resources for both packet and circuit-switched services.

2.5 TDMA Frame Structure

GSM uses a combination of TDMA and FDMA schemes to provide base stations with simultaneous access to multiple users. Each channel is time shared between as many as eight subscribers using TDMA.

Each of the subscribers uses the same a unique time slot (TS) per frame. Each TS has a time duration of 576.92 us as shown in Figure, and a single GSM TDMA Frame spans 4.615ms.

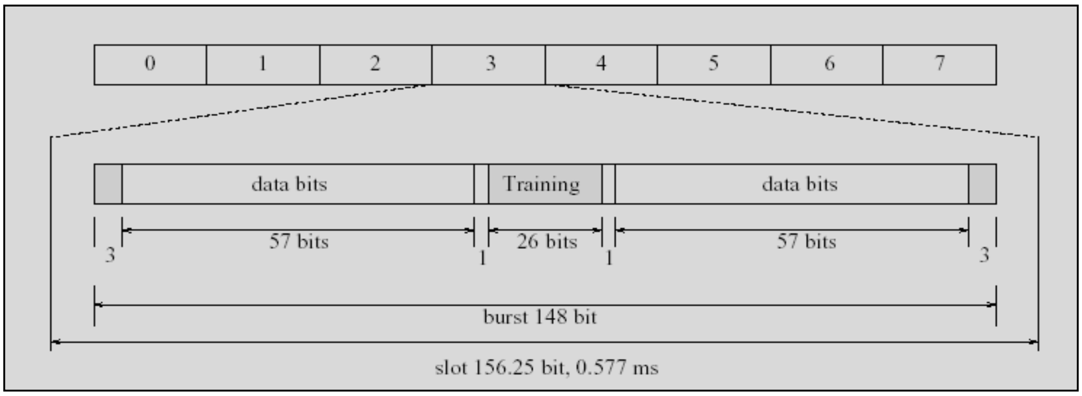


Figure 2.5 TDMA Frame Structure

2.6 Coding Scheme

Four GPRS coding schemes CS1, CS2, CS3 and CS4 are defined, whose characteristics are listed in Table 3. Initially only CS1 and CS2 were be developed. The table indicates that the GPRS channel coding schemes increase data rate at the cost of decreasing protection (correction capability). These coding schemes also reduce worst link budget and cell range. For GSM, the worst link budget is 142.5 dB and the maximum cell range is 730 m. On the other hand, the GPRS worst link budget is 128.5-135 dB and the maximum cell range is 290-450 m.

Table 2.1 Characteristics of the GPRS coding schemes.

Coding scheme	**CS1**	**CS2**	**CS3**	**CS4**
User data rate	9.05 Kb s^{-1}	13.4 Kb s^{-1}	15.6 Kb s^{-1}	21.4 Kb s^{-1}
Correction capability	Highest			None
Worst link budget	135 dB	133 dB	131 dB	128.5 dB
Maximum cell range	450 m	390 m	350 m	290 m

2.7 Limitation of GPRS Network

Despite its potential, GPRS remains constrained by the following limitations

- GPRS transmission rates are much lower than in theory. To achieve the theoretical maximum of about 170 kbit/s would require allocating eight time slots to a single user which is not likely to be allowed by network operators. Even if this maximum allocation was allowed, the GPRS terminals may be constrained by the number of time slots they can handle.
- GPRS relies on packet switching which means that data packets can traverse different routes and then be reassembled in their final destination leading to potential transit delays affecting the Quality of Service.
- GPRS relies also on re-transmission and data integrity protocols to ensure that data packets transmitted over the radio air interface are not lost or corrupted. This can affect even further the transit delay problem.
- GPRS allows the specification of QoS profiles using service precedence, delay, reliability, mean and peak throughput. Although these attributes are signaled in the protocols and are negotiated between the network and the MS, no procedures are defined to provide QoS differentiation between services. This causes a lack of uniformity with respect to QoS between manufacturers and potentially between operators. This issue is being addressed in later standard efforts.
- Although it is possible theoretically to specify a high QoS profile, traffic over radio imposes severe constraints on the Quality of Service.

The protocols between the BSS and the SGSN support mainly asynchronous data transfer applications making it a challenge to implement real-time interactive traffic.

CHAPTER III
LITERATURE SURVEY

There are many researcher interested topic that can improve capacity of GPRS/GSM network. One popular Topic is QoS on GPRS/GSM network. They are several topics. Such as, GQ-MAC Protocol, FEC/MAC Protocol, Slot level re-retransmission etc. We briefly describe these topics.

Many researches improve efficiency of QoS. We interest to improve GQ-MAC protocol. There are 2 Related Research that is interesting in QoS at Data link layer, GQ-MAC protocol and Link Adaptation in GPRS Network.

3.1 Guarantee QoS Medium Access Control (GQ-MAC Protocol)[1]

GQ-MAC protocol is the medium access control (MAC) on mobile wireless that can guarantee QoS. The GQ-MAC protocol supports bounded channel access delay for delay-sensitive traffic, bounded packet loss probability for loss-sensitive traffic, and dynamic adaptive resource allocation for bursty traffic with peak bandwidth allocation adapted to the current queue length.

Performance of the GQ-MAC protocol shows that it is capable of providing guaranteed QoS performances for the specified QoS profiles (streaming, conversational, interactive and background) over GPRS wireless links while optimizing channel resource utilization. We use this technique for develop our thesis work

- **QoS provisioning over GPRS wireless medium**

The wireless link is characterized by a broadcast mode in the downlink (BS to MS) direction and a multiple access mode in the uplink (MS to BS) direction. A medium access control (MAC) protocol is required to distribute packet transmission over the shared medium among all users. MAC protocols could be classified according to their method of resource sharing and the accompanying multiple access schemes. The GPRS standard specifies the FDD/TDMA multiple access with four radio access priorities, and some reference guidelines on the resource sharing method but its overall design will be decided by the developers.

Table 3.1 QOS Profiles Definition

GPRS QoS Class	**QoS Attributes**	**Implemented QoS Profiles**			
		Streaming	**Conversational**	**Interactive**	**Background**
Precedence	Congestion Packet Discard Probability	Tolerable (<10^{-2})	Tolerable (<10^{-2})	Loss sensitive (<10^{-5})	N/A
Delay	Latency	Bounded (<500 msec)	Bounded (<80 msec)	Less stringent than that of Conversational & Streaming	N/A
	Jitter	Stringent	Stringent	N/A	N/A
Reliability	Packet Loss Probability	Tolerable (<10^{-2})	Tolerable (<10^{-2})	Loss sensitive (<10^{-5})	N/A
Mean and Peak Throughputs	Throughput	Guaranteed	N/A	Guaranteed	N/A
	Burstiness	Low	High	Higher than Conversational	N/A

The GPRS packet data channels (PDCH) are classified as one of the following:

- Packet Random Access Channel (PRACH): This is the request access channel for uplink. It consists of two time-multiplexed channels: Signaling PRACH (S-PRACH) and User-data PRCH (U-PRACH).
- Packet Access Grant Channel (PAGCH): This is request acknowledgement channel for downlink. The BS uses PAGCH to broadcast the request status information.
- Packet Data Traffic Channels (PDTCH): The remaining PDCH's, on the uplink and downlink, act as PDTCH's. These are used for carrying the payload.

The request access priorities of the S-PRACH and U-PRACH are illustrated in table. Through the slotted Aloha random access protocol, the S-PRACH is multi-accessed by the signaling requests of new calls and handoffs to gain admission to the PDTCH. Handoff requests are assigned with higher access priority than that of new call requests. Admitted streaming traffic source enjoys per-session dedicated reservation of PDTCH resources. Admitted conversational and interactive traffic sources would have to perform on-demand reservation by multi-accessing the U-PRACH via slotted Aloha random access protocol. During the contention cycle, only the conversational traffic sources are allowed to multi-access the U-PRACH via the tree limited-contention access protocol. Consequently, conversational traffic class has a higher U-PRACH access priority than the interactive traffic class. On the other hand, the background traffic sources are allocated with unused PDTCH resources in a round robin fashion. The access protocols employed by the proposed GQ-MAC protocol are described as follows.

Table 3.2 Request Access Priorities

Request Access Type		**Access Priority**	**PRACH used**
Signaling	New Call	Low	S-PRACH
	Handoff	High	
User Data	Conversational Traffic	High	U-PRACH
	Interactive Traffic	Low	

- **Slotted Aloha**

This random access protocol is employed by the signaling requests of new calls and handoff to access the S-PRACH. Mobile stations with signaling packets transmit immediately on the first available slot. If a collision occurs, they transmit on the next slot with probability "P_a". Handoff requests are given higher priority by having higher "P_a" value as compared to new call initiation requests.

- **Tree Protocol**

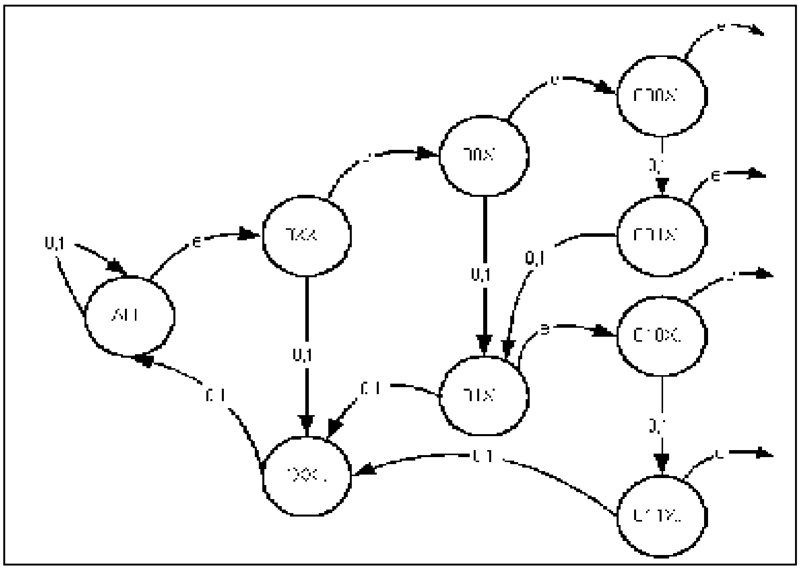
This limited-contention access protocol is used for in-session channel access request for conversational traffic because it provides a deterministic channel access time. A state transition diagram for the algorithm is illustrated in figure. Initially all the MS's in the cell can access the channel. If a collision occurs on any slot "k", the BS starts a contention cycle, which reserves slot "k+1" for MS's having MSB (Most Significant Bit) of their identifiers equal to "0". If a collision occurs again on slot "k+1", then slot "k+2" is reserved for MS's with first two MSB bits equal to "00". On the other hand if a request was successfully transmitted on slot "k+1" or if the slot were idle, the slot "k+2" would have been reserved for MS's having MSB bits equal to "1XX". This is continued according to the figure.

By allowing only conversational traffic sources to participate in the contention resolution cycle, it is possible to guarantee a bounded delay on channel access. The contention cycle can be showed to have a bounded length of: TDMA_FRAME_LENGTH * ($2^{(\log_2 j + 1)} - 1$); Where j is the number of conversational MS's, simultaneously trying to access the U-PRACH. If voice is assumed to be the only conversational source in the system, then using the popular On-Off model with a voice activity factor of 0.4, it can be shown that the probability of more than 5 MS's trying to access the U-PRACH simultaneously is very low. Hence we can expect the channel access delay to be limited to 20 msec, assuming a GSM frame length of 4.615 msec.

- **Modified Slotted Aloha**

This is used for U-PRACH access by interactive traffic sources. It is similar to slotted Aloha protocol discussed above, with following modifications:

- If a collision occurs, a binary exponential back-off algorithm is used which reduces the transmission probability, "P(x)", in the next slot by 0.5 (up to a lower limit). P(x) is calculated as a function of output queue length (§ II-B.3) and fraction of QoS that was satisfied.
- When a contention resolution cycle is in progress, P(x) is reduced to "0", i.e. Interactive MS's do not participate in the contention resolution cycle.



e → Collision, 0 → Idle Slot, 1 → Success

Figure 3.1 State diagrams for limited contention Tree Protocol.

- **QoS Support**

To initiate a new call, a Call Initiation (Call_Init) request is sent on the S-PRACH using the slotted aloha protocol. The Call_Init request contains one or more of the following: Desired service type (Conversational, Streaming, Interactive or Background) and Requested Data Rate (RDR). On successful reception of this request, the AP-CAC module determines if enough resources are available to support this new call, without degrading the service guaranteed to others, and if admissible, sets up two state variables for that session in the BS: RDR and ADR (Achieved Data Rate). A temporary buffer is also set up to hold packets for that session and a suitable MS identifier is generated, which is transmitted to that MS on the PAGCH. We now discuss how each traffic type can be supported in the system.

- **Streaming Traffic**

The system offers streaming as a dedicated service in multiples of quantized data rates, according to $C_i = i \times C/N$; where $i_{min} < i \leq i_{max}$, C_i is the quantized bandwidth requested, C is the total bandwidth that is available and N is total number of PDCH's.

i_{min} and i_{max} depend on the lowest and highest streaming rate that is offered by the system. Thus for a streaming rate "X", one full PDTCH is allocated, while for rate "0.5X", only half of the PDTCH (a slot in every alternate frame) will be allocated and so on. Since resources are permanently allocated to a streaming call, it faces the problem of multi-access only during call set up. Thus a streaming call, once admitted, is guaranteed a bounded packet delay, constant inter-packet delay (i.e. minimal jitter) and a guaranteed throughput. By using a suitable FEC scheme, the packet loss due to corruption on the wireless link is limited.

- **Conversational Traffic**

A conversational traffic source, after getting admitted, demands a channel resource only when it has data to send. The request is sent on the U-PRACH using the tree protocol. If no resources are available at that time, the BS will reallocate resources allocated to other sources (except streaming) to this conversational source. If all the resources are currently allocated to other such conversational sources, then the BS rejects the resource request, and the MS has to send another request, after discarding the first packet in the queue. When there is no data, an allocated resource is held for a channel holding time of 3 TDMA frames, following which an explicit release message is sent. Thus by using the tree protocol for channel access, packet delay for a conversational traffic is guaranteed to be bounded. Also since resource is reserved, till it is explicitly released, inter-packet delay is guaranteed to be constant.

- **Interactive Traffic**

For interactive traffic sources, a scheduling algorithm is required which can guarantee the required throughput. We propose to use a distributed scheduling algorithm for allocation of uplink PDTCH's. This algorithm has been inspired from, with modifications. In this algorithm, MS takes active part in the scheduling process on the uplink. For every interactive stream that is admitted into the cell, the BS and the corresponding MS maintain the following state variables.

- Requested Data Rate (RDR): The MS sends the RDR value in the Call-Init request packet.
- Achieved Data Rate (ADR): The ADR value is continuously updated by MS/BS as it sends/gets data packets.

Every MS maintains a queue at its output interface having a finite length. After getting admitted into the cell, the MS sends a Rate Request Packet (RRP) on the U-PRACH requesting some number of PDTCH's, closely matching its RDR value. The access probability for this first RRP varies according to:

$$
\begin{array}{ll}
P(x) = e^{\frac{\frac{x}{L_U}-1}{(1-\alpha)}} & 1 < x \le L_U \\
P(x) = 1 & x > L_U
\end{array}
$$

Where *P(x)* is the access probability with which the MS transmits the first RRP when there are *x* packets in the queue, L_U is a fixed upper threshold (100, say) and α is a factor that depends on the ADR/RDR value and changes the curve for *P(x)* as shown in figure 3.2.

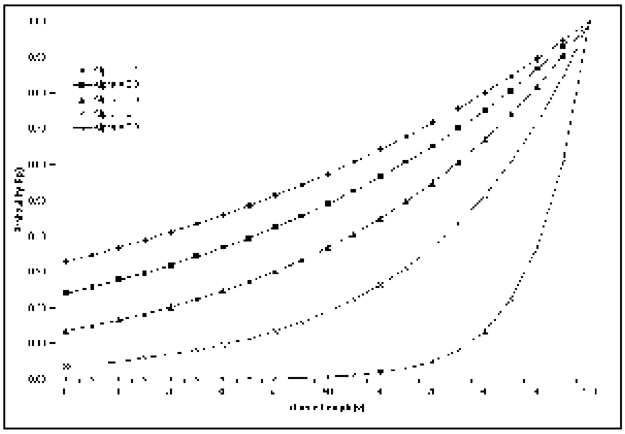


Figure 3.2 Access Probability V/s Queue length for different values of α.

By having α directly proportional to the ratio of ADR/RDR, we have a mechanism that gives higher priority to MS's that have not been able to achieve the requested data rate, to send an RRP. If queue length increases beyond the threshold L_U, the MS will send an RRP with probability 1. Multiple thresholds can be defined in the same way to allow the MS to demand PDTCH's in addition to the minimum required. The BS attempts to allocate, as many PDTCH's requested, as available. For this, it might even pre-empt other like sources, which have achieved ADR >= RDR.

If a MS is allocated additional PDTCH's in response to its RRP packet, it will start transmitting on these PDTCH's. When there is no more data to be sent on the additional PDTCH, it is released. The last PDTCH is held by the MS for the channel holding time of 3 TDMA frames. The entire operation is illustrated in figure 3.3

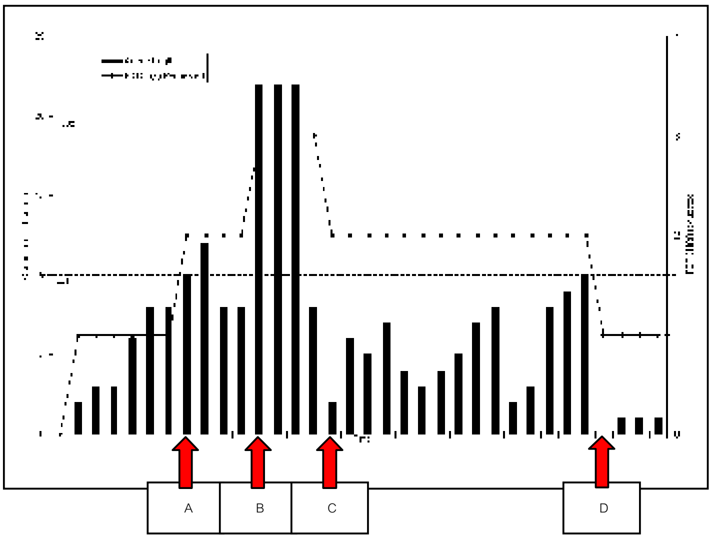


Figure 3.3 Relation between queue length and PDTCH's requested.

For ease of explanation we assume that the MS transmits an RRP only when the threshold is reached. Also multiple thresholds are shown (L_{U1} = 10 and L_{U2} = 20) in this figure as an example. Initially the MS has been allocated a single PDTCH. At time "A", the queue length in the MS increases beyond the threshold L_{U1} (=10). This will make the MS send an RRP demanding an additional PDTCH, which we assume is granted. Thus now the MS will start transmitting on two PDTCH's. Again at "B", the length of the queue increases beyond the second threshold L_{U2} (=20), which makes the MS send another RRP demanding an additional PDTCH, which is again assumed to be granted. Now the MS starts transmitting with 3 PDTCH's. At time "C", the MS has only two packets in its output queue. Thus no packet is available to be transmitted on the 3rd PDTCH.

As a result the MS has to relinquish the 3rd PDTCH and is left with just two PDTCH's. At time "D", the MS has 0 packets in its output queue. So it has no packets to be sent on both the PDTCH's. But the MS will relinquish one of them and hold the last PDTCH for the channel holding time.

- **Background Traffic**

The background traffic sources, after getting admitted and allocated an identifier, camp on the PAGCH to see which slots are allocated to them in the uplink frames. The BS allocates unused PDTCH's to background traffic sources in a round robin fashion.

3.2 Link Adaptation in GPRS Network [2]

In GPRS four coding schemes have been defined with different degrees of data protection. The selection of fixed one for a certain packet transmission would lead to a throughput loss if the channel quality conditions vary during the connection therefore, a link adaptation algorithm based on a BLER estimation is proposed and analyzed. The BLER is calculated from the acknowledgement messages reported by the receiver, so the acknowledgement frequency impacts the link adaptation, and therefore, is to be studied in the present article. The resulting link adaptation algorithm has been analyzed with a dynamic network simulator, under different frequency reuse patterns and frequency hopping strategies. Furthermore, the mean user throughput for the fixed coding schemes and for link adaptation has been estimated from GSM network reported measurements, and a network operator planning tool.

Into the way towards the Third Generation of Mobile Communications, GPRS (General Packet Radio Service) has become a key step for those applications that require wireless data communications. In the RLC/MAC layer of GPRS four coding schemes (CS1, CS2, CS3 and CS4) have been defined with different degree of data protection with the propose of adapting the transmission of the data radio blocks to the different quality conditions.

The performance of every coding scheme strongly depends on the channel quality conditions, so if a fixed coding scheme was selected the variation of the channel quality conditions during the connection would lead to a performance loss. The aim of link adaptation is to update the coding scheme according to the channel quality variations.

The proposed link adaptation algorithm is based on using the Block Error Rate as the channel quality indicator. The transmitter obtains the channel quality information from the acknowledgement messages containing information of the successfully received radio blocks. Thus, then acknowledgement messages frequency influences the link adaptation algorithm. But, besides link adaptation, the packet data transfer adds some constrains to the acknowledgement frequency. So the acknowledgement frequency must be studied in order to reach a compromise that fulfills the requirements of both.

The link adaptation algorithm is analyzed under different frequency reuse configurations, frequency hopping strategies and compared to the fixed coding schemes performance.

Coding in the Physical Link Layer

The RLC/MAC data block transfer is subject to bit transmission errors due to the radio channel conditions. Tith the propose of error detection and correction, the Physical Layer applies FEC coding to the RLC/MAC data bits. This coding is one of the main features of GPRS, and can be accomplished by means of four different channel coding schemes (CS1 to CS4). The coding schemes can be characterized as follows.

CS1 is the same coding as applied for the Slow Associated Control Channel (SACCH), and consists of half rate convolution coding.

CS2 and CS3 are punctured versions of the same coding applied for CS1

CS4 does not apply convolution coding at all.

The four coding schemes have been defined with different degrees of data protection with the propose of adapting the transmission of the RLC/MAC data blocks to the different channel quality conditions.

Table 3.3 Coding parameters for the GPRS coding schemes

Scheme	**Coding Tate**	**Payload**	**Max. Throughput**
CS1	½	181	9.05
CS2	2/3	268	13.4
CS3	¾	312	15.6
CS4	1	428	21.4

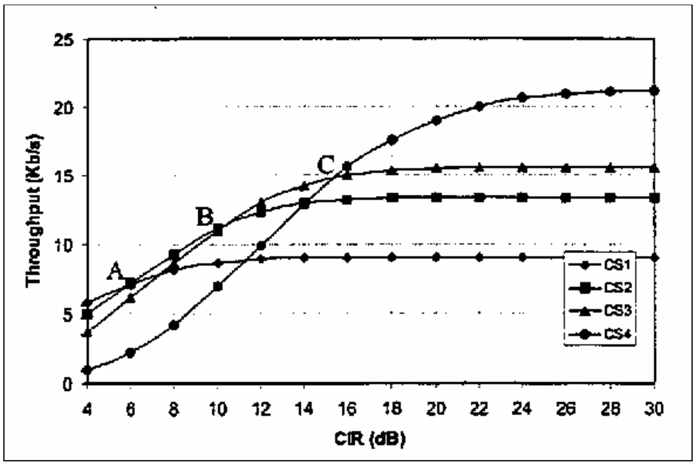


Figure 3.4 Throughput vs CIR for the 4 Coding Schemes with Ideal Frequency Hopping and TU3 km/h channel model

The more data protection is applied by the schemes the more redundancies is included and, therefore, more number of errors can be corrected. But, on the other hand, the more coding is applied the less quantity of data bits can be borne per data block and, thus, the lower throughput can be reached. Hence, there is a channel coding trade-off between low data protection and few retransmissions as a function of the CIR

BLER Based Link Adaptation

In the mobile radio channel the quality conditions vaty during the connection between the MS and the BSS. The aim of link adaptation is to select the most suitable coding scheme according to different channel conditions reaching for every CIR the maximum possible throughput.

For the proposed link adaptation algorithm, the employed parameter to estimate the channel quality is the Block Error Rate (BLER), since the CIR estimation can not be accurately reported by the MS in GSM.

Whenever an Ack/Nack message is received, the BLER is computed over a set of RLC/MAC data blocks called averaging window. The optimum averaging window size is closely related to the data blocks acknowledgement procedure, and will be treated further. After the Ack/Nack message reception, if the estimated BLER in the intersection points A, B, C in figure. For example if the transmitter (MS or BSS) is coding the RLC/MAC data blocks with CS4, and, after the Ack/Nack message reception, the estimated BLER in the last averaging window is higher than the threshold calculated for the point C, the coding scheme is updated to CS3 in the next RLC/MAC data blocks.

The threshold can be calculated using the following expression

$$BLER = (1 - Thr / Thr_{max})$$

Where Thr is the throughput of a coding scheme, and Thr_{max} is the maximum achievable throughput for that coding scheme. Therefore the BLER is

$$BLER = (1 - Thr / Thr_{max})$$

Filling the throughput at the intersection points and the maximum throughput for every coding scheme gives the thresholds. The resulting thresholds reach BLER values up to 40%, but in order not to increase excessively the number of retransmissions, which could lead to a transmission stalling, and the maximum threshold values are limited to 20%. Figure 3.5 depicts the coding scheme update, and the corresponding BLER thresholds.

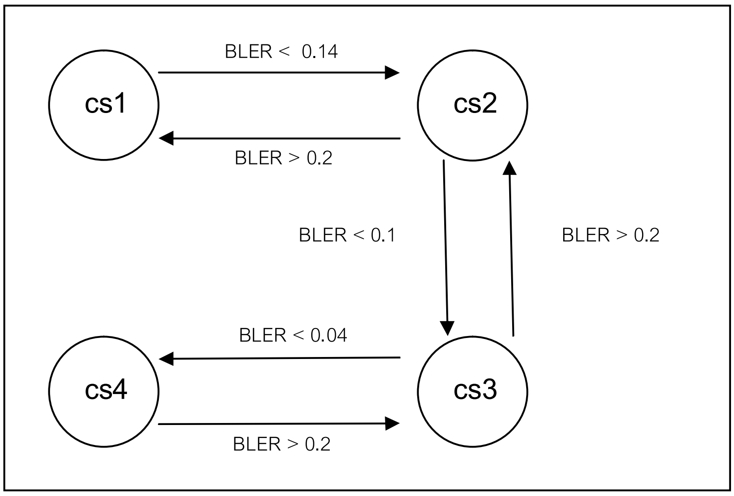


Figure 3.5 Coding Scheme Adaptation

3.3 Acknowledgment Operation in the RLC Layer (FEC/ARQ)[7]

This paper considers the acknowledgment procedure used to assure the delivery of packets on GPRS. The acknowledgment operation are analyzed its performance. The delay introduced by the acknowledgment procedure is evaluated. These papers propose additional hybrid FEC/ARQ mechanism, which can operate with the current one. This mechanism permits to decrease the number of control blocks used for RLC acknowledgment and thus reduce the delay.

Basically, there are two fundamental techniques for maintaining reliable and efficient communication over noisy channels: FEC and ARQ mechanisms. Since in FEC mechanisms, the transmitted information rate is constant regardless of the channel conditions, but the reliability falls as the channel degrades. On the other hand, transmitted information rate of ARQ mechanisms depends mainly on channel quality, but the reliability is almost independent of the channel error rate. The RLC-level

GPRS uses the selective repeat ARQ. In this paper modified RLC-lever with hybrid FEC/ARQ mechanism to improve the performance of sub layer.

Simulations results have shown the capacity of the modification to decrease the number of control messages (Ack/NAck) and reduce the delay of data delivery.

3.4 Performance of MAC Protocol in Dynamic TDD (T-MAC)[8]

This paper consider the future wireless communication systems, a mobile station needs to be able to switch in and out between an indoor pico cell and an outdoor micro/macro cell. To adapt to different transmission enables GPRS to work in low-tier TDD (Time Division Duplex). This architecture consists of a new GPRS MAC protocol that can dynamically assign the transmission sequence so that optimal channel utilization is achieved. The simulation show that proposed architecture gives high flexibility to GPRS in the TDD mode.

The original design of GPRS is specifically for high-tier, TDMA/FDD GSM networks. To make it interoperable with low-tier radio networks, this paper adapt a multi-mode model. This architecture is part of our broadband mobile wireless (BMW). T-MAC represents the TDD version of MAC protocol that makes a transparent transformation from the shared radio medium into GPRS logical channels. The simulation show MAC delay performance of using the above multi-frames and some problems in layer3 related IP addressing.

3.5 Performance Analysis of RLC/MAC (Slot level re-transmission)[9]

This paper evaluate the performance of the RLC(acknowledge mode) layer using block lever retransmission and compare it with that of using slot lever retransmission. The result show that slot lever retransmission at the RLC layer performs better than block level retransmission particularly when the channel error rate is high.

CHAPTER IV
PROPOSED MODEL

This chapter proposes a model that adapts the GQ-MAC Protocol for better data transmission at the data link layer. The technique used is called the Link Adaptation which adapts the traffic rate in accordance with changes in GSM/GPRS network environment at a particular point of time.

4.1 Overview of the Proposed Model

Figure 4.1 shows the overview of our proposed model that applies the concept of the queueing model to express the adaptive GQ-MAC protocol.

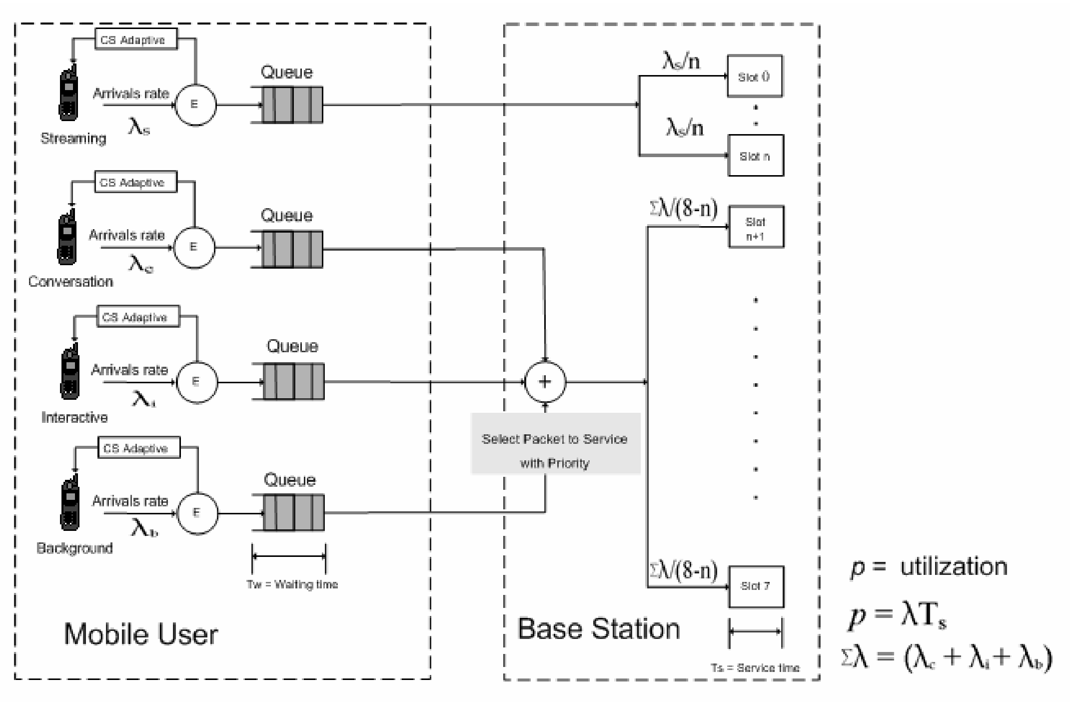


Figure 4.1 Proposed Model of Adaptive GQ-MAC

In our model, the incoming traffic is categorized into four types: streaming, conversation, interactive and background. Each traffic has its own queue when it arrives at a base station at different arrival rate. The base station releases a packet waiting for a service from the queue according to the TDMA of GPRS. In other words, as the time slot for getting a packet arrives, the packet will be put in that time slot which consists of eight slots from 0 to 7 since one TDMA frame has 8 slots.

The proposed model consists of two modules described as follows.

1. Mobile User

The mobile user module (MS) get a service from the TDMA's slot on an available base station site. The module will send a packet of different traffic type to the base station at distinct arrival rate specified in the following symbols.

λ_S : Arrivals rate for streaming traffic

λ_c : Arrivals rate for conversation traffic

λ_i : Arrivals rate for interactive traffic

λ_b : Arrivals rate for background traffic

Each packet will wait at its own queue for an available slot at the base station.

2. Base Station

The base station module (BS) provides a service for packets sent by mobile users, and manages the priority of each traffic before sending a packet to that service. The base station will use the GQ-MAC protocol for selecting a higher priority packet to be put into each slot. In order to obtain a suitable quality of services, we treat the streaming traffic different from other traffic as it has its n own slots for streaming service. Other types of traffic will be prioritized by the GQ-MAC protocol. The service rate of streaming packet in each slot can be calculated as λ_S/n where λ_S is the arrival rate of streaming traffic, and n is the number of slots giving services to streaming traffic. As a result, the service rate of the remaining traffic in each slot will be equal to $\sum\lambda/(8-n)$ where $\sum\lambda$ is the sum of the arrival rate of $\lambda_c + \lambda_i + \lambda_b$.

In addition, T_s is defined as the service time which a slot provides to one packet. Each frame will always start at slot 0, and the time of one frame is equal to 4.615 ms.

4.2 Adaptive Coding Schemes

Link adaptation is the proposed technique used to adjust the coding scheme of the traffic to suite with the network environment. Adjusting the coding scheme will affect the amount of data sent by users. Figure 4.2 shows different coding schemes used to communicate between a mobile and a base station. For example, the mobile station (MS) sends three packets with the coding scheme of CS4, CS2, and CS4, respectively. If the environment is good as bright weather or clear location, the MS can send a packet with CS4. If the environment is getting bad or worse due to the coming thunder storm, the MS cannot send a packet with CS4, and must send the packet with the lower CS such as CS3 or CS2 as shown in Figure 4.2.

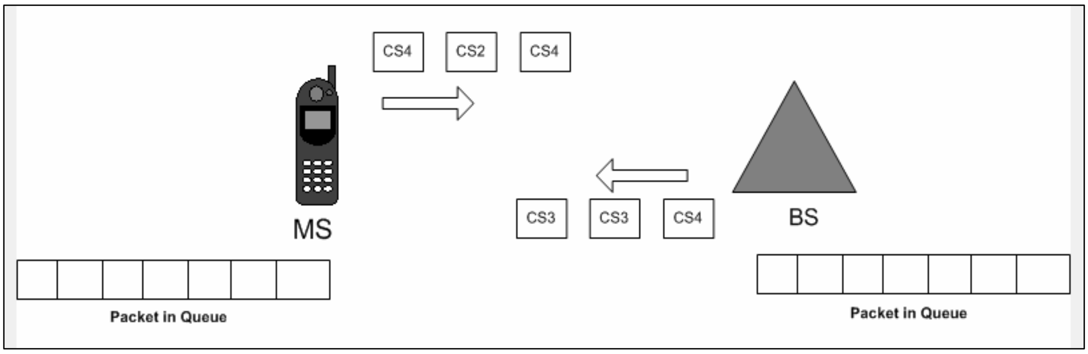


Figure 4.2 Coding Schemes used between Mobile and Base Station

Whenever the network environment does not support for sending high coding scheme, the coding level will be gradually decreased as shown in Figure 4.3. similarly, the higher coding scheme will be used in order to maximize the data transmission efficiently if the network environment supports. An example is shown in Figure 4.4.

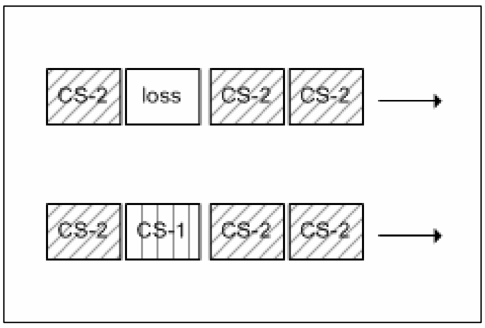


Figure 4.3 Reduced Coding Scheme

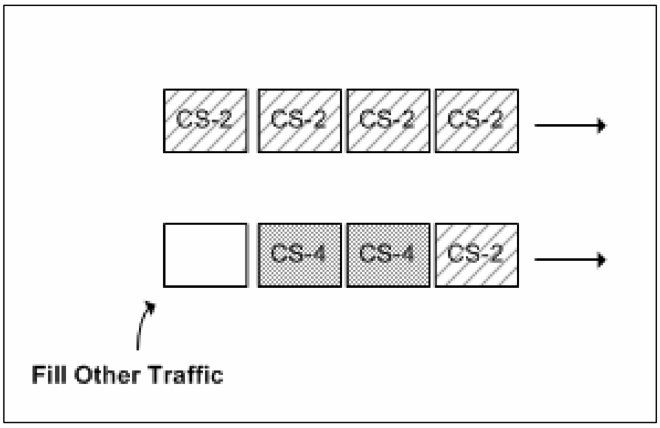


Figure 4.4 Increased Coding Scheme

In summary, the proposed model greatly emphasizes on enhancing the efficiency of the GQ-MAC protocol using adaptive coding schemes as they will be adjusted in order to harmonize with its surroundings.

4.3 Model Parameters

Our model consists of three parameters: Traffic parameters, System parameters, and Environment parameters as described in detail as follows.

4.3.1 Traffic Parameters

Each traffic type has different traffic parameters as shown in Table 4.1.

4.3.2 System Parameters

The system parameters used in our proposed model are given in Table 4.2.

4.3.3 Environment Parameters

If the environment changes while a mobile is transmitting packets to the base station, the quality of traffic will be affected. Thus, we generate a set of environment events that make the coding scheme changed for traffic transmission between a mobile and a base station periodically during the period of the simulation as shown in Figure 4.5. In the simulation, we specify the number of trigger Event denoted as NTrigger. Each event is triggered on an assigned time. For example, we specify the number of trigger events to 3. thus, the event will change the CSRate three times in an increasing or decreasing transmitted data rate according to the environment condition changes as described in Section 4.2. with more implementation details in Section 5.4.2. Each simulated event uses two parameters as described in Table 4.3.

Table 4.1 Traffic Parameters

Traffic	Parameter	Description
Streaming	λ_S	Arrival rate for streaming packet
	T_w	Waiting time at Streaming traffic queue. It defines the delay limitation on this traffic. But, packets may be lost and need to be resent.
Conversation	λ_c	Arrival rate for Conversation traffic modeled as an On-Off model.
	T_w	Waiting time at Streaming traffic queue. If it is 20 ms, each user can talk simultaneously with the same delay time.
	Access Priority	The GQ-MAC set the priority of this traffic to be high.
Interactive	λ_i	Arrival rate for Interactive traffic modeled using Poisson distribution
	T_w	Waiting time at Interactive traffic queue defined as ∞ (infinity)
	Access Priority	The GQ-MAC set the priority of this traffic to be low.
Background	λ_i	Arrival rate for Background traffic
	T_w	Waiting time for Background traffic queue defined as ∞ (infinity)
	Access Priority	The GQ-MAC set the priority of this traffic to be low.

* Poisson Distribution is an exponential distribution function used to generate traffic in our simulation [7].

Table 4.2 System Parameters

Parameter	Description
NSLOT	Number of reserved slots for streaming traffic which depends on Streaming bandwidth and the number of users.
NFRAMES	Parameter used to identify the number of slots for streaming traffic. A set of slots are set aside because it will affect services provided to the remaining traffic.
SLOT_TIME	Defining coding scheme when the environment has changed and will cause λ recalculation for each traffic.

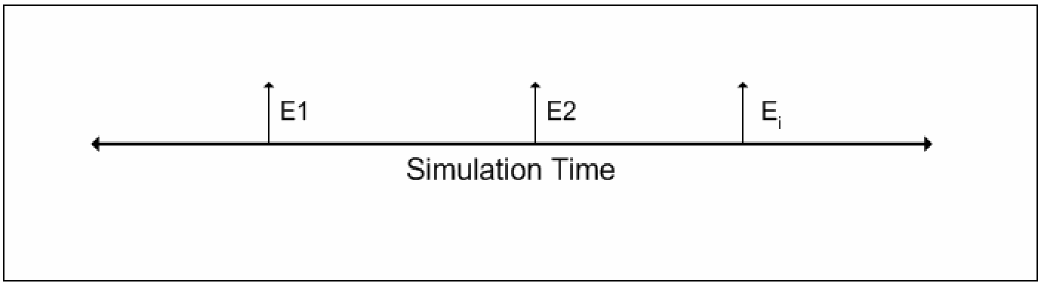


Figure 4.5 Events of Environment

Table 4.3 Environment Parameters

Parameter	Description
CSRate	Coding Scheme Rate adapted after an event.
Event(E_i)	Designated events used in simulation for changing CSRate.
NTrigger	Number of trigger events that affect to CSRate adaptation.

CHAPTER V
IMPLEMENTATION

This chapter presents the details of the implementation of the proposed model. First, we describe the system requirements, and the assumptions used in our simulation. Second, we explain the simulation tool used, and the overview of the simulation. Next, the flow diagram in each part of the simulation are presented. Finally, the format of the results is described.

5.1 System Requirements

The hardware and software used in this thesis are given as follows.

- Hardware Requirements
    - Intel Pentium 4 Processor of 1.8 GHz
    - Memory DDR 512 MB
    - Hard disk space 20 GB
- Software Requirements
    - Operating System: Windows XP SP2
    - Compiler : Visual C++ 6.0
    - Simulation tool : csim19
    - Result Presentation tool : Microsoft Excel XP

5.2 Assumptions of the Simulation

The proposed model assumes that users of each traffic type will transmit data. In addition, the model will not consider some related factors such as the hand-off mechanism, the delay time during the transmission and the retransmission between the mobile and the base station. The details of the model's assumptions are described as follows.

1) The model will consider only the data transmission period. In other words, we ignore the hand off which is practically taken place for data transmitted between base stations and a mobile.

2) The model will consider only the waiting time and the service time when computing the packet loss, and the delay time incurred due to waiting for an available slot in the GQ-MAC protocol. Moreover, for conversation traffic, the tree protocol is used, and a slot is reserved. Thus, the traffic must be delayed for receiving a service. However, the data transmission in GPRS has many other delays such as the retransmission time or the processing time originated from the coding scheme adjustment. In practice the delay time depends on the environment, hardware, software and other related factors.
3) In the simulation model, only one base station is used since we assume that the current mobile is located in the same cell site. Thus, the model does not cover the data transmission between two base stations.
4) Since the model has the coding scheme adjustment, in practice, the mobile must have the traffic creating capability for each coding scheme using additional hardware or software. However, the model does not concern with the possibility of how it can be supported by hardware.

5.3 Simulation Tool

In the implementation of the proposed model, we select the simulation tool called csim19 since it is suitable for the queueing model expressed in our proposed work of Chapter 4. The csim19 has several characteristics as follows.

- It has the complete facilities for simulating a queuing model.
- The simulation can be written in C and C++ language.

In this work, we do not compare the results of our simulation with the outputs of the original work [1] which proposed the GQ-MAC protocol since it used Opnet network simulator which we cannot acquire it. In addition, we do not simulate the GSM/GPRS network, but simulate the queueing model using the same parameters as the original work except that we consider the adaptive coding schemes.

5.4 Simulation Overview

The simulation model consists of four main components: traffic generation, the GQ-MAC protocol, the slot consumption and time recording, and result recording. Each component is processed step-by-step as shown in Figure 5.1, and described in detail in the next section.

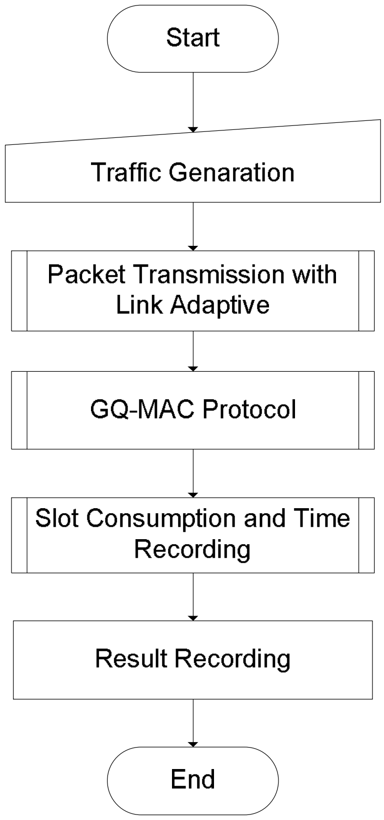


Figure 5.1 Simulation Process Overview

5.4.1 Traffic Generation

This process prepares traffic data used in the experiment. Three traffic types will be generated, and each traffic uses different distribution model. All traffic types except background traffic will be generated. The streaming traffic uses the uniform distribution [6]. The conversation traffic uses the on-off model [1] which is the most popular model for voice data. The interactive traffic uses the Poisson distribution [6].

The parameters of each distribution are assigned at the beginning of the simulation. Figure 5.2 shows the flow diagram for generating the traffic.

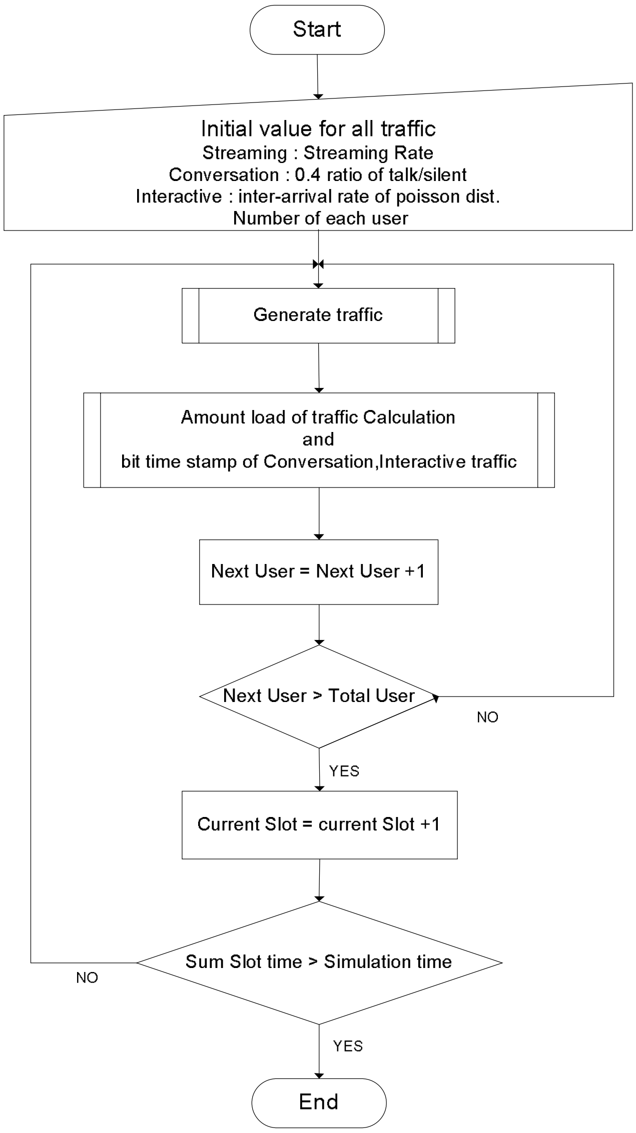


Figure 5.2 Traffic Generation Process

5.4.2 Packet Transmission with Link Adaptive

This process is responsible for making a decision to transmit a packet with a suitable CS. During the simulation time, the MS always generates the traffic. In addition, the environment parameters will be used to determine an event change which will make the current CS adaptive to another CS according to the environment condition. When the total number of bits in the generated traffic is more than the number of bits per slot with the current CS, the packet will be sent to the base station on the assigned slot. But, if the generated traffic is less than the number of bits per slot, the system will wait for more traffic to be transmitted in the queues.

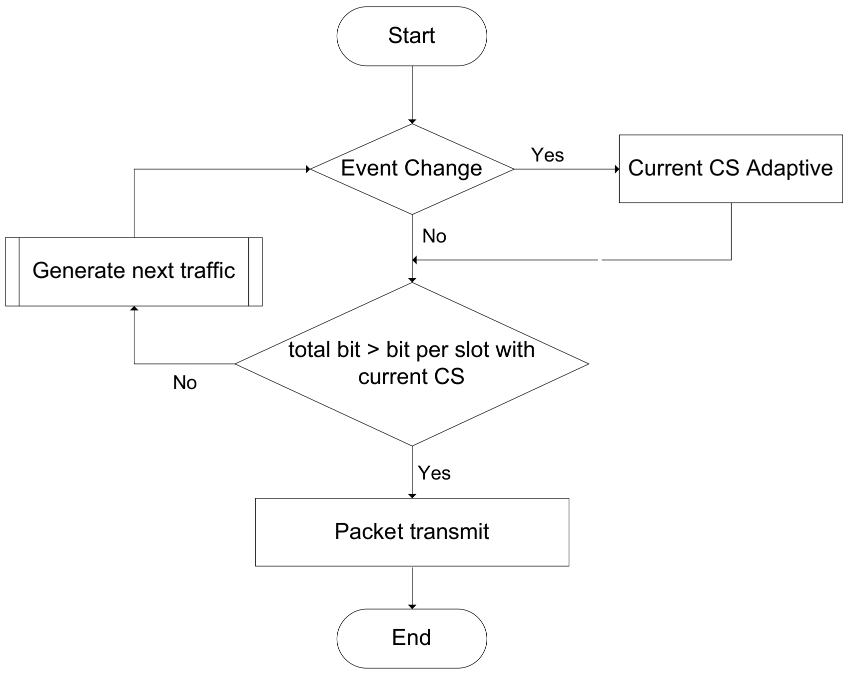


Figure 5.3 Packet Transmittion with Link Adaptive

5.4.3 GQ-MAC Protocol

This process is the main procedure that assigns a set of channels or slots for a particular traffic. The GQ-MAC protocol manages the three traffic differently, and they are described as follows.

1. *Streaming Allocated Slot*

The system allocates multiple slots for streaming traffic. The number of reserved slots, NSLOT, assigned is calculated using the following formula:

$$\text{NSLOT} = \text{Streaming Rate/CS Rate}$$

where CS Rate is current coding scheme used for transmission, and Streaming Rate is the total data rate that all streaming users required. The system will reserve slots for streaming traffic if they are available, and the GQ-MAC protocol gives the first priority to the streaming traffic. Thus, the available slots for other traffic would be reduced as well.

2. *Tree Protocol*

The Tree Protocol as its state transition diagram shown in Figure 3.1 is used to reserve slots for conversation traffic. In addition, we have to assign a timestamp of each packet since we need to use packet's timestamp for ordering the reserved slots manually. Figure 5.4 shows the flow diagram of the tree protocol implementation.

3. *Distribution Scheduling Protocol*

This protocol guarantees the required throughput for interactive traffic as shown in Figure 3.2 that illustrates the access probability versus queue length that derived from P(x) of algorithm. In addition, we define the same parameters as those in the original paper [1] since it does not mention how these parameters such as Achieved Data Rate (ADR) and Requested Data Rate (RDR) are acquired in detail. However, we adjust the calculation of the simulation parameters similar to the trend of P(x) in [1]. Figure 5.5 shows the flow diagram of this protocol.

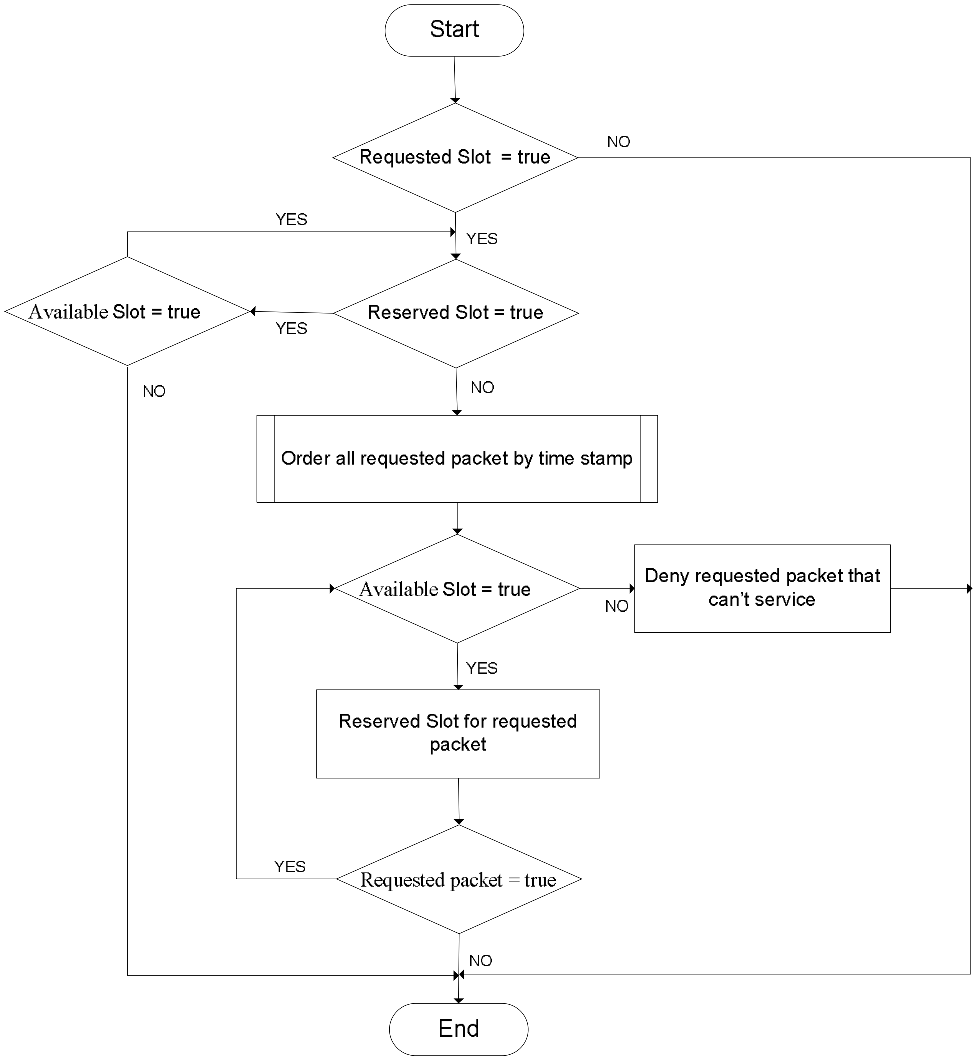


Figure 5.4 Tree Protocol Process

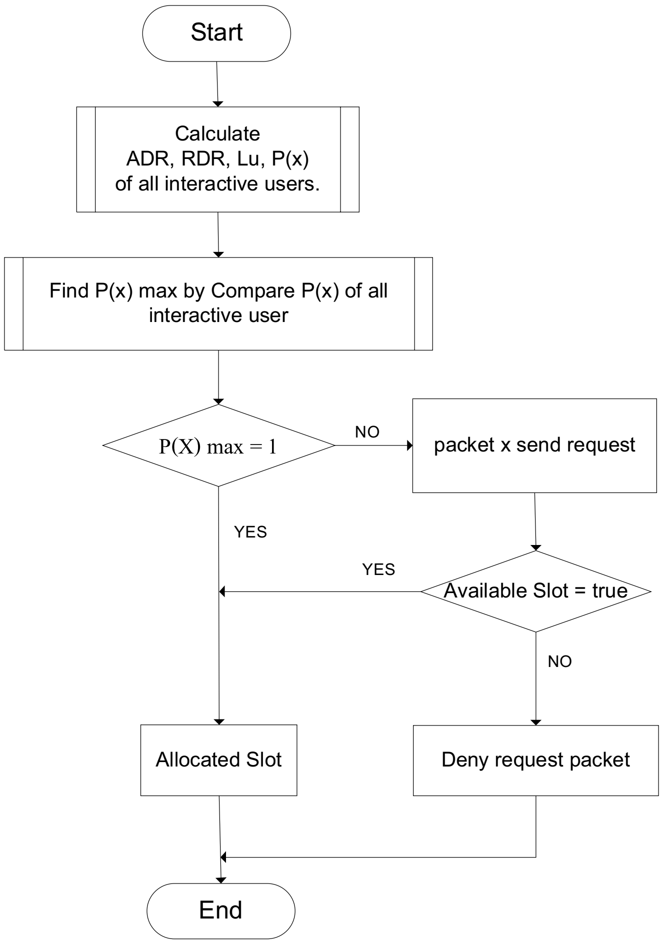


Figure 5.5 Distributed Scheduling Protocol Process

5.4.4 Slot Consumption and Time Recording

This process is for putting packets into allocated slots. We have to identify the assigned slot number and record the timestamp of using the slot. We use the facilities defined in CSIM as slot resources, and use CSIM's simulation time for recording the timestamp. The functions and facilities in CSIM used in the proposed model are given as follows.

- **Slot Declaration and Slot Usage**
  - **FACILITY Slot[8]** is the declaration 8 slots used for one time frame.
  - **reserve(slot[i]);** is the function for reserving a slot i. Whenever this function is operated by one packet, another packet cannot use the slot.
  - **hold(TIME);** is the function for advancing the time in simulation.
  - **release(slot[d]);** is the function for releasing a slot i. When this function is operated by one packet, it releases the slot and another packet can use the slot.
- **Time Recording**
  - **TABLE tblDelay[MS];** is the declaration of a table for recording time statistics.
  - **record(DelayTime, tblDelay[MS]);** is the function for recording the delay time into the table declared.
- **Random Numbers**
  - **Poisson(interarrivalRate);** is the function to generate the interarrival time between two packets of incoming traffic. It usually operates with the hold function as in hold(poisson(time Value));

5.4.5 Output Recording and Presentation

CSIM provides a way to record results generated in simulation automatically. The function called *report();* is used to write the results into an output file. We can also add other additional results into the output file in case CSIM does not record the results such as the percentage of packet dropping computed from the number of dropped packets over the total transmitted packets.

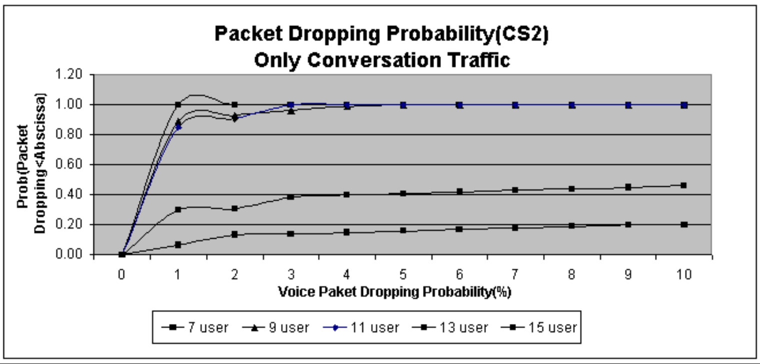
For the output presentation in this thesis, we use the facilities of Microsoft Excel. The data from the CSIM output file are put into Excel and some formulas are set for plotting a graph. Figure 5.6 shows computed packet dropping probability obtained from the CSIM output file, and the sheet in Microsoft Excel and its corresponding graph are shown in Figure 5.7 and 5.8, respectively.

```
Convt MS: 11 users
Convt BW: 5360 bit/s
Inter Arrival Rate: 270 us

Event Trigger: None
Simulation Time: 13000 frames* 8 slot * 576.92 us = 1.00 min
============================================
Convt MS 1: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 2: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 3: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 4: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 5: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 6: Packet Dropping Prob: 0/2851 = 0.0000
Convt MS 7: Packet Dropping Prob: 21/2851 = 0.0074
Convt MS 8: Packet Dropping Prob: 145/2851 = 0.0509
Convt MS 9: Packet Dropping Prob: 317/2851 = 0.1112
Convt MS 10: Packet Dropping Prob: 495/2851 = 0.1736
Convt MS 11: Packet Dropping Prob: 659/2851 = 0.2311
Convt MS 12: Packet Dropping Prob: 835/2851 = 0.2929
Convt MS 13: Packet Dropping Prob: 1005/2851 = 0.3525
      CSIM Simulation Report (Version 19.0 for MS Visual C/C++)

  Experiment4:Packet Dropping Probability of Conversation with Tree Protocol

                Sun Apr 01 11:54:27 2007
```

Figure 5.6 Sample CSIM Output File

13 user	%drop(>60ms)		
1	0		
2	0		
3	0		
4	0		
5	0.0001	0.01%	
6	0.043	4.3%	
7	0.1537	15.37%	
8	0.3401	34.01%	
9	0.5247	52.47%	
10	0.7083	70.83%	
11	0.8863	88.63%	
12	0.9789	97.89%	
13	0.9989	99.89%	
%drop	user	sum	Prob(PckDrop<Abscissa)
<2%	4	4	0.3077
<3%	1	5	0.3846
<10%	1	6	0.4615

Figure 5.7 Partial CSIM Output in Microsoft Excel



7 user		9 user		11 user		13 user		15 user	
PckDrop(%)	Prob(PckDrop<Abscissa)	PckDrop(%)	Prob(PckDrop<Abscissa)	PckDrop(%)	Prob(PckDrop<Abscissa)	PckDrop(%)	Prob(PckDrop<Abscissa)	PckDrop(%)	Prob(PckDrop<Abscissa)
0	0.0000	0	0.0000	0	0.0000	0	0.0000	0	0.0000
1	1.0000	1	0.8889	1	0.8500	1	0.3000	1	0.0667
2	1.0000	2	0.9300	2	0.9091	2	0.3077	2	0.1333
3	1.0000	3	0.9600	3	1.0000	3	0.3846	3	0.1400
4	1.0000	4	0.9900	4	1.0000	4	0.4000	4	0.1500
5	1.0000	5	1.0000	5	1.0000	5	0.4100	5	0.1600
6	1.0000	6	1.0000	6	1.0000	6	0.4200	6	0.1700
7	1.0000	7	1.0000	7	1.0000	7	0.4300	7	0.1800
8	1.0000	8	1.0000	8	1.0000	8	0.4400	8	0.1900
9	1.0000	9	1.0000	9	1.0000	9	0.4500	9	0.2000
10	1.0000	10	1.0000	10	1.0000	10	0.4615	10	0.2000

Figure 5.8 Graph Plotting of CSIM Output in Microsoft Excel

CHAPTER VI
EXPERIMENTS AND RESULTS

In this chapter, the experimental results of the adaptive GQ-MAC protocol are presented. We describe the objectives of the experiments, explain the experiments, discuss the experimental results, and finally give the summary.

6.1 Objectives of the Experiments

1. To evaluate the performance of the GQ-MAC Protocol with adaptive coding scheme.
2. To investigate the effects of having a variety of mixed traffic on the performance of the adaptive GQ-MAC protocol we proposed.

6.2 Experiments and Results

In this thesis, we simulate three types of traffic. They are streaming, conversation, and interactive. We do not generate background traffic because it has no quality of services guaranteed. A single type of traffic and their combinations are tested in our experiments. Particularly, we conduct four experiments in total described as follows.

- Experiment 1: Streaming Traffic
- Experiment 2: Conversation Traffic
- Experiment 3: Conversation and Interactive Traffic
- Experiment 4: Streaming and Conversation Traffic

The experiments are designed in this sequence because we want to evaluate the performance of the proposed model from high to low priority traffic. Streaming, conversation, and interaction traffic has the high, moderate, and low priority, respectively. In the first two experiments, we measure the performance of a single traffic type consisting of only streaming or conversation traffic. In the last two experiments, we measure the performance of the combined traffic. For all experiments, we use the common parameters as given in Table 6.1.

Table 6.1 Common Simulation Parameters

Parameter	Value
Number of Base Stations	1
TDMA frame duration	4.615 ms
Slot duration	576 us
Number of Channels(Slot)	8 Slots
Simulation time	130,000 frames(approx. 10 mins)
Maximum Conversation Delay	60 ms
Maximum Other Traffic Delay	N/A

6.2.1 Experiment 1: Streaming traffic

In this experiment, the performance of streaming traffic transmission is evaluated to examine whether the system can guarantee QoS of all streaming users. The number of users is varied from 1 to 8 users for each coding scheme. During the simulation, the system will reserve a set of available slots for streaming traffic. The maximum bit rate of each coding scheme is recorded, and the results are plotted as a graph shown in Figure 6.1.

Table 6.2 Parameters of Experiment 1

Parameter	Value
Streaming Data Rate	9.05 Kbps to 171.2 Kbps
Number of Users	1 to 8 users
InterArrival Rate	700 us
CS Rate	CS1 to CS4

Table 6.3 shows the obtained maximum data rate when there is only one user allocated to all slots. Since CS2 is the commonly used coding scheme, we want to compare the maximum transmitted data rate of CS3 and CS4 with CS2 having the maximum data rate of 107.2 Kbps. Hence, for the case of only one user, CS4 with the

maximum data rate of 171.2 Kbps has better performance than CS2 for about 60%. Similarly, CS3 with the maximum data rate of 124.8 Kbps has better performance than CS2 for about 16%.

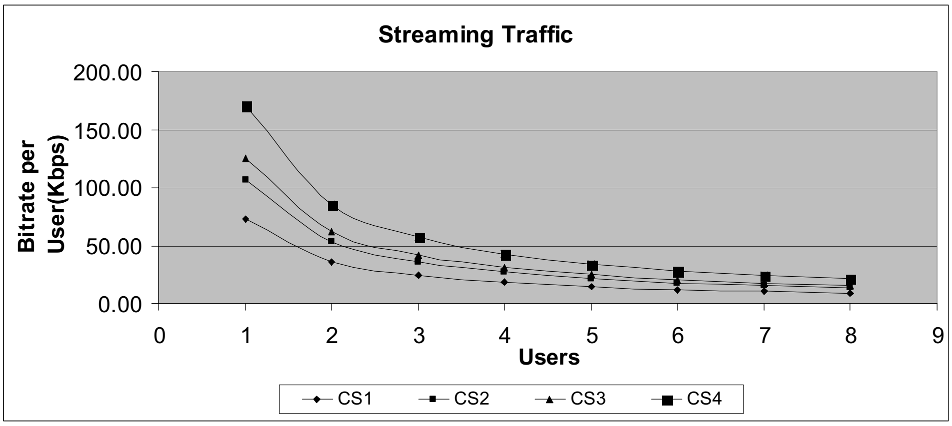


Figure 6.1 Capacity of Streaming Traffic

Table 6.3 Maximum Data Rate of One Streaming User

Coding Scheme	Maximum Data Rate
CS1	72.4 Kbps
CS2	107.2 Kbps
CS3	124.8 Kbps
CS4	171.2 Kbps

Table 6.4 shows the obtained maximum data rate of each user when there are 8 users using the system concurrently. The results also illustrate the efficiency of CS3 and CS4. In other words, each CS3 and CS4 user can transmit data at the rate of 15.6 and 21.4 Kbps, respectively. Thus, they have the performance gain of 1.16 and 1.6 over CS2 which has the data rate of 13.4 Kbps. In conclusion, streaming traffic also shows the similar performance whether a user uses the maximum or lower number of slots.

Table 6.4 Individual Maximum Data Rate of 8 Streaming Users

Coding Scheme	Maximum Data Rate
CS1	9.05 Kbps
CS2	13.4 Kbps
CS3	15.6 Kbps
CS4	21.4 Kbps

6.2.2 Experiment 2: Conversation Traffic

In this experiment, the performance of conversation traffic transmission is evaluated to examine whether the system can guarantee QoS of all conversation users. Since voice packets must be delivered in real time with the maximum delay time of less than 60 ms, the delay time can be used to calculate the packet dropping rate of each user. Moreover, to get a good Mean Opinion Score (MoS) for voice, the MoS should be limited between 1 to 2% for conversation traffic transmission.

For the simulation in this experiment, the on-off model is used for conversation traffic generation with the voice activity factor of 0.4 with allowing talk length of 1 ms and silent length of 1.35 ms. The voice traffic is generated until the simulation is ended. The coding scheme in this experiment is varied from CS2 to CS4 while the number of users can be varied for each experiment, and the packet dropping rate is measured. We divide this experiment into three parts describe as follows.

(a) Experiment 2.1: Conversation Traffic in CS2

The parameters used in this experiment are shown in Table 6.5. The results of packet dropping probability and average packet delay are shown in Figure 6.2 and 6.3, respectively.

Table 6.5 Parameters of Experiment 2.1

Parameter	Value
Conversation Data Rate	On-Off model with 0.4 ratio
Number of User	7 to 15 users
CS Rate	CS2 - 13.4 Kbps

The results illustrate that 11 is the maximum number of voice users that can transmit voice packets with 95% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 11 voices, the system can guarantee the QoS. In contrast, if the number of users is higher than 11 voices, the system cannot guarantee the QoS. For example, 13 voice users can transmit voice packets with 30% of all packets experienced packet dropping of less than 2% while the remainder of 70% experienced packet dropping of higher than 2%. In addition, Figure 6.3 shows that 11 users with 95% of all packets experienced the delay time of less than 56 ms.

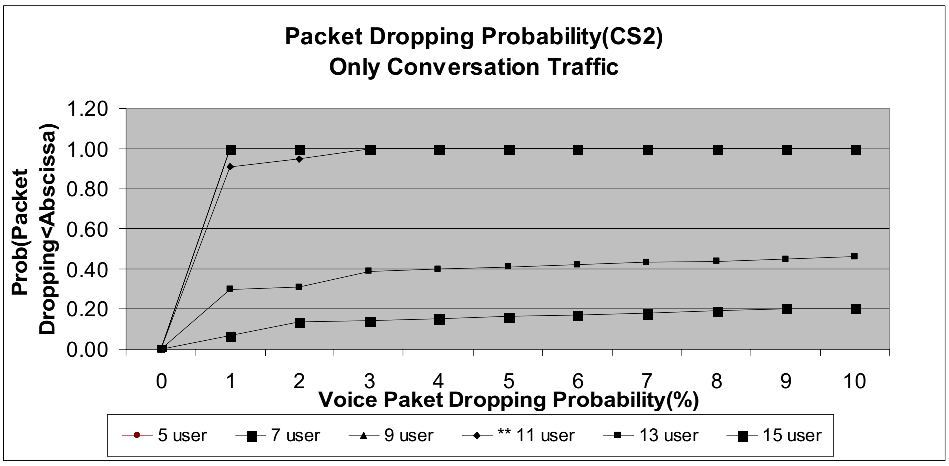


Figure 6.2 CDF of Voice Packet Dropping Probability of Conversation Traffic (CS2)

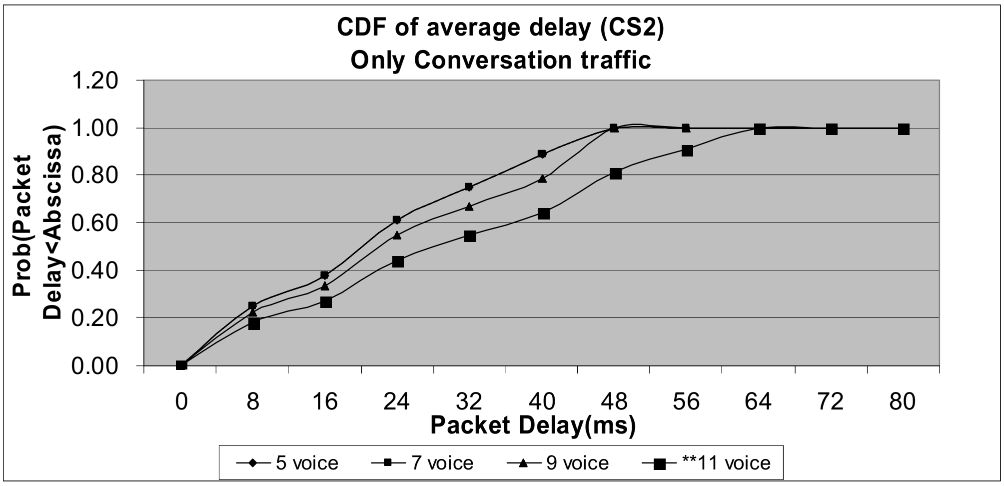


Figure 6.3 CDF of Average Packet Delay of Conversation Traffic (CS2)

In summary, the system can support the maximum number of users at 11 voices with guaranteed QoS. The capacity gain of 1.38 is obtained when compared with the system that does not implement the tree protocol. When the number of users becomes higher than 11, the performance is obviously decreased, and the system can not guarantee QoS for the conversation traffic. In addition, the average delay of all packets is about 56 ms which is less than the maximum limited delay of 60 ms. Thus, the number of users does not significantly affect the delay time. Moreover, this experimental results show the similar output with those in the original work [1] since we use the same parameters in our experiments.

(b) Experiment 2.2: Conversation Traffic in CS3

The parameters used in this experiment are shown in Table 6.6. The results of packet dropping probability and average packet delay are shown in Figure 6.4 and 6.5, respectively.

Table 6.6 Parameters of Experiment 2.2

Parameter	Value
Conversation Data Rate	On-Off model with 0.4 ratio
Number of User	11 to 15 users
CS Rate	CS3 of 15.2 Kbps

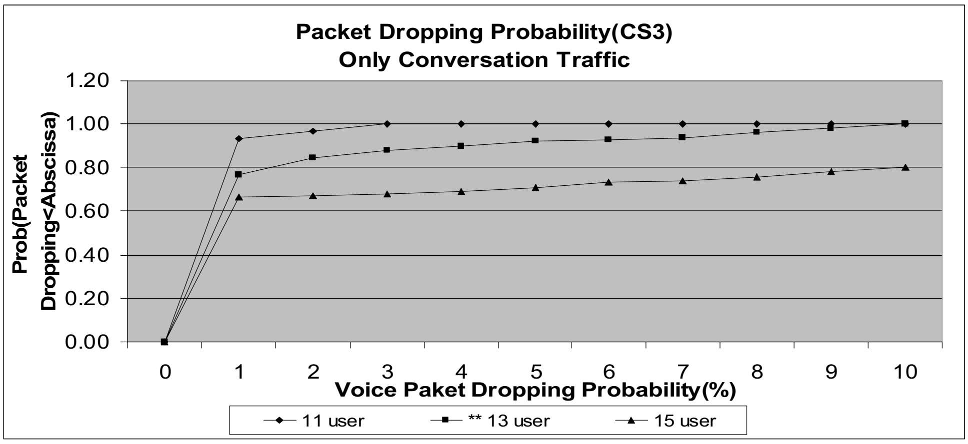


Figure 6.4 CDF of Voice Packet Dropping Probability of Conversation Traffic (CS3)

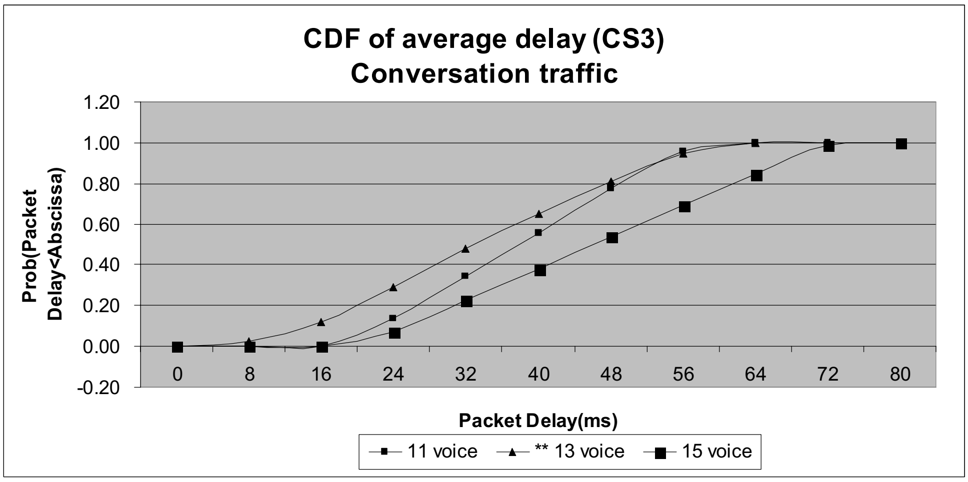


Figure 6.5 CDF of Average Packet Delay of Conversation Traffic (CS3)

The results illustrate that 13 is the maximum number of voice users that can transmit voice packets with 90% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 13 voices, the system can guarantee the QoS. In contrast, if the number of users is higher than 13 voices, the system cannot guarantee the QoS. In addition, Figure 6.5 shows that the users of less than 13 have 95% of all packets experienced the delay time of less than 56 ms.

In summary, the system can support the maximum number of users at 13 voices with guaranteed QoS. The capacity gain of 1.63 is obtained when compared with the system that does not implement the tree protocol. When the number of users becomes higher than 13, the performance is significantly decreased, and the system can not guarantee QoS for the conversation traffic. Thus, using CS3 makes the maximum number of users to 13, and that is about 18% higher than the case of CS2. Moreover, the most delay of all packets in CS3 is the same as in CS2. That is, the delay is about 56 ms which is less than the maximum limited delay of 60 ms. Thus, the number of users does not significantly affect the delay time neither.

(c) Experiment 2.3: Conversation Traffic in CS4

The parameters used in this experiment are shown in Table 6.7. The results of packet dropping probability and average packet delay are shown in Figure 6.6 and 6.7, respectively.

Table 6.7 Parameters of Experiment 2.3

Parameter	Value
Conversation Data Rate	On-Off model with 0.4 ratio
Number of Users	15 to 21 users
CS Rate	CS4 of 21.4 Kbps

The results illustrate that 18 is the maximum number of voice users that can transmit voice packets with 95% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 18 voices, the system can guarantee the QoS. In contrast, if the number of users is greater than 18 voices, the system cannot guarantee the QoS. In addition, Figure 6.7 shows that the users of less than 18 have 95% of all packets experienced the delay time of less than 56 ms.

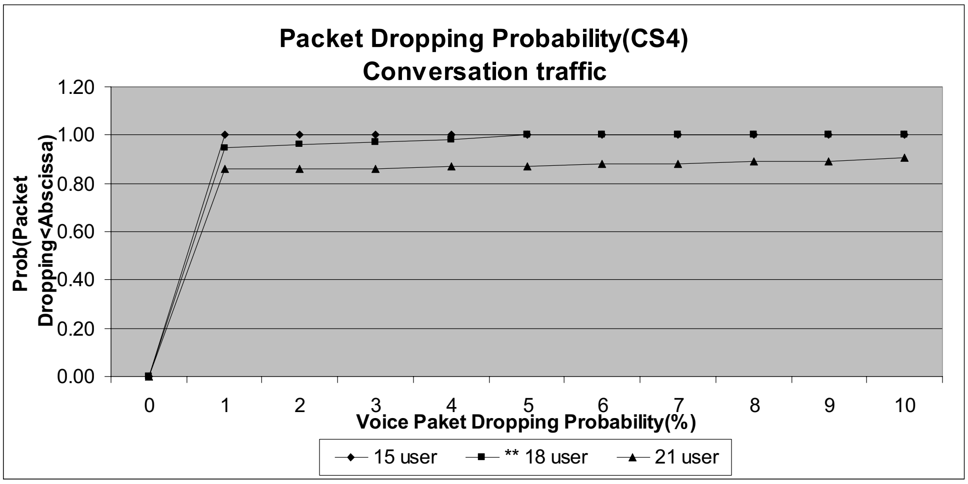


Figure 6.6 CDF of Voice Packet Dropping Probability of Conversation Traffic (CS4)

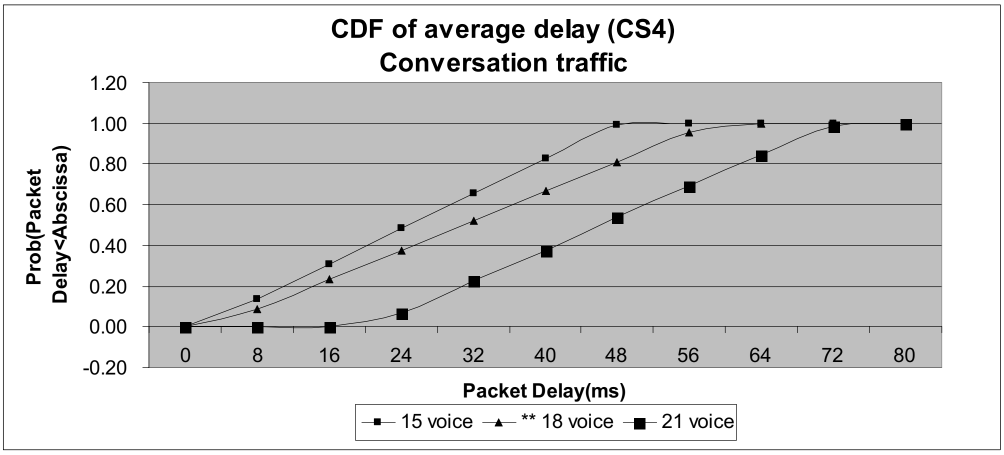


Figure 6.7 CDF of Average Packet Delay of Conversation Traffic (CS4)

In summary, the system can support the maximum number of users at 18 voices with guaranteed QoS. The capacity gain of 2.25 is obtained when compared with the system that does not implement the tree protocol. When the number of users becomes higher than 18, the performance is significantly decreased, and the system can not guarantee QoS for the conversation traffic. Thus, using CS4 makes the maximum number of users to 18, and that is about 64% higher than the case of CS2 which allows the maximum number of users up to 11 only. Similar to other cases, the most delay of all packets in CS4 is the same as in CS2 and CS3. That is, the delay is about 56 ms which is less than the maximum limited delay of 60 ms. Thus, the number of users does not significantly affect the delay time neither.

6.2.3 Experiment 3: Combined Conversation and Interactive Traffic

In this experiment, the performance of the combined conversation and interactive traffic is evaluated while maintaining the system QoS. The number of interactive users is varied from 0 to 7. In addition, the Poisson distribution is used to generate interactive traffic with the data rate between 22.4 and 56 Kbps while the conversation traffic is simulated with a limited delay time of 60 ms and the Mean Opinion Score (MoS) between 1 to 2% which are the same as the previous experiment. We divide this experiment into three parts with parameters as defined in Table 6.8 and described as follows.

Table 6.8 Parameters of Experiment 3

Experiment	**Coding Scheme**	**Number of Voice Users**	**Number of Interactive Users**
3.1	CS2	11	0 to 7
3.2	CS3	13	0 to 7
3.3	CS4	18	0 to 7

(a) Experiment 3.1: Combined Conversation and Interactive Traffic (CS2)

The results of packet dropping probability and average packet delay are shown in Figure 6.8 and 6.9, respectively.

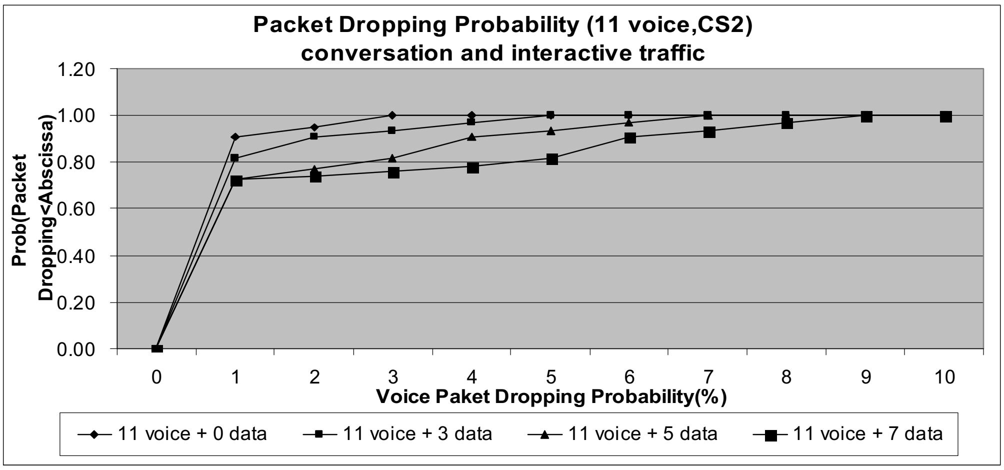


Figure 6.8 CDF of Voice Packet Dropping Probability of Experiment 3.1 (CS2)

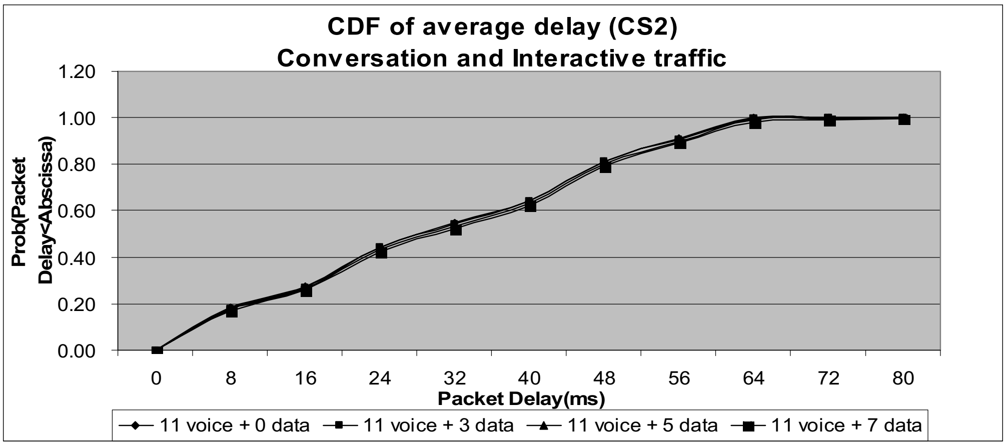


Figure 6.9 CDF of Average Packet Delay of Experiment 3.1 (CS2)

The results illustrate that 11 of voice users and 3 of interactive users can transmit voice packets with 95% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 11 voices with 3 interactive users, the system can guarantee the QoS. Otherwise, the system cannot guarantee the QoS. In addition, Figure 6.9 shows that all cases have most of the delay time about 56 ms, and the number of users does not significantly affect the delay time.

In summary, the system can support the maximum number of voice users at 11 voices and 3 interactive users with guaranteed QoS. The performance is significantly degraded when the number of interactive users becomes 5, and there are 70% of all packets experienced packet dropping of less than 2%, and the system cannot guarantee the QoS of voice users. Moreover, the number of interactive users does not significantly affect the delay time of voice packets.

(b) Experiment 3.2: Combined Conversation and Interactive Traffic (CS3)

The results of packet dropping probability and average packet delay are shown in Figure 6.10 and 6.11, respectively.

The results illustrate that 13 of voice users and 3 of interactive users can transmit voice packets with 75% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 13 voices with 3 interactive users, the system can guarantee the QoS. Otherwise, the system cannot guarantee the QoS. In addition, Figure 6.11 shows that all cases have most of the delay time about 60 ms, and the number of users does not significantly affect the delay time.

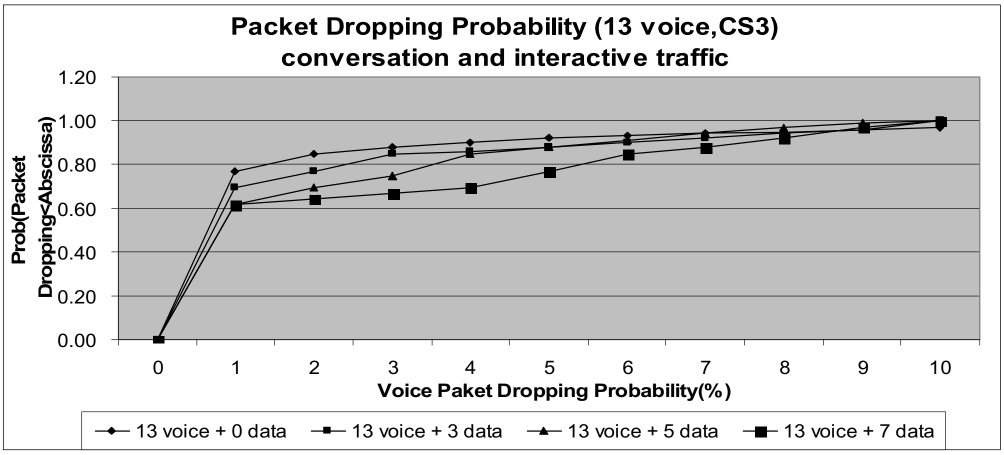


Figure 6.10 CDF of Voice Packet Dropping Probability of Experiment 3.2 (CS3)

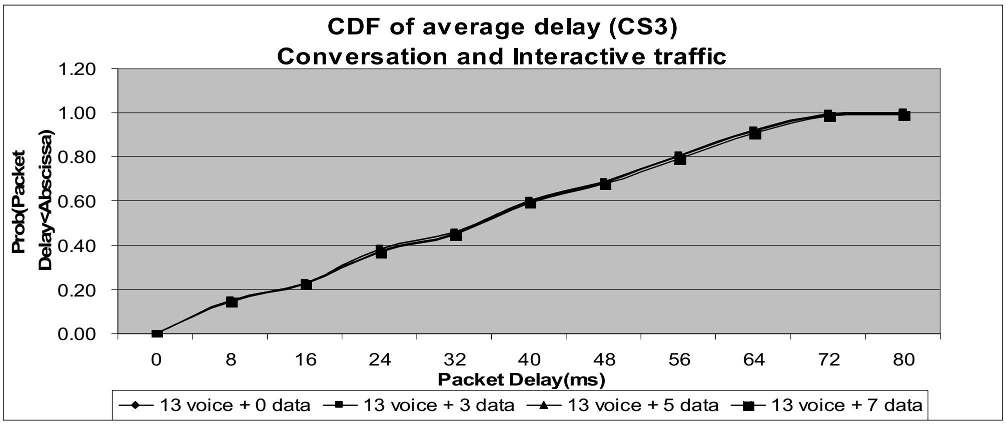


Figure 6.11 CDF of Average Packet Delay of Experiment 3.2 (CS3)

In summary, the system cannot support the maximum number of voice users at 11 voices and 3 interactive users with guaranteed QoS. There are 75% of all packets experienced packet dropping of less than 2%, and the system cannot guarantee the QoS of voice users. This result indicates that CS3 cannot transmit any data because of too high packet dropping of conversation traffic. However, the number of interactive users still does not significantly affect the delay time of voice packets.

(c) Experiment 3.3: Combined Conversation and Interactive Traffic (CS4)

The results of packet dropping probability and average packet delay are shown in Figure 6.12 and 6.13, respectively.

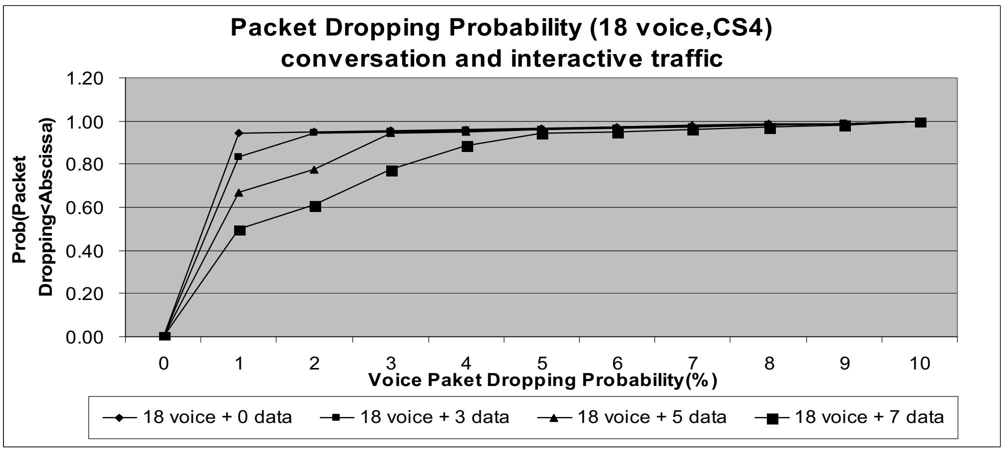


Figure 6.12 CDF of Voice Packet Dropping Probability of Experiment 3.3 (CS4)

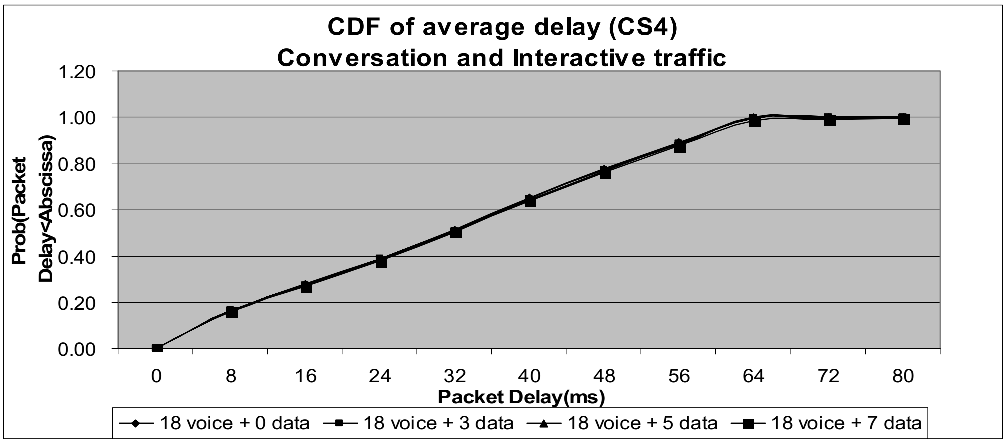


Figure 6.13 CDF of Average Packet Delay of Experiment 3.3 (CS4)

The results illustrate that 18 of voice users and 3 of interactive users can transmit voice packets with 95% of all packets experienced packet dropping of less than 2%. If the number of users is lower than 18 voices with 3 interactive users, the system can guarantee the QoS. Otherwise, the system cannot guarantee the QoS. In addition, Figure 6.13 shows that all cases have most of the delay time about 56 ms, and the number of users does not significantly affect the delay time.

In summary, the system cansupport the maximum number of voice users at 18 voices and 3 interactive users with guaranteed QoS. The performance will be degraded greatly when the number of interative users becomes 5, and there are 75% of all packets experienced packet dropping of less than 2%, and the system cannot guarantee the QoS of voice users. Similarly, the number of interactive users still does not significantly affect the delay time of voice packets.

6.2.4 Experiment 4: Combined Streaming and Conversation Traffic

In this experiment, the performance of the combined streaming and conversation traffic with guaranteed QoS is evaluated. The maximum streaming data rate of 107.2 Kbps using CS2 will be used as the baseline. This experiment simulates the maximum streaming rate and varies the number of conversation users in CS3 and CS4. We measure how much the capacity of the conversation traffic would be increased while transmiting the streaming traffic with the maximum CS2 data rate in CS3 and CS4. Similar to the previous experiment, the conversation traffic is simulated with the limited delay time of 60 ms and the Mean Opinion Score (MoS) between 1 to 2%. We divide this experiment into two parts with parameters defined in Table 6.9 and described as follows.

Table 6.9 Parameters of Experiment 4

Experiment	Coding Scheme	Number of Voice Users	Streaming Data Rate	Conversation Data Rate
4.1	CS3	1 to 3	107.2 Kbps	On-Off with 0.4 Ratio
4.2	CS4	3 to 5	107.2 Kbps	On-Off with 0.4 Ratio

(a) Experiment 4.1: Combined Streaming and Conversation Traffic (CS3)

The results of packet dropping probability and average packet delay are shown in Figure 6.14 and 6.15, respectively.

The results illustrate that the system has to reserve 7 slots for 107.2 Kbps streaming rate and has the remaining one slot for supporting a voice user. In other words, the system can support one voice user with no packet dropping. If there are two voice users transmit packets with 50% of all packets experienced packet dropping of less than 2%, then the system cannot guarantee the QoS. In addition, Figure 6.15 shows that only one voice user has 90% of all packets experienced the delay time of less than 60 ms. If the number of voice users is greater than one, 60% of all packets will experience the delay time exceeding 60 ms.

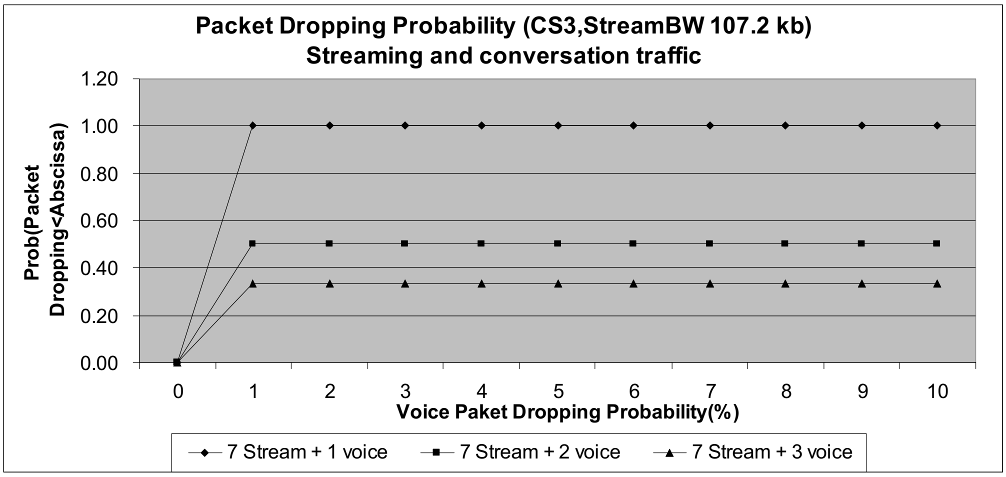


Figure 6.14 CDF of Voice Packet Dropping Probability of Experiment 4.1 (CS3)

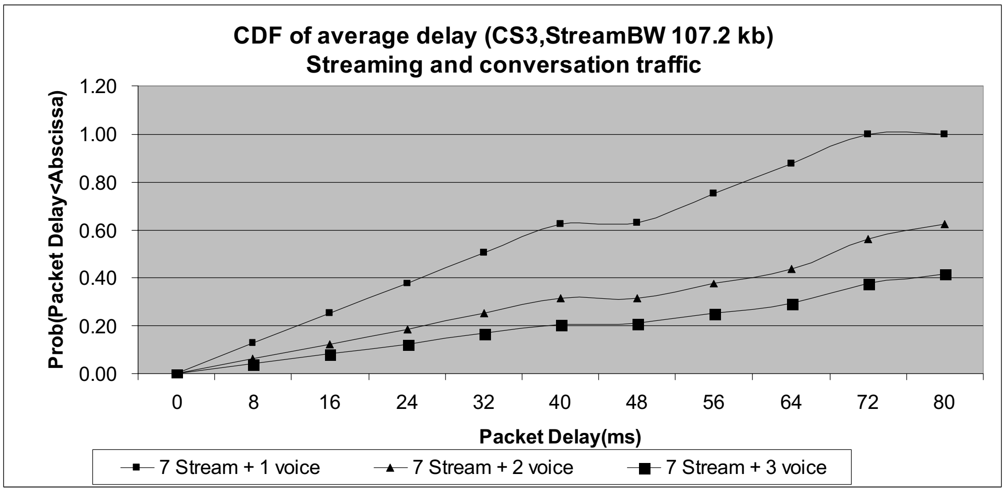


Figure 6.15 CDF of Average Packet Delay of Experiment 4.1 (CS3)

In summary, the system can support only one voice user. There is no increase in the capacity of the system, and the performance is obviously degraded when the system has two voice users. In addition, there are 50% of all packets experienced packet dropping of less than 2%, and thus indicating that the system cannot guarantee QoS of voice users. Since the Tree protocol does not reserve a slot when the system has only one available slot, the system performance cannot be improved for supporting voice users. Moreover, voice users experience 60 ms delay time which is higher than 56 ms of the previous experiment. However, it does not exceed the limited delay of 60 ms.

(b) Experiment 4.2: Combined Streaming and Conversation Traffic (CS4)

The results of packet dropping probability and average packet delay are shown in Figure 6.16 and 6.17, respectively.

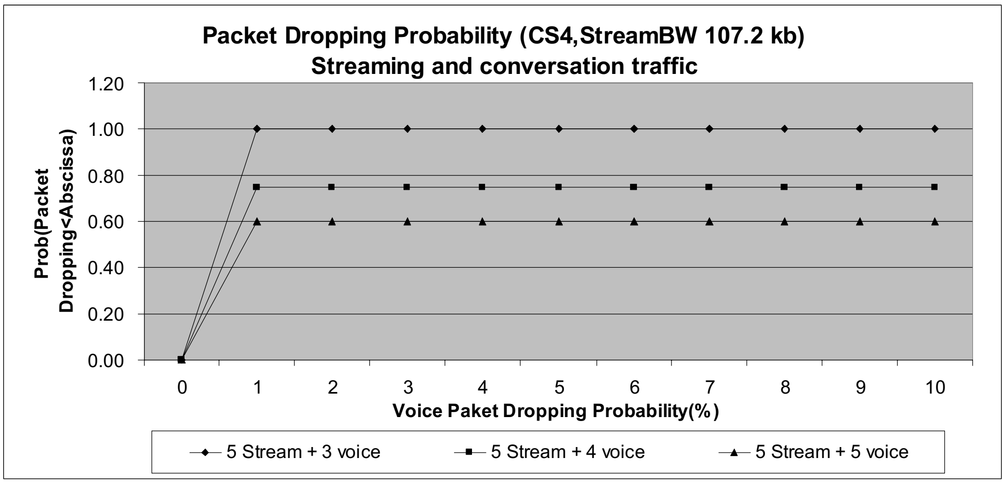


Figure 6.16 CDF of Voice Packet Dropping Probability of Experiment 4.2 (CS4)

The results in Figure 6.16 show that the system has to reserve 5 slots for 107.2 Kbps streaming rate and has the remaining 3 slots for supporting voice users. In other words, the system can support three voice users with no packet dropping. If there are 4 voice users transmit packets with 75% of all packets experienced packet dropping of less than 2%, then the system cannot guarantee the QoS. In addition, Figure 6.17 shows that three voice user has 90% of all packets experienced the delay time of less than 56 ms. If the number of voice users is greater than three, 80% of all packets will experience the delay time exceeding 60 ms.

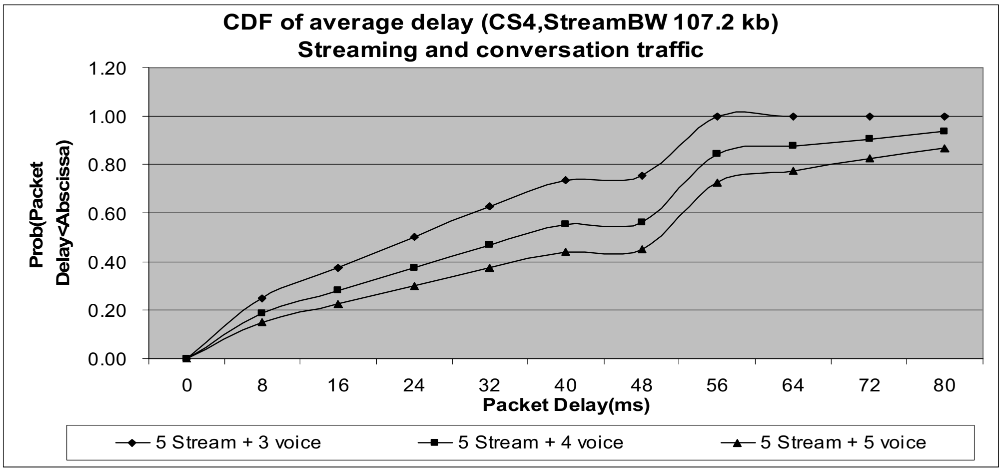


Figure 6.17 CDF of Average Packet Delay of Experiment 4.2 (CS4)

In summary, the system can support three voice users, and the system can support almost four voice users. Thus, the performance is increased to 33% with 4 voices using 3 slots, and the delay time does not exceed 60 ms. If the system has three available slots, the Tree protocol can reserve two slots, and the performance has been much improved. Furthermore, most voice users experience 60 ms delay time which is higher than 56 ms delay time of the previous experiment. However, it does not exceed the limited delay of 60 ms.

6.3 Summary of Experimental Results

The simulation results show that applying CS adaptation can increase the overall performance of the GQ-MAC Protocol for both the single traffic and the combined traffic cases. However, the performance of each case does not show the same value as summarized in Table 6.10 below. In addition, the original experiment of the GQ-MAC Protocol [1] simulated only CS2, and in our experiments, we obtain the similar trend in our output. But, we do not compare our results with those in the original work [1] since we use different simulation tools even though we mostly use the same parameter values with them.

Table 6.10 Comparison of All Experimental Results

Experiment	Summary	Performance
1. Streaming Traffic	Maximum bit rate at 124.8 Kbps with CS3.	Increased 16% from CS2.(I)
	Maximum bit rate at 171.2 Kbps with CS4.	Increased 60% from CS2.(I)
2.Conversation Traffic	Support maximum 11 voice users with CS2.	Increased 38% from maximum 8 voices (II)
	Support maximum 13 voice users with CS3.	Increased 62.5% from maximum 8 voices.(II)
	Support maximum 18 voice users with CS4.	Increased 125% from maximum 8 voices.(II)
3. Conversation and Interactive Traffic	Support maximum 11 voice users and 3 interactive users with CS2.	Increased 3 data users.(III)
	Support maximum 13 voice users and 3 interactive users with CS3.	Not increased data users.(III)
	Support maximum 18 voice users and 3 interactive users with CS4.	Increased 3 data users.(III)
4. Streaming and Conversation Traffic	7 slots reserved for 107.3 Kbps streaming rate and support 1 voice user	Increased 1 voice users.(IV)
	5 slots reserved for 107.3 Kbps streaming rate and support 3 voice users	Increased 3 voice users.(IV)

Note that the Roman number in the last column of Table 6.1 has specific meanings as follows.

(I) Compared with Streaming traffic using the maximum data rate in CS2 at 107.3 Kbps.

(II) Compared to the system that does not implement the Tree protocol at the maximum 8 voice users.

(III) Compared with the results of Experiment 2.

(IV) Compared with the results of Experiment 1.

CHAPTER VII
DISCUSSION AND CONCLUSION

In this chapter, we discuss the experimental results presented in Chapter 6. The topics to be discussed include the performance, the limitations, and the implementation of the proposed model. Finally, we give the conclusions of our work and suggests some future work.

7.1 Discussions

From the experimental results, we discuss the issues given as follows.

- **Performance**

  The performance of the proposed model is in accordance with the idea to improve the efficiency of the GQ-MAC protocol. The link adaptation technique is put into the GQ-MAC protocol and the model is simulated. In other words, when the coding scheme is changed, all traffic will adaptively transmit packets accordingly. In summary, all experiments show better performance more or less at different degree of improvement.

- **Limitations of the Proposed Model**

  The adaptive GQ-MAC protocol proposed has some limitations as shown in Figure 7.1 that as the coding scheme is reduced, the capacity of the system will be decreased during packet transmitting. The result will affect users using the service at that time. For example, there are 13 conversation users getting services with CS3. If the coding scheme is adjusted to CS2, 95% of conversation users will experience packet dropping more than 2%. As a result, some users may be denied in gettting a slot, and certainly the service too.

  All traffic types except interactive traffic are affected because interactive packets do not limit the delay time. Thus, the interactive traffic can still guarantee the throughput with distributed scheduling algorithm. In practice, the coding scheme and hardware modification are essential issues. However, the link adaptation in practice is very complicated. For practical use, the system, a mobile station and corresponding hardware must be modified.

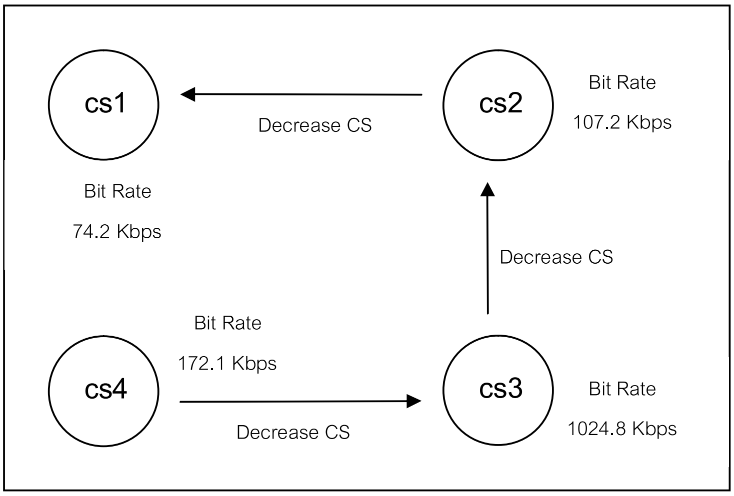


Figure 7.1 Reduction of Coding Scheme for Link Adaptation

- **Implementation**

In the original GQ-MAC protocol, the popular OPNET network simulator is used in the simulation. OPNET is a network simulation tool that provide network functions and facilities such as the GSM uplink-downlink Slot or even the Slot ALOHA protocol. However, we use the CSIM generic simulation tool which does not provide any network facilities. Thus, we cannot compare our experimental results with those in the original work.

7.2 Conclusions

The conclusions of the proposed work can be drawn as follows.

1. The adaptive GQ-MAC protocol proposed mostly performs better than the original work in changing environment. But, in some cases, there is no improvement.
2. The adaptive coding scheme is difficult to be implemented in practice since the modification of hardware, mobiles and algorithms are necessary.
3. Other parameters in the implementation is needed in several aspects so that the results close to the real systems are possible.

7.3 Suggestions for Future Work

1. The Tree protocol can be modified for having fair reserved slots. If the average delay time of each user is close to the delay of all users, the performance may be better or the number of users would be increased.
2. The Slot ALOHA protocol should be included in the implementation for the sake of the model completion.
3. If some buffer management is included when handling streaming traffic, the performance will be better, and the CS adaptation would be more flexible.

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