Radio Resource Allocation and Hybrid Multiplexing of Voice and Data over IP in a GSM/GPRS Cellular Network

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A research report submitted to the Faculty of Engineering and the Built Environment, University of the Witwatersrand, Johannesburg, in partial fulfilment of the requirements for the degree of Master of Science in Engineering.

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Declaration

I declare that this research report is my own, unaided work, except where otherwise acknowledged. It is being submitted for the degree of Master of Science in Engineering to the University of the Witwatersrand, Johannesburg. It has not been submitted before for any degree or examination in any other university.

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David Emmanuele Vannucci.

Abstract

In this study, a first order investigation of the issue of resource allocation between circuit voice, packet data and packet voice was completed. The study was done with reference to the GSM / GPRS air interface. To study the allocation of resources, suitable traffic source models were developed to represent the nature of the traffic offered to the base station subsystem. Circuit voice and packet data were represented using Markovian arrivals and exponentially distributed holding times. Voice over IP was modelled using a two-state Markov modulated Poisson process. The base station subsystem was modelled as a continuous time controller with eight channels (one GSM / GPRS TDMA frame). The radio propagation environment was considered by means of a large-scale propagation model, which would merely alter the load presented to the developed simulator package. From the results of the simulations, it was found that insufficient data resources lead to similar packet delay regardless of the packet size. It was found that if capacity on demand is used, then the data resources could equal the load. In the case of sufficient data channels, with capacity on demand, additional channels have a greater effect on average delay than the probability of it occurring. Prioritisation of VoIP packets did not significantly alter the probability of delay but affected the average packet delay. Packet size had a greater effect on average delay than the probability of delay. In the case of all eight channels being used for VoIP and data, the combined load should not exceed seven erlang, indicating that a higher voice load could be supported with VoIP than with circuit switched voice.

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Chapter 1

Background

In the past, circuit switched voice and packet switched data have been operated by a majority of service providers as two separate networks, having completely different structures. Today's networks are rapidly converging to become a single, homogeneous next generation network capable of satisfying all user requirements simultaneously. Whilst this network is slowly becoming a reality with entities such as the soft switch, the concept of combining voice and data is not new. This topic has been of considerable interest since the advent of packet switching and its multiplexing gains.

The Integrated Services Digital Network (ISDN) is a prime example of a product which has evolved from these ideas. ISDN supports voice communications whilst transmitting and receiving data, provides the advantages of a single access, high data rates and the flexibility to operate both services simultaneously (Jr. Andrews 1984). Although the manner in which this is done does not entirely create a converged network, due to the fact that the only integrated feature ISDN possesses is the physical path from the exchange to the customers' premises (Turner 1986). Nevertheless, a substantial amount of useful queueing theory was applied to understanding these networks and their properties, as discussed in (Schwartz 1987, pg. 661–668) and (Higginbottom 1998, pg. 329–334). Gimpleson (1965) analyses the mixture of wide and narrow band traffic as both a blocked calls lost network and a blocked calls queued network, with the conclusion that the economy which results from combining traffic types has to be greater than the cost of the queueing facility and increased trunking capacity. With current processing power and rapid evolution in high speed communication links, the processing complexity and required bandwidth is becoming less of an issue. However, when the case of wireless systems is considered, the radio resources are once again limited to the allocated frequency bandwidth. Much deliberation on the distribution of resources has to be performed, to reduce resource wastage and increase the usage efficiency of those frequencies allocated to a particular station.

Services such as banking, trade, tourism, education and entertainment are increasingly making use of Information and Communication Technologies (ICT) by means of the Internet, mobile phone transactions, point to point connections and virtual private networks. Cellular networks are rapidly adopting suitable network models for supporting packet switched data services and as such, are expected to be able to support a variety of packet switched data services such as Internet access and e-mail (Koodli and Puuskari 2001). As users are becoming increasingly familiar with the use of technology, they are demanding greater flexibility and mobility, performing transactions out of the office and at any time of day. GSM phones offer the most convenient medium of access, with the ability to provide transactions over a highly secure interface, Internet access through WAP, or as a wireless modem for the user's notebook. With the introduction of GPRS to GSM, the support of packet based billing allows the user to always be connected to the network with an always-on IP connection. By combining voice and data onto a common IP platform, the user can benefit from additional services such as push to talk, Voice over IP which allows various bandwidths and thus options in call cost, click-to-talk services and voice advertising during mobile transactions. In addition to the perceived user benefits, the operator would also have a consolidated management platform, and also possibly benefit from increased efficiencies due to the statistical multiplexing of packets, especially if voice is transmitted as VoIP in the core of the network, where a large degree of aggregation could occur.

1.1 Combining Voice and Data

The criteria for handling the transmission of voice and data are extremely different, especially where tolerable data loss and delay are concerned. For example, data is usually far more loss sensitive than voice, but relatively delay insensitive (although according to Mitrou et al. (1993) voice should not have a bit error rate (BER) exceeding 10^{-3}), whereas voice requires delays of less than 300 ms (Mitrou et al. 1993). Voice is also extremely sensitive to out of sequence delivery, and fluctuation in delay (known as jitter). Thus the understanding of bandwidth allocation issues is extremely important in order to provide an acceptable Grade of Service with such different services.

Bandwidth allocation usually falls into three categories (Mitrou et al. 1993):

1. *Circuit switching* - Resources are granted to a connection for the entire duration of the connection and may not be reassigned for other use. This dedicated allocation of resources results in a predictable grade of service but is extremely inefficient if data is not being continuously transmitted over the link.

2. *Burst or Packet switching* - Resources are allocated for the duration of a data burst or talkspurt period. However the prediction of the duration of the burst can be inaccurate and resources are not guaranteed for bursts which follow the initial burst. *Packet switching* is a refinement of burst switching where resources are allocated to serve the immediate requirement. This makes the most efficient use of resources but requires very fast reallocation of resources, as is the case with the General Packet Radio Service over GSM (Bettstetter et al. 1999), when resources are allocated on a case by case basis.

3. *Hybrid switching* - Hybrid switching supports both circuit switching and burst switching, combining the strategies in three different manners:

    - Fixed Boundaries - In such a scheme, a dedicated portion of the available resources are allocated for a particular service, usually in a circuit switched manner, whilst the remainder are freely allocated to other services.

    - Moving Boundaries - The allocation of resources can be freely altered but one particular service is guaranteed a certain amount of resources as in a fixed boundary, whilst other services can use those resources if unallocated.

    - No boundaries - This allocation allows all services of all available resources, usually in a packet switched manner.

Intuitively, hybrid switching provides both the benefits of circuit switching and packet switching, with moving boundaries making the best use of the available resources whilst still being able to provide a guarantee of resources to a particular service. No boundaries is similar to a moving boundary except that no service receives a greater degree of guaranteed resources than another. The access control or bandwidth allocation method is the fundamental constraint of the network when considering factors such as quality of service and performance. Each access control performs differently, depending on the environment, and is usually tailored to a certain expected traffic behaviour. A hybrid switching protocol is discussed by

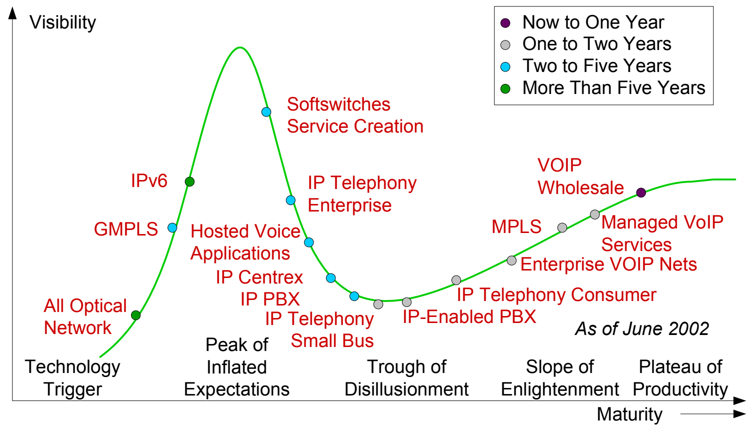


Figure 1.1: The VoIP hype cycle (Winogradoff 2002)

Mahdavi, Edwards and Cvetkovic (2001) and Okuda et al. (2002) in which resources are shared between users for all services, with circuit switched voice having priority and preemption. In their system the last voice user is delayed for an imperceptible time, which decreases the mean delay in the buffer. Miyake (1988) considers an ISDN user interface shared between several terminals. He finds that when the number of terminals is small the call congestion is small, and that the stationary distribution of the number of calls in a system with a general service time distribution depends only on the mean of the service time. With the growing use of Voice over Internet Protocol (VoIP) (whereby a user's voice is converted into a digital signal, compressed and then broken down into a series of packets (Varshney et al. 2002)), allowing voice to be transmitted over packet networks such as the Internet. It is increasingly likely that the benefits of VoIP will be carried through to the mobile operator. The problem of performance analysis with priority queues and correlated arrivals, as with a VoIP codec, is analysed by Ali and Song (2003), modelling a multiplexer to understand the effect of priority mechanisms.

The idea of VoIP was introduced in the early 1970's (Schulzrinne 1999), but due to the then generally low bitrates it was not a viable alternative until the mid 1990's. With the current rapid growth of the Internet and the widespread deployment of an IP based infrastructure, VoIP is becoming a viable technology (Winogradoff 2002; Hafner 2003), and as such it is being heavily developed especially in the wholesale market (see figure 1.1 for a general prediction of the status of VoIP on the Gartner hype cycle), resulting in increasingly superior codecs and call signalling and interworking functions between the conventional circuit switched network and packet switched networks.

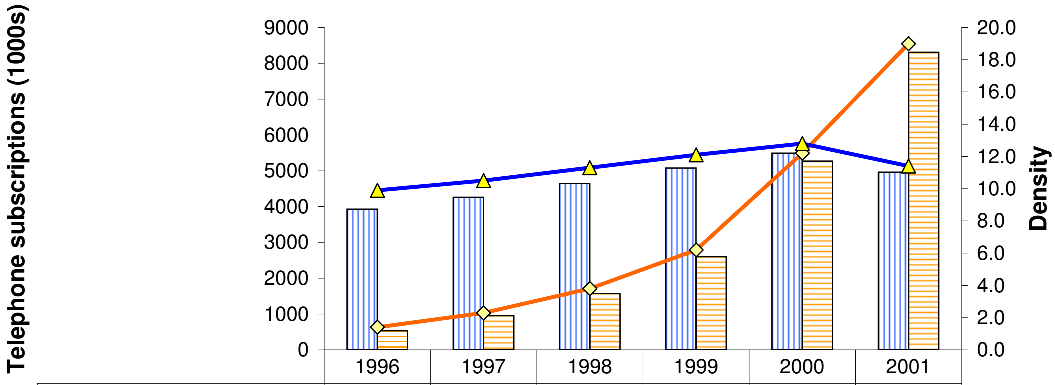
1.2 Wireless Voice and Data

Mobile telephony in African countries has proven to be far more effective than fixed-line telephony for a number of reasons:

- Installation is faster - individual copper pairs do not have to be installed in customer premises
- Prepaid contracts allow telecommunication services to be offered to those who would not be able to qualify for post-paid services - and facilitate asymmetrical access, allowing those who can afford the basic GSM handset a means of receiving calls, and sending text messages.
- Coverage areas are greater using wireless - the size of the network can be reconfigured as capacity or coverage limited, allowing a large amount of users to be serviced with a single base station.
- Fraud and illegal use of telecommunication services is reduced. The possibility of making fraudulent calls is reduced since the air interface is more secure than that of a fixed line.

In addition, the management of accounts within a mobile network is easier due to each subscriber having a unique identity number stored within both the phone and the subscriber identity module. The suitability of mobile telephony is illustrated by the fact that in a study done by the ITU (2001), it is predicted that by 2005 there will be three times as many mobile subscribers as fixed-line subscribers. This trend is believable, considering the growth of mobile subscribers in South Africa compared to fixed-line subscribers, as shown in figure 1.2.

Wireless links introduce a number of problems which would be added to those of a wireline environment. For example radio channels experience propagation disturbances such as frequency dependent fading (Mitrou et al. 1993), reflection (Neskovic et al. 2000), diffraction (Rappaport 2002, pg. 126), delay spread due to multipath effects and signal decay as a function of separation distance (Rappaport 2002, pg. 138). To minimise these errors, a number of techniques are implemented such as interleaving to reduce burst errors, antenna diversity to minimise multipath interference, frequency hopping to reduce frequency dependent fading, as well as encoding techniques such as forward-error-correction and advanced encoding. There are also a number of delays introduced into wireless environments (Balaji et al. 2000), for example the processing delays, the channel allocation delays and propagation delay.



	1996	1997	1998	1999	2000	2001
Fixed	3926	4259	4645	5075	5493	4962
Mobile	535	953	1574	2600	5269	8308
Mobile per 100 inhabitants	1.4	2.3	3.8	6.2	12.2	19.0
Fixed per 100 inhabitants	9.9	10.5	11.3	12.1	12.8	11.4

Figure 1.2: Fixed and mobile telephone subscribers in South Africa (1000s) (ITU 2001)

Chaskar and Madhow (2001) analyse statistical multiplexing on wireless links with interleaving. They consider two sources of delay, 1) the queueing delay at the base station buffer and packet loss due to buffer overflow, and 2) link shaping and coding delay. From this study they develop a framework for quality of service (QoS) provisioning over the wireless link. Various resource allocation schemes are also considered by Moorhead et al. (2000), and the authors find that the mechanism used to reserve resources for the VoIP packets is central to the use of VoIP over GPRS.

Mitrou et al. (1993) investigate the integration of voice and data over an air interface, in order to facilitate the efficient sharing of network resources among various services, thereby reducing the overall cost and improving the flexibility of the network management and configuration. A number of benefits from the integration of voice and data over an air interface are enumerated below (Bi et al. 2001):

- Efficient usage of limited radio bandwidth due to statistical multiplexing.
- Contributing to the development of an understanding and implementation of a converged network supporting a variety of different services.
- A simplified channel management due to homogeneous services.
- Flexible and faster service creation due to a common platform.

1.3 Global System for Mobile Communication (GSM), Research Objectives

In this dissertation, hybrid multiplexing of voice and data is considered with respect to the GSM network. GSM as a second generation technology provides circuit switched capabilities for both voice and data. To support packet switching, the GSM network can be upgraded to support General Packet Radio Service (GPRS), which adds additional software and hardware to the network, as discussed further in Chapter 2. The radio resources of the air interface between the mobile terminal and base station are shared using hybrid allocation methods (Brasche and Walke 1997; Bettstetter et al. 1999).

The standard used in GSM/GPRS is circuit switched voice and packet switched data, supporting services such as multimedia messaging (MMS) and Internet browsing. The use of Voice over IP has been a subject of intense interest as it offers the possibility of furthering the benefits of statistical multiplexing and making better use of the radio resources.

Whilst such a service is currently not implemented using GPRS, a future implementation is not inconceivable, especially as it would provide the last step towards a completely unified voice and data network using the Internet Protocol, a step seen as the inevitable future despite the challenges created by protocol overhead (Bi et al. 2001). Thus there is a requirement by operators to understand the needs of users, and the behaviour of the network under varying resource allocations. This dissertation provides an analysis of such a situation (hybrid multiplexing of voice and data), using the GSM/GPRS operation as a case study.

This work has a number of objectives some of which are:

- To analyse the performance of a hybrid multiplexing scheme, with circuit switched voice, packet switched data and packet switched voice, with varying allocated resources over a common air interface.
- To develop acceptable models for the operational environment for such a network.
- To propose resource allocation bounds from the results of the studies.

1.4 Overview of Research Report

To achieve the goals outlined in the previous section, this report is structured as follows: The GSM/GPRS network is discussed briefly in Chapter 2 within the scope of the details that are required in order to model the operation of the GSM/GPRS network. In Chapter 3, the requirements and considerations for VoIP operation are discussed with respect to boundaries and performance in a limited resource data switched environment. Chapter 4 introduces the models developed and the basic queueing theory applied to the investigation, application of the queueing theory is shown for the case of fixed boundary hybrid multiplexing. The verification and validation of the developed simulator is discussed in Chapter 5, and the functioning of the simulator is illustrated in Chapter 6. The results of the simulations are presented in Chapter 7. An overview of the results and recommendations made from the findings are reported in Chapter 8.

1.5 Key Achievements of the Study

The key achievements of this study are as follows:

- Traffic source models were developed to simulate the allocation of resources in a hybrid resource allocation environment such as GSM/GPRS.
- A computer simulation was designed using these models to predict the performance under varying loads. The source code for the simulator is contained in the compact disc, in the Appendix.
- Using the developed simulator, recommendations and conclusions could be made as to the behaviour of the modelled GSM/GPRS base station controller under varying loads and resource allocations.

The following chapter describes the basics of the GSM/GPRS operation and the components within the network.

Chapter 2

GSM and GPRS Operation

This chapter introduces the components, and operation of a GSM network necessary for understanding the details of the study.

2.1 A Brief Background on GSM

In the mid 1980's Wireless mobile communication was already well underway with the Advanced Mobile Phone System (AMPS) widely deployed as a proven technology. However, demand was increasing at a rate AMPS could not sustain, thus new second generation systems were developed to address the increased capacity requirements. Three technologies were standardised and deployed in the USA (Halonen et al. 2002): IS-95 (a 1.25 MHz CDMA carrier scheme), IS-136 (a TDMA technology utilising the AMPS 30 kHz structure), and GSM (a European 200 kHz TDMA standard at 900 MHz). GSM was defined by the standardisation body "Group Special Mobile" within the *Conference Europeenne des Postes et Telecommunications* (CEPT) (Halonen et al. 2002). The major GSM milestones are shown in table 2.1 (Bates 2002, pg. 4). From the 1990's most GSM development was conducted by the European Telecommunications Standards Institute (ETSI) Special Mobile Group.

GSM was rapidly proving to be the leader in these three standards, due to its roaming and diverse implementation. As further countries adopted GSM, it benefitted from large economies of scale for its equipment and software. With GSM currently used in 197 countries and 863.6 million subscribers [1] it truly stands for the Global System for Mobile Communications (GSM-Association 2003).

[1] As of May 2003

Table 2.1: Some milestones in GSM history

Year	Event
1982	CEPT establishes a GSM group for pan-European cellular mobile.
1985	Recommendations generated by CEPT are accepted.
1986	Field tests are performed to test the different proposed radio techniques for the air interface.
1987	TDMA/FDMA is chosen as the access method. The initial *Memorandum of Understanding* is signed by telecommunication operators representing 13 countries in Copenhagen, Denmark.
1988	GSM system is validated.
1989	The responsibility of the GSM specifications is passed to ETSI.
1990	GSM Phase 1 delivered.
1991	Commercial launch of GSM phase 1.
1992	First roaming agreement signed between Telecom Finland and Vodafone UK.
1993	GSM subscribers break through the 1 million mark.
1995	GSM Phase 2 standardisation completed. Fax, video and data communication demonstrated.

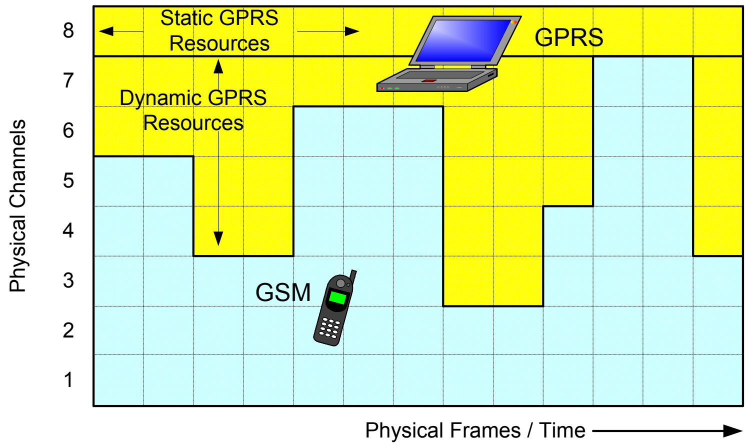


Figure 2.1: GSM/GPRS resource allocation

2.1.1 GPRS - General Packet Radio Service

General Packet Radio Service (GPRS) is an upgrade to the GSM network, increasing the number of connections per bearer by using the physical channels more efficiently than the previous Phase 2 service (Walke 2002, pg. 287). GPRS is the first implementation of packet switching within GSM on the evolutionary path to a 3G network.

The benefits of GPRS are most notably the efficient use of radio resources, as with a packet switching technology radio resources are only used when radio resources are actually sending or receiving data. Thus the radio resources are more effectively shared between several stations. GPRS also provides multiplexing of logical connections on GSM channels, providing a variable bit rate due to flexible channel capacity (Brasche and Walke 1997; Granbohm and Wiklund 1999) on a case by case basis, as illustrated in figure 2.1. In this figure, one GSM/GPRS TDMA frame is shown with eight time slots, in which one has been reserved for data and any unused channels are dynamically allocated to GPRS.

Due to GPRS's statistical multiplexing and flexible resource allocation, it is extremely efficient in its use of scarce spectrum resources (Bettstetter et al. 1999) . Since long packets reduce the effect of multiplexing gain, GPRS is most suitable for two traffic types (Walke 2002, pg. 286):

1. Frequent, regular transmission of short packets of length up to 500 bytes.
2. Irregular transmission of packets of length up to a few kilobytes.

2.2 Introduction to GSM Architecture

This section introduces the GSM architecture, the basic components and functions. Several functional entities exist within a GSM network: the *mobile station* which the subscriber uses, the *Base Station Subsystem* (BSS) which contains the radio control and the *Network Subsystem* (NSS) which forms the core logic and control of the network. A basic diagram of the GSM network with additional GPRS components is shown in figure 2.2. In this research, only the radio resource allocation was considered, thus only the MS, and BSS were investigated as these were considered to pose the bottleneck for user access as found by Lindemann and Thümmler (2003).

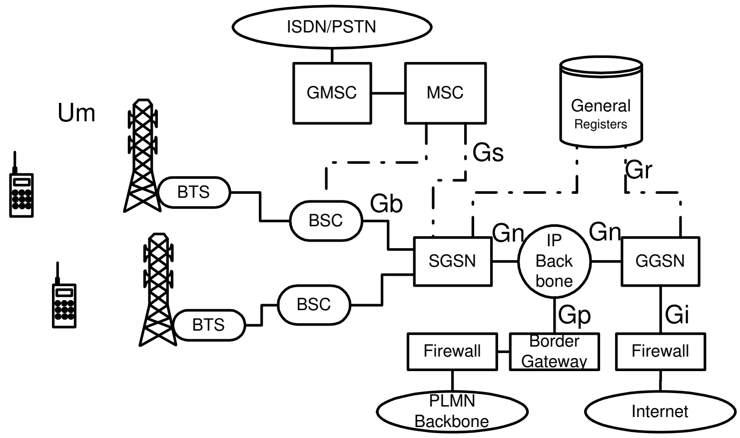


Figure 2.2: Basic GSM/GPRS architecture

2.2.1 Mobile Station

The MS consists of the handset and *Subscriber Identity Module* (SIM). The SIM uniquely identifies the subscriber within the network by means of an *International Mobile Subscriber Identity* (IMSI), and is portable, allowing the subscriber to use any mobile handset. Each handset contains an unique *International Mobile Equipment Identify* (IMEI), used to identify the handset.

2.2.2 Base Station Subsystem

The BSS consists of the *Base Transceiver Station* (BTS), and the *Base Station Controller* (BSC). Connections originating from or terminating on the fixed networks are routed through the Gateway Mobile Switching Centre (GMSC). GSM networks have a hierarchical structure, with at least one administrative region (comprising multiple location areas) assigned to a particular Mobile Switching Centre (MSC) (Bettstetter et al. 1999). Among the general registers used for call control are the: Home Location Register (HLR), Visitor Location Register (VLR), the Authentication Centre (AUC) and Equipment Identity Register (EIR) (Brand and Aghvami 2002, pg. 100). Permanent and temporary data of registered users is stored in the HLR. At all times the HLR knows the position of the subscriber, and is queried in order to route calls to the correct base station controller. The VLR is responsible for a number of location areas and the subscribers in these area, maintaining temporary HLR information for faster access.

Base Transceiver Station

The BTS contains the radio modems, and radio link protocols. The BTS is visibly identified as the mast, antennas and enclosure containing the equipment. Each BTS serves a single cell and can also perform the following functions (Bates 2002, pg. 19–20):

- Coding, multiplexing, modulating and encrypting the RF signals.
- Transcoding[2] and rate adaption.
- Time and frequency synchronising
- Timing advances
- Channel measurements

Four Coding schemes are implemented in GPRS, Coding Scheme 1 (CS-1) to CS-4, for the radio link control data blocks. As can be seen from table 2.2, CS-1 has a high level of error correction, whilst CS-4 has no error correction but a higher data rate. Each coding scheme is paid for by the operator individually, with CS-1 being mandatory for GPRS operation, whilst CS-2 and CS-3 are most commonly implemented. Since GPRS Phase II, the packet's encoding is determined dynamically based on factors such as the measured link quality, and as such an average bit rate has to be assumed for the sake of simulation, as discussed in Chapter 7.

[2] The direct digital-to-digital conversion from one encoding scheme to a different encoding scheme without returning the signals to analogue form (American National Standard 1996)

Table 2.2: The four GSM coding schemes (Bates 2002)

Coding Scheme	Code Rate	RLC/MAC Block data size (bytes)	RLC/MAC max throughput (Kbps)
CS-1	1/2	20	8
CS-2	2/3	30	12
CS-3	3/4	36	14.4
CS-4	1	50	20

Base Station Controller

A single BSC controls a number of BTSs and manages the radio resources. The BSC handles radio channel setup, frequency hopping, and handovers (Bates 2002). The BSC translates 13 Kbps voice used over the radio link to the standard 64 Kbps channel used over the PSTN and controls the power levels of attached BTS and MS in the cells. The BSC assigns and controls the allocation of time slots and frequencies. One BSC can serve up to nine fully loaded BTSs (Bates 2002), or more if only partially loaded.

2.3 Towards an All IP Infrastructure

The use of an all IP network has a number of advantages: for example the use of generic data networking equipment reduces the infrastructure cost, service creation on a common platform is faster and a greater number of services can be offered to users. In (Chitamu and Vannucci 2002) another two key reasons are given for implementing a packet switched data-centric network: a packet switched network allows an operator to provide a wide range of varying bandwidth services, such as e-mail, voice, video and Internet browsing. Packet switching also facilitates effective use of resources, especially necessary in lowering the average cost per user. The support of multimedia services such as video and packet voice are two key applications in an all IP network, as these are services that subscribers will use frequently. Fabri et al. (2000) analysed the performance of MPEG-4 video using GPRS. They found that with ten frames per second, at an average throughput of 32 kbit/s, the quality was comparable to that of a wireline environment.

The transportation of voice over IP is considered as a possible method of increasing

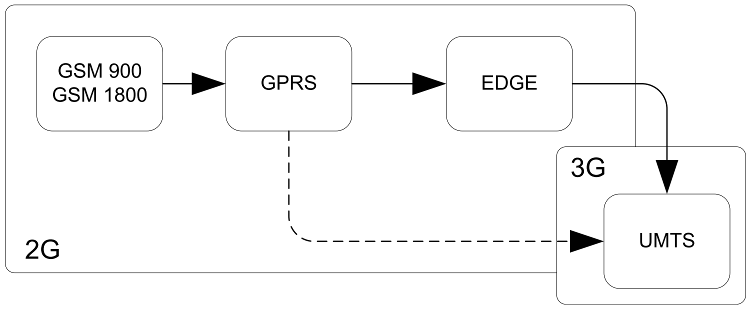


Figure 2.3: Evolutionary path of GSM (Bates 2002)

the capacity of the GSM network. As such, methods to enable VoIP services have received much attention. A mechanism to enable VoIP is discussed by Chang et al. (2003) in which the costly MSC is replaced with a VoIP softswitch routing voice through the GPRS network to a H.323 Gateway, as opposed to the public circuit switched telephone network. Another scenario similar to *Telecommunications and Internet Protocol Harmonisation over Network* (TIPHON) (ETSI 1998) and *GSM on the Net* (Granberg 1998) is discussed by Rao et al. (2000), where VoIP is provided as a service without any additional access equipment.

The use of the Internet Protocol is viewed as a fundamental component of any third generation network, and the shift to an all IP network is seen as a part of the natural evolution of the network. One evolutionary path proposed for GSM is that of GPRS and later Enhanced GPRS (EGPRS), or alternatively from GPRS to Enhanced Data rates for Global Evolution (EDGE), as shown in figure 2.3. EGPRS can be implemented without modifying the logical link control layer for GSM/GPRS but the radio link control/medium access control has to be modified to allow for the more efficient multiplexing and link adaption (Bates 2002, pg. 339). Due to the cost of upgrading GSM to support GPRS, it is understandable that the operator would like to leverage most benefit from the existing resources, thus being able to transport voice over IP using GPRS would allow the offering of some 3G-like services. Should this be possible it could extend the lifespan of GSM/GPRS in the presence of 3G technologies, perhaps even mitigating the need to implement an outright upgrade to a 3G technology.

In the following chapter VoIP and its requirements are reviewed with respect to some of the challenges facing its implementation in a GSM/GPRS network and environment.

Chapter 3

VoIP and its Application to GSM/GPRS

This section describes what Voice over Internet Protocol is, the fundamental traffic theory, statistical models, applications and analysis of VoIP, as applied in a network. This information is used for the simulation of the VoIP sources in Chapters 4 and 5 where the voice activity detection is taken into account as well as factors such as header compression.

Benefits of VoIP VoIP has the potential to provide a unified IP network through GPRS by combining voice and data over IP. Some of the benefits of using VoIP as opposed to conventional circuit switched voice are:

- variable bandwidth allocation on a hop-by-hop basis assists the control of congestion and caters for differing user requirements.
- reduces bandwidth with high compression codecs
- potential reduction in call costs due to common infrastructure
- silence suppression allows bandwidth to be used by other data during pauses in conversation.
- allows a flexible network hierarchy due to the ability to route VoIP packets independently through the network.

As listed by Blakely et al. (2000) there are a number of issues to consider when integrating voice and data in a wireline network, such as the codec used, the silence

Table 3.1: VoIP codec properties (Blakely et al. 2000; Cisco Systems 2002)

Algorithm	Voice BW (kbps)	Frame size (bytes)	VoIP payload (bytes)	Packets per second	RTP/UDP /IP header (bytes)	cRTP header (bytes)	Layer 3 BW (kbps)
G.711	64	80	160	50	40	-	80.0
G.711	64	80	160	50	-	2	64.8
G.729	8	10	20	50	40	-	24.0
G.729	8	10	20	50	-	2	8.8
G.723.1	6.3	30	30	26	40	-	14.56
G.723.1	6.3	30	30	26	-	2	6.656

suppression, VoIP packet compression and loss sensitivity. Likewise these issues have to be considered in a wireless network.

VoIP Codecs Many VoIP codecs are available, requiring different bandwidth, processing power and providing different voice quality. Some of the standard codecs used are G.711, G.723.1, G.726, G.728, and G.729. Each of these codecs can also have different implementations such as G.729a and G.729b. The codec used determines the ultimate bandwidth required, since the codec determines the size of the packets used to encode the speech. Table 3.1 shows some codecs and their associated data characteristics (Blakely et al. 2000; Cisco Systems 2002). Wong and Mack (1996) discusses some of the mean opinion scores associated with various codecs.

The rate of packet transmission and the payload size can also be altered so as to vary the bandwidth and packet size. For example, by increasing the payload, the number of packets sent is reduced and thus the bandwidth is reduced due to lower protocol overhead. In this research, an average packet size of thirty bytes was used for simulation purposes.

Voice Activity Detection Voice conversations can consist of as much as 60% silence (Wang et al. 1998), with VoIP as with standard circuit switched traffic both voice and silence is encoded. With the use of Voice Activity Detection (VAD), in which only the conversation is encoded, the bandwidth required is approximately reduced by the amount of silence removed. The G.729 Annex-B codec has a built-in VAD, but in all other respects has identical performance to G.729 (Cisco Systems 2002). It was assumed for all simulation purposes that VAD is used for VoIP.

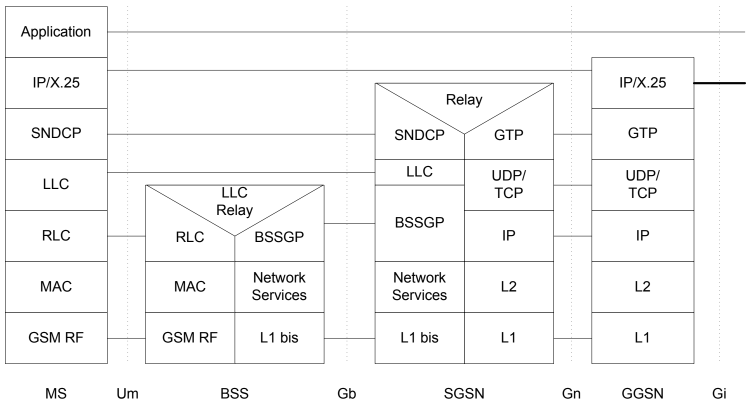


Figure 3.1: The GPRS protocol stack (Bates 2002; Halonen et al. 2002)

RTP Header Compression The Real-Time Protocol (RTP) provides applications with end-to-end network functionality for the transmission of real-time data, such as voice (Network Sorcery 2003). All VoIP packets are encapsulated by a RTP header, UDP and IP, which together form a forty byte header, the VoIP codec, RTP and UDP would form part of the Application layer in the GPRS protocol stack as shown in figure 3.1. When this header is compared to that of the default twenty byte sample of a G.729 packet, these headers are extremely inefficient. RTP header compression can compress the header size from 40 bytes to 2-4 bytes, resulting in a further considerable saving in the required bandwidth and improved throughput. A number of header compression schemes exist such as cRTP, ROHC, and ROCCO. Robust header compression schemes can permit a Bit Error Rate (BER) of 10^{-4} (Eriksson et al. 2000).

VoIP and sensitivity to Packet Loss VoIP is generally loss insensitive to a degree, as opposed to data which is loss sensitive. Random packet loss can be substituted with silence or for better performance white (background) noise, which can then be interpolated by the human brain to maintain the conversation coherency (Hassan et al. 2000). Figure 3.2 shows the user perceived quality for a 16bit linear, 8kHz DVI4 voice codec as a function of packet loss (Watson and Sasse 1997). In the paper by Blakely et al. (2000) a burst packet loss of no more than 15% is recommended. Persistent packet loss would result in the VoIP conversation becoming unintelligible. As with wireline VoIP persistent delays due to overload is aggravated by the lack of flow control within the IP/UDP/RTP protocol stack (Blakely et al. 2000). Thus **for an acceptable conversation, the packet loss for VoIP should be kept below 20%**.

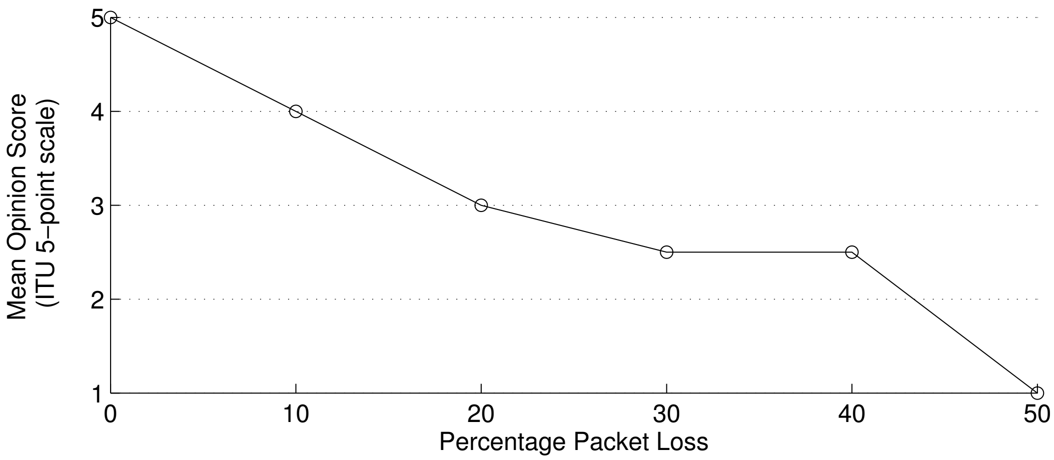


Figure 3.2: Mean opinion score versus packet loss (Watson and Sasse 1997)

Chapter 4

Theoretical Models for Traffic Simulation

This section describes the various models used for the simulation study, and their key parameters. The traffic source models are described with respect to a GSM situation.

4.1 Traffic Source Models

There are three primary sources of traffic: voice, data and packetised voice. For each source a suitable model was used to reflect its load on the GSM/GPRS network. Each of these models is discussed in further detail with respect to the equations used and the properties of the model.

4.1.1 Circuit Switched Voice

The model used for circuit switched voice was a Markovian arrival process with a Markovian departure process. Thus the holding times are exponentially distributed as well as the inter-arrival times as in (Lindemann and Thümmler 2003). Exponentially distributed processes indicate a seemingly random arrival pattern. A *Poisson process* describes the nature of arrivals at a certain point in time. The n-th interarrival time A_n is described as exponentially distributed (Hlavacs et al. 1999)

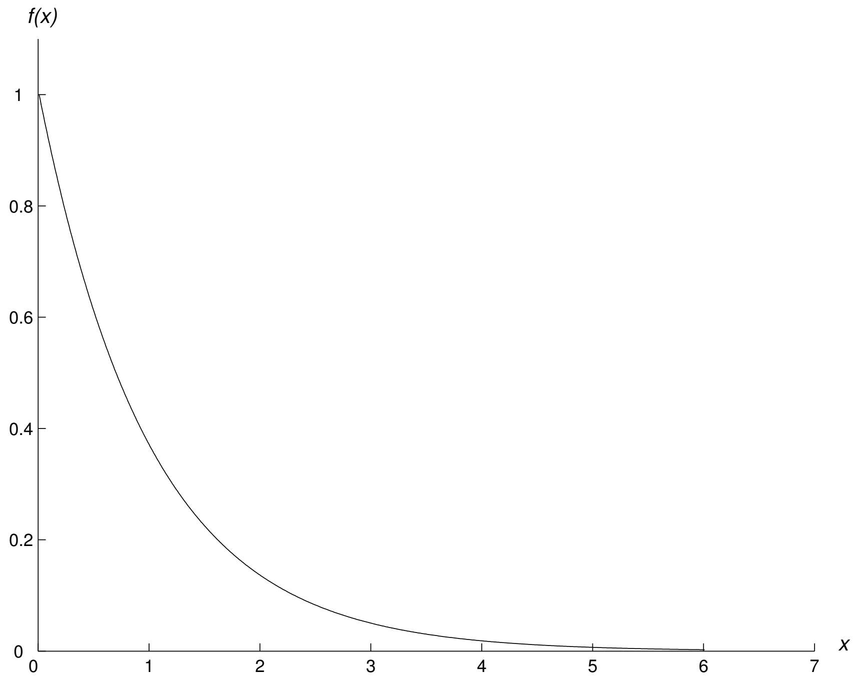


Figure 4.1: Exponential density function: equation 4.3

$$P\{A_n \leq \tau\} = 1 - e^{-\lambda\tau} \tag{4.1}$$

An *exponential distribution* is continuous and its function is given by equation 4.2 (Higginbottom 1998, pg. 138), where λ is the average arrival rate.

$$F(x) = \begin{cases} 1 - e^{-x/\beta} & \text{if } x \geq 0 \\ 0 & \text{otherwise.} \end{cases} \tag{4.2}$$

The *density function* of equation 4.2 as shown in figure 4.1, is given by equation 4.3, where β is a scaling factor.

$$f(x) = \begin{cases} \frac{1}{\beta}e^{-x/\beta} & \text{if } x \geq 0 \\ 0 & \text{otherwise.} \end{cases} \tag{4.3}$$

This density function shows the probability of a call lasting for the duration shown on the x axis, as in figure 4.1 in which a mean of one (i.e. a β of one) was used for illustrative purposes. All voice calls were assumed to have a mean of two minutes, ie a β of two, given higher cellular call costs compared to those of a wireline network.

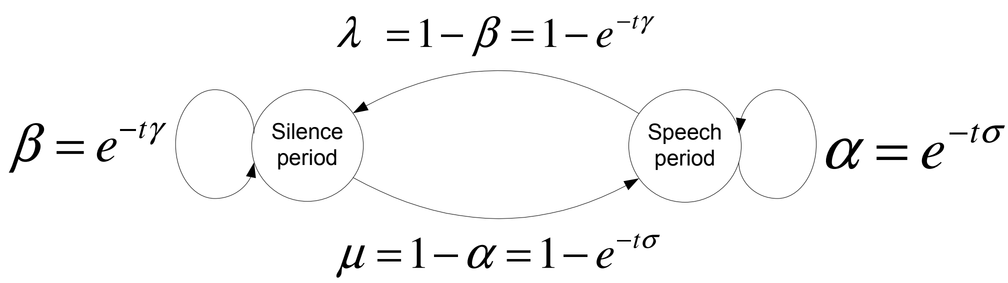


Figure 4.2: VoIP model

4.1.2 Packet Switched Data

The data was modelled as an Poisson arrival process as in the case of the voice, but a mean duration of 300 bytes was assumed. The duration of the packet was calculated by determining the transmission time for a set baud rate. An average baud rate of 13.2 kbps was used for the simulations. Thus if a packet size of 450 bytes is assumed, the packet duration is $450 \times 8/(13.2 \times 1024) = 266.3$ ms. **Given that one data channel reserved for one hour could carry one hour of voice traffic, the relationship between erlang and data can be established as the channel occupancy time**. Thus, if the data channel rate is assumed to be 13.2 kbps, a transmission of approximately six mega bytes would be equivalent to one erlang of data traffic which we here name as a *data erlang*.

4.1.3 Packet Switched VoIP

The VoIP was modelled as a two-state Markov modulated Poisson process. The talk-spurts and gaps were modelled using exponential distributions as done by Yang and Yantorno (1996) and Habib and Saadawi (1992). The distributions of the talk-spurts and gaps are heavily dependant on the type of silence detector used (Jiang and Schulzrinne 2000), with an exponential model being the generally accepted model for a large number of voice sources. However Jiang and Schulzrinne (2000) found from experimentation that the spurt/gap distributions are often not exactly exponential. We use the exponential model as a first order estimator of performance for VoIP behaviour patterns and trends. The two-state Markov model adapted from Sarker and Halme (2000) is shown in figure 4.2, where the talk duration is given as t seconds, with a mean of $1/\sigma$ and the distribution is $\alpha = e^{-t\sigma}$. The silence period has a mean of $1/\gamma$ and the distribution is $\beta = e^{-t\gamma}$. The transition from the silence period to the talk spurt is $\lambda = 1 - \beta = 1 - e^{-t\gamma}$.

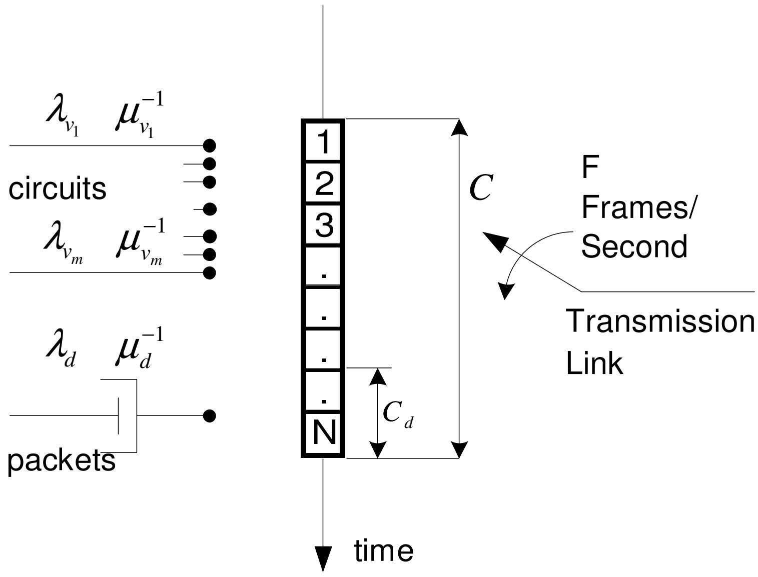


Figure 4.3: A generic TDM heterogeneous channel

4.2 BSS Resource Allocation Modelling

The Base Station Subsystem resource allocation was modelled as a controller with eight channels (one GSM/GPRS TDMA frame). A number of resource allocation strategies were considered, the most feasible of which was the strategy in which circuit switched Voice would receive a higher priority than data. VoIP packets would receive a higher priority than standard data packets.

It was assumed that when a voice call arrives and no resources are available, the call is discarded. For GPRS operation, if a data packet arrives and no resources are available, it is queued in a buffer until it may be serviced.

Since there is a combination of both voice and data, a heterogeneous traffic case had to be considered. As shown in figure 4.3 (Schwartz 1987, pg. 664), hybrid multiplexing is used to combine different traffic types. In this case, circuit switched voice and packet switched data is multiplexed onto a common transmission link. Figure 4.3 shows circuit switched voice with arrival rate λ_v and holding time μ_v^{-1} being combined with data packets onto a common time division multiplexing (TDM) channel of total capacity C bits/second. The TDM channel has a total of C time slots per frame (F), with C_d reserved for data and $C - C_d$ for voice.

In our model of the BSS, all circuit switched users are considered to be one class of traffic and packet switched data is assumed to be queued in a common queue.

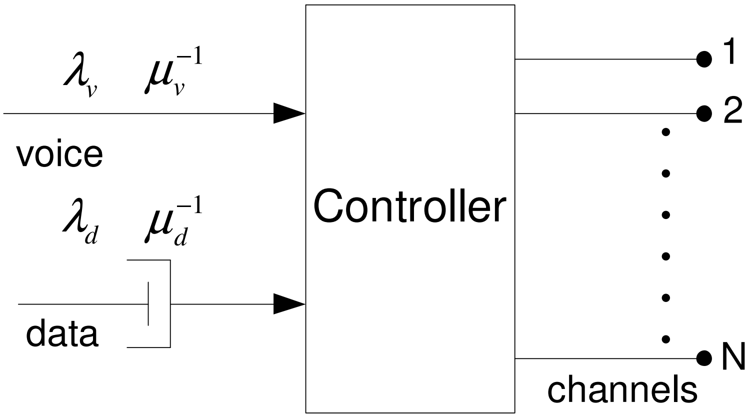


Figure 4.4: Continuous time model, with voice and data

A circuit-switched call is assumed to keep its channel assignment, until such a time as the call is completed. The call is blocked if no free channels are available, thus resulting in a blocked-calls-lost network. Data is queued if no channels are available. For our analysis, we used continuous time analysis in which the discrete time slots of the TDMA frame are ignored, and rather the concept of channels is used as in (Lindemann and Thümmler 2003; Mahdavi, Edwards and Ivey 2001) and (Schwartz 1987, pg. 672). With continuous-time analysis, the model can now be viewed as N channels operating in parallel, as in figure 4.4. The controller represents the resource allocation. The voice calls have an arrival rate of λ_v and mean duration μ_v^{-1}, while the data is assumed to have an arrival rate λ_d and mean duration μ_d^{-1}. It is assumed that priority is given to voice traffic whenever possible, and that the channel controller operates in a First In First Out manner. In addition, GSM/GPRS capacity-on-demand operation is modelled, resulting in any unused voice channels being allocated to data, as in a *movable-boundary* strategy (Schwartz 1987, pg. 67).

A rigourous analytical proof was not chosen for a number of reasons:

1. The solution of such analytical equations becomes increasingly complex as the number of channels increases,

2. The analytical solution would have to be recalculated for each distribution and would be extremely difficult for two-state models, such as those used to model the VoIP.

3. The analytical equations would have to be recalculated if a different queueing strategy was used.

4. Incorporating any delay or prioritisation would further increase the complexity of the analysis.

4.3 The Effects of the Propagation Environment

Use of a propagation model characterises the radio path, allowing some consideration of the Radio Frequency environment and the attenuation of the radio signal as it transverses the air interface. This information allows the coverage area to be established for each area, and predict performance. Mobile propagation models can be classified in terms of *large-scale* or *small-scale* effects (Tabbane 2000, pg. 53).

Large scale propagation models characterise signal strength at stationary or near stationary speeds, with the signal strength being log-normally distributed. Small scale fading describes the rapid fast fading or rayleigh fading component of changes in signal strength for a mobile terminal in motion. Large-scale models neglect the influence of near-field effects and do not consider them to alter the mean path loss (Rappaport 2002, pg. 139).

Large scale path propagation models serve to determine the received power based on signal reflection, diffraction and scattering. Standard measurement and theory has shown that the received power, measured in dB, decays logarithmically as a function of distance as given in equation 4.4. When plotted on a log-log scale the path loss is a straight line with a slope of $10n$ dB per decade (Rappaport 2002, pg. 139)(Chitamu et al. 2002). $\overline{PL}(d)$ represents the average path loss at a distance d[m] from the transmitter, n is the path loss exponent and $\overline{PL}(d_o)$ is the average path loss at a distance of one hundred metres. Typical path loss exponents for different environments are shown in table 4.1 (Rappaport 2002, pg. 139).

$$\overline{PL}(d) = \overline{PL}(d_o) + 10nlog(\frac{d}{d_o}) \tag{4.4}$$

Due to the fact that any given point a distance d from the base transceiver station the signal strength is different, a random distribution has to be applied to the mean value to account for these fluctuations. Thus, the field strength at a particular distance is log-normally distributed around the predicted average value, following a Gaussian (normal) distribution (Blaunstein 2000, pg. 163). Thus the path loss can be written as in equation 4.5 (Rappaport 2002, pg. 139), where X_σ is a zero mean Gaussian distributed random variable with a standard deviation of σ. The received power ($P_r(d)$) can be expressed as the transmitted power (P_t) minus the path loss, as in equation 4.6, or alternatively in terms of an average, as shown in equation 4.7 (Chitamu et al. 2002).

Table 4.1: Path loss exponents for various environments (Rappaport 2002)

Environment	Path Loss Exponent, n
Free Space	2
Urban area cellular radio	2.7 to 3.5
Shadowed urban cellular radio	3 to 5
In building line-of-sight	1.6 to 1.8
Obstructed in building	4 to 6
Obstructed in factories	2 to 3

$$\overline{PL}(d)[db] = \overline{PL}(d_o) + 10nlog(\frac{d}{d_o}) + X_\sigma \tag{4.5}$$

$$P_r(d)[dBm] = P_t[dBm] - PL(d)[dB] \tag{4.6}$$

$$\overline{P_r(d)} = \overline{P_r(d_o)} - \overline{PL(d)} \tag{4.7}$$

As the received power is randomly distributed about the distance dependant mean, the Q-function or error function (erf) can be used to determine the probability of the received signal level being above a particular level γ (Rappaport 2002, pg. 140). Although these equations are not used in the simulator they are presented for completeness. The probability of the received level being above γ is (Rappaport 2002, pg. 140):

$$Pr[P_r(d) > \gamma] = Q\left(\frac{\gamma - \overline{P_r(d)}}{\sigma}\right) \tag{4.8}$$

where Q is defined as:

$$Q(z) = \frac{1}{\sqrt{2\pi}} \int_z^\infty \exp\left(\frac{-x^2}{2}\right) dx = \frac{1}{2}\left[1 - erf(\frac{z}{\sqrt{2}})\right] \tag{4.9}$$

The values of the path loss exponent (n) and standard deviation (σ) are computed from measured data and vary according to the environmental conditions, termed morphology. Morphologies can be broadly separated into four components (C. Smith 2002, pg. 388):

1. rural - sparsely populated open areas with structures not exceeding two storeys

2. suburban - large number of residential houses of one to two storeys and some businesses with one to five storeys
3. urban - building structures from five to ten storeys
4. dense urban - metropolitan areas with buildings above ten storeys.

The following section describes how the source models were used.

4.4 Traffic Source Modelling and Queueing

Queueing models for radio systems can be split into two major systems, Blocked/Lost Calls Cleared (BCC) and Blocked/Lost Calls Delayed (BCD) (Stallings 1987, pg. 364–377).

In a *Blocked Calls Cleared* system, if a call arrives and no channels are available to service the call, the call is discarded. The caller would have to try again at a later time. Calls are assumed to arrive with a Poisson distribution, that is the duration between calls are exponentially distributed, and it is also assumed that there is a sufficiently large number of users to represent an infinite population. The *Erlang B* formula describes the chance of a call arriving and being blocked for a given traffic load and resources, i.e. the Grade of Service (GOS). In order to hold with the Poissonian arrivals it is assumed that a blocked caller will retry at a random time. This model is accurate for a large system with many channels, typically five or more channels are sufficient (Rappaport 2002, pg. 556), and many users with similar calling patterns (Rappaport 2002, pg. 555).

In a *Blocked Calls Delayed* system, calls not receiving immediate service are held in a queue for service at the next available opportunity. For a delayed system it is necessary to know the chance of a call arriving and being delayed due to unavailable resources. It is also necessary to know how long the call is delayed before being serviced. The chance of a call being delayed is given by the *Erlang C* formula. The GOS is measured by the probability of the call having a delay greater than t seconds. The Erlang C formula together with the service distribution is used to analyse the GOS. For a BCD system, as in a BCL system, it is assumed that there are a nearly infinite number of users present in the system, and that calls in the queue are eventually serviced.

In our investigation we considered the allocation of eight time slots, one GSM/GPRS TDMA frame, as illustrated in figure 4.3 where $N = 8$. The number of channels allocated to GSM circuit switched data is N_V and the number of channels for GPRS packet data channels is N_D. The voice and data arrivals are assumed to be uniformly distributed amongst the respective available resources (Vannucci and Chitamu 2003a). The voice calls are assumed to have an exponential holding time with mean duration two minutes and an exponentially distributed arrival rate λ_V. Data terminals are assumed to generate exponentially distributed data bursts at a rate of λ_D. The data burst arrival rate is assumed to be a superposition of newly arriving and retransmitted data bursts as assumed by So and Cho (2001).

Two combinations of voice and data integration were investigated:

1. GSM circuit switched voice and GPRS packet switched data
2. GPRS packet switched Voice over IP (VoIP) and GPRS packet switched data

4.4.1 Circuit Switched Voice and Packet Switched Data

In this mode the resources required to adequately support data were investigated for a given voice intensity level. It was assumed that if there are insufficient resources to accept a voice call, it is dropped, i.e. blocked-calls-lost. In the case of insufficient resources for a GPRS data packet, it was assumed that the packet is queued in a buffer until it may be serviced. Thus we represent the air interface and the BSS as shown in figure 4.5.

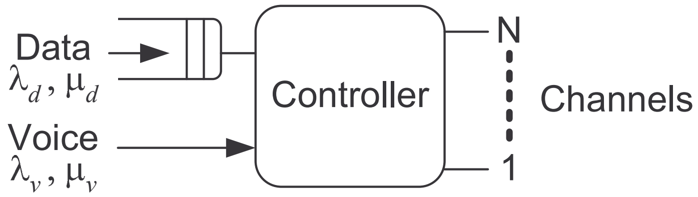


Figure 4.5: GSM/GPRS model

The possible number of calls and packets in the system, and the transitions between them is represented in a state space diagram as shown in figures 4.6 and 4.7. Figure 4.6 shows the state space model for the case of fixed resources, without capacity-on-demand and six channels reserved for voice and two channels reserved for data, whilst figure 4.7 shows the state space model with capacity-on-demand in operation.

Each state has a probability of occurrence and $P_{i,j}$ is the probability of there being i voice calls and j data packets in progress in the system. Transitions from one state to another follow the transition rates indicated in the diagram, where λ_i is the arrival rate, and μ_i is the service rate. When there are $i = N_v$ voice calls in the system further calls are blocked, and when there are $j = N_D$ packets in the system further packets are queued. The probability of a voice call being blocked and lost is derived from balancing the state equations calculated using the transitions (vertical transitions referring to diagram 4.6) as these receive priority over packet switched data and are thus independent of them. The use of the Erlang B formula is based on the following assumptions, which corresponds to GSM/GPRS operation:

1. Calls are initiated in a random manner with each being independent of another.
2. All channels are fully available for servicing, i.e. they all have an equal chance of being used, unless occupied.
3. The duration of a call (also known as service time) is exponentially distributed.
4. There are a finite number of channels available.
5. Call arrivals are described by a Poisson distribution, i.e. the interarrival times are exponentially distributed.
6. The number of busy users is equal to the number of busy channels.

The two main measures of performance considered are the probability of a voice call being blocked, and the average delay of a data packet. The probability of a voice call being blocked due to insufficient resources is given by the Erlang B formula, equation 4.10, and the probability of a data packet being delayed is given by the Erlang C formula, equation 4.11.

$$\Pr[\text{Blocking}] = \frac{\frac{A^C}{C!}}{\sum_{k=0}^{C} \frac{A^k}{k!}} \tag{4.10}$$

$$\Pr[\text{Delay} > 0] = \frac{A^C}{A^C + C!\left(1 - \frac{A}{C}\right)\sum_{k=0}^{C-1} \frac{A^k}{k!}} \tag{4.11}$$

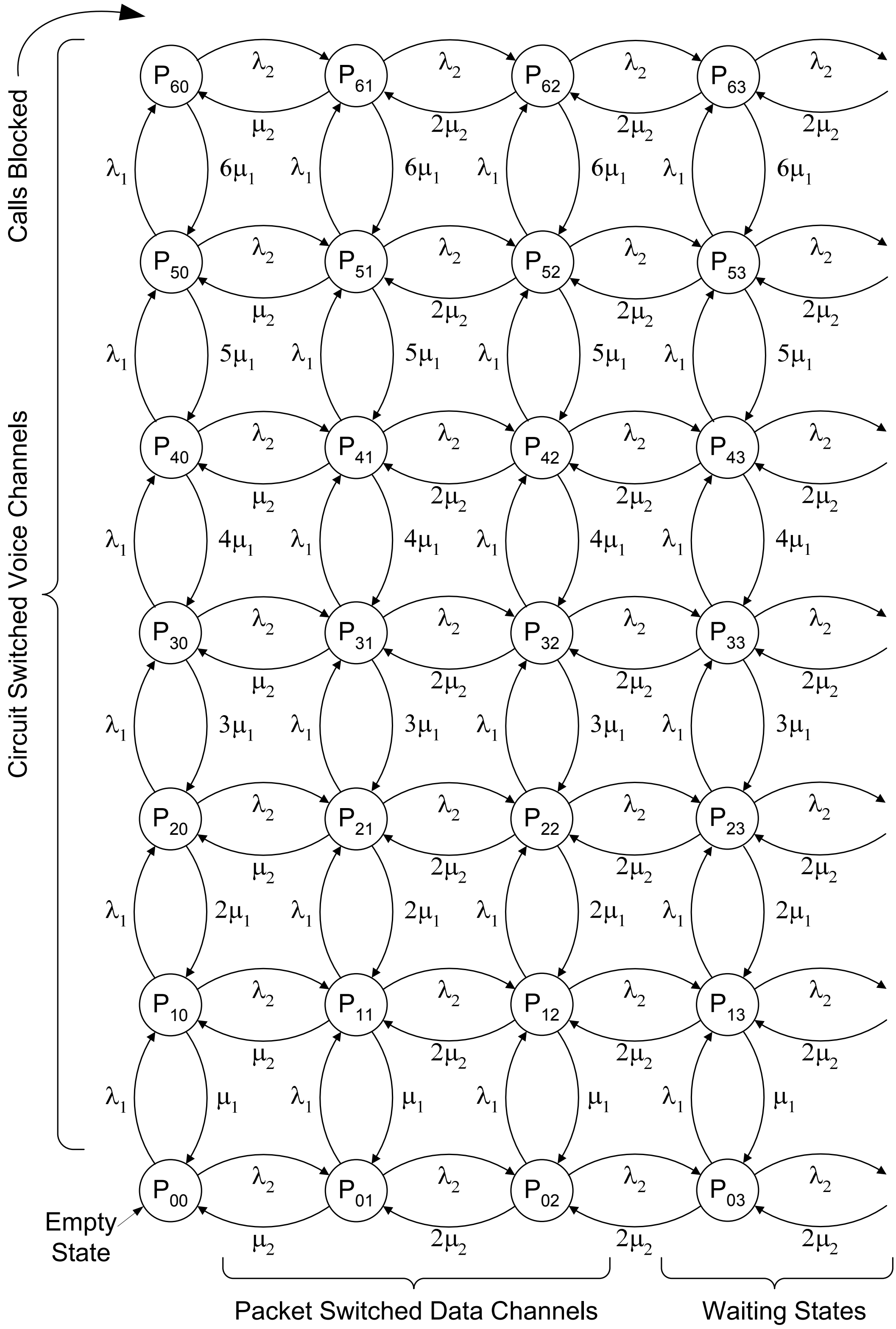


Figure 4.6: Fixed resources state space

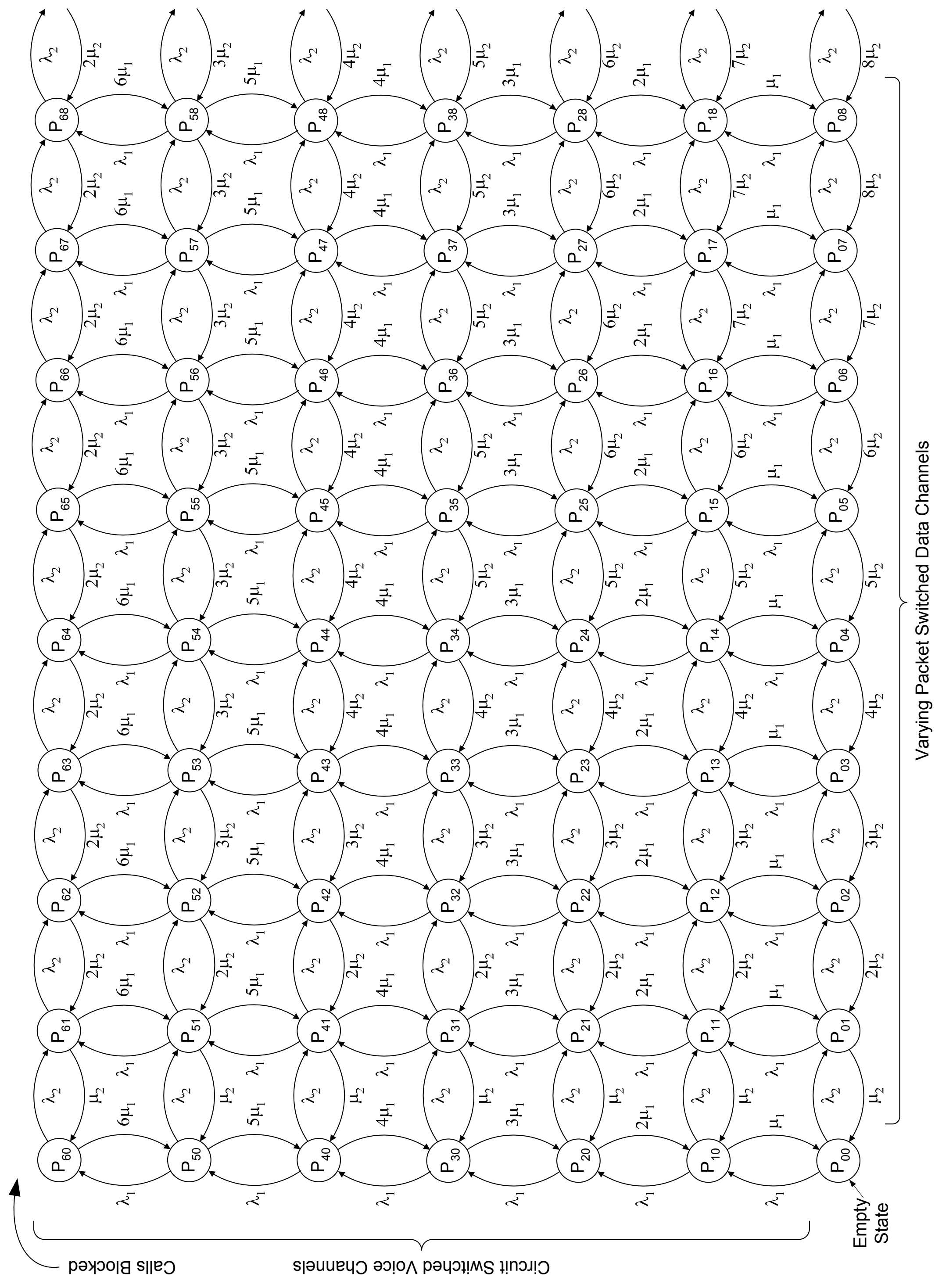


Figure 4.7: Variable resources state space

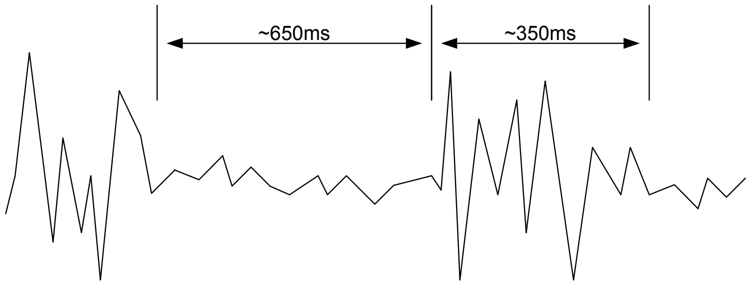


Figure 4.8: Illustration of talksputs and gaps during conversation

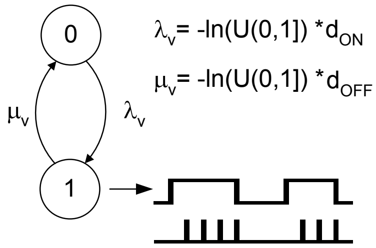


Figure 4.9: Two-state VoIP model

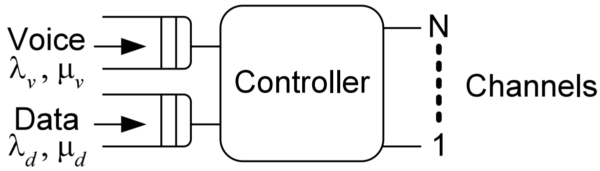


Figure 4.10: VoIP GPRS model

A is the offered traffic in erlang. An erlang is the call arrival rate (arrivals/s) multiplied by the average call holding time. C is the number of channels available. The average delay for a data packet is given by the probability of a packet being delayed multiplied by the average holding time in the queue, as in equation 4.14. The link shaping and coding delay is assumed to be a constant and thus was discounted since the probability of the data packet being delayed is dependant on the resources and load.

4.4.2 Packet Switched Voice and Packet Switched Data

Packetised voice is represented as an encoded variable data rate source made up of talk-spurts and gaps, as shown in figure 4.8. A two-state ON-OFF Markov model was used to generate voice segments which are then broken into VoIP packets, as in figure 4.9. The On and Off states were exponentially distributed (using the natural log of a random number between 0 and 1) with respective means of d_{ON}=352 ms and d_{OFF}=650 ms (Habib and Saadawi 1992).

VoIP packets are treated as normal data, except that the VoIP packets are prioritised, and will receive priority service whenever possible. Thus the VoIP and data packets can be modelled as in figure 4.10.

The probability of a data packet being delayed can be calculated using the Erlang C formula, if the amount of resources is known explicitly. The Erlang C formula is derived from the assumption that a single queue is used to hold all requested calls which cannot be immediately assigned a channel. That is, referring to figure 4.6, the Erlang C formula is derived from balancing state equations calculated using the horizontal transitions as these are independent of the voice calls in a fixed resource situation. If no channels are available, the packet is delayed and held in a queue, until it can be serviced. The chance that a queued packet is forced to wait more than t seconds is given by equation 4.12.

$$\Pr[\text{Delay} > \text{t}|\text{delayed}] = e^{-\frac{(C-A)}{H}t} \tag{4.12}$$

where C is the total number of channels, t is the delay time of interest, and H is the average duration of a call or packet in seconds. Note that one assumes an exponential distribution. If this is not the case, the probability of delay would change. The chance of any call waiting for a duration greater than t is equation 4.12 multiplied by equation 4.11 as given in equation 4.13

$$\Pr[\text{Delay} > \text{t}] = \Pr[\text{Delay} > 0] \times e^{-\frac{(C-A)}{H}t} \tag{4.13}$$

The average delay D for all packets in a BCD system is given by

$$D = \int_0^\infty \Pr[\text{Delay} > 0] \times e^{-\frac{(C-A)}{H}t} dt = \Pr[\text{Delay} > 0] \times \frac{H}{(C-A)} \tag{4.14}$$

4.4.3 Determining the Grade of Service from the Queueing Models

The Erlang B and Erlang C formula can be used collectively to estimate the Grade of Service (GoS) for voice and data for a given load and resources. The number of fixed allocated packet data channels (PDCH) for the data is used in the Erlang C formula to predict the average delay and grade of service for data packets. While the number of channels minus the fixed PDCH for GPRS would be used in the Erlang B formula to determine the grade of service for the voice traffic. A Microsoft Excel implementation of the Erlang formulas is shown in the appendix.

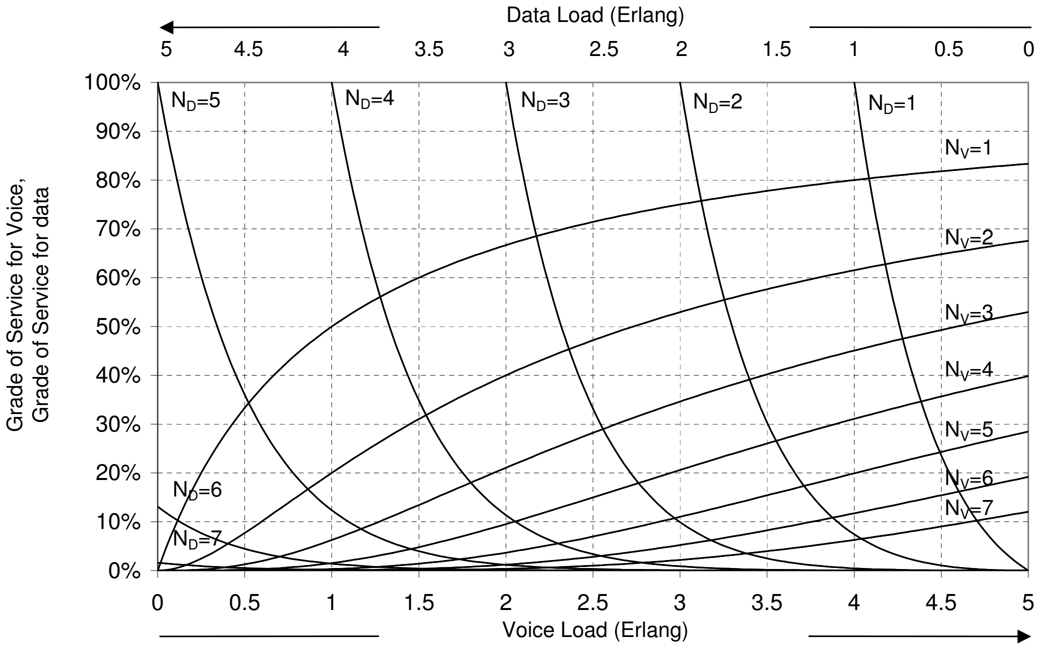


Figure 4.11: Theoretical Grade of Service for voice and data for varying loads and resources

If the load is assumed to vary from zero to five erlang, then a graph can be plotted of the GoS for both the voice and data, as shown in figure 4.11. *Note that only the GoS for a fixed resource can be predicted from the Erlang equations*, and the instantaneous GoS for a capacity-on-demand mode of operation has to be obtained by changing the demand model in real time, and averaging the result to determine the average grade of service over the busy hour.

Using figure 4.11, it is possible to predict the number of channels required for a given load and GoS. For example, if seven channels are reserved for voice, and a GoS of 5% is required, then from the graph one can determine that theoretically a load of 3.5 erlang could be supported. With the remaining channel reserved for data, at a 5% GoS, a load of approximately 0.2 erlang could be supported. From figure 4.11 one can also observe the effects of packet multiplexing and the greater efficiency of packet switching over circuit switching. For example, consider the case of five channels, for a 5% GoS for packet switching the load would be approximately 3.5 erlang, as opposed to 2.25 erlang for circuit switching traffic operation.

The next chapter describes the verification of the computer simulation, and how the simulator was validated before being used for the simulation of more complex models, such as the two-state VoIP model and the capacity on demand functionality of the controller.

Chapter 5

Simulation Environment Specification, Validation and Verification

Modelling a communication network can take two forms: a mathematical analysis, using mathematical methods such as the probability theory discussed in the previous section, allowing exact answers to particular questions to be gained by way of an *analytic* solution (Law and Kelton 1982), or computer *simulation* (Balaban 1984). However, mathematical analysis can often become intractable if a large number of initial parameters are considered, and thus often becomes exceedingly complex as the number of variables and model complexity increases. Computer simulation, the second accepted method used to analyse communication networks, uses the computer to evaluate the model *numerically* over a period of time and provide an *estimated* solution. Due to the use of high level languages, this approach is considered to be far more flexible.

There are a number of advantages and some disadvantages to computer simulation, some of which as reported by (Law and Kelton 1982, pg. 8–9):

+ Most real-world systems with stochastic elements cannot be accurately described by an analytical model
+ Simulation allows the operating conditions to be varied and the performance under the varying conditions to be altered. An analytical model often assumes a fixed set of operating conditions.
− Simulation models are time consuming to develop and require multiple simulations to provide an acceptable estimated answer, whilst an analytical model produces one deterministic result.

The computerised model developed in this study uses a combination of discrete and continuous time simulation, known as *combined discrete-continuous simulation* (Law and Kelton 1982, pg. 47). In this section, the required simulation functionality is discussed as well as the features of the developed simulator.

A simulation of the system incorporating the models developed during the problem analysis phase was created to solve the problem recursively. The computerised model was created in Java, using object orientated techniques. This section describes the computerised model functionality, capabilities, components and verification.

5.1 Simulator Specification

The problem entity description is of great importance when evaluating the computational model, since the conceptual model is based on the description of the real world system, and likewise the computerised model is based on the conceptual model, thus the description of the real system leads is fundamental to the *modelling and simulation*[1].

One of the first considerations undertaken in the development of the model of the GSM/GPRS system was the validity of the model, the degree to which it corresponds to the real world system. Validity can be portrayed symbolically as in equation 5.1 (Zeigler 76, pg. 5), where validity is expressed in degrees of strength.

$$\text{real system data} \stackrel{?}{=} \text{model generated data} \tag{5.1}$$

The objective of this computerised model was to present a first order *predictively valid* model, i.e. a model that could match the general operation of the real system *before* data is acquired from the real system (Zeigler 76, pg. 5).

The reason for the modelling was to gain an *understanding* of the operation, observe its behaviour and suggest possible improvements which would warrant further investigation.

After analysis of the real world system, it was decided to simplify the model to retain only what was relevant to the study: the GSM/GPRS resource allocation methodology and the traffic source models. The hybrid multiplexing methodology

[1] "*modelling* deals primarily with the relationships between the real system and models; *simulation* refers primarily to the relationship between computers and models." (Zeigler 76, pg. 3)

was modelled together with the offered traffic nature. Due to the complexity of GSM/GPRS operation, other features such as signalling, temporary block flow allocation, coding scheme variations and other supervisory functions and protocols were not simulated.

By simulating only the radio resources, the resulting patterns could be analysed so as to make recommendations for further analysis of a more realistic system.

5.1.1 Simulator Functionality

As the simulator was created to implement the BSS model, all aspects of the initial traffic could be varied to determine the impact of those variables under different conditions. For all traffic types, the mean holding time and arrival rate could be altered to vary the traffic intensity. For the VoIP traffic, the mean ON and OFF durations could be altered.

5.1.2 Simulator Output

The output of the simulation is an entry into a comma separated text file, specifying the results of the simulation in terms of the delay for the packets, the average call blocking, the average probability of delay. In order for a statistically acceptable averages, these parameters had to be combined by means of further processing in Matlab as described in the following paragraph.

5.1.3 Simulator Output Processing

Due to the results of the simulation being stored in a text file, the **output files had to be further processed to collate similar simulations and produce an average simulation result**. The results were entered into a SQL database and then collated and exported to a text delimitated file for processing within Matlab. Within Matlab the results were averaged, and graphs of the results plotted. All of the Matlab scripts are contained on the compact disc in the Appendix.

5.1.4 Simulator Random Number Generator

By their basic nature, simulations are models for stochastic processes, and hence one of the key functions of a simulator is the ability to generate uncorrelated random variables (Higginbottom 1998, pg. 96). A Mersenne twister Portable Random Number Generator (PRNG) was used to produce repeatable random numbers of a high quality as reported by Luke (2000); Matsumoto and Nishimura (1998). Such a portable random number generator allows a set of random numbers to be reproduced given the same initial seed. This is very useful for testing to reduce the effects of stochastic variation and observe if the simulation output is repeatable.

5.2 Simulation Validation and Verification

This section discusses the verification and validation (V&V) of the developed computerised model. The recommended procedures for V&V are defined and their application is discussed with respect to the resulting developed simulator. Model validation is defined by Schlesinger et al. (1979) as "substantiation that a computerised model within its domain of applicability possesses a satisfactory range of accuracy consistent with the intended application of the model". Likewise, model verification is defined as "ensuring that a computerised model represents a conceptual model within specified limits of accuracy" (Schlesinger et al. 1979). It is generally accepted that a model is developed for a specific purpose and its validity should be determined for that purpose (Sargent 1994). As part of development process (Sargent 1994), several versions of the model were developed prior to obtaining a satisfactorily valid model. Determining the validity of the model for the entire stated domain was deemed too time consuming. Instead, tests and evaluations were conducted until a sufficient body of evidence existed to develop an acceptable confidence in the simulator, a standard practice considering the abstracted nature of the model (Sargent 1994). The relationship between cost/time of the model validation and the perceived value of the model to a user is shown in figure 5.1 (Sargent 1994). From this graph we can see that even a moderately validated model has a value to the user, and the rate of return on model value decreases as the validation cost increases.

Sargent (1994) and (Schlesinger et al. 1979) illustrate a model validation and verification modelling process. This diagram, shown in figure 5.2, has been adapted to further simplify the relationship between the problem entity and the computerised

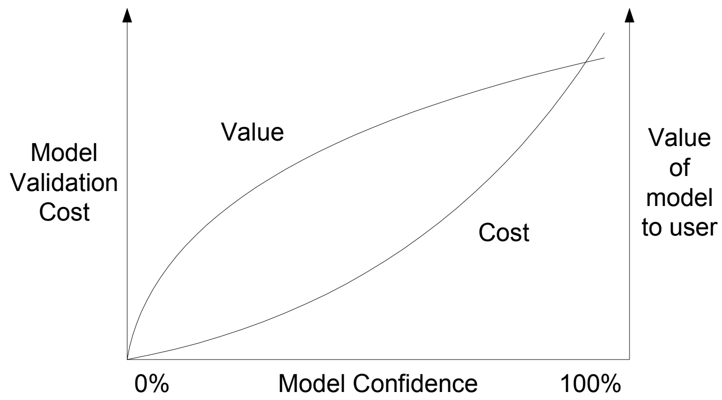


Figure 5.1: General model confidence (Sargent 1994)

model with only one way relationships, and to enhance the relationship between the computerised model and conceptual model with a two-way relationship which better illustrates the methodology used in the development of this computer simulation.

Figure 5.2 shows the development of the computer simulation, represented by the computerised model, and the intermediate verification and validation that occurs between iterations. The *problem entity* is the system being modelled, the *conceptual model* is the developed representation, the *computerised model* is the conceptual model implemented using a high level programming language (Sargent 1994; Schlesinger et al. 1979), which in this case was Java. The dotted arrows indicate the design phases. It was also found that there was an iterative process between the conceptual model and the computerised model, with the introduction of more complex traffic source models such as the VoIP codec described in Chapter 3.

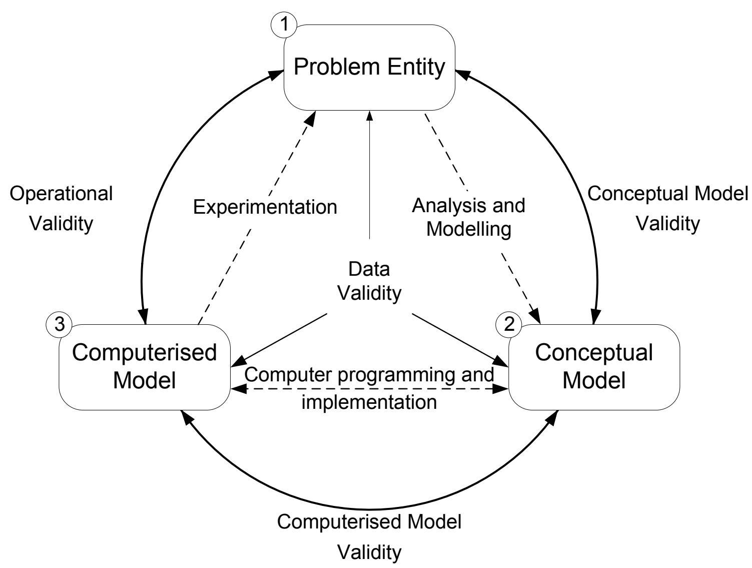


Figure 5.2: Adapted version of the Modelling Process

5.2.1 Validation

There are three fundamental approaches to determining the validity of a computerised model, as outlined in (Sargent 1994). Each validation process is part of the V&V that takes place at each iterative stage of development. The first and most common method to determine validity is based on the subjective decision of the developers themselves, and whilst this decision is based on the results of various tests and evaluations during the development process, it can be biased. Thus a preferred method is the process of independent verification and validation (IV&V) (Sargent 1994). IV&V requires the review of an independent third party, ranging in detail from simple reviewing to a involved V&V effort. As part of the IV&V process, the developed models for this simulation were discussed at multiple conferences in which they were peer-reviewed (Vannucci and Chitamu 2003a;b;c). The third validation approach is to use a scoring model. This method of validation was found by Sargent (1994) to be impractical for a number of reasons as listed below, and thus was not used in the validation of this simulation model:

1. Individual scores are determined subjectively by the developer of the model.
2. The final total score at which to accept a model is ambiguous and varies between models.
3. A model might be defective and still have an acceptable score.
4. A high score might cause an overconfidence in the model and be used to compare simulation models with different objectives.

(Law and Kelton 1982, pg. 338) cite the paper by Naylor and Finger (Naylor and Finger 1967) as one of the fundamental papers in validation literature. In (Naylor and Finger 1967), an alternative three-step process, as below, is described for validating a simulation model. This is further discussed by Law and Kelton (1982).

The first step is developing a model with a high face validity, a form or rationalism (Naylor and Finger 1967), i.e. a system that seems *reasonable*. This can be accomplished in a number of ways, conversations with experts (Chitamu 2003), existing theory and practice (i.e. previous GSM/GPRS simulations) and general knowledge (i.e. the application of queueing theory). The second step is *empirical testing*, determining the validity of simulation results by comparing them to known real world results. Due to the inability to experiment on a live system, empirical testing could only be verified for some of the source models for which such testing had already been performed by independent parties. The third step is to determine how *representative* the simulation output is, by comparing the output to a more detailed simulation of the system or doing a statistical comparison with the real system behaviour. This was not done for this computer simulation due to the excessive validation cost that this would incur (see figure 5.1) for a model which was only meant to evaluate one aspect of a GSM/GPRS system.

5.2.2 Verification

In *Simulation Modeling and Analysis*, Law and Kelton (1982) identify five techniques to assist in the verification of the simulation model. These techniques are (Law and Kelton 1982, pg. 334–336):

1. *Write and debug the computer program in modules.* This technique was used extensively due to the object orientated programming techniques used, allowing the simulator objects to be tested and detail added as each function was verified, gradually increasing the complexity of the model.

2. *Have more than one person evaluate the computer program.* This technique was used during the initial design and construction of the simulation, by consulting an experienced programmer (Hodgkinson 2002) who assisted in implementing the core functionality and processes of the computational engine.

3. *Use a trace whilst debugging the program.* All major high level programming languages allow the system variables to be inspected at any stage of the program. This proved to be crucial in successfully debugging the program. The use of fixed seeds and a portable random number generator allowed deterministic events to be inputted, providing a known event set for computation.

4. *The model should be run under simplifying assumptions in which the characteristics can be easily computed.* The output of the simulator was tested under constrained circumstances and the results were compared to those predicted by theory, namely the Erlang B and Erlang C formulas. The simulator testing is limited to the assumptions of the Erlang formulas, for which exponentially distributed inter-arrivals are assumed. Thus confidence in the simulator operation was established before further more complex models were implemented such as the two-state Markov modulated Poisson process. The simulator output was verified under different simulation durations and traffic loads.

5. *Graphically display the progress of the simulation.* This feature was implemented by setting the debug flag in the simulation engine, so that the progress of the simulation and the event handling could be observed. This allowed the cause of infrequent errors to be observed by simultaneously recording the debug output in a text file.

In addition to the simulator verification, it is extremely important to validate the simulations performed, since experimentation with a simulation is a *proxy* for experimenting with a existing or proposed system. Thus one of the primary goals in simulation is the *relevancy* of the simulation, to ensure that the model developed can be used to predict the performance of the real system, were such changes feasible and possible. However, one should always keep in mind the constraints of a simulation and the fact that it is merely an *approximation*, thus rather referring to the degree to which the model agrees with the system (Law and Kelton 1982, pg. 337), and likewise keeping in mind the *overall objective* of the simulation study.

Verification Results

The results of multiple computerised model verification tests under varying conditions are shown in figures 5.3, 5.4, 5.5, 5.7. Verification of the computerised model is of utmost importance in ensuring that the computer programming and implementation of the conceptual model are correct (Sargent 1994). Before the final simulation was completed, basic testing was done on the simulation functions, such as the randomness of the Mersenne Twister, the generation of events, the chronological sorting of events, and other subsections of the program, as described in Chapter 6. By varying the initial conditions, the dynamic load testing of the computerised model could be done, as shown in the following figures. In addition, some critical components of the program, such as the sorting and queueing methods, were **redone** to improve the speed of the computerised model, allowing internal consistency tests to be performed.

In figure 5.3 the data load was varied from 0.9 to 1.5 erlang and the voice load from 1 to 4 erlang, capacity-on-demand was in operation. Similarly in figure 5.4 the data load was varied from 1.5 erlang to 1.9 erlang.

It can be seen that the voice percentage blocking does not vary with different data loads. This corresponds to the theoretical operation due to voice being prioritised.

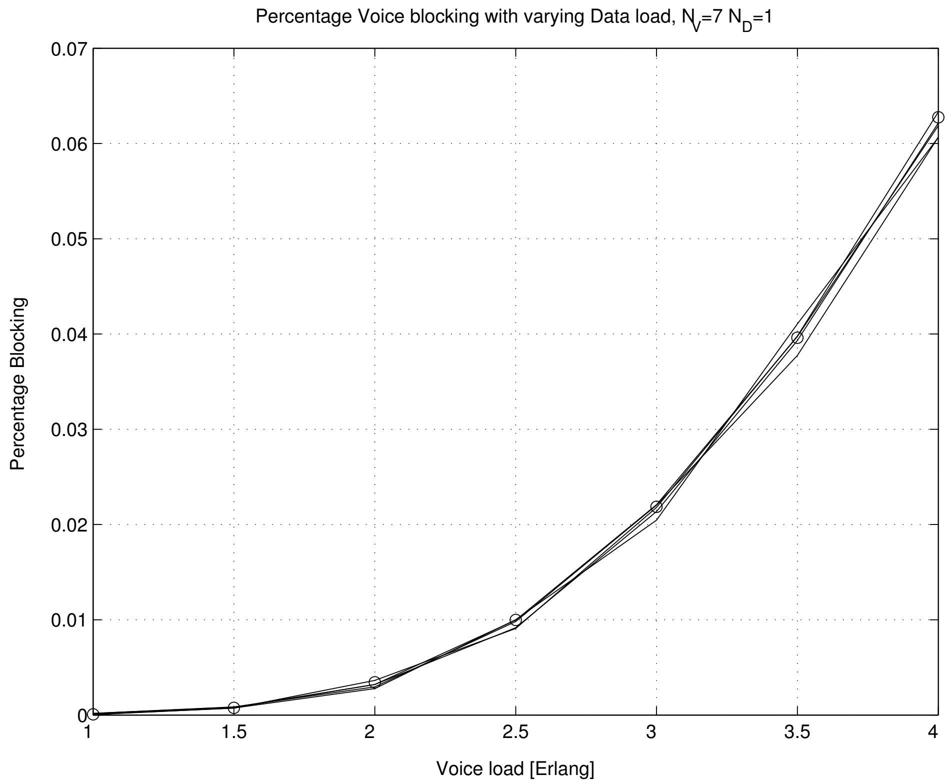


Figure 5.3: Voice blocking probability with seven voice channels

In addition, in figure 5.3 and 5.4 the theoretical blocking probability is marked by a circle as calculated by the Erlang B formula, and the **simulation results correspond with the calculated Erlang B values**.

Figure 5.5 shows the probability of a data packet being delayed as calculated by the simulator (solid line), compared to the Erlang C formula (circles). The percentage difference between the two graphs is show in figure 5.6. One can see the percentage difference at low traffic values is large, the reason for this is: At low traffic intensity values the probability of delay is extremely small. For example with seven data channels, and a load of 1.5 erlang, the probability of delay is 0.00096%, requiring a very long simulation time to produce accurate results.

As the load increases, the accuracy increases, as can be clearly seen in figure 5.6. The simulator inaccuracy at traffic intensity values of less than 1.5 erlang for seven channels is acceptable, since grade of service in these cases is high (poor) and no investigation in these regions is warranted.

The simulator output for seven data channels and one voice channel, with capacity-on-demand is shown in figure 5.7, and the corresponding percentage error is shown

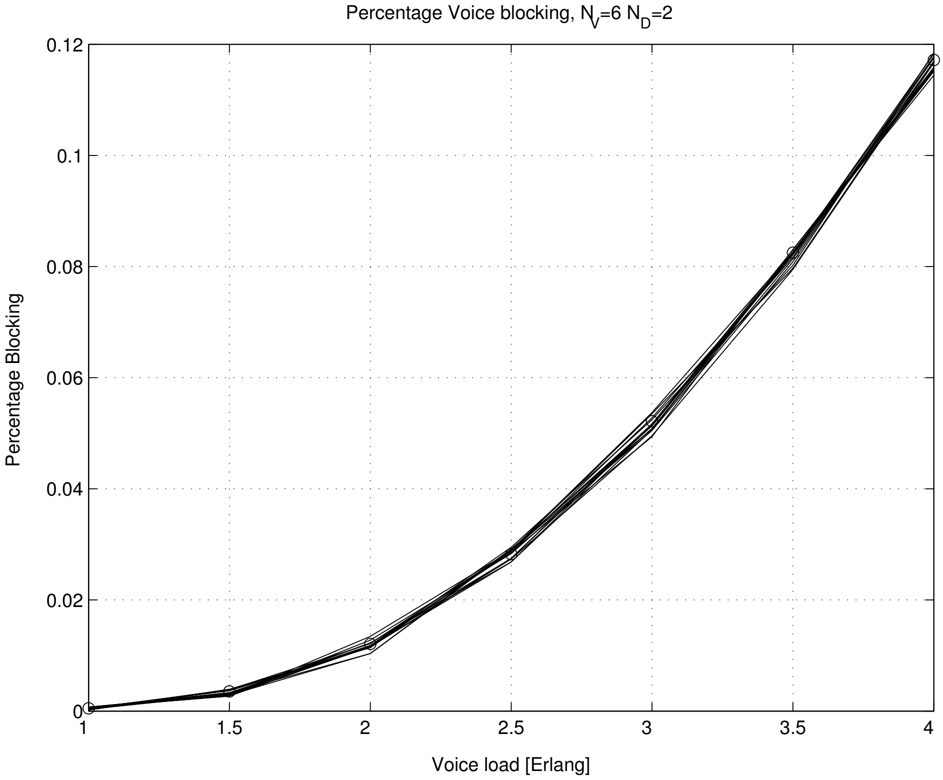


Figure 5.4: Voice blocking probability with six voice channels

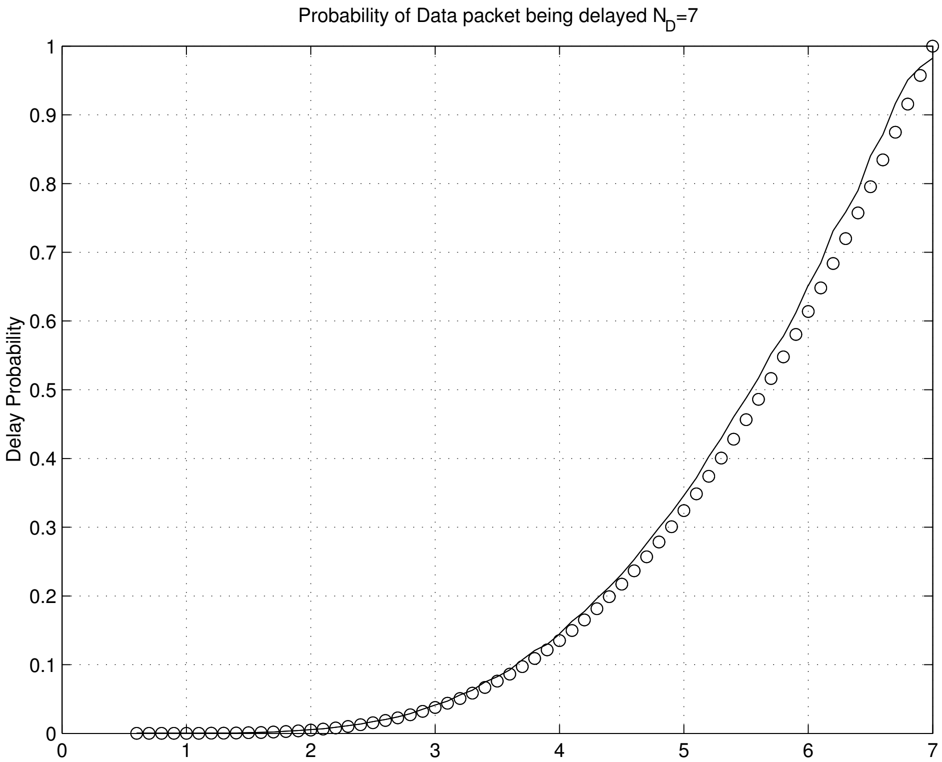


Figure 5.5: Probability of data delay with seven data channels, no COD

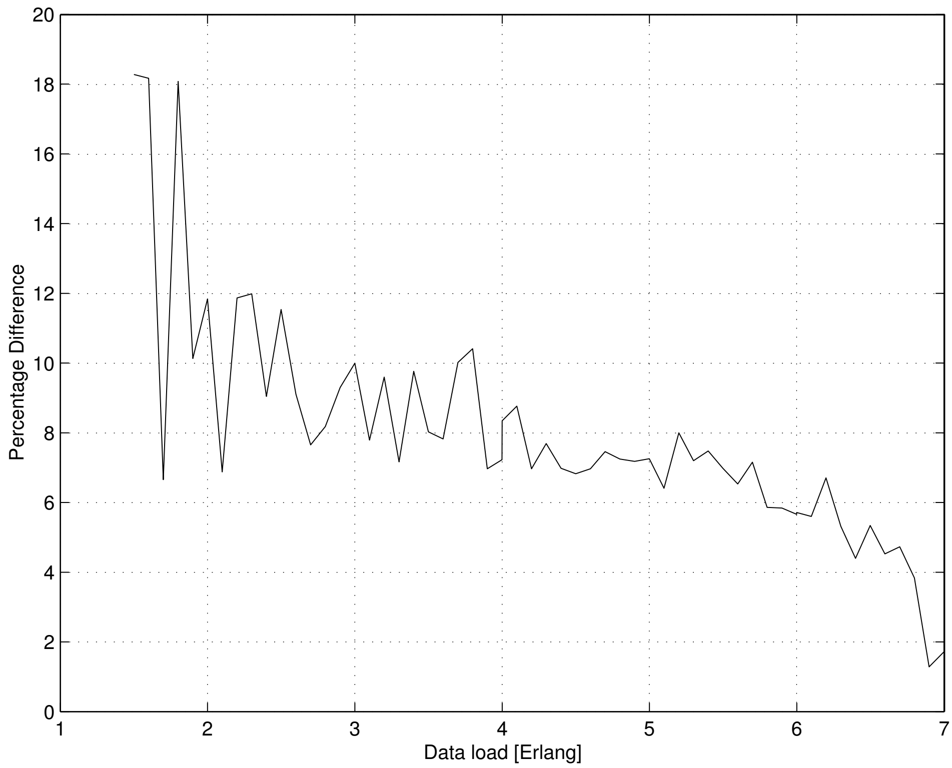


Figure 5.6: Percentage difference between Erlang C and simulated results for seven effective channels

in figure 5.8. It was found that if the expected blocking was low, then the duration of the simulation had to be large for an adequate statistical mean to be found. This was found to lead to problems when combining circuit switched voice (of a duration in hundreds of seconds) with data packets (of a duration of hundreds of milliseconds). Thus the event set for the voice would be insignificantly small when compared to the data. If a simulation of duration ten minutes was required for an adequate circuit switched voice result, the associated set for the data would be large, resulting in extremely long simulation times.

The number of simulations required to achieve a statistical mean is shown in figures 5.9 and 5.10. The two graphs show the results of repeated simulations to calculate the average delay for data given a load of six and seven erlang with eight packet data channels. From figures 5.9 and 5.10 it can be seen that the lower the delay probability, the longer the required simulation to obtain a statistical average.

The following chapter describes the core simulator logic and source models as implemented in the simulator.

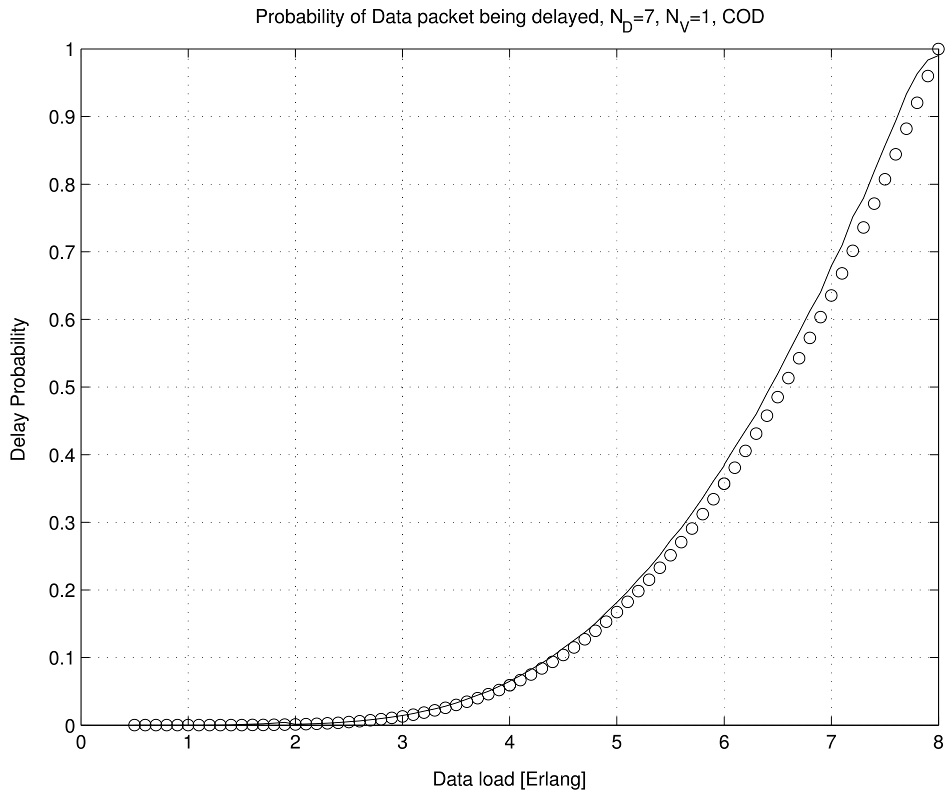


Figure 5.7: Probability of data delay with seven PDCHs, one voice channel, COD

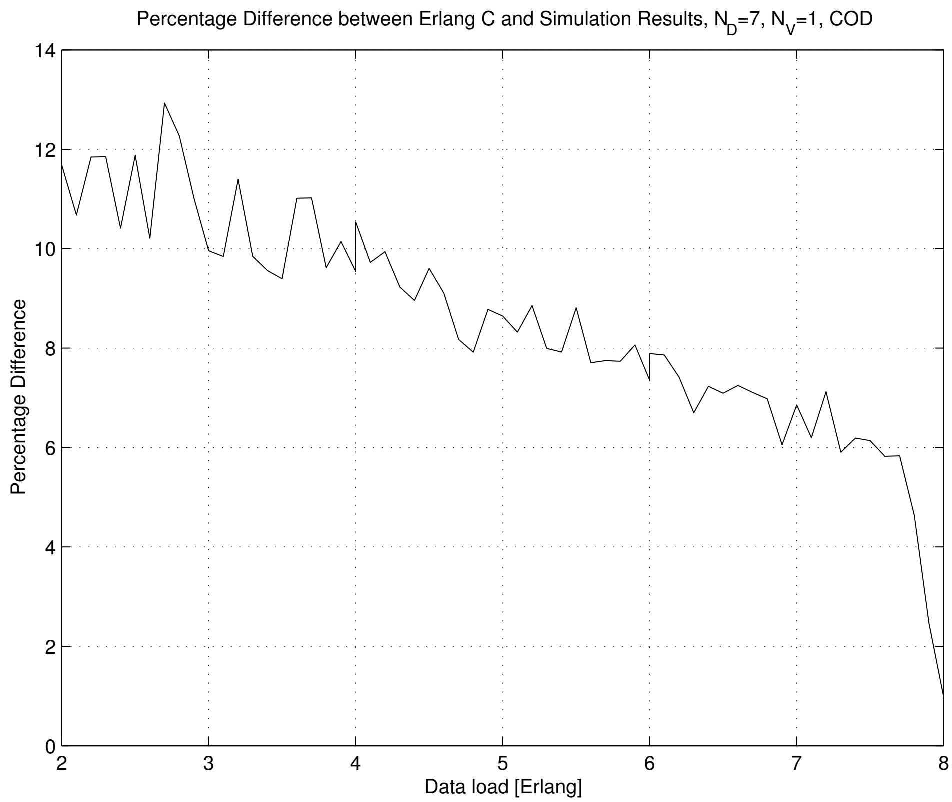


Figure 5.8: Percentage difference between Erlang C and simulated results for eight effective channels

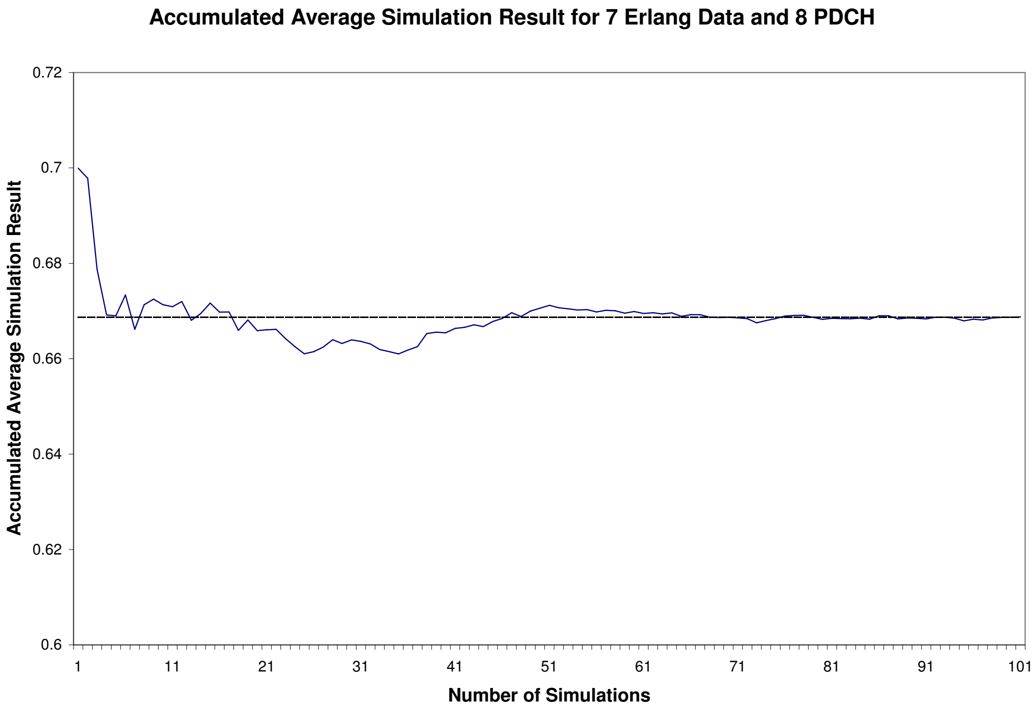


Figure 5.9: Simulation convergence to a mean value, seven data erlang

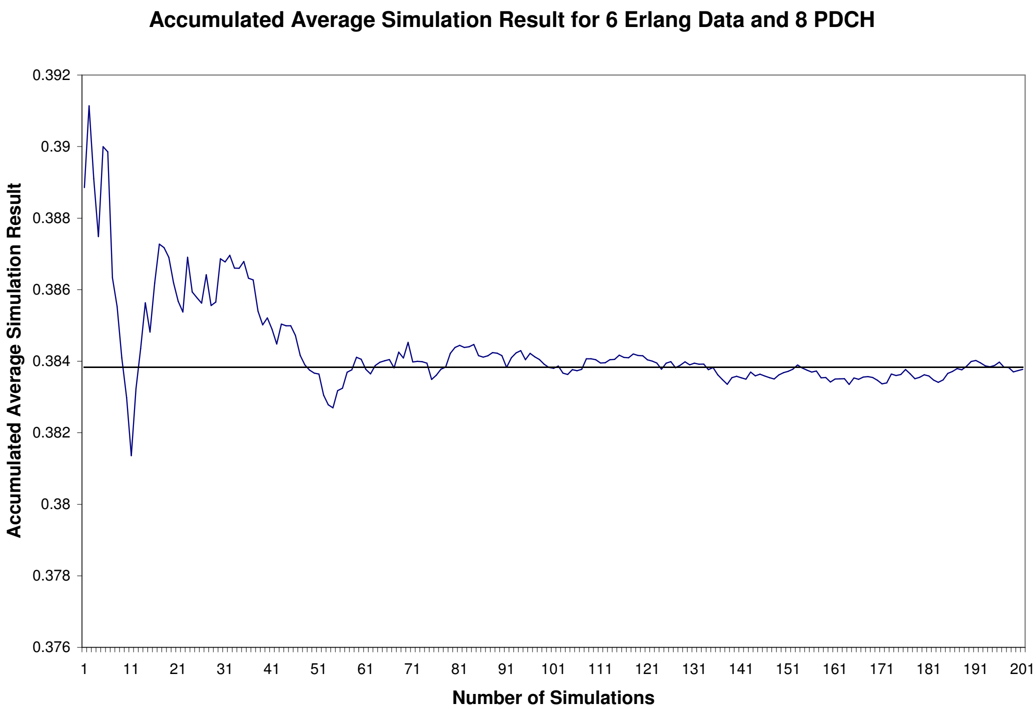


Figure 5.10: Simulation convergence to a mean value, six data erlang

Chapter 6

Object Orientated Simulator Structure

This chapter describes the construction of the simulator in order to simulate the GSM/GPRS operation.

An Object Orientated Design (OOD) was used for the models and the programming of the simulator.

The objects used in the simulator are as follows:

- `ArrayEvent` - A class used to label events and compare them
- Event - An event can be either voice, data, or VoIP.
- `EventQueue` - This represents the buffer, allowing events to be added to or removed from the queue
- `FileHandler` - This records the results of the simulation to a file
- `MersenneTwister` - This class provides the random number generator developed by Sean Luke (Luke 2000).
- `OnOffTrafficGenerator` - Generates talk spurts and gaps from a given voice call
- `SimulationEngine` - The core computational engine of the simulator
- Tester - The main class in which the simulation parameters are configured
- `VoIPTrafficGenerator` - This class breaks a talkspurt into VoIP packets

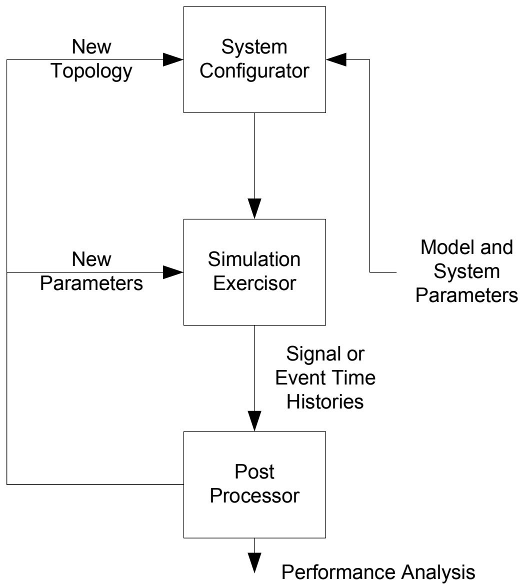


Figure 6.1: Operational structure and flow of simulation package

The computerised model structure was similar to that proposed by Balaban (1984), where the modular structure and process flow is as in figure 6.1. The system configurator is responsible for initialising the correct parameters, the simulation exercisor (`SimulationEngine`) is responsible for the actual simulation, and the output of the post processor (`FileHandler`) is what is analysed. The results of each simulation lead to further investigations and iterations of the topology and system parameters.

6.1 Generation of Events

The events generated for the simulation are of type voice, data or VoIP. The traffic intensity for each traffic type is specified in the initial simulation setup, and the traffic types are either generated natively in the Simulation engine or with assistance of the OnOffTrafficGenerator and VoIPTrafficGenerator.

The parameters necessary for voice and data are:

- The traffic intensity of the voice or data measured in erlang
- The mean duration of the voice call or data packet

From these two values the arrival intensity is simply given as the traffic intensity divided by the mean duration. The logical flow chart of the voice generation process

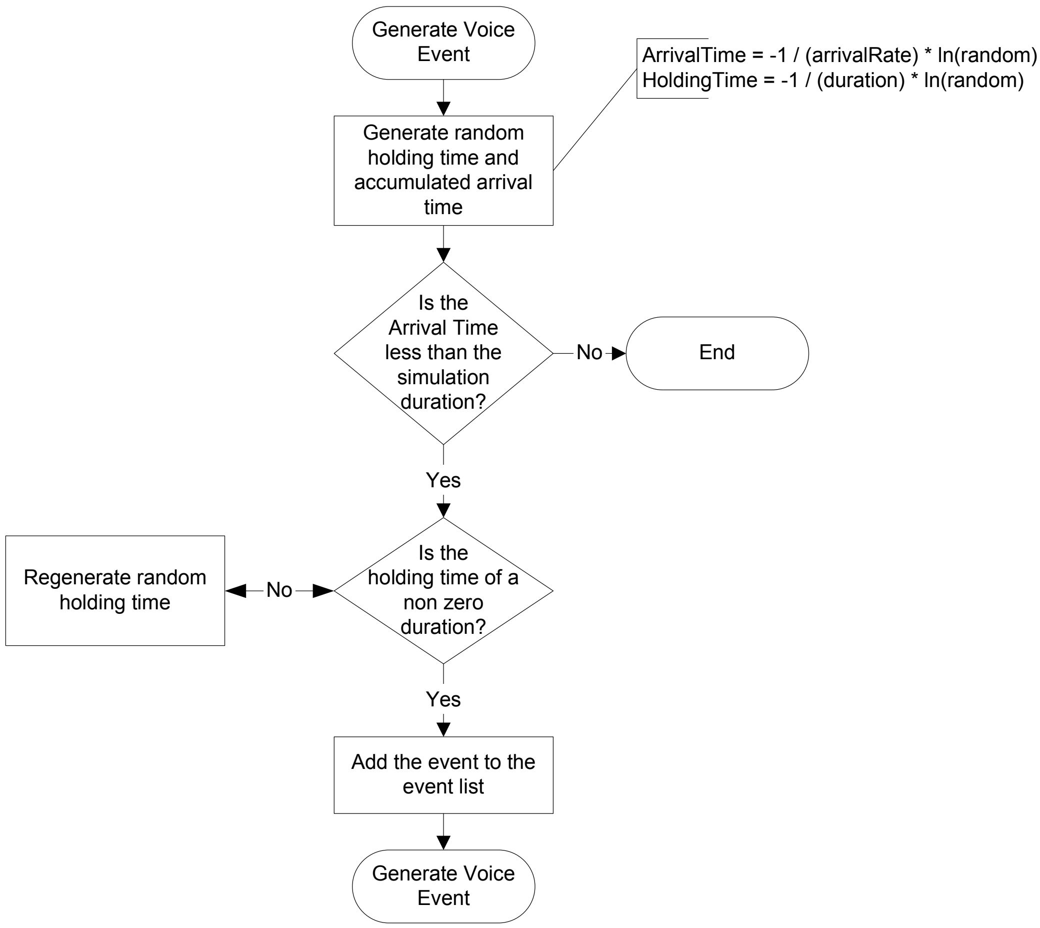


Figure 6.2: Logical flow chart of voice generation

is shown in figure 6.2.

The parameters necessary for VoIP are the following:

- The traffic intensity of the voice or data measured in erlang
- The mean duration of the voice call
- The VoIP Talk-spurt duration
- The VoIP Talk-gap duration
- The duration of the VoIP packet
- The number of VoIP packets per second

With this information, VoIP packets are generated by creating a voice conversation, decomposing it into talk spurts and gaps, by passing the voice Events to the

OnOffTrafficGenerator, and encoding each talkspurt into a number of VoIP packets using the VoIPTrafficGenerator (it was assumed that voice activity detection is used, as described in Chapter 3), and the process is illustrated in figure 6.3.

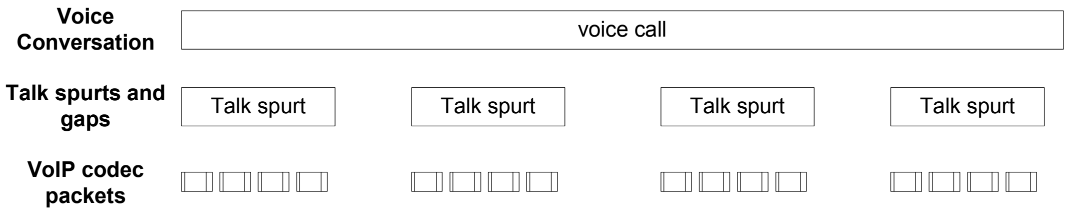


Figure 6.3: VoIP packet generation

Due to the stochastic nature of the call duration, occasionally a single voice call was of such a small duration that it was incorrectly processed in the simulator, due to rounding during number conversion and processing. **This challenge was overcome by simply testing the value of the duration before committing the event to the event list**. If the duration was too insignificant, then the duration was recalculated until a suitable value was obtained. The average mean of the generated events was found to be unchanged by this process due to the accepted duration being of an extremely small value in itself.

All generated events were stored in a one-dimensional array as opposed to a vector [1] to decrease the processing time during event searches (Singh 2003). An ArrayEvent object was created for each Event, the purpose of the ArrayEvent object was to allow the Event to be separated into a start event and a stop event, allowing the event list to be sorted chronologically. This separation of a Event into start and stop events assisted in the orderly processing of events, as described by (Higginbottom 1998, pg. 109). Once all of the generated events were added to the event list, the resulting array could be processed by the simulation engine. An example of the event list is shown in figure 6.4.

Data	Data	Voice	Data	Data	Data	Voice	Data
1	2	1	2	1	3	1	3
Start	Start	Start	Stop	Stop	Start	Stop	Stop

Figure 6.4: Illustrative list of voice and data events

[1] The Vector class implements a *resizable* array of objects, to accommodate the addition and removal of items (Sun Microsystems 2002)

6.2 Processing the Event List

The `SimulationEngine` object was responsible for the processing of the Event List. Once all the events are generated, the list is processed. There are six possible events:

- Voice Start
- Voice Stop
- Data Start
- Data Stop
- VoIP Start
- VoIP Stop

The logical flow of the simulation engine is shown in diagram 6.5. To reduce the simulation time an event was removed from the event list once it was completed, thus reducing the length of the event list as the simulation progressed. This removal of events consumed a finite amount of time and thus a tradeoff had to be made between the time to process a list and the time to remove events that were no longer required from the list. A suitable tradeoff point was found by processing the list once a voice event had been completed, thus resulting in two voice events being removed, but hundreds of frequently occurring data packet events being removed. Once any event was removed from the queue, a new effective starting time had to be reassigned to it so that the time continuum could be preserved. The new starting time required the event list to be resorted, so a modified insertion sort was used for this purpose, since the event to be sorted was known, and it allowed a faster sort than a quick sort does.

The next chapter presents the results of the simulation.

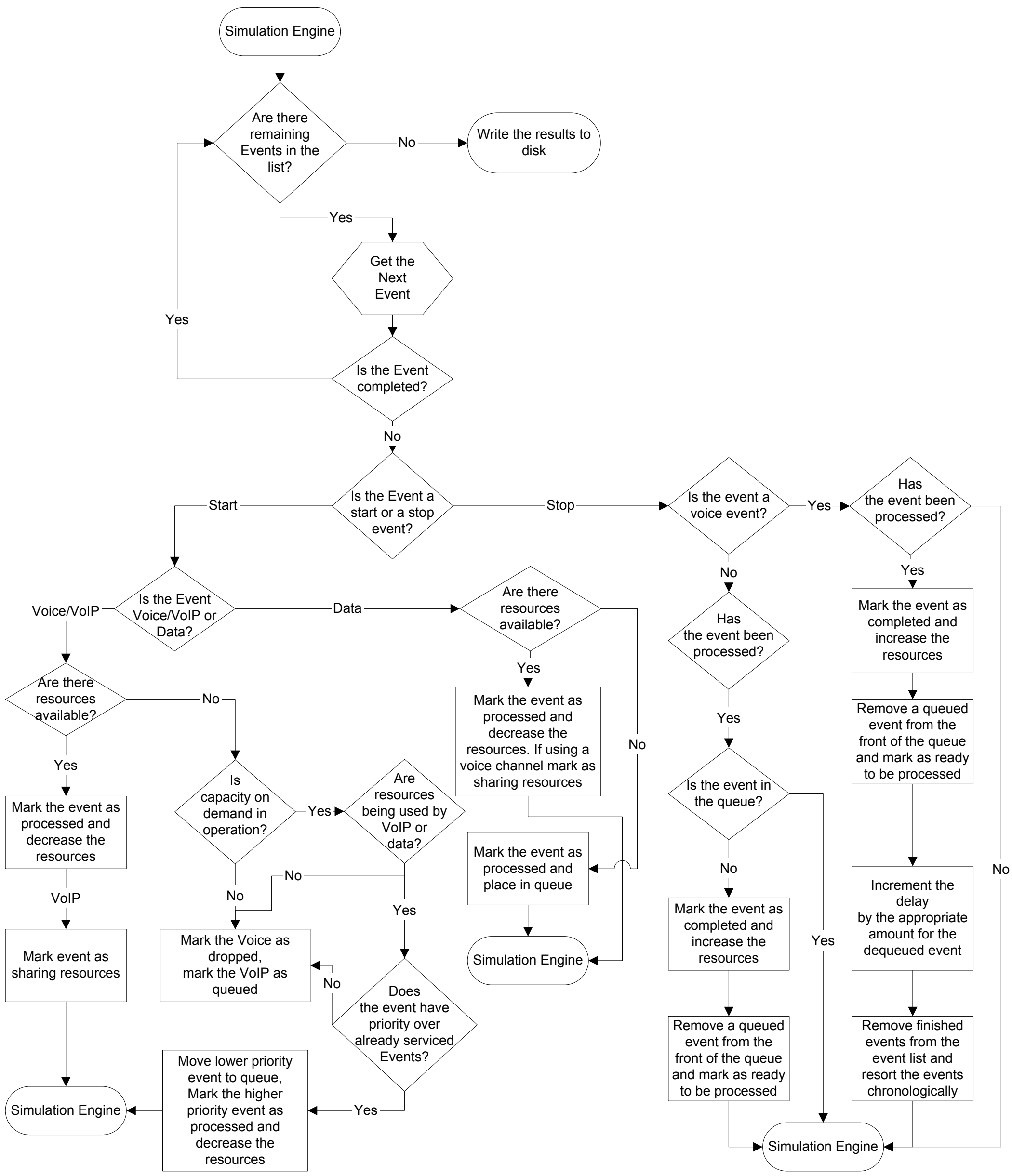


Figure 6.5: Logical flow diagram of Simulation Engine logic

Chapter 7

Simulation Results

This chapter deals with the results of simulations performed with the aid of the developed computer simulation tool. Varying loads and resource allocations, as well as different multiplexing configurations were considered to provide an insight into the patterns and trends of such configurations.

7.1 Effect of Insufficient Capacity

In this test, the effect of insufficient capacity was investigated for varying packet sizes. The purpose was to determine the theoretical delay and performance of various packet sizes under varying load in a condition of insufficient capacity. Three packet sizes for the data were considered in this experiment: 100 bytes, 200 bytes and 300 bytes. One erlang of data traffic was applied with no channels reserved for the data, but capacity-on-demand was in operation. As can be seen from figure 7.1, the average delay is similar for all three packet sizes. This is also the case for a load of two data erlang as in figure 7.2. From these simulations we can conclude that in the case of insufficient resources the delay is similar, regardless of packet size (Vannucci and Chitamu 2003a;c). This result is intuitive since in the case of insufficient resources, where the service rate is less than the arrival rate, the buffer would overflow and all packets would receive the same service regardless of the packet size.

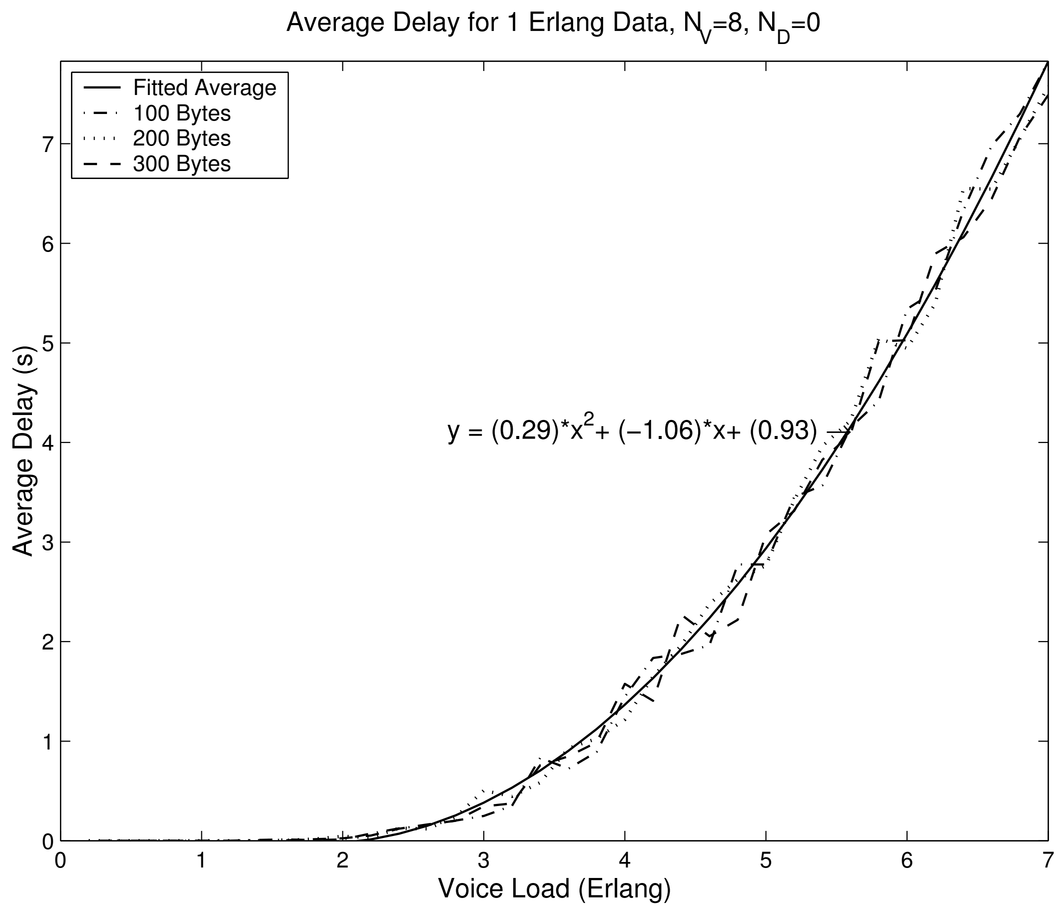


Figure 7.1: Average delay of various packet sizes with one data erlang load and no fixed resources.

7.2 Packet Delay: Capacity-on-Demand vs Fixed Channel Allocation

In this test the effect of capacity-on-demand is investigated. The purpose of this test was to determine the effect that Capacity-On-Demand (COD) has on the probability of a data packet being delayed. Different PDCH allocations were considered with COD in operation, and the delay probability was compared to that predicted in the same situation with a fixed resource allocation by using theoretical calculations. Due to capacity-on-demand, as the voice load increases the average resources available to the data decrease. Since the available resources in a capacity-on-demand situation is constantly fluctuating, it was taken that the average resources available to the data would be the total resources less the average voice load, thus the theoretical probability of delay is shown in the figures (by means of solid dots) for eight channels minus the voice load. Note that **the number of channels has to be an integer value**.

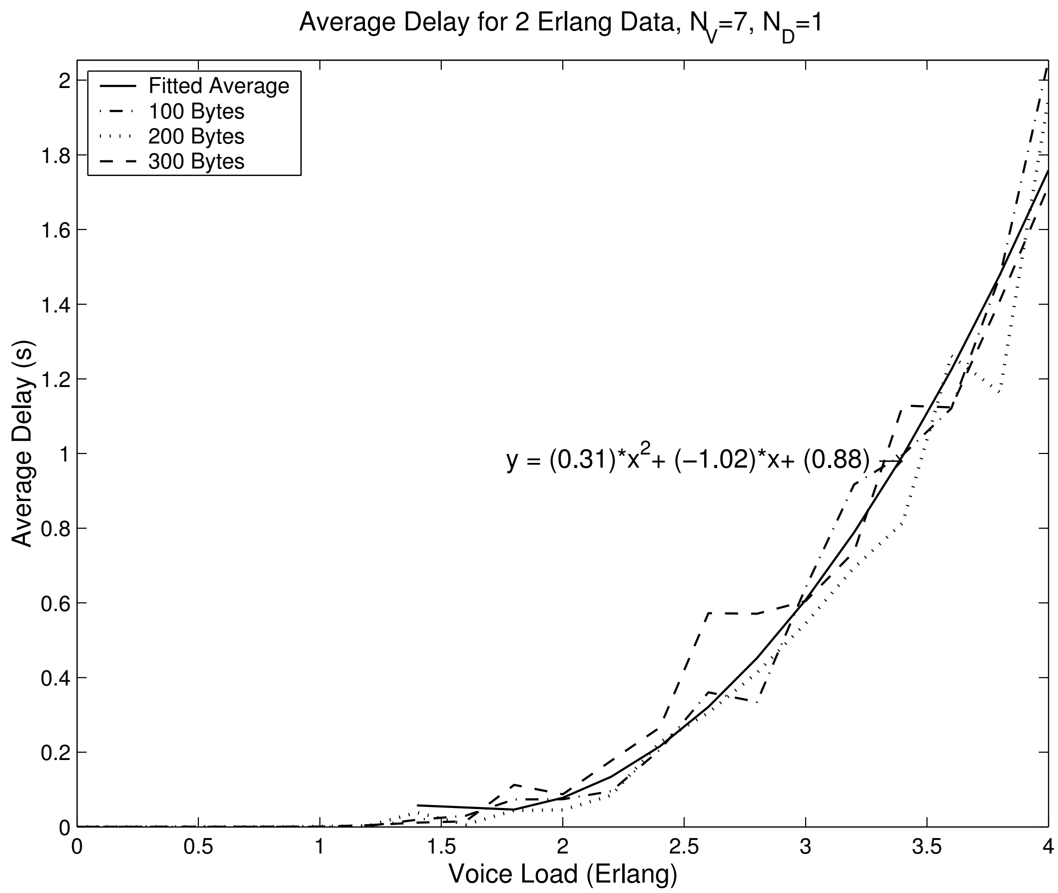


Figure 7.2: Average delay of various packet sizes with two data erlang load and one PDCH.

Figure 7.3 shows the case when there is a load of two data erlang and a variable voice load, the theoretical probability of delay is shown with the solid circles, at integer channel values, whilst the results of the simulated effect of COD are shown with a solid line. Since there are only six channels available for voice traffic, one would not expect a voice load in excess of 3.75 erlang, assuming that the voice GoS should be less than ten percent. Nevertheless, for the purpose of determining the pattern of use it can be seen in figure 7.3 for the case of five erlang of voice, the effective delay probability is less than that predicted if there was an additional packet data channel without capacity-on-demand in operation. At a voice load of just over three erlang, the delay probability is very similar to that predicted by four dedicated packet data channels. And similarly at a voice load of just over two erlang, the delay probability is very similar to that predicted by five dedicated packet data channels. At loads of one to two erlang of voice traffic, the probability of delay is 5% or less, whilst at high voice loads of five erlang, the probability of delay is approximately 32 percent (for the case of fixed resources with two data erlang and two data channels there would be a constant 100% probability of delay).

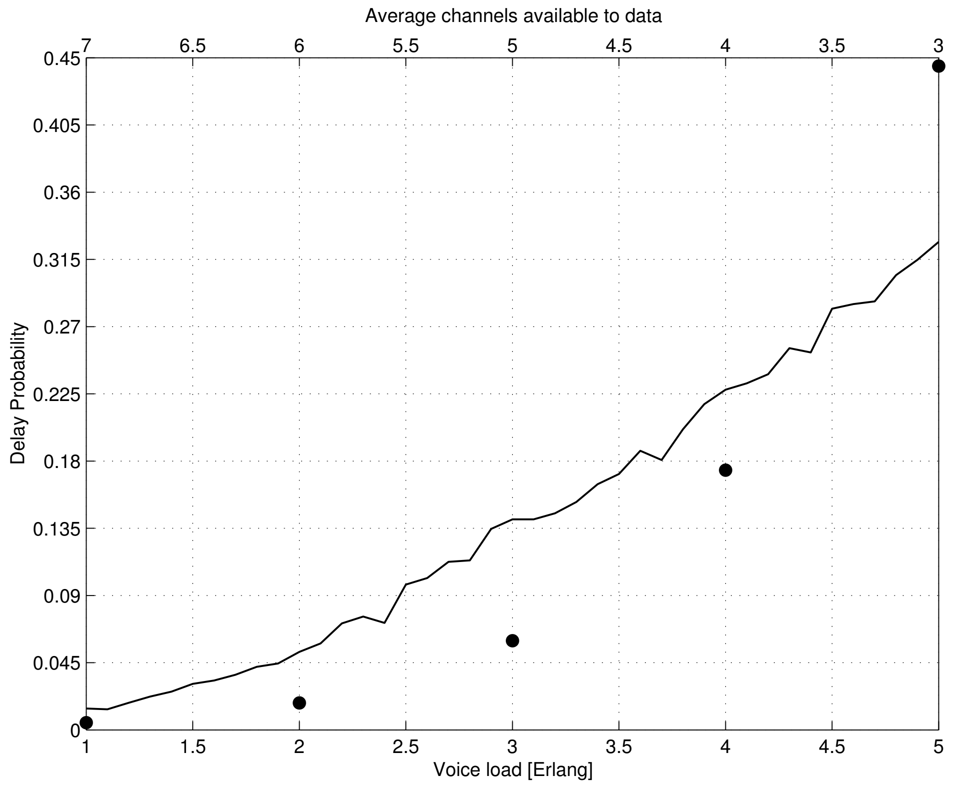


Figure 7.3: The effect of COD,Delay Probability, two data erlang, two PDCH

The associated average delay is shown in figure 7.4. Figure 7.5 shows the effect of COD in a case where there are three channels reserved for data, considering that the region of operation would be three erlang or less, the gain is not so much in the probability of delay as it is in the delay itself. Whereas for the case of three erlang, as shown in figure 7.6, the delay is far less than that of the case in which there are only two fixed packet data channels as in figure 7.4. One can conclude that capacity-on-demand provides a significantly better grade of service to data packets, effectively providing more than one additional data channel under the worst case scenario.

By comparing figures 7.7 (in which two dedicated channels have been reserved for the data), 7.9 (in which three channels have been reserved for data), and 7.11 (in which four channels have been reserved for data) it can be seen that the probability of delay is decreasing slowly as expected, but the average packet delay is decreasing considerably with each additional packet data channel (as seen in figures 7.8, 7.10 and 7.12), indicating that the additional packet data channel does not significantly decrease the data delay probability, but would have a measurable effect on the average delay of any packets that are blocked. This indicates that in the situation of capacity-on-demand, the average delay is the dominant factor of consideration.

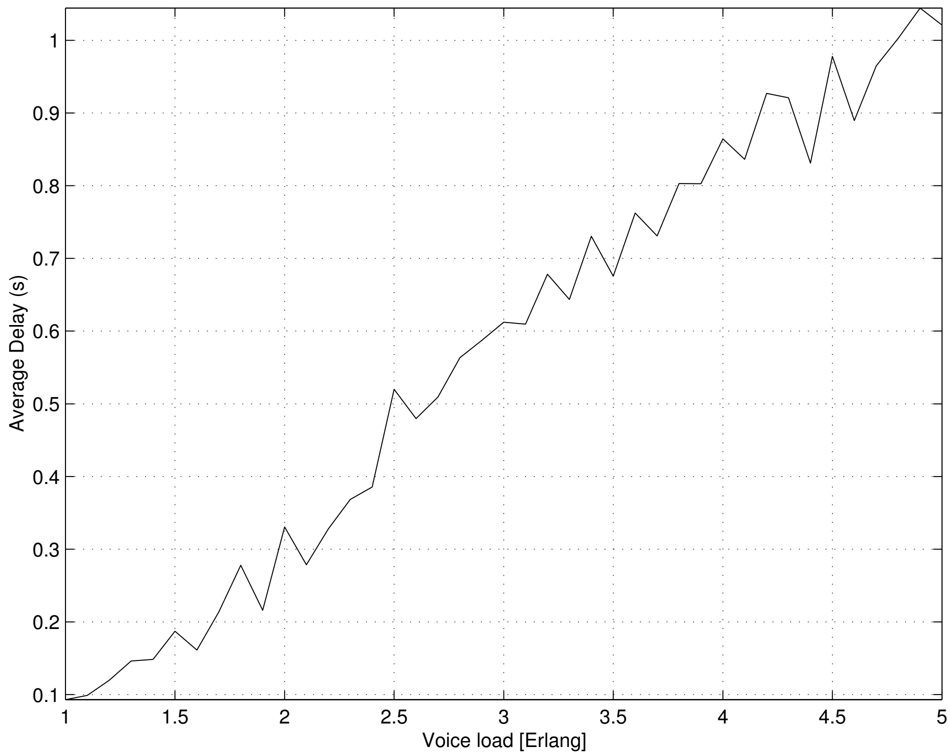


Figure 7.4: The effect of COD,Average Delay, two data erlang, two PDCH

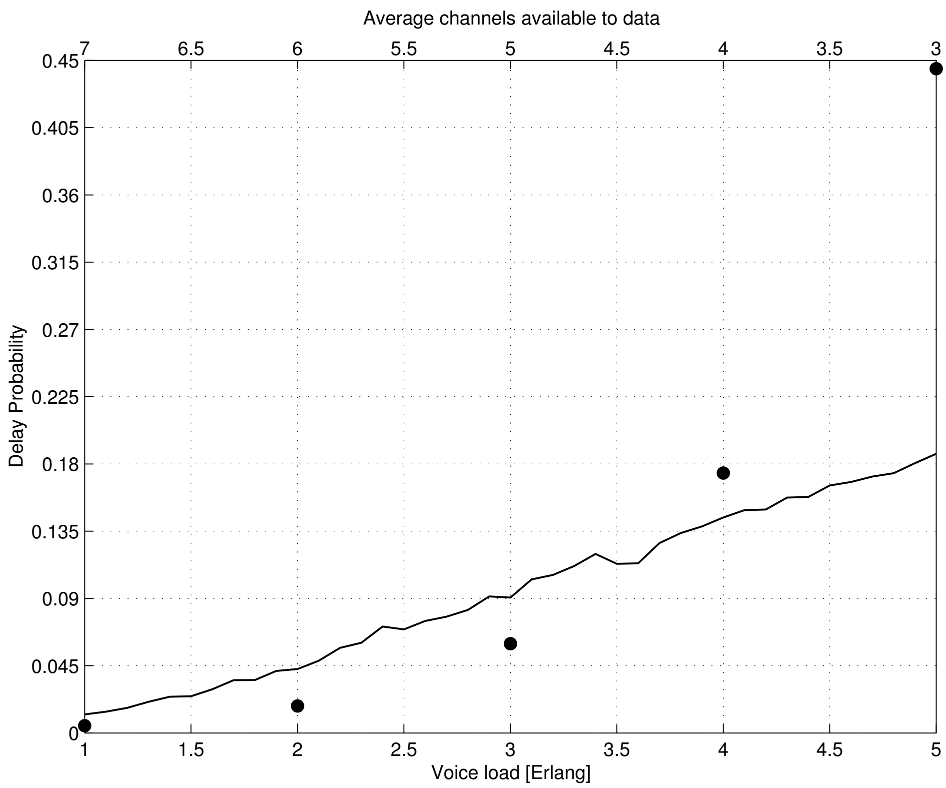


Figure 7.5: The effect of COD,Delay Probability, two data erlang, three PDCH

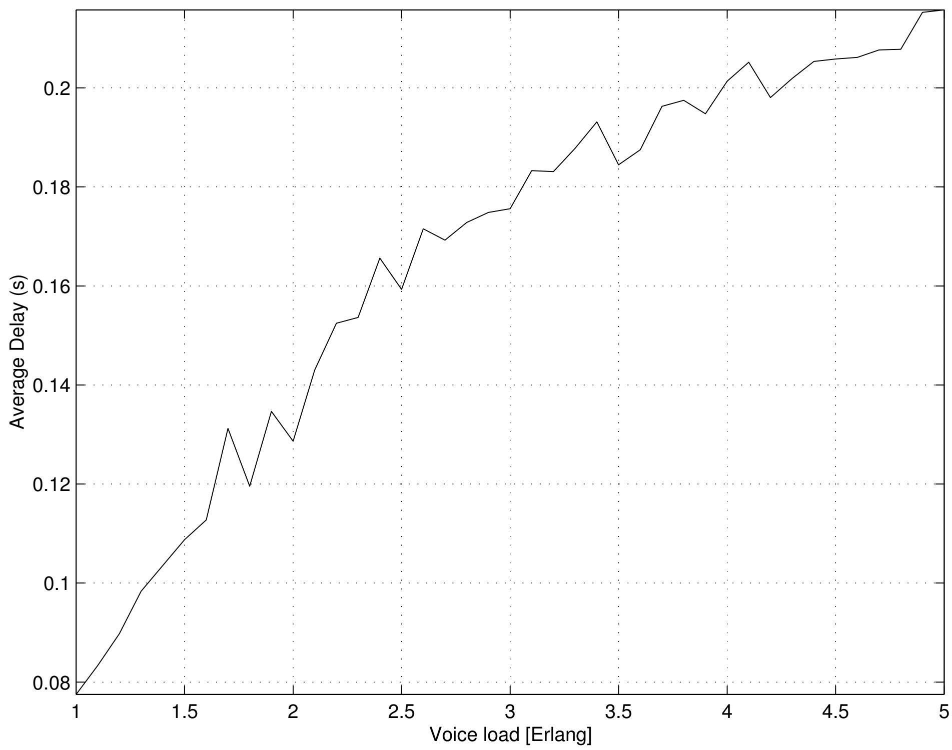


Figure 7.6: The effect of COD,Average Delay, two data erlang, three PDCH

One can observe that the effect of capacity-on-demand is to provide a flexible delay, as opposed to the static delay of approximately 2.5 seconds for three fixed packet data channels, no capacity-on-demand and 100% probability of packet delay.

7.3 Packet Delay: Delay vs. Load

In this packet delay test of delay vs load, the average delay for a packet for a given load was investigated. In figure 7.13 one channel was reserved for data with COD, a fixed voice load was applied, and the data load varied from zero to seven erlang. Likewise in figure 7.14 the simulation is repeated for two channels reserved for data. The purpose of this test was to determine the probability of delay for a given voice load and resource allocation. For example, from the graph one can see that there is a 30 percent chance of delay for the data when there is a load of five data erlang and one voice erlang. However when there is a voice load of four erlang, only three data erlang can be supported with the equivalent probability of delay.

One can see that the probability of delay does not alter between the two graphs for low voice traffic loads, indicating that there are sufficient free voice resources to

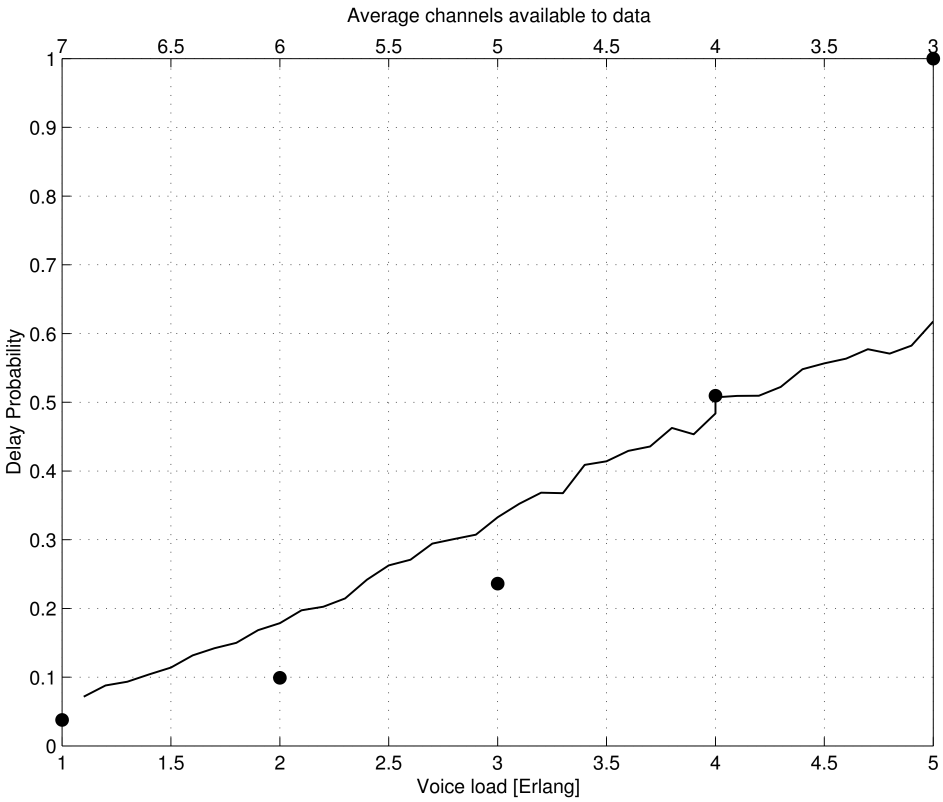


Figure 7.7: The effect of COD,Delay Probability, three data erlang, two PDCH

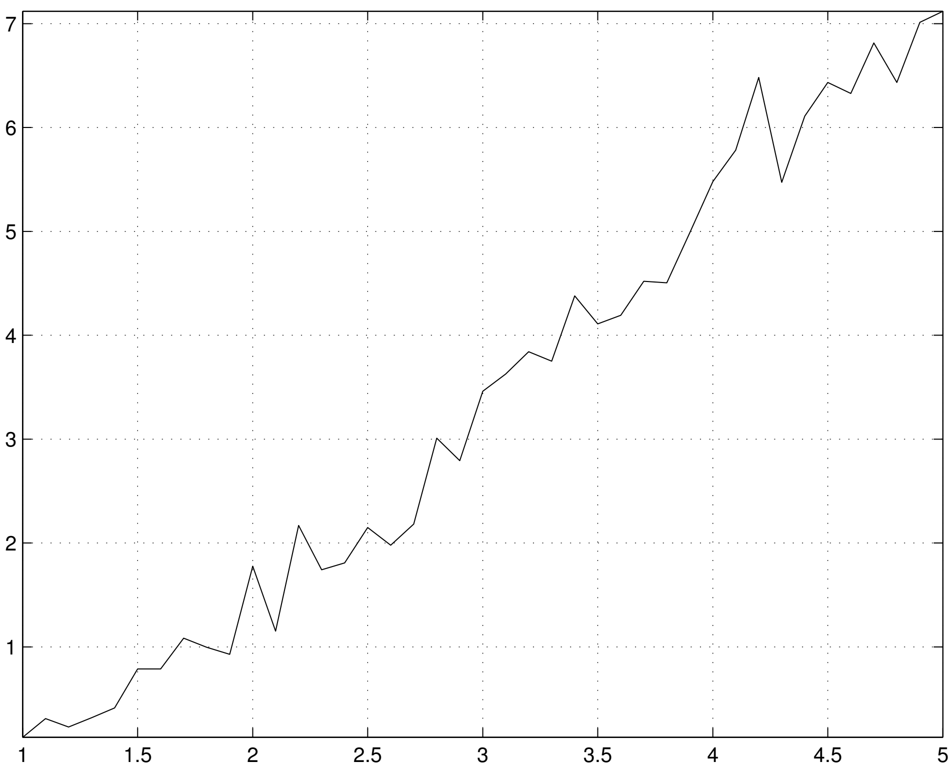


Figure 7.8: The effect of COD,Average Delay, three data erlang, two PDCH

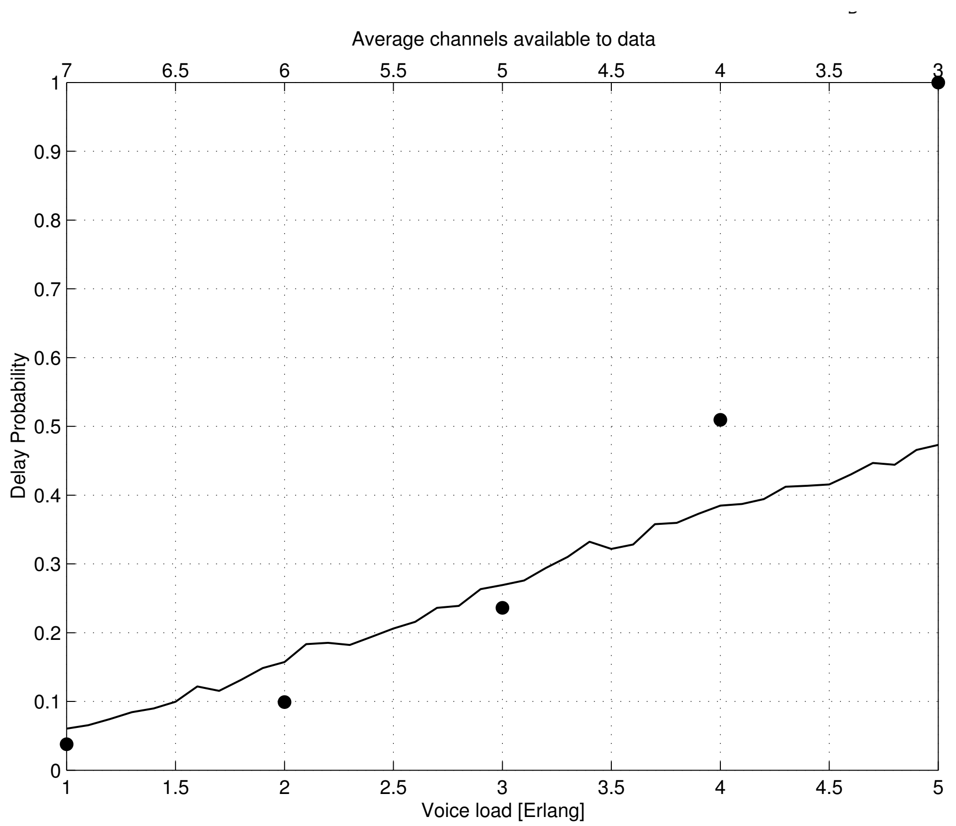


Figure 7.9: The effect of COD,Delay Probability, three data erlang, three PDCH

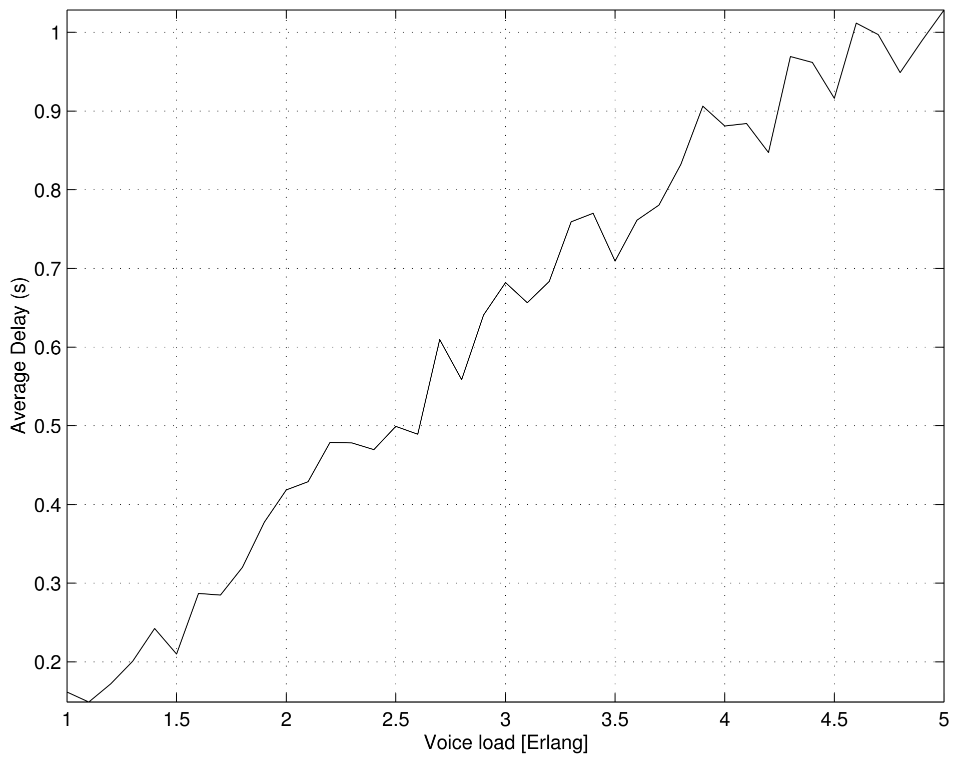


Figure 7.10: The effect of COD,Average Delay, three data erlang, three PDCH

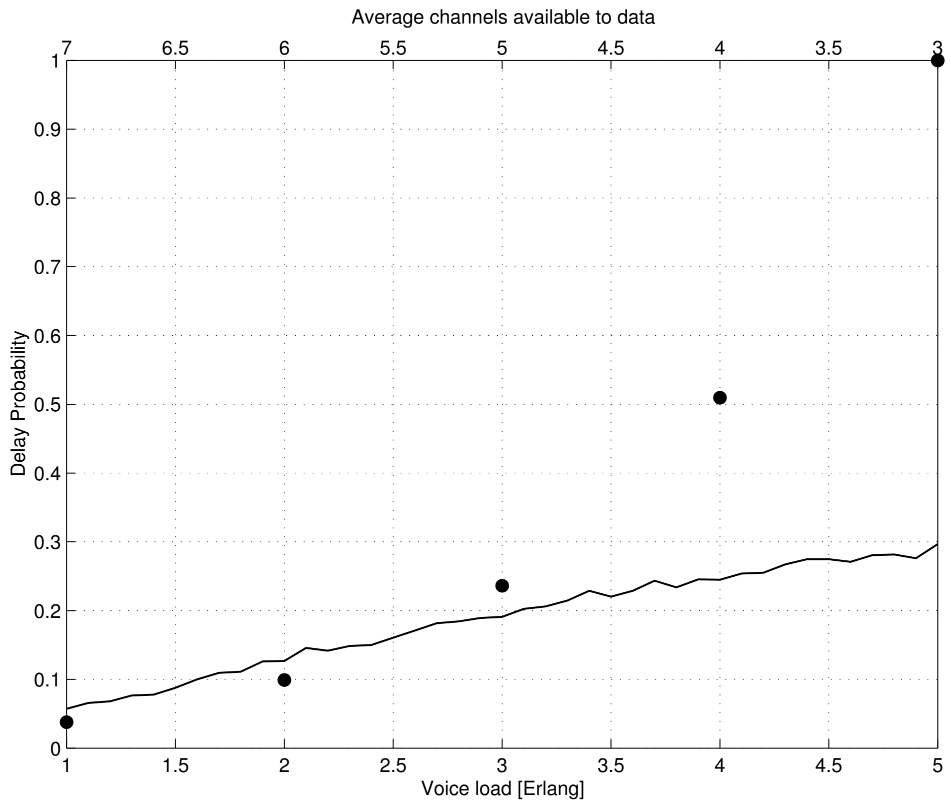


Figure 7.11: The effect of COD,Delay Probability, three data erlang, four PDCH

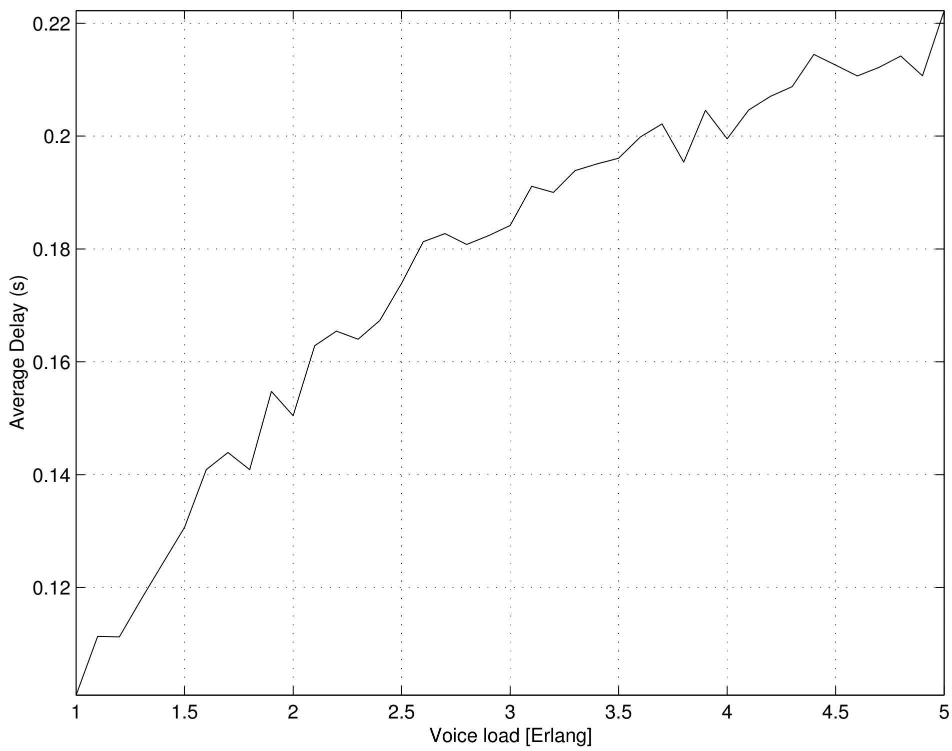


Figure 7.12: The effect of COD,Average Delay, three data erlang, four PDCH

have an equivalent delay probability. As the voice load increases, at higher data loads, the probability of delay becomes constant, indicating that there is a constant number of delayed packets, having an unacceptable grade of service.

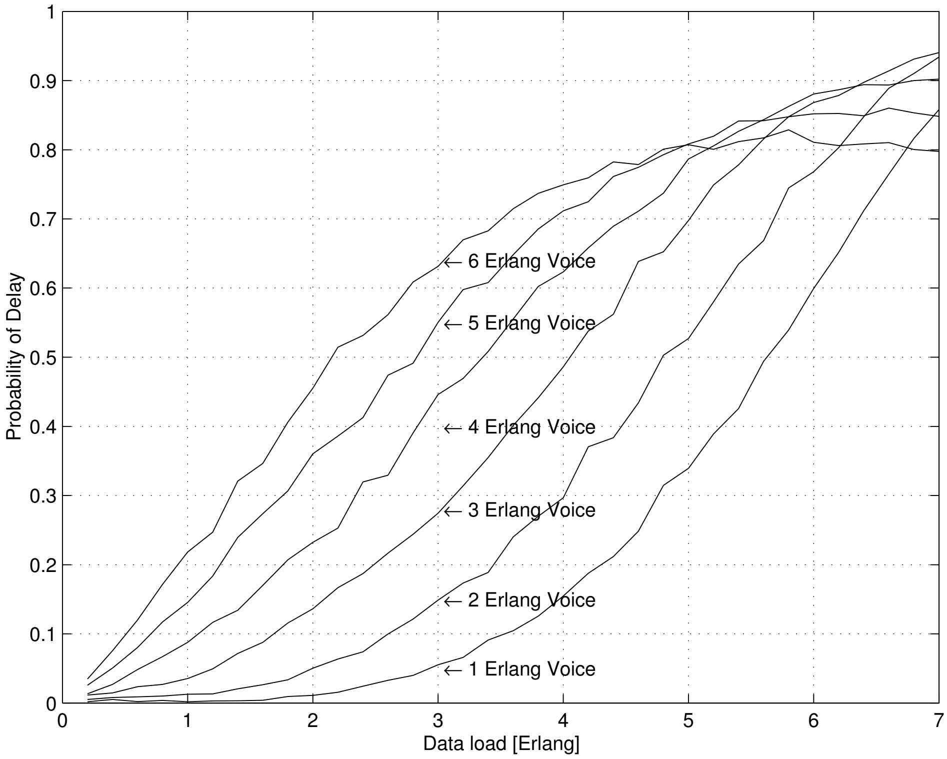


Figure 7.13: Probability of delay for various voice loads, $N_V = 7$

7.4 VoIP and Data: GoS

In this VoIP and data test, VoIP was generated using the two-state Markov model. The VoIP load was fixed whilst the Data Load was varied. Figure 7.15 shows the probability of a data packet being delayed for a given VoIP load and figure 7.16 shows the equivalent VoIP probability of delay. One can see that at low data loads the probability of delay is similar despite the significant packet size difference (30 byte VoIP packets compared to 450 bytes). It is only at high loads, above a combined load of five data erlang, that the difference in delay probability changes marginally. This indicates that, despite the VoIP prioritisation, the probability of delay is equivalent, which indicates that prioritisation has a greater effect on the average delay than on the probability of it occurring.

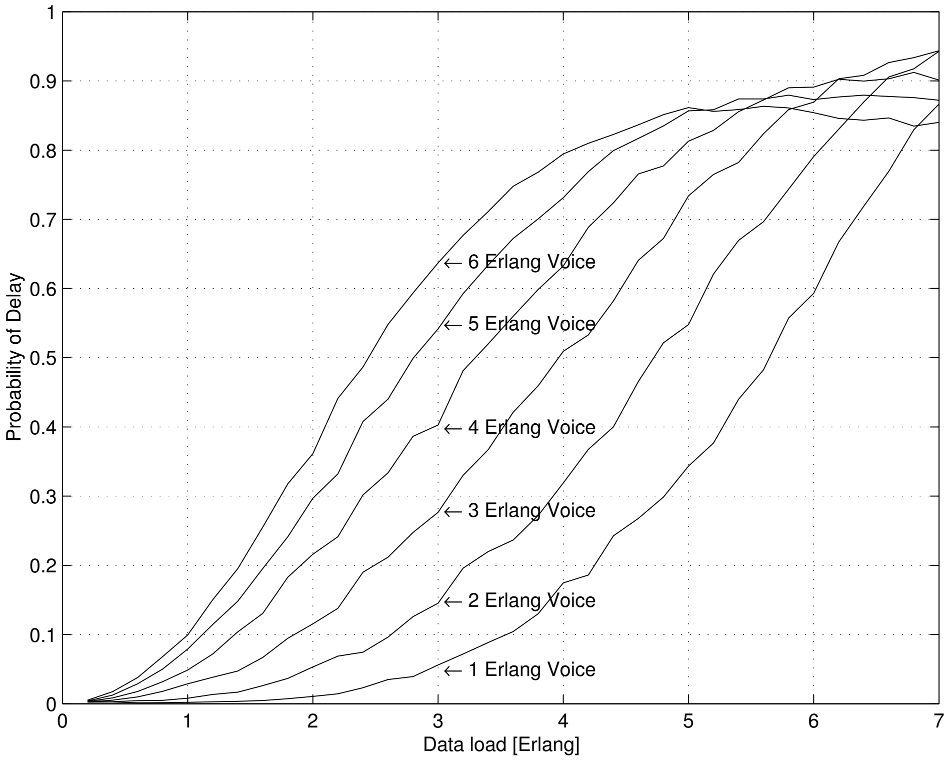


Figure 7.14: Probability of delay for various voice loads, $N_V = 6$

7.5 VoIP Exponentially Distributed and Data: GoS

In this experiment VoIP packets were represented using a simple exponential model. The VoIP packets were assumed to be 30 bytes, representing 20 ms of speech. It is further assumed that approximately 65% of the conversation was silence, and thus one VoIP packet occurs every 60 ms. In figures 7.17 and 7.18, the data was assumed to consist of 450 byte packets. In figures 7.19 and 7.20 the data was assumed to consist of 150 byte packets. It can be seen that figures 7.17 and 7.19 are similar, thus indicating that the size of the data packets is not a large factor in the delay probability, but rather affects the average delay. Figure 7.18 and 7.20 show the corresponding average delay for the given VoIP load. From the simulation one can see that at loads of four erlang and above, the average delay is similar in both graphs, whilst at lower VoIP loads, the smaller 150 byte packets suffer less delay, as expected. A conclusion which can be made from such behaviour is that smaller packets whilst having a similar probability of delay as larger packets, do not have as large a delay under lower loads and thus would provide a better grade of service for an application.

In figure 7.22, the average delay for exponentially distributed VoIP packets is shown

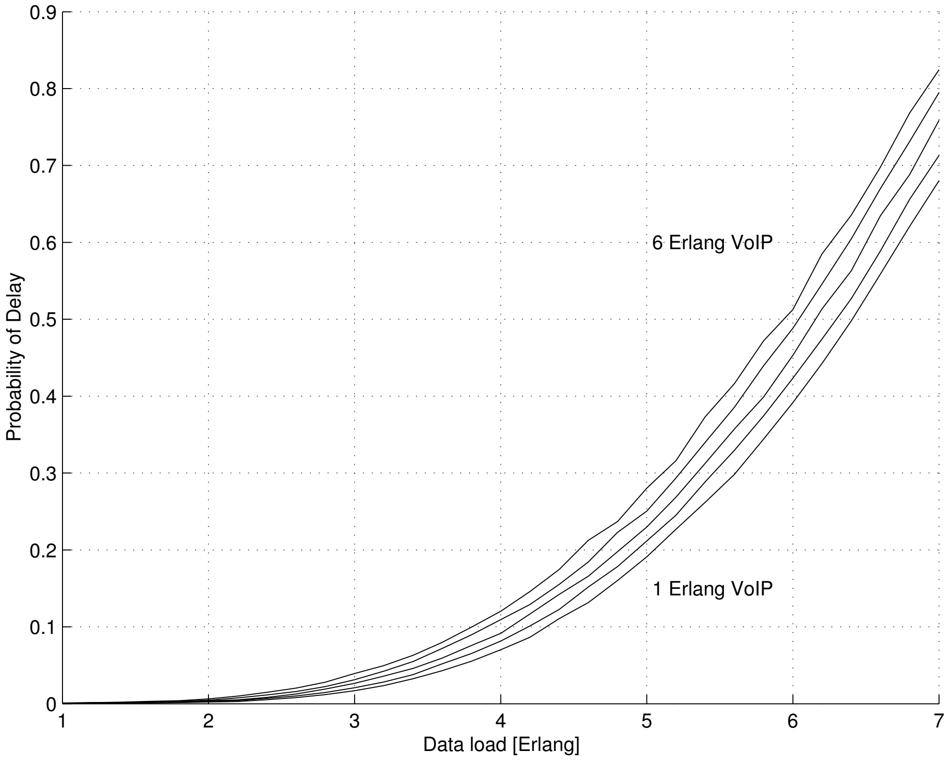


Figure 7.15: Two-state VoIP and data, Probability of data delay

for loads of one to six data erlang. From this figure one can see that if a delay of less than 150 ms is to be used, then with three data erlang, only four data erlang of VoIP can be supported, assuming no further delays. One can observe that a combined load of no more than seven erlang should be present for eight channels if the VoIP delay is to be kept at 150 ms. In figure 7.21 one can see that the probability of a VoIP packet being blocked is consistently increasing as the data load increases, thus with three data erlang and five data erlang of VoIP the probability of a VoIP packet being delayed is approximately 27%.

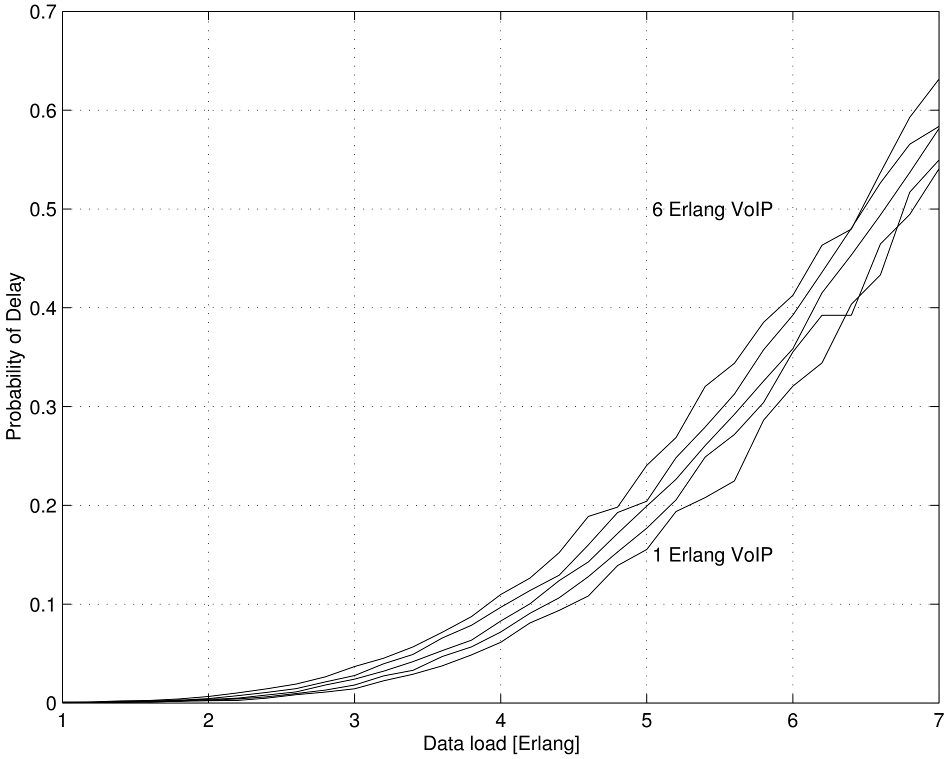


Figure 7.16: Two-state VoIP and data, Probability of VoIP delay

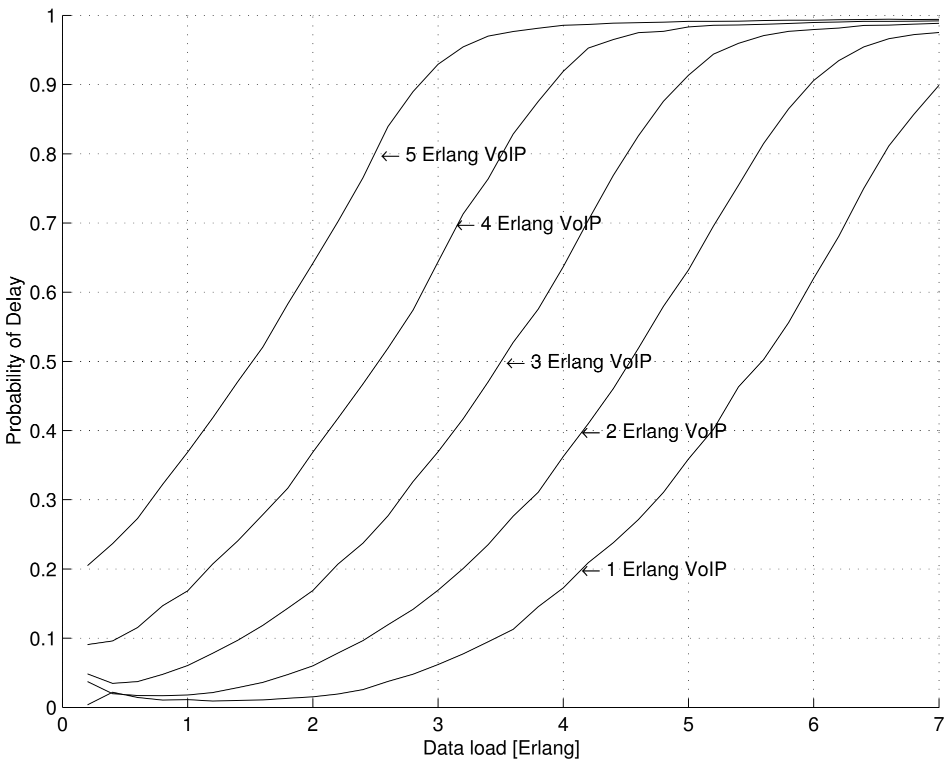


Figure 7.17: Probability of data delay for various exponential VoIP loads

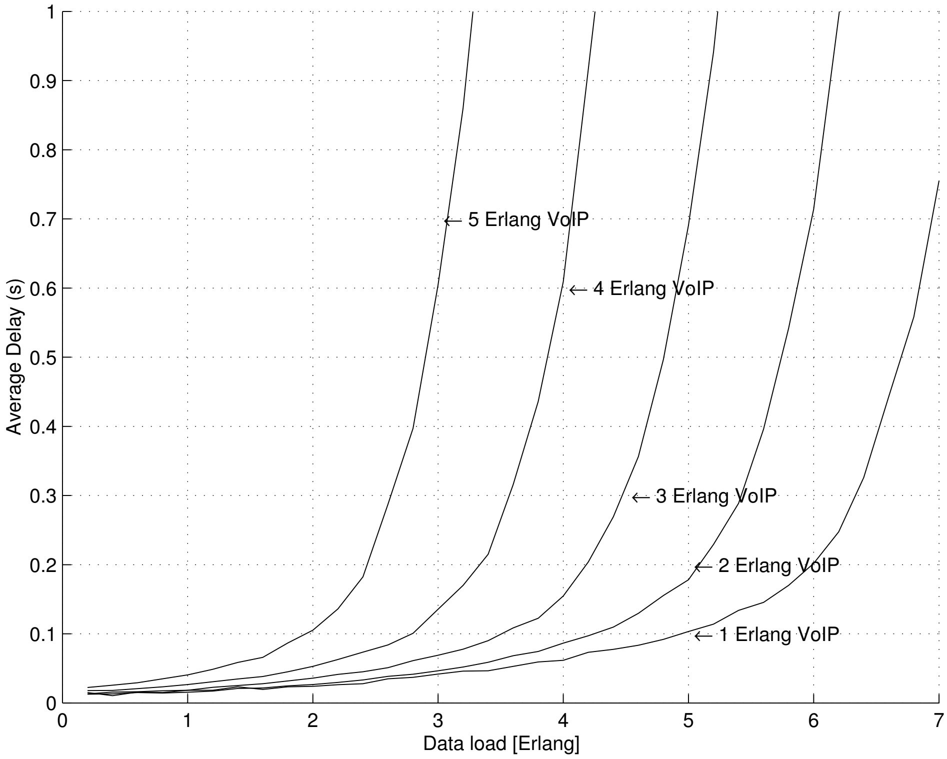


Figure 7.18: Average data delay for various VoIP loads

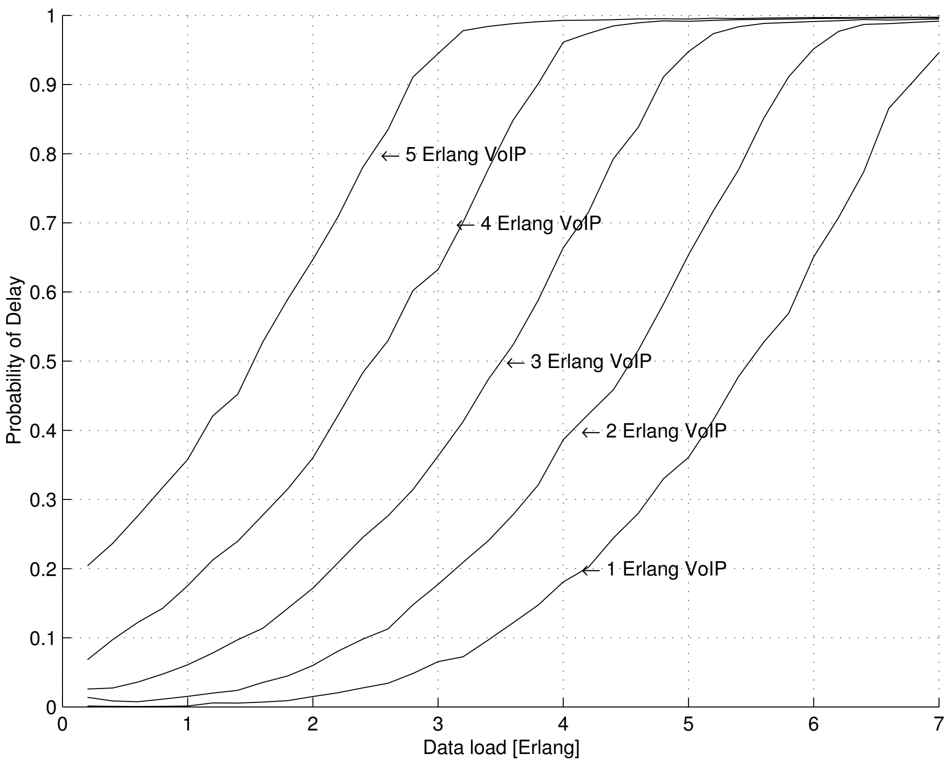


Figure 7.19: Probability of data delay for various voice loads

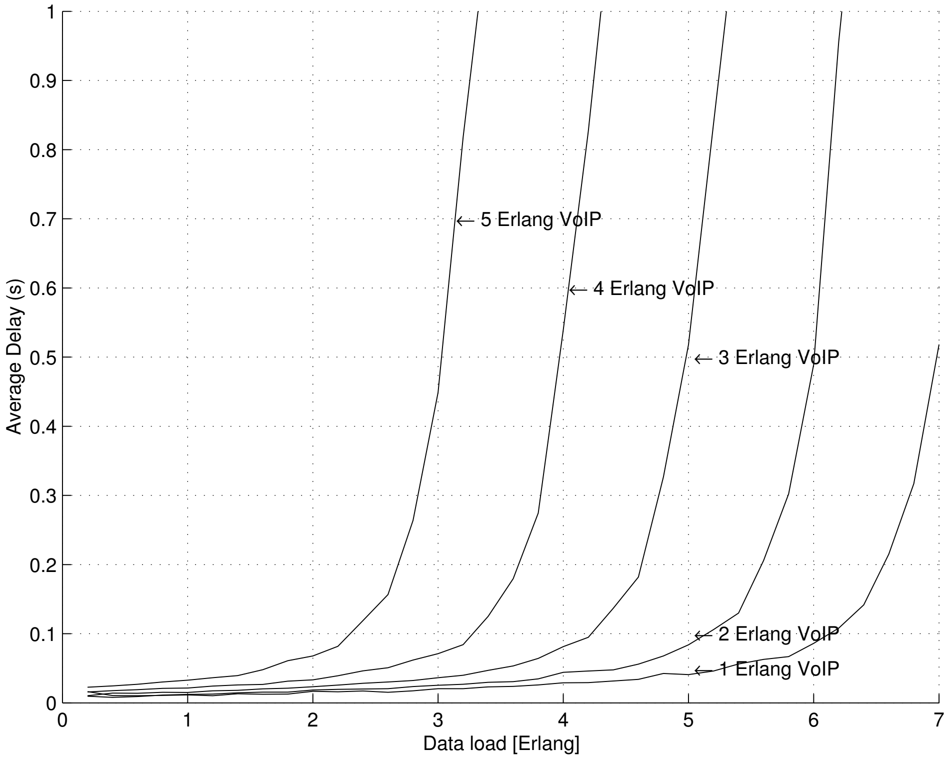


Figure 7.20: Average data delay for various VoIP loads

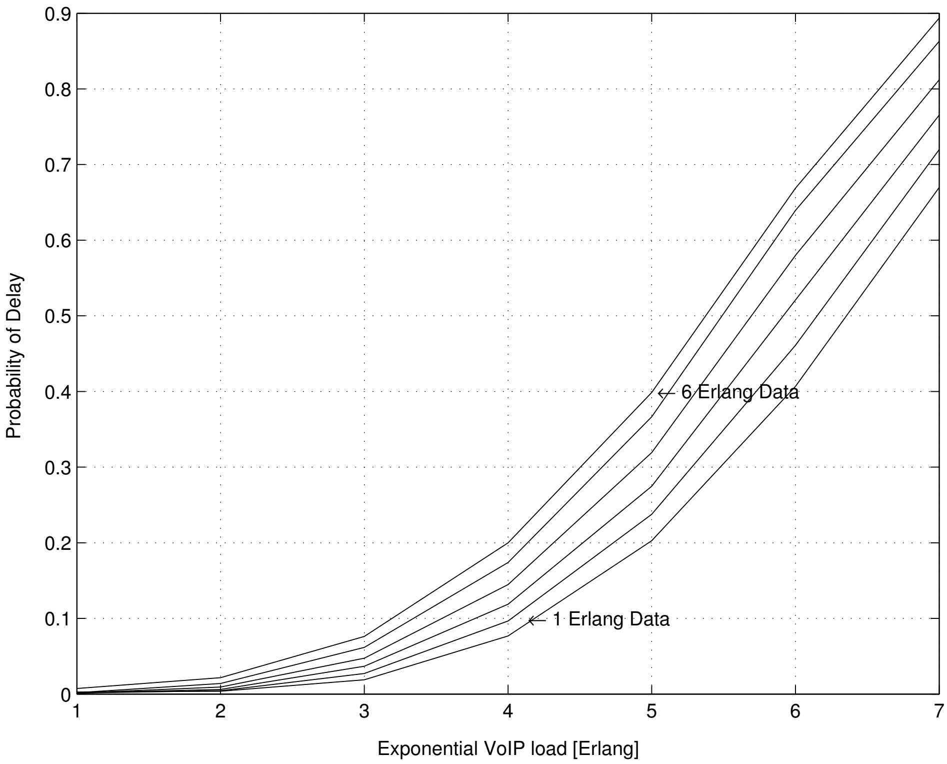


Figure 7.21: Probability of exponentially distributed VoIP delay for various data loads

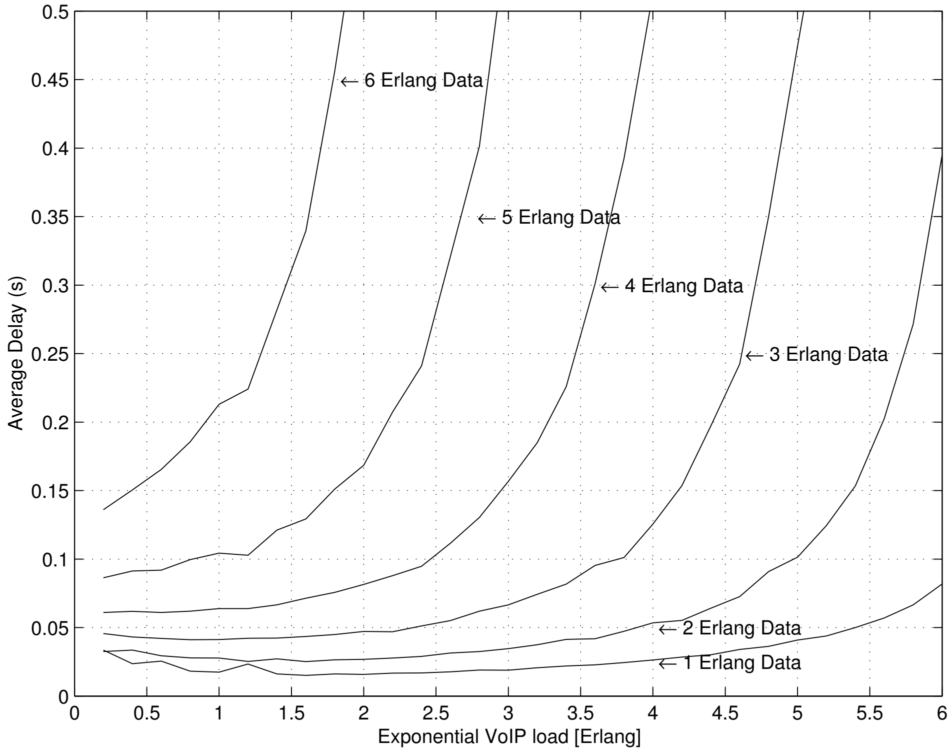


Figure 7.22: Average exponential VoIP packet delay for various data loads

Chapter 8

Review and Conclusions

It is clear that the current communication networks are rapidly converging towards one homogeneous network capable of fulfilling many different requirements. The key factor in such a convergence is the ability to support both voice and data simultaneously, even though both services have very different requirements. Wireless communications is seen as the most suitable medium for communication in Africa, due to the large coverage areas and low average population densities. This is clearly demonstrated by the growth of mobile telephone subscribers on the continent. However, with wireless communications the bandwidth is a limited resource and thus has to be carefully conserved and managed to ensure the best possible allocation for a particular service and area. Thus the resource allocation between voice and data was examined in this study, with particular reference to the GSM/GPRS cellular network.

8.1 Review

GSM, together with GPRS, offers the possibility of serving both circuit switched voice and packet switched data over a common air interface. The multiplexing of resources can take on two forms in a GSM/GPRS network, a static resource allocation for both circuit switched voice and packet switched data, or a superior flexible moving boundary resource allocation. Both resource allocations were considered in this study, as well as the possibility of using Voice over IP in place of circuit switched voice.

The use of VoIP is seen as a key feature in the move towards an all IP based packet network. Such a network would offer a number of benefits such as the use of generic data networking equipment, common service creation and more flexible network configurations. The possibility of transporting voice using VoIP would allow GSM/GPRS networks to be able to support an all IP operation, thus allowing the GSM/GPRS network's life to be extended, since some 3G network services could be offered immediately. In addition to this, is is possible that there could be a capacity gain since the majority of a voice conversation is silence, and in GSM operation even though the mobile station does not transmit, during silence the channel is still reserved, whilst using GPRS those resources could be used for another purpose.

VoIP has a large number of requirements that have to be overcome before VoIP could be implemented in a GPRS network. For example VoIP is more sensitive to delay than standard data, but is less sensitive to loss. The VoIP header has a large overhead due to the many protocols used, and there are a large amount of different codecs. A number of these issues have been addressed in previous studies and solutions exist to satisfy a number of the requirements such as header compression, voice activity detection, and low variable-bit-rate codecs, allowing VoIP to be seen as an accepted technology.

To study the allocation of radio resources at the GSM/GPRS air interface, traffic source models had to be developed which would suitably represent the nature of the offered traffic. For this purpose traffic models were created for circuit switched voice, packet switched data and packet switched voice. Circuit switched voice and packet switched data were represented using Markovian models, Markovian arrivals and exponentially distributed holding times. VoIP was modelled using a two-state Markov modulated Poisson process, which better models the talk-spurts and talk-gaps in a conversation. The base station subsystem was modelled as a controller with eight channels (one GSM/GPRS TDMA frame). For the voice operation, a blocked-calls-lost approach was used, and for the data a blocked-calls-queued approach was used. All the circuit switched users were assumed to be in one class of traffic whilst, packet switched data was assumed to be queued in a common queue. Continuous time analysis was used in which the channels can be seen as operating in parallel, rather than discrete time slots. Capacity-on-demand was modelled as in a moveable boundary resource allocation. The radio propagation environment was considered by means of a large scale propagation power outage model, which would merely alter the load presented to the simulator.

Using the Erlang formulae, the Grade of Service for a given voice and data load was estimated and the results presented in a graph that allows for intuitive inspection.

However the Erlang formulae require a static number of channels to evaluate the GoS. In order for the GoS to be evaluated for a capacity-on-demand situation an average of the real-time instantaneous GoS would have to be obtained, over the duration of the business day busy hour.

A complete validation and verification of the computer simulation was performed, developing the simulation from the given problem entity to the conceptual model and implementing that model using the Java high level programming language. The models were independently verified and validated and found to be acceptable for the given scenarios. The simulator underwent extensive verification whenever possible, and the results were compared to those predicted by theory, thus allowing confidence to be established in the simulator.

The models were implemented using logical flow charts to facilitate the programming and description of the computer simulation. Specific parts of the simulator were described to facilitate an understanding of the workings of the simulator and some of the optimisations implemented were discussed to give insight into some of the problems of constructing a computer simulation.

8.2 Conclusions

This study has presented the results of an investigation into, and simulation of the air interface radio resource allocation of a GSM/GPRS network. Various different resource allocations for voice, VoIP and data were investigated, under varying loads. The results of the simulations lead to a number of conclusions:

1. In the case of insufficient resources allocated to data, the average packet delay is similar regardless of the size of the data packet. It was found that if capacity-on-demand is in operation, then the allocated data resources should be equal to the expected load or more, to ensure delay is kept to a tolerable limit.
2. Investigations into fixed resource allocation have shown that in the case of sufficient statically assigned data channels with capacity-on-demand being used, additional packet data channels were found to have a greater effect on the average delay than on the probability of delay. Capacity-on-demand was found to be equivalent to an additional channel under a high voice load scenario, and provided an increasing benefit as the voice traffic decreased. In addition,

capacity-on-demand was found to provide an average delay probability well below that predicted in a fixed resources scenario. Of greater importance, the average delay is varying and less than that predicted with a static resource allocation.

3. From the analysis of the case of VoIP and data, it could be seen that the prioritisation of the voice packets did not significantly alter the probability of delay but rather had a more pronounced effect on the average data packet delay.

4. The size of the data packet was found to be a significant factor in the average packet delay. However it does not significantly affect the delay probability. Thus one can deduce from the analysis of the GSM/GPRS operation, smaller packet sizes would provide a better grade of service than large packets.

5. Simulations of VoIP and data have shown that the combined load should not exceed seven erlangs (for the case of eight channels), if the average delay is to be kept below one hundred milliseconds. This leads to the conclusion that when transporting voice over IP, a higher average load can be tolerated than the case of circuit switched voice and packet switched data.

8.3 Recommendations and Future Improvements

From the results of the investigation, transporting voice over packet switched data appears to offer greater traffic carrying capacities through the use of statistical multiplexing. However as a study that was intended to investigate the trends in the resource allocation, it is by no means a completed study. Improvements that could be made are as follows:

- Internet traffic and packet switched data is often heavy-tailed and an exponentially distributed Markovian model, whilst adequate for a first order approximation, does not suitably represent all data traffic types, especially given that the multimedia messaging and services often have peculiar characteristics.

- A large scale propagation power outage model was considered which would merely alter the load presented to the base station controller. However mobile station movement and the effects of interference were not considered, so the results of the simulations could be slightly optimistic due to the absence of these factors.

- By modelling the base station subsystem as a continuous time controller with eight channels, the essence of the radio resource allocation was captured. However such a model does not consider the intricate resource reservation and allocation signalling of a GSM/GPRS network, a factor which would be critical in any in-depth study on such a topic.

- The computer simulation was found to be slow when extremely large loads were applied to the simulator. This is due in part to the Java virtual machine. Whilst Java is a programming language that is extremely object orientated, it is not able to compile natively into byte code, and as such is always slower than executable languages. It is clear that Java is not the optimal language to use when such computer intensive simulation packages are created, and C++ would have provided additional speed. However Java offers all of the functionality of other programming languages, so only this one factor was not in its favour.

In conclusion, this study was intended as a first order investigation of the issues of resource allocation between circuit switched voice, packet switched data and packet switched voice. The developed simulator was found to produce results which compared to those predicted by theory, and further used to produce results for more complex cases of resource allocation. The methodology used in this investigation will hold for cases of other networks where resources are similarly allocated.

References

Ali, M. M. and Song, X.: 2003, A Performance Analysis of a Multiplexer with Priority Queues and Correlated Arrivals, *Proceedings of the 18th International Teletraffic Congress – ITC-18: Providing Quality of Service in Heterogeneous Environments*, Vol. 5a, International Teletraffic Congress, Berlin, Germany, pp. 481–490.

American National Standard: 1996, Transcoding: Definition, *Federal Standard 1037C: Glossary of Telecommunications Terms* . `http://glossary.its.bldrdoc.gov/fs-1037/dir-037/_5538.htm`.

Balaban, P.: 1984, Computer-Aided Modeling, Analysis, and Design of Communication Systems: Introduction and Issue Overview, *IEEE Journal on Selected Areas in Communications* **2**(1), 1–7.

Balaji, A., Gonsalves, T. A. and Kumar, R.: 2000, The Performance of corDECT Voice/Data Wireless Local Loop, *IEEE IETE Journal of Research* **46**(6), 489–504.

Bates, R. J.: 2002, *GPRS (General Packet Radio Service)*, McGraw-Hill, New York, USA.

Bettstetter, C., Vögel, H. J. and Eberspächer, J.: 1999, GSM Phase 2+ General Packet Radio Service GPRS: Architecture, Protocols, and Air Interface, *IEEE Communications Surveys* **2**(3), 2–14.

Bi, Q., Zysman, G. I. and Menkes, H.: 2001, Wireless Mobile Communications at the Start of the 21st Century, pp. 110–116.

Blakely, K., Gregson, S. and Mulvey, M.: 2000, Unified IP Networks, *BT Technology Journal* **18**(2), 44–56.

Blaunstein, N.: 2000, *Radio Propagation in Cellular Networks*, Artech House, Massachusetts, USA.

Brand, A. and Aghvami, H.: 2002, *Multiple Access Protocols for Mobile Communications GPRS, UMTS and Beyond*, John Wiley and Sons, New York, USA.

Brasche, G. and Walke, B. H.: 1997, Concepts, Services, and Protocols of the New GSM Phase 2+ General Packet Radio Service, *IEEE Communications Magazine* **35**(8), 94–104.

C. Smith, D. C.: 2002, *3G Wireless Networks*, McGraw-Hill, New York, USA.

Chang, M. F., Lin, Y. B. and Pang, A. C.: 2003, vGPRS: A Mechanism for Voice over GPRS, *Wireless Networks* **9**(2), 157–164.

Chaskar, H. M. and Madhow, U.: 2001, Statistical Multiplexing and QoS Provisioing for Real-Time Traffic on Wireless Downlinks, *IEEE Journal on Selected Areas in Communications* **19**(2), 347–354.

Chitamu, P. J.: 2003, private communication.

Chitamu, P. J., Kennedy, I. G. and van Olst, R.: 2002, Technology Options to support Telecommunications in Countries with Low Tele-densities, *Proceedings of the ISCIT 2002: The International Symposium on Communication and Information Technology*, Pattaya, Chonburi, Thailand, pp. 289–292.

Chitamu, P. J. and Vannucci, D. E.: 2002, Networking Africa: A Case for VoIP, *Proceedings of the IEEE Africon 2002 Conference*, Vol. 1, George, South Africa, pp. 361–361.

Cisco Systems: 2002, Traffic Analysis for Voice over IP, *Cisco White Papers* . `http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/ta_isd.pdf`.

Eriksson, G. A. P., Olin, B., Svanbro, K. and Turina, D.: 2000, The challenges of voice-over-IP-over-wireless, *Ericsson Review* (1), 20–31.

ETSI: 1998, Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Description of technical issues, *Technical Report TR 101 300 v1.1.5*, European Telecommunications Standards Institute.

Fabri, S. N., Worrall, S. T. and Kondoz, A. M.: 2000, Video Communications over Mobile Networks, *Proceedings of Communicate 2000, Online conference*, London, United Kingdom.

Gimpleson, L. A.: 1965, Analysis of Mixtures of Wide- and Narrow-Band Traffic, *IEEE Transactions on Communication Technology* **13**(3), 258–266.

Granberg, O.: 1998, GSM on the Net, *Ericsson Review* (4).

Granbohm, H. and Wiklund, J.: 1999, GPRS–General packet radio service, *Ericsson Review* (2), 82–88.

GSM-Association: 2003, Membership and Market Statistics as at end of May 2003. http://www.gsmworld.com/news/statistics/stats_03.pdf.

Habib, I. W. and Saadawi, T. N.: 1992, Multimedia Traffic Characteristics in Broadband Networks, *IEEE Communications Magazine* **30**(7), 48–54.

Hafner, B.: 2003, IP Telephony Can Revolutionise Business Communications, *Research Note SPA-20-1210*, Gartner.

Halonen, T., Romero, J. and Melero, J. (eds): 2002, *GSM, GPRS and EDGE Performance: Evolution Towards 3G/UMTS*, Wiley.

Hassan, M., Nayandoro, A. and Atiquzzaman, M.: 2000, Internet Telephony: Services, Technical Challenges, and Products, *IEEE Communications magazine* **38**(4), 96–103.

Higginbottom, G.: 1998, *Performace Evaluation of Communication Networks*, Artech House, Massachusetts, USA.

Hlavacs, H., Kotis, G. and Steinkeller, C.: 1999, Traffic Source Modeling, *Technical Report TR-99101*, Institute of Applied Computer Science and Information Systems, University of Vienna.

Hodgkinson, W. T.: 2002, private communication.

ITU: 2001, Africa Reaches Historic Telecom Milestone: The unwiring of a continent, *Indicators update*, ITU Telecommunication Indicators Update. http://www.itu.int/ITU-D/ict/update/pdf/Update_3_01.pdf.pdf.

Jiang, W. and Schulzrinne, H.: 2000, Analysis of On-Off Patterns in VOIP and Their Effect on Voice Traffic Aggregation, *Proceedings of The 9th IEEE International Conference on Computer Communication Networks*, Las Vegas, Nevada, USA.

Koodli, R. and Puuskari, M.: 2001, Supporting Packet-Data QoS in Next-Generation Cellular Networks, *IEEE Communications Magazine* **39**(2), 180–188.

Law, A. M. and Kelton, W. D.: 1982, *Simulation Modeling and Analysis*, first edn, McGraw-Hill, New York, USA.

Lindemann, C. and Thümmler, A.: 2003, Performance analysis of the general packet radio service, *Proceedings of Computer Networks*, number 41, pp. 1–17.

Luke, S.: 2000, Mersenne Twister and MersenneTwisterFast. http://www.math.keio.ac.jp/~matumoto/emt.html.

Mahdavi, M., Edwards, R. M. and Cvetkovic, S. R.: 2001, Policy for Enhancement of Traffic in TDMA Hybrid Switched Integrated Voice/Data Cellular Mobile Communications Systems, *IEEE Communications Letters* **5**(6), 242–244.

Mahdavi, M., Edwards, R. M. and Ivey, P. A.: 2001, Performance Evaluation of a Data Subsystem in GSM/GPRS Using Complete Sharing of Available Bandwidth, *Proceedings of London Communications Symposium*.

Matsumoto, M. and Nishimura, T.: 1998, Mersenne Twister: A 623-dimensionally equidistributed uniform pseudorandom number generator, *ACM Trans. on Modeling and Computer Simulation* **8**(1), 3–30.

Mitrou, N. M., Lyberopoulos, G. L. and Panagopoulou, A. D.: 1993, Voice and Data Integration in the Air-Interface of a Microcellular Mobile Communications System, *IEEE Transactions on Vehicular Technology* **42**(1), 1–13.

Miyake, K.: 1988, Traffic Study on High Speed/Multiplexed ISDN User-Network Interface, *Proceedings of the Fifth ITC Seminar :Traffic Engineering for ISDN Design and Planning*, Vol. 9, Lake Como, Italy, pp. 487–490.

Moorhead, C., Hardman, V. J. and Lane, P. M.: 2000, Resource Allocation Schemes to Provide QoS for VoIP over GPRS, *Proceedings of the 1st IEE International Conference on 3G Mobile Communication Technologies*, London, United Kingdom, pp. 138–142.

Naylor, T. H. and Finger, J. M.: 1967, Verification of Computer Simulation Models, *Management Science* **14**(2), 92–101.

Neskovic, A., Neskovic, N. and Paunovic, G.: 2000, Modern Approaches in Modelling of Mobile Radio Systems Propagation Environment, *IEEE Communications Surveys* pp. 2–12.

Network Sorcery: 2003, RTP, Real-Time Protocol. `http://www.networksorcery.com/enp/protocol/rtp.htm`.

Okuda, T., Ando, Y., Ideguchi, T. and Tian, X.: 2002, A Simplified Performance Evaluation for Delay of Voice End-User in TDMA Integrated Voice/Data Cellular Mobile Communications Systems, *Proceedings of IEEE GLOBECOM 2002: The World Converges*, Taipei, Taiwan, R.O.C.

Rao, H. C. H., Lin, Y. B. and Cho, S. L.: 2000, iGSM: VoIP Service for Mobile Networks, *IEEE Communications Magazine* **4**(38), 62–69.

Rappaport, T. S.: 2002, *Wireless Communications: Principles and Practice*, second edn, Prentice Hall, New Jersey, USA.

Sargent, R. G.: 1994, Verification and Validation of Simulation Models, *Proceedings of the 26th conference on Winter simulation*, Orlando, Florida, United States, pp. 77–87.

Sarker, J. H. and Halme, S. J.: 2000, An Analysis of Channel Efficiency With(out) Voice and Data Traffic Activity Factors, *Proceedings of The Twelfth Annual International Conference on Wireless Communications, Wireless 2000*, Calgary, Alberta, Canada, pp. 618–626.

Schlesinger, S., Crosbie, R. E., Gagné, R. E., Innis, G. S., Lalwani, C. S., Loch, J., Wright, R. D., Kheir, N. and Bartos, D.: 1979, Terminology for model credibility, *Simulation* **32**(3), 103–104.

Schulzrinne, H.: 1999, Service for telecom: Version II, *IEEE Internet Computing* **3**(3), 40–43.

Schwartz, M.: 1987, *Telecommunication Networks: Protocols, Modeling and Analysis*, Addison-Wesley, Massachusetts, USA.

Singh, T.: 2003, Things you can do to speed up things, *Technical report*. `http://www10.brinkster.com/geniusportal/speedjava.html`.

So, J. W. and Cho, D. H.: 2001, Access Scheme for Integrated Voice/Data Transmissions over Common Packet Channel in 3GPP, *IEEE Communication Letters* **5**(2).

Stallings, W.: 1987, *Local Networks*, third edn, Macmillian Publishing Company, New York, USA.

Sun Microsystems: 2002, Java 2 Platform Standard Edition JDK API Documentation, *Technical Report v1.4.1*, Sun Microsystems.

Tabbane, S.: 2000, *Handbook of Mobile Radio Networks*, Artech House, Massachusetts, USA.

Jr. Andrews, F. T.: 1984, ISDN '83, *IEEE Communications Magazine* **22**(1), 6–10.

Turner, J. S.: 1986, New Directions in Communications (or Which Way to the Information Age?), *Proceedings of Digital Communications: New Directions in Switching and Networks*, pp. 25–31.

Vannucci, D. E. and Chitamu, P. J.: 2003a, Efficient Radio Resource Allocation in a GSM and GPRS Cellular Network, *Proceedings of the 7th World Multiconference on Systemics, Cybernetics and Informatics*, Florida, USA.

Vannucci, D. E. and Chitamu, P. J.: 2003b, Investigation of Radio Resource Allocation for GSM/GPRS, *Proceedings of the 2003 International Conference on Electrical Engineering and Technology*, Dar es Salaam, Tanzania.

Vannucci, D. E. and Chitamu, P. J.: 2003c, Radio Resource Allocation and Multiplexing of Voice and Data over IP in a GSM/GPRS Cellular Network, *Proceedings of Southern African Telecommunication Networks and Applications Conference (SATNAC): IP Generation leading the charge for convergence*, George, South Africa.

Varshney, U., Snow, A., McGivern, M. and Howard, C.: 2002, Voice over IP, *Communications of the ACM* **45**(1), 89–96.

Walke, B. H.: 2002, *Mobile Radio Networks - Networking, Protocols and Traffic Performance*, second edn, John Wiley and Sons, New York, USA.

Wang, J., McCann, P. J., Gorrepati, P. B. and Liu, C. Z.: 1998, Wireless Voice-over-IP and Implications for Third-Generation Network Design, *Bell Labs Technical Journal* **3**(3), 79–97.

Watson, A. and Sasse, M.: 1997, Multimedia Conferencing via Multicast: Determining the Quality of Service Required by the End User, *Proceedings of International Workshop on Audio-Visual Services over Packet Networks AVSPN '97*, Aberdeen, Scotland, pp. 189–194.

Winogradoff, A.: 2002, VoIP in Public Services: Facts, Fiction and Myth, *Article Top View AV-18-6958*, Gartner.

Wong, W. T. K. and Mack, R. M.: 1996, Low rate speech coding for telecommunications, *BT Technology Journal* **14**(1), 28–44.

Yang, W. and Yantorno, R.: 1996, Development of a burst model for speech transmission with cell losses in an ATM network, *Proceedings of the 1996 1st Annual Conference on Emerging Technologies and Applications in Communications, etaCOM*, pp. 20–23.

Zeigler, B. P.: 76, *Theory of Modelling and Simulation*, John Wiley & Sons, Massachusetts, USA.

Appendix

8.4 Calculating the Erlang Formula in Microsoft Excel

The Erlang B and Erlang C values can be calculated in Microsoft Excel using the `Poisson` function. The Erlang B formula is

```
=+(POISSON(<circuits>,<traffic>,FALSE)/EXP(-<traffic>))/
  (POISSON(<circuits>,<traffic>,TRUE )/EXP(-<traffic>))
```

and the Erlang C formula is

```
=+POISSON(<circuits>,<traffic>,FALSE)/
 (POISSON(<circuits>,<traffic>,FALSE)+
 (1-<traffic>/<circuits>)*
 POISSON(<circuits>,<traffic>,TRUE ))
```

8.5 Structure of the Compact Disc

- Masters CD
  - Dissertation
    - Misc
      - CODvsFixedCH
      - DelayVsLoad
      - ExponentialVoIPandData
      - Insufficient Capacity
      - Simulator Verification
      - statespace
      - VoIPandData
  - Matlab Code
    - Dissertation Matlab Code
    - Initial Matlab Code
    - Multi-channel call concentrator
    - Old Simulation Runs
  - Simulator
    - GSMsim
      - bak
      - gsmsim
    - classes
      - gsmsim
      - package cache
    - doc
      - gsmsim
    - src
      - gsmsim
    - sym
      - gsmsim

Figure 8.1: Structure of the Compact Disc