Call Admission Control and Resource Utilization in WCDMA Networks

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Abstract

Unlike FDMA or TDMA systems, CDMA is interference limited and has a soft capcity that changes depending on the interference felt at the base station at a given time. Admitting a new call and user movement increases the interference level in the system. Therefore a robust Call Admission and Power Control Mechanism is needed.

This thesis discusses the main approaches mentioned in the literature on Call Admission Control and Power Control and analyses two modern solutions, namely the QoS aware Power Control and Handoff Prioritization scheme introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] and the Received Power based simulation model discussed in [A. Capone and S. Redana 2001], in greater detail. Then we proceed to recommend improvements that are then tested in a MATLAB simulation environment. The recommended changes improve the overall dropping and handoff loss probabilities. The impact of the NRT overload mechanism discussed in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] is also investigated. The investigations determined the optimum solution achievable with the NRT overload parameter settings.

As the final task, a discrete time dynamic feedback control system that aims to keep the dropping and handoff loss rates for RT services below a target value regardless of the traffic dynamics or the bandwidth requirements is designed. A simple Integral Feedback controller is chosen for this task because a controller that is capable of reducing steady state error is required. The controller is used for the NRT overload mechanism while the NRT error rate is left as best effort. The controller parameters are tuned using simulations and the final result is benchmarked against two algorithms that have fixed NRT overload parameters by simulating in environments under various Poisson call arrival rates and traffic loads. The NRT overload mechanism with our controller performed best by holding the RT error rate at the required target value while producing comparatively lower NRT error rates.

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Chapter 1. Introduction

1.1 Background and Motivation

The rapid expansion of the mobile market over the past few years has seen cellular communications move away from just voice to a host of multimedia services. Users are now demanding their handsets to be packed with more features while at the same time being lighter and more power efficient [Englewood Cliffs 1998]. As a result third generation (3G) Wideband Code Division Multiple Access (W-CDMA) mobile communications are gearing up to deliver the kind of flexible services wanted.

Much research has been conducted in this field by research groups around the world. CDMA supports variable bit rates and hence is the ideal mode of communication for future cellular networks. To support various integrated services with a certain quality of service (QoS) requirement in these wireless networks, resource provisioning is a major issue [Grillo, Skoog, Chia, and Leung 1998], [Hong and Rappaport 1986]. The Universal Mobile Telecommunication Systems (UMTS) supports QoS provisioning through four basic classes of service [ETSI 23.107 v5.9.0 (2003 – 2006)] and [ETSI 25.401 v6.3.0 (2004 – 2006)]:

- Class 1: Conversational (high sensitivity to delay and jitter).
- Class 2: Streaming (medium sensitivity to delay and high sensitivity to jitter).
- Class 3: Interactive (low sensitivity to delay, high sensitivity to round trip delay time and Bit Error Rate (BER)).
- Class 4: Background (no delay sensitivity, high sensitivity to BER).

Call admission control (CAC) is a provisioning strategy to limit the number of connections into the networks in order to reduce the network congestion and call dropping. In previous generation networks such as AMPS, GSM, GPRS, the decision of accepting a new call was a relatively easy one, since the available number of channels in a cell is known. CDMA on the other hand is interference limited and the number of calls cannot specify the capacity of the system. A user will be granted access to the network only if this action will not cause the other users to experience a drop in quality or affect system stability. In wireless networks, another dimension is added. Call dropping is possible due to the users' mobility. A good CAC scheme has to balance call blocking and call dropping in order to provide the desired QoS requirements.

The goal of this project was to devise an interactive call admission control algorithm which would minimize dropping and handoff losses while maintaining a high server usage.

1.2 Research Objectives

The main objectives of this thesis are:

- Survey causes of unsuccessful Call Admission and Resource Allocation over WCDMA networks. In particular issues of soft capacity and service prioritization.

- Survey some of the improvements to Call Admission and Resource Allocation mechanisms proposed in the literature.

- Investigate the performance of modern schemes and identify areas for improvement.

- Introduce and develop improvements to the existing schemes to further reduce loss rates and further prioritize services according to user requests.

- Investigate through simulations the performance achieved by the suggested improvements.

- Investigate possible applications of Control Theory in Resource Reservation mechanisms.

- Discuss the strengths and weaknesses of the proposed controller mechanism.

1.3 Thesis Overview

This section provides an overview of the thesis structure and briefly discusses the main points of each chapter.

Chapter 2 presents an overview of the WCDMA network and some of the issues related to call admission and resource management schemes. This leads to a discussion on the main aspects that need attention when attempting to modify the call admission and resource allocation schemes. Important issues such as power control, soft handoff, SIR and causes of congestion are treated in detail.

In Chapter 3, the main approaches mentioned in the literature on Call Admission Control are presented. The main advantages and drawbacks of the schemes are also highlighted. We discuss in detail the two Call Admission and Resource Managements schemes that are of greatest interest to us. Here we discuss in detail the propagation and power control models that will be used in our study and we also introduce the reader to Quality of Service Parameters such as the Subscriber Degrade Descriptor and mechanisms such as the NRT Overload Mechanism.

In Chapter 4 we implement ideas discussed in Chapter 3 (particularly our simulation environment) before discussing their strengths and weaknesses and identifying possibilities for a better solution through this thesis.

In Chapter 5 we introduce our extensions to the existing algorithms discussed in previous chapters. We identify the parameters that can be altered and discuss methods of further reducing the rate of dropping calls after admission. We aim to maintain the strengths of the existing algorithms and obtain an overall better result with our extensions.

In Chapter 6, we describe classical feedback control theory to adaptively control the NRT overload mechanism in the Call Admission and Power Control algorithm. We introduce a discrete time dynamic feedback control system that aims to keep the dropping and handoff loss rates for RT services below a target value regardless of the traffic dynamics or the bandwidth requirements. We investigate the performance of the controller through simulations.

Chapter 7 concludes the thesis by highlighting the main findings made throughout it and the strengths and weaknesses of the proposals made, based on the analysis of the simulation results. Suggestions for future work to further develop the methods studied in this thesis are also presented in this chapter. It is hoped that this discussion will provide directions to the continuation of the work in this thesis based on the lessons learned. In particular this thesis only investigates improvements in the downlink direction of WCDMA transmission. Similar improvements are plausible and should be considered for the uplink in the future.

Chapter 2. WCDMA and Call Admission Issues

Cellular networks have developed at an astounding speed during the past few decades. The cellular concept arose from the need to share the spectrum which is a limited resource in mobile communication. Before discussing the actual design of the project it is useful to understand some background on the subject.

2.1 Second Generation (2G) Networks

Second generation of cellular networking solutions began to emerge in the mid-1980s when cellular providers foresaw a need for additional capacity. Once digitized, the human voice can be modeled and encoded using mathematical algorithms. These algorithms effectively compress the amount of digital data needed in a voice transmission and open the door for more efficient spectrum utilization. Besides increasing capacity, providers were able to reap the benefits of a wide array of revenue generating features like caller ID and short message service with the implementation of Time Division Multiple Access Technology (TDMA) and more recently Code Division Multiple Access (CDMA) cellular [Prasad, Mohr and Hauser 2000].

2.2 Third Generation (3G) Networks

Universal Mobile Telecommunications System (UMTS) provides broadband, packet-based transmission of text, digitized voice, video, and multimedia at data rates up to and possibly higher than 2 megabits per second (Mbps), offering a consistent set of services to mobile computer and phone users no matter where they are located in the world. Based on the Global System for Mobile Communications (GSM) standard, UMTS is endorsed by major standards bodies and manufacturers and is the planned standard for mobile users around the world. Once UMTS is fully implemented, computer and phone users can be constantly attached to the Internet as they travel and, with roaming service, have the same set of capabilities no matter where they travel. Wideband code division multiple access (WCDMA) is the standard to be used for multiple access in these networks.

2.3 CDMA verses Other Multiple Access Techniques

Frequency Division Multiple Access (FDMA) – Figure 2.3-1 (a) allocates a single channel to one user at a time. Essentially, FDMA splits the allocated spectrum into many channels. When a FDMA cell phone establishes a call, it reserves the frequency channel for the entire duration of the call.

TDMA – Figure 2.3-1 (b) is a digital transmission technology that allows a number of users to access a single radio-frequency (RF) channel without interference by allocating unique time slots to each user within each channel. TDMA builds on FDMA by dividing conversations by frequency and time. GSM fits eight digital conversations into an FDMA channel.

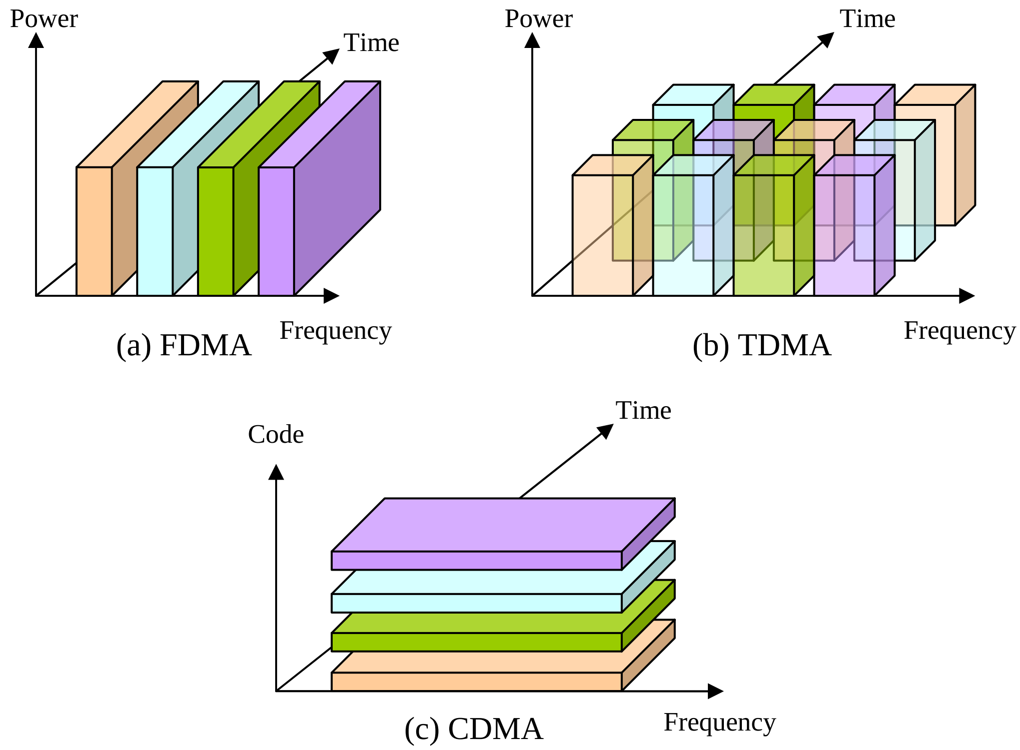


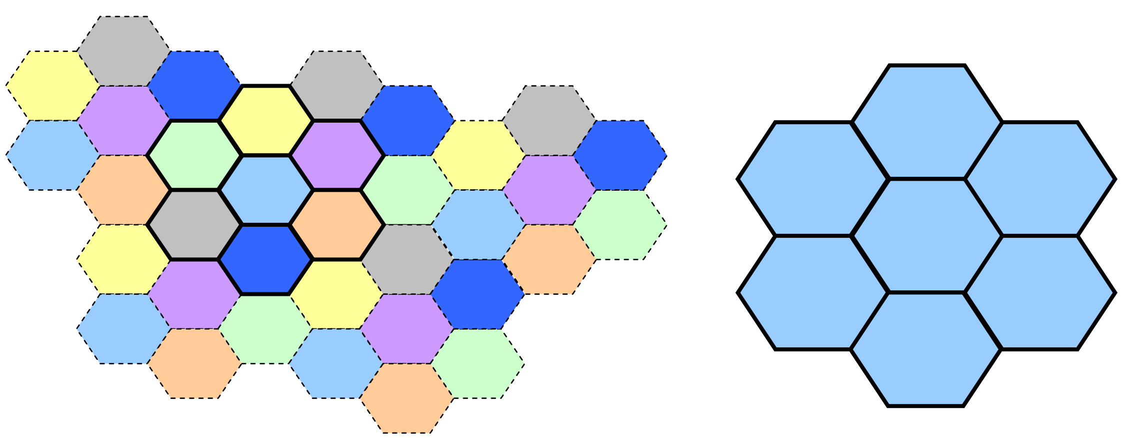
Figure 2.3-1: Bandwidth usage of different systems, spectrum and time. A colored block represents a piece of bandwidth.

CDMA - Figure 2.3-1 (c) systems encode each call as a coded sequence across the entire frequency spectrum. Each conversation is modulated, with a unique code (called a pseudo-noise code) that makes it distinguishable from the other calls. From the perspective of one call, upon extracting the signal, everything else appears to be low-level noise. As long as there is sufficient separation between the codes (said to be mutually orthogonal), the noise level will be low enough to recover the digital signal. Each signal is not, in fact, spread across the whole spectrum but is spread across 1.25 MHz "pass-bands" [Keith W. Ross 1995]. Since CDMA offers far greater capacity and variable data rates depending on the audio activity, many more users can be fit into a given frequency spectrum and higher audio quality can be provided. The current CDMA systems boast at least three times the capacity of TDMA systems.

2.3.1 Frequency Reuse

Frequency reuse is a measure of how often the same frequency spectrum can be used in neighbouring cells. A TDMA system (figure 2.3.1-1 (a)) uses a typical reuse pattern know as 7 cell reuse. Cells of the same color share the same frequency band. The further away the nearest cell with the same frequency band the better in terms of interference.

On the other hand in a CDMA system all cells share the same frequency band as shown in figure 2.3.1-1 (b). This means that any transmission in a neighbouring cell can be received by a mobile host or by the base station of the home cell, assuming it is strong enough to be heard.



(a) TDMA with 7 cell reuse (b) CDMA

Figure 2.3.1-1: Frequency Reuse. Same color represents the same frequency spectrum.

2.3.2 Signal to Interference Ratio

SIR is similar to the quantity known as Signal to Noise Ratio (SNR) in communications and signal processing applications. It is defined as in Equation 2.3.2-1 below.

$$SIR = \frac{SignalPower}{Total\ \ Interference\ Power} \qquad \textbf{Equation 2.3.2-1}$$

Figure 2.3.2-1 shows the two types of interference which can occur in a CDMA system. A frequency division duplex (FDD) link is used to communicate between the mobile terminal and the base station. All uplink connections are on one frequency band and the downlink connections on another. Figure 2.3.2-1 (a) highlights interference caused by the uplink channels of mobiles in the vicinity and figure 2.3.2-1 (b) shows downlink interference to a mobile caused by neighboring base stations. This type of interference is unique to a CDMA system due to the frequency reuse issues discussed above.

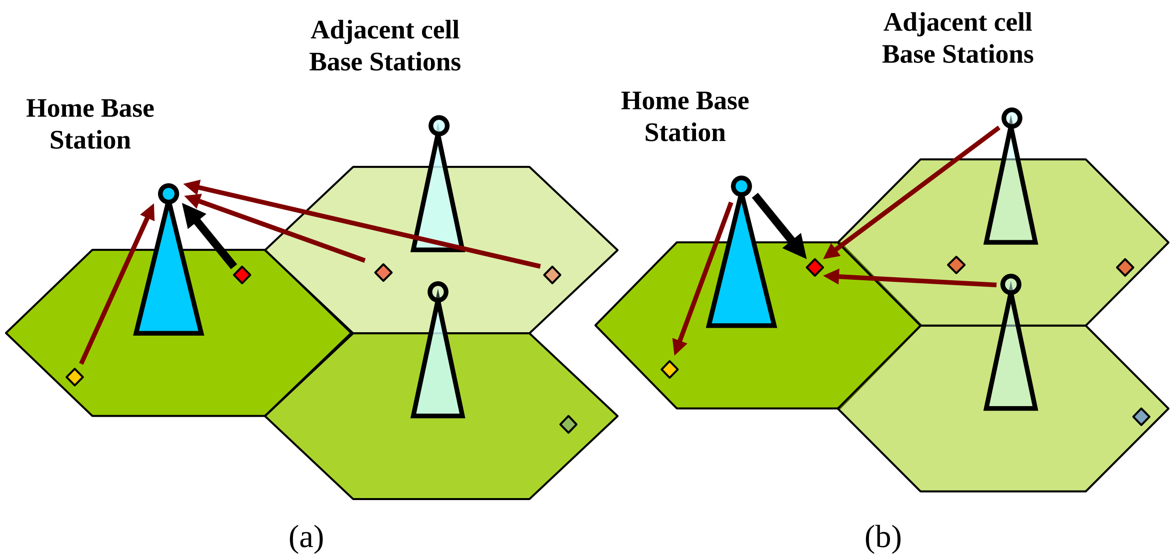


Figure 2.3.2-1: Interference experienced by a base station and a mobile host.

The *SIR* Equation 2.3.2-1 can be expanded as follows – Equation 2.3.2-2

$$SIR = SF.\frac{P_r}{I_{intra} + I_{inter} + P_N} \qquad \textbf{Equation 2.3.2-2}$$

where,

- P_r is the received signal strength.
- P_N is the thermal noise power assumed equal to -99dBm in downlink, and -103dBm in uplink.
- I_{inter} is the sum of signal powers received from other cells.
- I_{intra} is the sum of signal powers due to other transmissions within the same cell.
- SF is the spreading factor for a given call type.

The Spreading Factor (*SF*) or processing gain is defined as,

$$SF = \frac{Bandwidth}{Information\ Rate} = \frac{ChipRate}{Bitrate} \qquad \textbf{Equation 2.3.2-3}$$

It can be seen from the above equation that the *SF* is inversely proportional to the information rate (bit-rate) of the call for a given system bandwidth. Calls with a lower information rate have a better *SIR* which implies that services with higher capacity requirements need more power to maintain a given *SIR*.

2.4 WCDMA Networks

The WCDMA system goes a step further in CDMA technology by using a 5MHz wide radio signal and a chip rate of 3.84Mcps, which is about three times higher than the chip rate of CDMA2000 (1.22Mcps) [Ericsson Radio System 2001]. The main benefits of this wideband carrier with a higher chip rate are:

- Support for higher bit rates.
- Higher spectrum efficiency.
- Higher Quality of Service (QoS).

WCDMA system and the GSM system have many similarities because WCDMA Radio Access Network (RAN) and the GSM Base Station Subsystem (BSS) are both connected to the GSM Core Network [Ericsson Radio System 2001] (figure 2.4-1). Plus both GSM BSS and WCDMA RAN systems are based on the principles of a cellular radio system. The GSM Base Station Controller (BSC) corresponds to the WCDMA Radio Network

Controller (RNC). Examples of WCDMA-specific functions are fast power control and soft handoff.

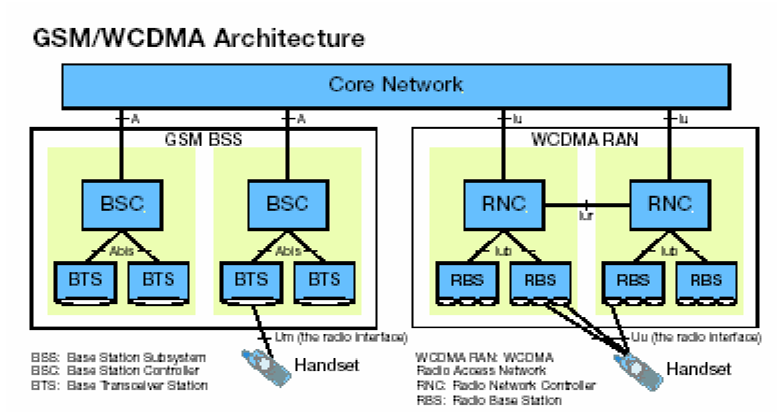


Figure 2.4-1: GSM/WCDMA Architecture

2.4.1 Power Control in WCDMA

The power control regulates the transmit power of the terminal and base station, which results in less interference and allows more users. WCDMA employs fast closed-loop Power Control [E. Dahlman and P. Bening]. SIR-based power control is used where the receiver compares the estimated received *SIR* with a *SIR* target value and commands the transmitter to increase or decrease power accordingly.

The target *SIR* values are controlled by an outer power-control loop. This outer loop measures the link quality, typically a combination of frame and bit error rates (BER's) depending on the service and adjusts the *SIR* targets accordingly. Ensuring the lowest possible *SIR* target is used at all times results in maximum capacity.

2.4.2 Soft Handoff

With the soft handoff functionality the handset can communicate simultaneously with two or more cells in two or more base stations. This flexibility in keeping a connection open across base stations results in fewer lost calls. Soft handoff enables the handset to maintain the continuity and the quality of the connection while moving from one cell to another. During Soft handoff the handset momentarily adjusts its power to the base station that requires the smallest amount of transmits power and the preferred cell may change very quickly. Soft handoff is illustrated in figure 2.4.2-1 [Ericsson Radio System 2001]. In a well designed radio network 30% to 40% users are regularly in soft handoff.

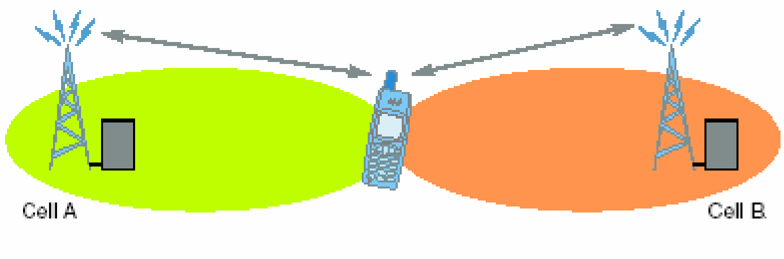


Figure 2.4.2-1: Soft Handoff in Mobile networks.

2.4.3 Admission Control

Capacity esitimation in CDMA systems is an important issue which is closely related to traffic charcteristics, power control, radio propagation, sectorisation and other factors. Unlike in an FDMA or TDMA system the number of users in the system does not have a fixed upper bound as no “channels” are present. Instead the system is interference limited and has a soft capcity which changes depending on the interference felt at the base station at a given time. If interference increases beyond an acceptable level the system becomes unstable and may lead to call dropping [J. Knutsson, P. Butovitsch, M. Persson and R.D Yates 1997].

Admitting a new call always increases the interference level in the system. Hence a robust method of accepting or blocking potential users, i.e. a Call Admission Control (CAC) technique, is required. The basic strategy under heavy congestion is to protect ongoing calls by denying a new user access to the system because dropping an ongoing call is considered to be far worse than blocking a new call. But Handoff calls are also a very important element of mobile communications networks and a mechanism is needed to maintain connectivity during the call as the user migrates between cells. From the user perspective, a connection terminated in the middle of a call because of a handoff failure is equivalent to a dropped call.

Admission Control is required in both the uplink and the downlink in order to provide different services. Different services also demand different capacity as well as different qualities of service. Therefore service dependent Admission Control algorithms are required. These service dependent threshholds should depend on load estimates, for instance the received power level as an uplink estimate and the total downlink power from a base station as a downlink estimate [E. Dahlman and P. Bening 1998]. The measured values are obtained from the base station where the Admission Control algorithms are implemented and admission decisions are made.

2.4.4 Congestion Control

Even with efficient Admission Control, congestion could still be caused, mainly by users moving from one area within the cell to another area. When affected by congestion the output powers are rapidly increased by the fast closed-loop power control until one or several transmitters are operating at their maximum power. The connections unable to achieve their required *SIR* levels are considered to be useless and are only adding interference to the system. Therefore a procedure to reduce congestion by removing such users is required. Congestion can be controlled with several methods.

- Lowering the bit-rate of one or several services that are insensitive to increased delays.
- Performing inter frequency handoff.
- Removing useless connections.

The same measurements used for Admission Control can be used for Congestion Control [E. Dahlman and P. Bening 1998]. But these measurements have to be updated regularly because the considered values change rapidly during heavy congestion. Once the connections to alter are identified Congestion Control schemes can be utilized.

2.5 Summary

An overview of aspects relevant to call admission and resource management of WCDMA networks was presented in this chapter. Important issues such as power control, soft handoff, *SIR* and causes of congestion were detailed.

It was discussed that the CDMA systems do not have a hard limit when it comes to capacity but instead has a soft capacity. Interference caused by users and power availability in the system were singled out as major capacity limiters and the significance of appropriate Call Admission Control was emphasized.

User movement from one area to another area (mobility) within a cell was discussed as the second biggest cause of congestion. Several common methods of congestion control were introduced and the need for a robust congestion control scheme was emphasized.

Chapter 3. Survey of Call Admission Algorithms for WCDMA Networks

3.1 Introduction

Now that the main issues concerning WCDMA and its Call Admission considerations have been discussed, a survey of existing call admission algorithms is presented. Some remarks about the strengths and weaknesses of these proposals, based on experimental research reported in the literature, are also presented.

This chapter is a basis for the development and evaluation of the modifications proposed in this thesis. The Call Admission models and concepts directly relevant to our research are discussed in greater detail.

3.2 Different Call Admission Schemes

For the purposes of review the Admission schemes in existence can be classified as either Interactive or Non-Interactive schemes.

3.2.1 Interactive Call Admission Schemes

Ideally a Call Admission scheme accepts a new call only if the closed-loop Power Control mechanism is able to reach a new equilibrium where all connections observe a target *SIR* to ensure good quality. Interactive Call Admission scheme behavior is very close to ideal Admission Control because it allows new connections to transmit for a trial period during which it takes measurements to determine weather the connection can be tolerated.

Unfortunately the procedure required for such a scheme is too complex considering that during the trial period the scheme must ensure the new call does not affect the quality of the ongoing calls. Also taking measurements and making decisions with Interactive Admission schemes can be very time consuming. The other drawback is its inability to work with inactive connections. Interactive schemes can only work with always active connections and cannot exploit discontinuous transmission, which is very important in UMTS. [M. Andersin, Z. Rosberg, and Zens Zander 1997] and [D. Kim 2000] are examples of Interactive Call Admission Schemes and provide further detail on the subject.

3.2.2 Non-Interactive Call Admission Schemes

Unlike Interactive schemes, Non-Interactive schemes only estimate the network load by measuring a few system parameters. The decisions on call admission are based on the estimates. The total interference measured at the base station is generally considered a good load index since the ability of the power control mechanism to keep *SIR* at the target level depends on the interference level.

The measured interference includes both intra-cellular and inter-cellular interference. Therefore the admission decision can be based on interference experienced in the cell of the base station as well as in neighboring cells. The measured values are compared with a threshold and is only accepted if the threshold is not exceeded.

The acceptance thresholds are tuned to limit the dropping probability. The simple Receive Power based Admission Control schemes do not consider the additional load due to the new call. The threshold tuning must take into account that the load increase is highly varying depending on mobile terminal position and propagation conditions towards its Base Station and others. The acceptance threshold must be kept low in order to tolerate the worst possible scenarios and to minimize dropping probability. As a result the acceptance probability will be much lower in Non-interactive schemes than those of near ideal schemes. Examples are given in [C. Y Huang and R.D. Yates 1996], [J. Knutsson, P. Butovitsch, M. Persson and R.D Yates 1997] and [J. Knutsson, P. Butovitsch, M. Persson and R.D Yates 1998].

3.2.3 Predictive Receive Power Based Call Admission Control

The safety margin of admission thresholds used by Non-Interactive schemes can be reduced by using a scheme to estimate the additional interference due to the new call. Predictive schemes discriminate between calls requested by mobile terminals with different propagation conditions. This approach produces a non-uniform accepted traffic distribution where terminals close to the base station are more likely to be accepted. This is similar to ideal call admission, which exploits this effect to increase accepted traffic. Interference increase estimation algorithms are given in [H. Homa, J.Laakso 1999] and [J. Outes, L. Nielson, K. Pederson, P. Morgensen 2001].

3.2.4 Handoff Prioritization

Provisioning QoS over WCDMA cannot be fulfilled with just proper Admission Control and efficient scheduling [W. K. Wong, H. Zhu and V. C. M Leung 2003]. This is due to mobility and fading channels [T. S. Rappaport 2001], high error rates [Ericsson Radio System 2001], low and varying bandwidth, but mainly due to the unexpected handoff requests [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004].

Most issues mentioned above have been catered for using closed-loop power control mechanisms that operate solely on the basis of channel gain, but that are not aware of QoS requirements of underlying connections. This blind mode of operation does not necessarily yield optimal power utilization, especially when other non-premium connections in the system are willing to be degraded, that is they are capable of adaptation and willing to have their required bit-rate/power reduced. The issue of unexpected handoffs has been solved using reservation and prediction schemes [W. Soh and H. S. Kim 2003].

3.3 QoS aware Power Control and Handoff Prioritization for WCDMA

In this thesis research the QoS aware Power Control and Handoff Prioritization scheme introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] was studied and analyzed extensively. This scheme shows that user willingness to be degraded can be used to augment both traditional closed loop control mechanisms for congestion handling, as well as, to improve handoff by reducing the rate of blocking handoff requests. This section has been organized to describe the scheme in greater detail.

3.3.1 Subscriber Degrade Descriptor (SDD)

Subscriber Degrade Descriptor (SDD) is presented in [O. Lataoui, T. Rachidi and L. G. Samuel 2000] as a Lucent patented framework for modeling user willingness to be degraded. SDD is a number between 0 - 5 and the larger the SDD is the more willing a user is to be degraded and to eventually be dropped.

[T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] uses SDD together with service classes and the bit-rates as enabling QoS parameters in systems capable of minimizing handoff call blocking and new call blocking.

3.3.2 NRT Overload Admission Strategy

In the traditional strict admission strategy a new call is accepted in the system at an instant only if the power required by all users do not exceed the total power available and if the QoS requirements of the other users are not lowered.

But in the NRT overload admission strategy the base station is allowed to accept connections even if the total power required by all users exceed the available power. Here the NRT connections are backed off and delayed by a scheduler. Specifically, a new connection is accepted in the system at instant t if and only if:

$$\sum P_{i/RT}(t) \le P_{\max} \qquad \textbf{Equation 3.3.2-1}$$

Where $P_{i/RT}(t)$ is the power required by existing real-time connection *i*, (that is class 1 and 2 connections in the system including eventually the new connection).

and

$$\sum P_i(t) \le (1+\alpha)P_{\max} \qquad \textbf{Equation 3.3.2-2}$$

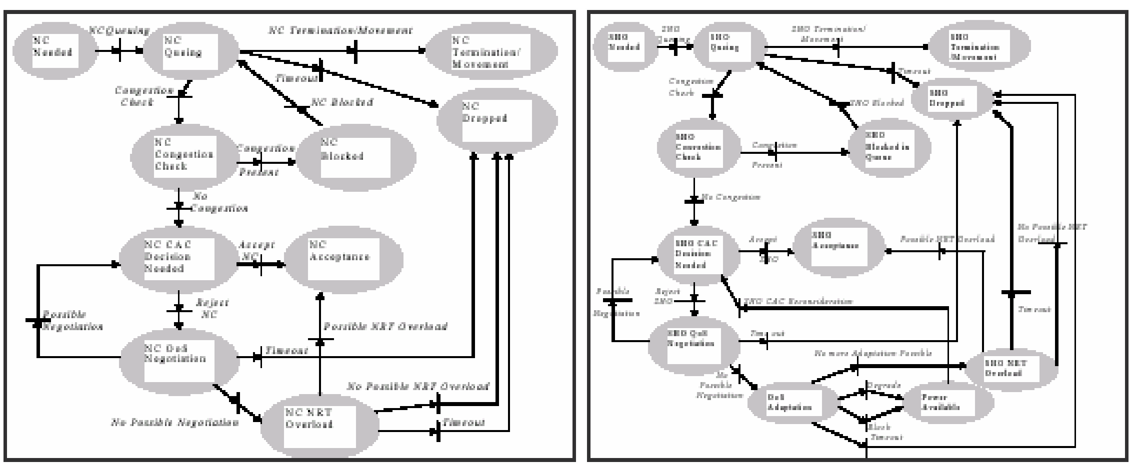
Where $0<\alpha<1$ indicates the maximum overload allowed for NRT connections, and *i* spans across all existing connections including the one under the admission decision. The

NRT overload strategy is used for admission decisions in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004].

3.3.3 The Complete Call Admission and Handoff Strategy

Figure 3.3.3-1(a) shows the new call admission handling process in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. A backed off and queued NRT call can be dropped from the queue if it reaches its timeout. It also shows that negotiation of QoS requirements take place in the Admission Controller entity. If negotiation fails and a NRT overload strategy is chosen, the new call can be accepted if it belongs to an NRT class, and reconsidered later when power is available.

Figure 3.3.3-1(b) shows the handoff admission handling process in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. This process is very similar to New Call handling except unlike a new call, there is no negotiation and QoS adaptation (Section 3.3.4) is triggered to provide the handoff request with the necessary bandwidth at the expense of existing connections.



(a) New Call Handling process (b) Handoff Call Handling Process

Figure 3.3.3-1: Call Admission in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]

3.3.4 QoS Adaptation Algorithm

This algorithm was utilized in both their call admission mechanism and the Power Control mechanism of [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. This algorithm resolves congestion in two phases. The two phases are applied differently in case of congestion handling and in case of handoff admission. The QoS profile carried by each user comprises of the traffic class, SDD, and the bit-rate.

- The Degradation Phase

  This phase is based on the SDD. Iteratively, the active user with the highest SDD is the user that degraded in terms of its bandwidth requirements. Calls are degraded according to Table 3.3.4-1.

Original Bit-rate	Degraded Bit-rate
384Kbps	144Kbps
144Kbps	64Kbps
64Kbps	16Kbps
16Kbps	Not Degraded Further

Table 3.3.4-1: Degradation Schema

- Dropping Phase

  The dropping phase takes place when willing connections are degraded, but congestion persists. Dropping is based on:

$$F_i(t) = SDD_i * P_i(t) \qquad \textbf{Equation 3.3.4-1}$$

Where $P_i(t)$ is the power required by connection i at time t. Calls with the highest $F_i(t)$ are the first to be dropped. $F_i(t)$ will be large for calls with high bandwidth requirements and high SDD values.

The QoS adaptation algorithm is invoked to provide the necessary bandwidth for Handoff requests that normally would not be accepted due to the lack of resources. Two cases are distinguished depending on the class of service. RT connections are accepted into the network during congestion by first degradation and then by dropping lower priority users. The NRT connections will not be accepted by dropping other users.

3.4 Simulation Model

In the approach used by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004], power is considered to be the only limiting source and other system resources such as spreading codes and buffering capacity are considered to be available in plenty. The cost of a connection is computed according to the following formulae.

$$C_i(t) = \frac{Eb}{No} \cdot \frac{1}{w} \cdot \frac{I_i(t)}{H_i(t)}$$ **Equation 3.4-1**

Energy to noise ratio is represented by Eb/No. Intercellular interference is not taken into account in this model. The chip-rate is represented by w and $I_i(t)$ represents the sum of interferences exerted by existing users at a given time within the same cell. The central limit theorem is used to model $I_i(t)$ as a Gaussian process with zero mean and a given variance σ^2. The channel gain at a given time $H_i(t)$ follows a Rayleigh distribution. $C_i(t)$ is the cost (power per bit) for maintaining connection i in a interference limited environment. The total average power required by connection i operating at bit-rate R_i is given by:

$$P_i(t) = C_i(t) * R_i$$ **Equation 3.4-2**

We believe the simulation model discussed in [A. Capone and S. Redana 2001] is a better and a complete model to test Call Admission schemes than the one mentioned above and used in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. [A. Capone and S. Redana 2001] has a better path loss model that also considers Intercellular interference. The model in [A. Capone and S. Redana 2001] is designed purely to test voice calls but it will be adjusted as part of this research.

This approach also considers power to be the only limiting source and *SIR* experienced by each user is used as the QoS measure. Others resources such as spreading codes and buffering capacity are considered to be readily available. The Base Stations are assumed to be located at the center of the cell with omni-directional antennas. The following sections will be used to describe the model from [A. Capone and S. Redana 2001] in detail.

3.4.1 Propagation Model

The relationship between the received power P_r and the transmitted power P_t is given by:

$$P_r = P_t \alpha^2 10^{\varepsilon/10} \frac{1}{L} \qquad \textbf{Equation 3.4.1-1}$$

Where L is the path loss, $10^{\varepsilon/10}$ accounts for the loss due to slow shadowing, ε being a normal variate with zero mean and α^2 represents the gain, with an exponential distribution of unit mean, due to fast fading. The path loss is expressed as:

$$10\log L = 128.1 + 37.6\log r \quad (db) \qquad \textbf{Equation 3.4.1-2}$$

Where r (in meters) represents the distance between the mobile and the base station. Furthermore the shadowing standard deviation is assumed to be 5dB.

3.4.2 The Receiver Model

At the receiver side the *SIR* after dispreading is evaluated for each transmission as:

$$SIR = SF \frac{P_r}{I_{\mathrm{int}ra} + I_{\mathrm{int}er} + P_N} \qquad \textbf{Equation 3.4.2-1}$$

Where P_r is the receiver signal strength, P_N is the thermal noise power assumed equal to –99dBm in the downlink and –103dBm in the uplink. $I_{\mathrm{int}er}$ is the sum of signal powers received from other cells, $I_{\mathrm{int}ra}$ is the sum of signal powers due to other transmissions within the same cell and SF is the spread factor. The spread factor is calculated as:

$$SF = \frac{3.84 Mchips}{Bitrate} \qquad \textbf{Equation 3.4.2-2}$$

3.4.3 Power Control Model

This power control mechanism is based on the procedures defined in the UMTS specifications for the dedicated channel (DCH). The transmitted power is adjusted at each algorithm iteration to maintain the *SIR* at the target value, SIR_{tar}. In this model a power control iteration is executed periodically depending on the type of traffic. The new power level is calculated as:

$$P_{new} = P_{old} \frac{SIR_{tar}}{SIR_{curr}}$$ **Equation 3.4.3-1**

Each uplink and downlink has separate power restrictions. The overall transmitting capability of the base station also has power restrictions. If the power control requires a power level higher than the maximum value, the maximum value is adopted. If the sum of powers required by the downlink channels exceeds the base station's maximum power, all the powers are proportionally reduced to limit the total power to the maximum value.

After each power control iteration the actual *SIR* values experienced by the each user is evaluated. If the *SIR* is lower than the target value, SIR_{tar} , the call is dropped.

3.5 Summary

In this chapter the main approaches mentioned in the literature for Call Admission Control were presented. The main advantages and drawbacks of the schemes were also highlighted. The schemes used to build my project on were discussed in greater detail.

The QoS aware Power Control and Handoff Prioritization scheme introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] and the Received Power based simulation model discussed in [A. Capone and S. Redana 2001] are chosen as the base for this project research. The simulation model of the latter was considered to be better than that of the former. The following chapters will look to improve the simulation model of [A. Capone and S. Redana 2001] and to test the call admission scheme introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004].

Chapter 4. Simulation Model Implementation

4.1 Introduction

This chapter describes the main aspects and the limitations of the simulator described in [A. Capone and S. Redana 2001], and the modifications made to it. The specific simulated environment and the performance metrics used to measure and compare performance are also described. It also looks at the performance of an admission scheme that uses ideas introduced in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] and compares with a simple admission algorithm.

4.2 Simulation Model

In our model Power is considered to be the only limited resource and other resources such as spreading codes and buffering capacity are assumed to be available in plenty. A single caller is assumed to request a single service. The Base Station is assumed to be located at the centre of a cell with an omnidirectional antenna. The model was implemented in a MATLAB simulation environment.

User requests are processed on a “first come, first served” basis. The cell radius is 300m. The decision of accepting or rejecting a request is based on the QoS profile attached to the request, the maximum power available in the system and the QoS requirements of users being served. In the uplink direction the total interference measured at the base station is used as the load indicator. In the downlink direction the only power level that needs to be considered is that emitted by the base station.

The propagation model follows the scheme used by [A. Capone and S. Redana 2001] and described by equation 3.4.1-1. The path-loss L is expressed by equation 3.4.1-2. Four classes of call (Chapter 1) are considered in the simulation, namely Conversational, Streaming, Interactive, and Background. Conversational and Streaming classes (Class 1 and 2 respectively) are Real-Time services. Examples of Conversational and Streaming classes are Voice calls and Video streams respectively. Interactive and Background classes are Non-Real Time services. Examples are Web traffic and e-mail respectively.

The traffic model adopted has each user requesting a single service and user arrivals according to a Poisson process with intensity $\lambda = 1$. Table 4.2-1 shows the experimental parameters obtained from [ETSI 23.107 v5.9.0 (2003 – 2006)] and [ETSI 25.401 v6.3.0 (2004 – 2006)]. The initial position in the cell of a new call and its SDD are generated randomly. For each call the bit-rate, speed, call duration and the target *SIR* are assigned according to the class shown in Table 4.2-1.

Class	**Bit-rate**	**Duration**	**Velocity**	SIR_{tar}
1	384 Kbps	120 s	16.7 m/s	4 dB
1	144 Kbps	120 s	27.8 m/s	4 dB
1	64 Kbps	16 s	33.3 m/s	4 dB
1	16 Kbps	256 s	44.4 m/s	4 dB
2	384 Kbps	120 s	0 m/s	7 dB
3	144 Kbps	120 s	27.8 m/s	5 dB
3	64 Kbps	16 s	33.3 m/s	5 dB
3	16 Kbps	256 s	44.4 m/s	5 dB
4	64 Kbps	16 s	33.3 m/s	5 dB
4	16 Kbps	256 s	44.4 m/s	5 dB
Chip-Rate	3.84 Mcps			
$\mathbf{P_{max}}$	35 W			

Table 4.2-1: Main Experimental Parameters

At the receiving end the *SIR* after dispreading is evaluated for each transmission according to equation 3.4.2-1. The spread factors were calculated according to Equation 3.4.2-2. I_{intra} was calculated by summing the signal powers of other users. I_{inter} was assumed to be 12% of I_{intra} [UMTS 30.03 v 3.2.0].

The power control mechanism follows that of Equation 3.4.3-1. In this model the power control iteration was executed every 10 ms. Initially for the downlink, each Real-Time channel was set, so the requirements cannot exceed a transmitted power of 5 Watts and each Non-Real-Time channels was set to limit transmit power requirements to 2 Watts.

For the uplink, the transmission was limited up to 2 Watts and the Uplink power of a user is reduced if the interference at the base station is significant. In [A. Capone and S. Redana 2001], If the sum of powers required by the downlink channels exceeds the base station maximum power all the powers are proportionally reduced to limit the total power at the maximum value. Some changes were made to that part of the algorithm in order to make use of the users that are accommodated with a higher than required SIR_{tar}. After each power control iteration the actual SIR values experienced by the each user is evaluated. If the SIR is lower than the target value, SIR_{tar}, the system increases its transmitted power provided sufficient Base Station Power is available. If the system resources are insufficient, that call is dropped. The users with the lowest SDD values and Real-Time services are considered first.

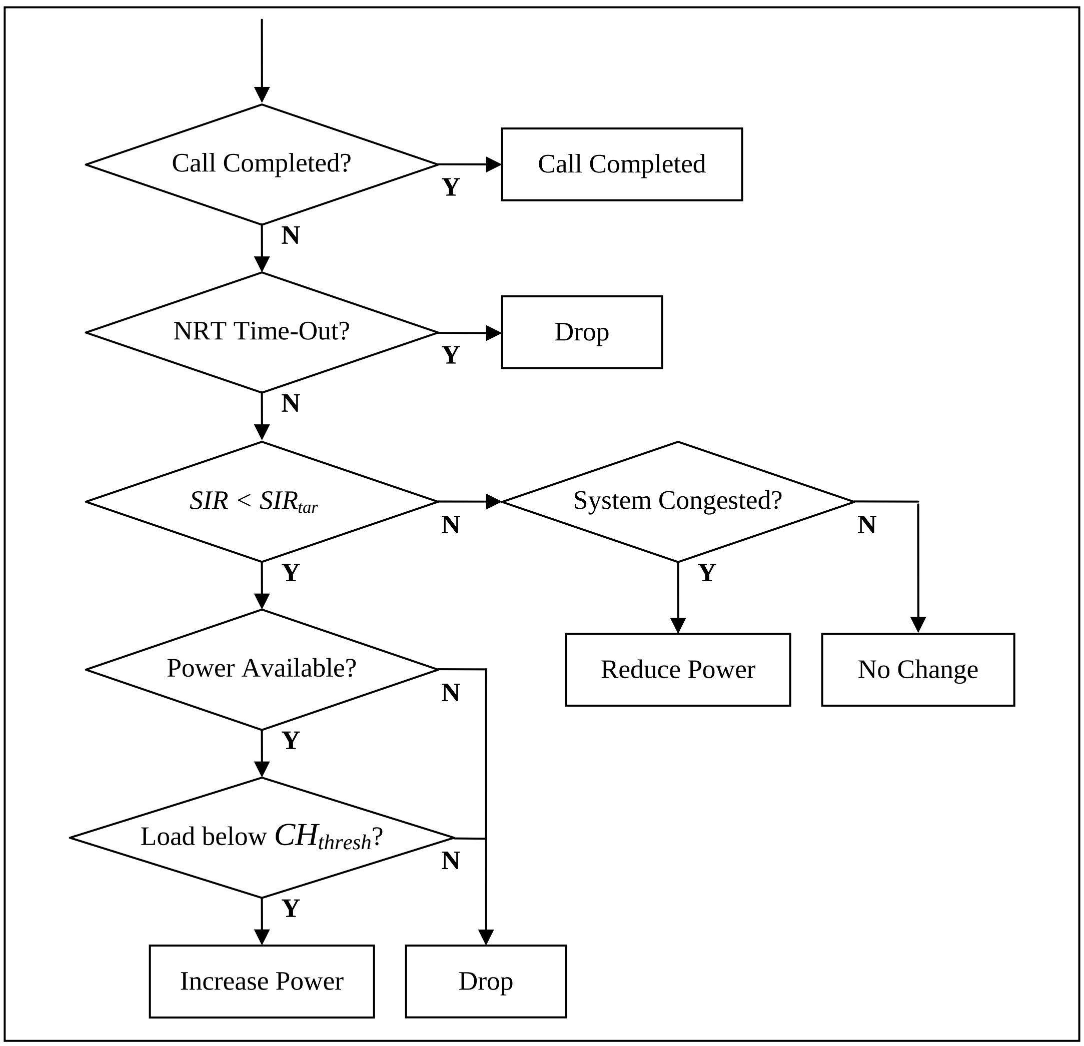


Figure 4.2-1: The Power Control Mechanism

The users with high SDD values and Non-Real Time services are considered last. If a user has a *SIR* value higher than SIR_{tar} the transmit power allocated to that user is reduced. The Downlink Power Control mechanism is summarised in Figure 4.2-1 below. Note that call termination due to user movement out of the cell is considered a completed call.

4.3 Admission Schemes

Two existing admission schemes are investigated in this chapter. Firstly the simple Received Power Based Admission scheme mentioned in [A. Capone and S. Redana 2001] for the down link direction and the more complex scheme mentioned in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. The former works by setting a Power threshold when it comes to admitting traffic. Therefore a user with a power requirement of more than P_{thresh} is not admitted into the system. The scheme described in [A. Capone and S. Redana 2001] is for voice calls only. Therefore the scheme was modified for this thesis research to accommodate different P_{thresh} values for different classes. All users are admitted with transmit power allocations that accommodates them at the SIR_{tar} . The receive power at the mobile required for each type of request is calculated using Equation 3.4.2-1 and the SIR_{tar} specified for that particular service Table 4.2-1. The transmit power required to support that caller is calculated using Equations 3.4.1-2 and 3.4.2-2. The system, first checks to see if there is sufficient power available at the base station to support the service that is requested. If there is insufficient power at the base station, the call in question is blocked from entering the system. The scheme is summarised in Figure 4.3-1.

Chapter 3 has the scheme used by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] described in detail. Some minor changes were made to the original Admission process by not queuing a connection at the beginning of the admission process and by not dropping calls with high SDD values in order to accommodate a new user. The reasoning behind the latter is mainly to reduce complexity of the simulation model. We also believe the ratio of the amount of Power recovered to the number of calls dropped would be very small because the dropping phase in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] is invoked after the degradation phase and the NRT overload phase and calls that are degraded or backed off would only consume a small amount of resources and those will be the first to be dropped in the dropping phase.

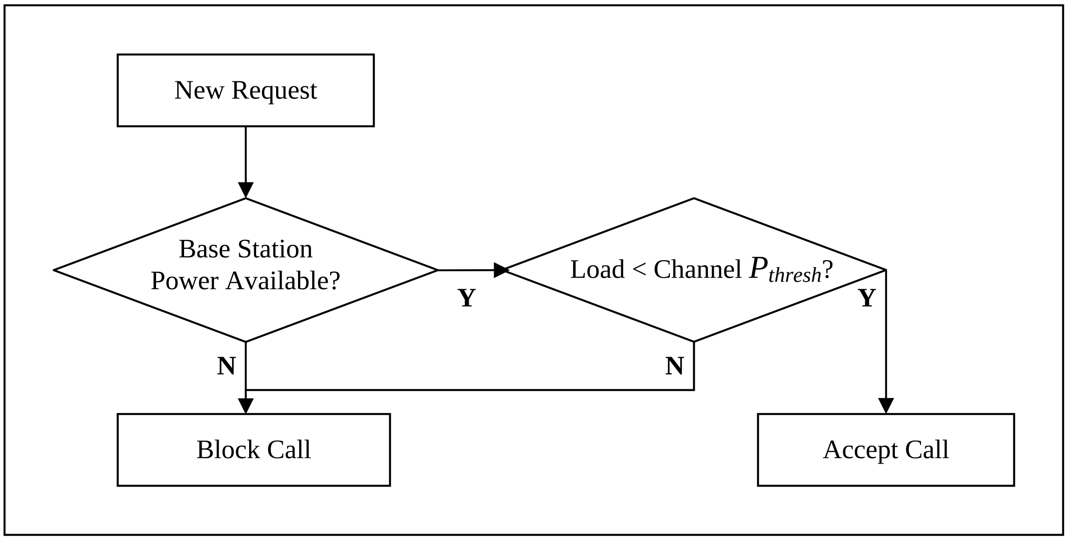


Figure 4.3-1: Simple Call Admission Process

In our modified algorithm, if a new request or a handoff does not find sufficient resources after all the admission considerations the request will be blocked instead of queuing for a certain time period. Only the backed off Non-Real Time calls are queued. A modified version of the diagrams (from Figure 3.3.3-1) is shown in Figure 4.3-2 in a simplified manner and is considered in this thesis. Here the required downlink transmitted power is first calculated using the same procedure as the Simple Admission Scheme discussed earlier. Then the system first checks to see if there is sufficient power available at the base station to support the service that is requested. If there is insufficient power the call in question is degraded and if congestion persists, the other callers in the system are degraded according to their SDD parameter. If congestion is still present the system will adopt the NRT overload scheme (Refer Section 3.3.2).

Our model holds a delayed NRT call for 8 seconds before timing out and dropping. If congestion is present after considering degradation and NRT overload schemes, the call in question is blocked from entering the system. Initially an NRT overload parameter α, of 10% was considered. Like [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] other users are not degraded to accommodate a New Request (Non Handoff).

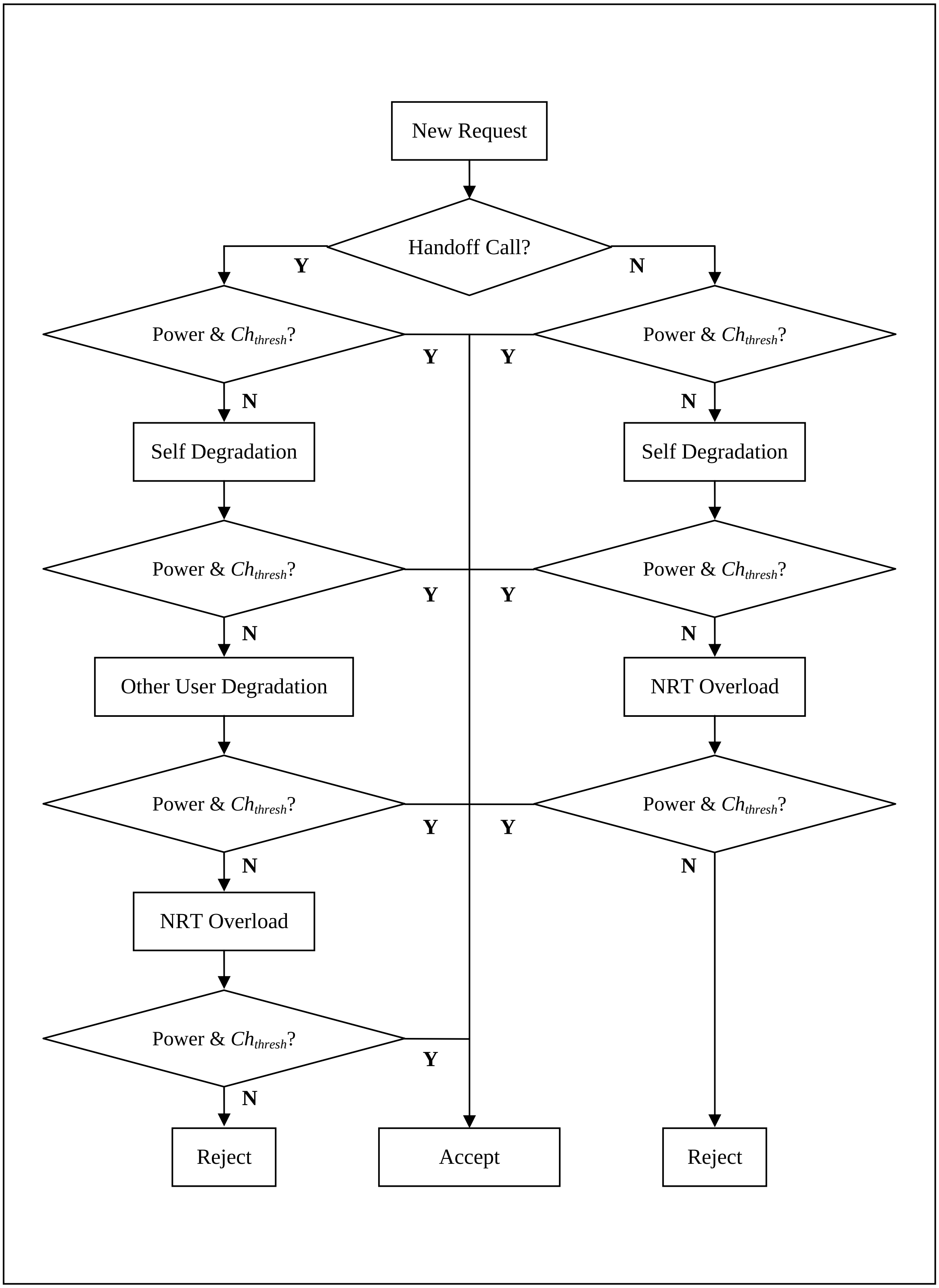


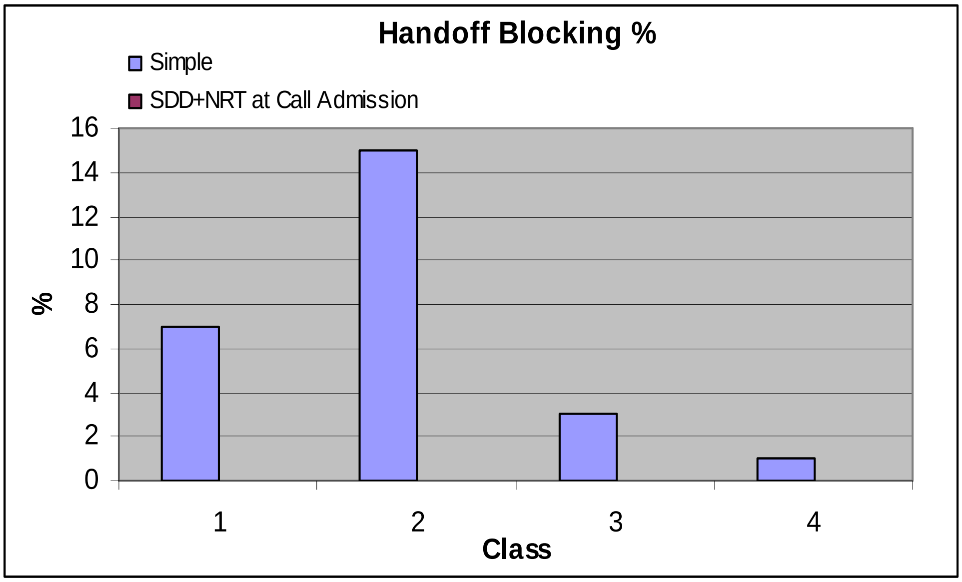
Figure 4.3-2: Complex Call Admission using SDD and NRT Overload

4.4 Analysis of the two Schemes

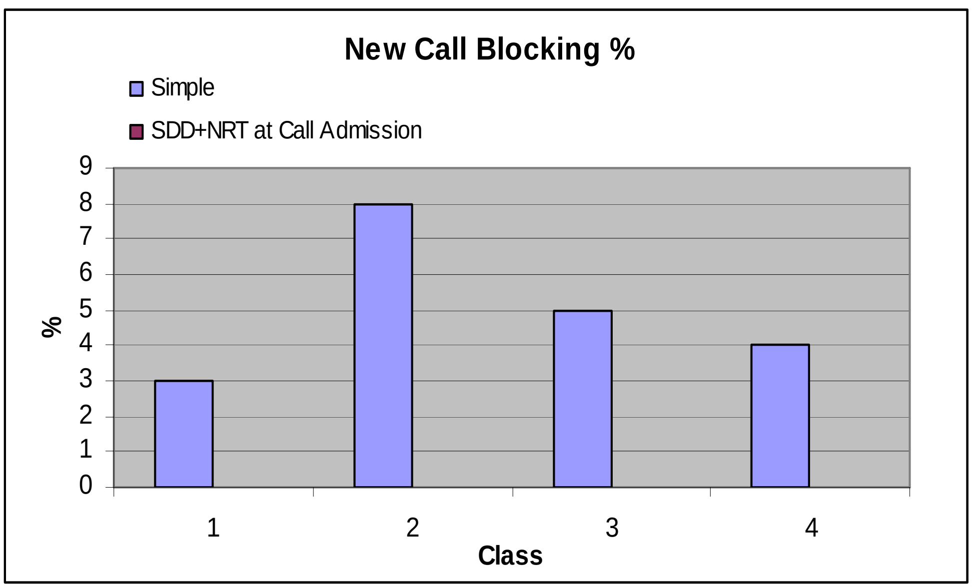
The experiment consisted of generating 2000 users with different QoS and service requirements and arriving according to a Poisson process. The simulation terminates after the admission or rejection of the 2000th user considered. Twenty such simulation experiments were conducted and the results were averaged. All results presented in this thesis are within 95% level of confidence (see Appendix). The traffic mix was set at 15% class 1, 5% class 2, and the remaining 80% were NRT calls. A high arrival rate of one arrival per second was considered. The handoff blocking percentages and the new call blocking percentages for the two Admission schemes are shown in Figure 4.4-1.

The intention of [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] was to create a call admission scheme that would minimise handoff failures. Looking at the simulation results of Figure 4.4-1 it further proves that SDD based degradation and NRT overload schemes are very effective in minimizing handoff losses. The Admission scheme that used SDD based degradation and NRT overload schemes produced 0% handoff and new call blocking for all classes while the simple algorithm produced a significantly higher failure rate for handoff calls and new calls.

The effect on the ongoing call dropping percentage from the two call admission schemes is shown in Figure 4.4-2. Although Degradation plus NRT overload works well in minimizing Handoff losses, there is a negative effect on the dropping percentages. In fact it performs even worse than the simple admission algorithm. Particularly the RT service (Class 1 and 2) dropping rates are poor. The reason is that these two Classes contain services that require very high bit-rates (and spread factors) and they have a higher chance of having power requirements that cannot be supported by system if the mobile terminals move closer to the edge of the cell when the system is experiencing heavy congestion. The degradation and overload scheme accepts more users than the simple scheme and therefore has a higher dropping percentage. Improvements have to be made to the Power Control mechanism in order to reduce the dropping percentages of ongoing calls.



(a) Handoff Blocking Percentage (nearest 1%).



(b) New Call Blocking Percentage (nearest 1%).

Figure 4.4-1: Handoff Blocking and New Call Blocking performance for the two Call admission schemes.

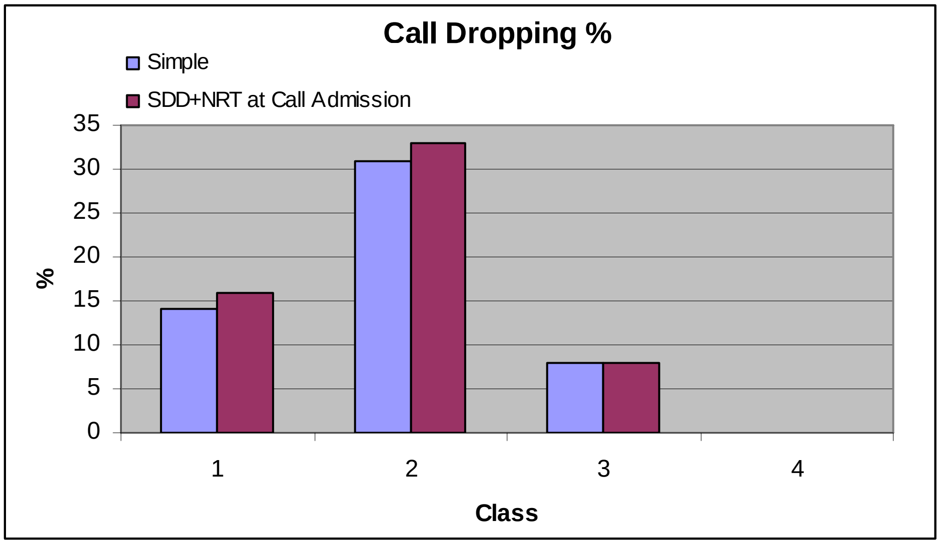


Figure 4.4-2: Dropping Percentages of ongoing calls (nearest 1%).

4.5 Summary

This chapter looked at the performance of an admission scheme that used ideas introduced in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] in the simulation environment introduced in [A. Capone and S. Redana 2001]. Some minor changes were made in our experiments to the schemes introduced by these authors.

The scheme that used Degradation and overload techniques performed superior to the simple admission scheme considered in reducing handoff losses and new call blocking. But both schemes produced high dropping percentages for the ongoing calls. It was concluded that improvements are required to the Power Control Mechanism in order to reduce the dropping rate.

Chapter 5. Enhancement Mechanisms

5.1 Introduction

This chapter introduces the mechanisms used to enhance the performance of the Call Admission and Power Control schemes discussed in Chapter 4. Comparisons are made between the performance of all three algorithms and results are presented.

5.2 Power Control Enhancements

The same degradation and overload techniques used at call admission can be used as a possible technique for Power Control performance enhancement. The suggested algorithm with improvements is summarised in Figure 5.2-1.

In this scheme, the system first performs a check to see whether the required Downlink Power level is within the threshold for that channel. If it is and if sufficient power is available in the system, the power level is increased to achieve the desired *SIR* value. If the required load is not below the threshold, the user is asked to consider transmission at a degraded bandwidth. If the power requirement is still unfulfilled, the bandwidths of the users being served are degraded according to their SDD value in the QoS profile. If that is still insufficient, the NRT connections being served are backed off and the resources are offered to the call in question if it is a RT service. If it is an NRT service or if the NRT overload scheme is unsuccessful in obtaining sufficient power, the call in question is dropped. If the call in question has a power requirement that is within the threshold, but is unable to obtain sufficient resources from the system, it is asked to degrade itself before considering any of the other techniques. The reason being, transmission at a lower bandwidth is capable of providing a lower power requirement.

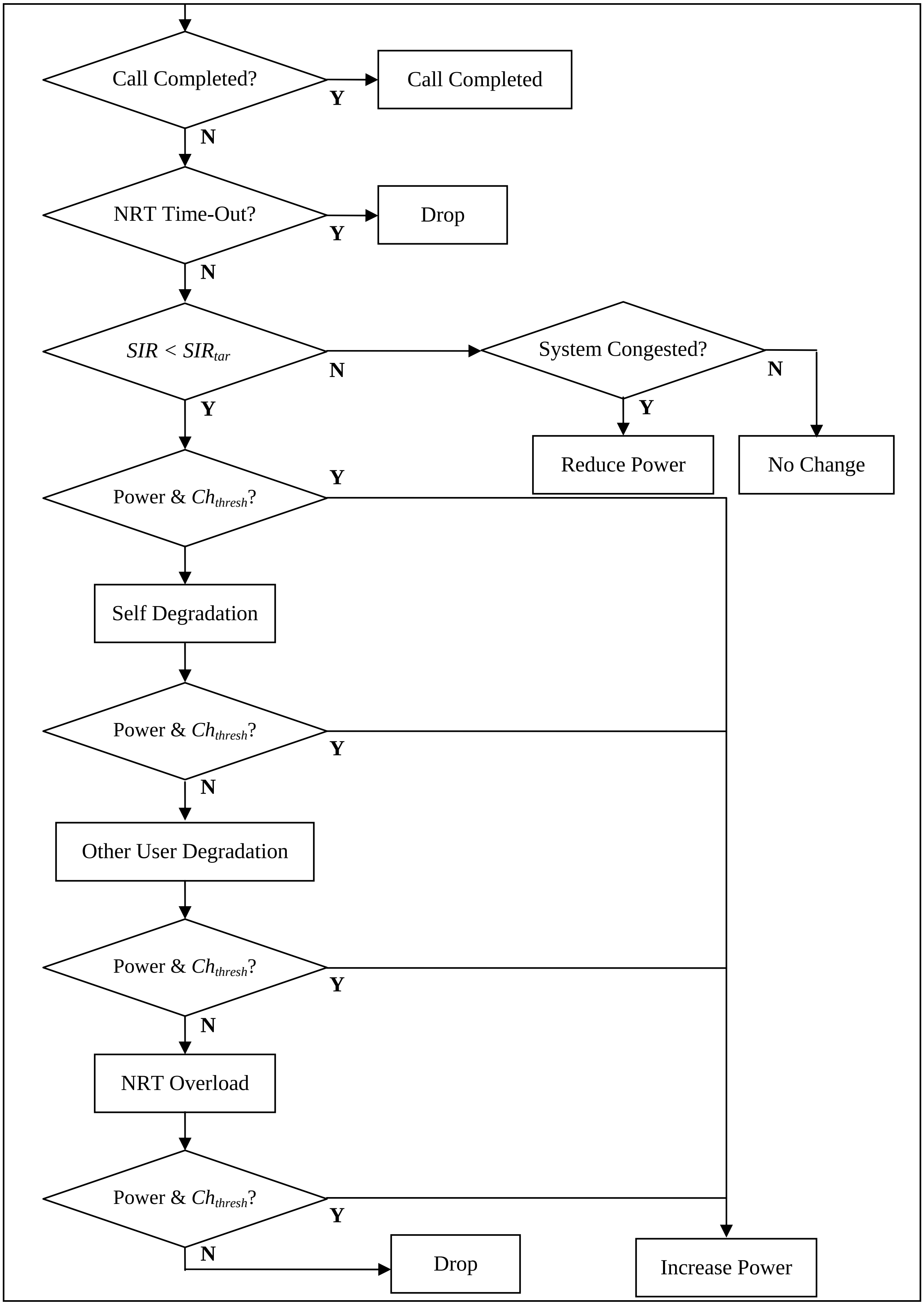


Figure 5.2-1: Enhanced Power Control mechanism

Like in the earlier algorithm, if the current *SIR* is better than the target, the power allocations for that user will be reduced to achieve the target *SIR* if the system is in congestion.

5.3 Enhancements to the Call Admission Scheme

Call dropping is considered to be far worse than call blocking. The algorithms discussed before in Chapter 4 produced low blocking percentages at the expense of high dropping percentages. The primary objective of a call admission scheme should be to ease the burden on the system as much as possible and reduce dropping rates. Reducing blocking rates should be considered secondary.

In order to produce that outcome the NRT overload consideration at New Call admission was removed. In this modified algorithm the NRT overload admission scheme is only applied to handoff calls. When a request to admit a new call arrives, admission takes place only if power is readily available or if the power level requested by the user can be negotiated down to an acceptable level through self degradation (Figure 5.3-1).

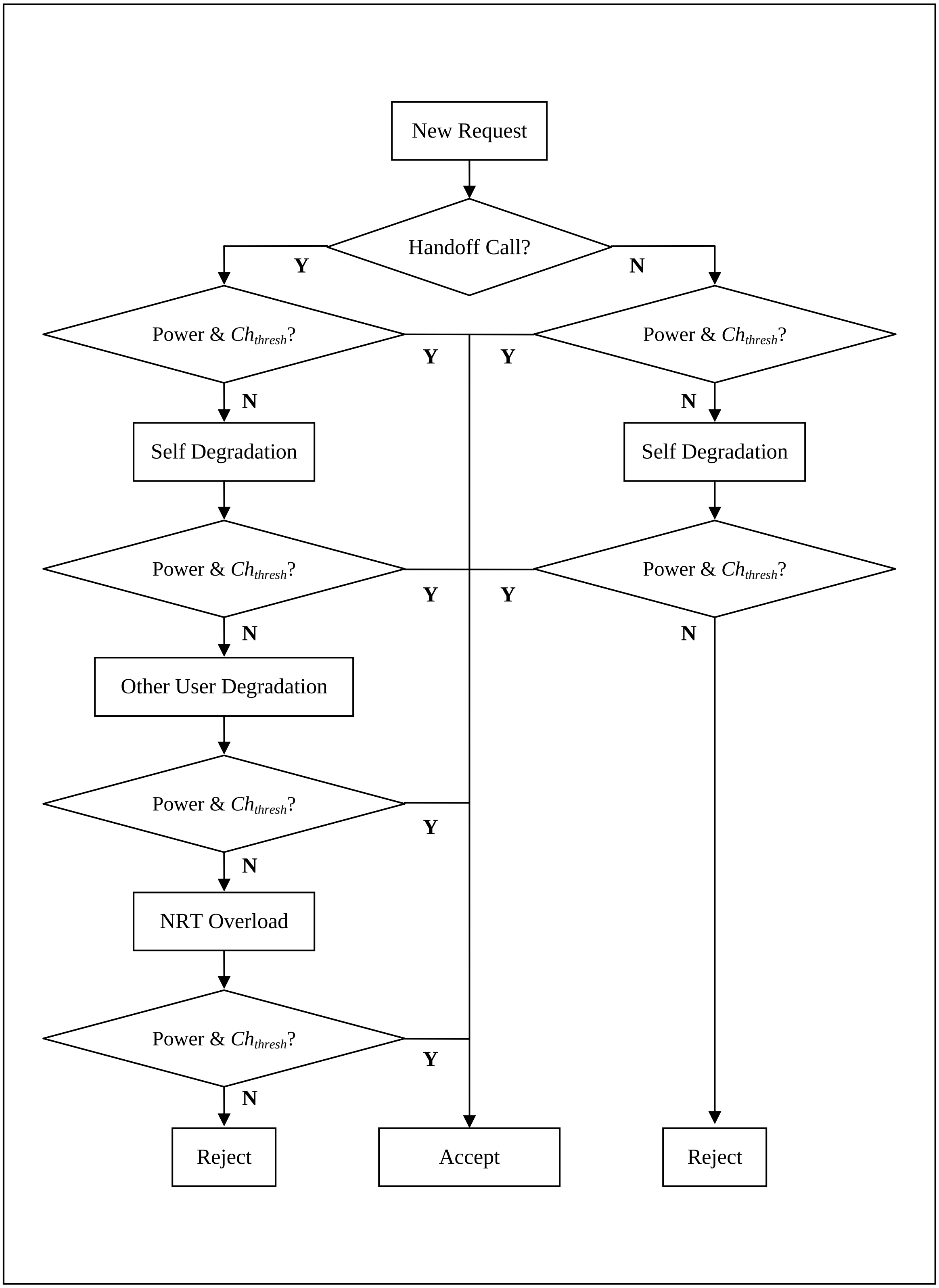


Figure 5.3-1: Complex Call Admission using SDD and NRT Overload

5.4 Analysis of the New Algorithm

Figure 5.4-1 showing the dropping percentages indicates the effectiveness of the modifications. The new algorithm has produced significantly lower dropping percentages for the real-time classes. Compared to the previous degradation and overload algorithm, the algorithm has increased the performance of class 1 services by almost 16% and the performance of the class 2 services by a mammoth 30%. Only the NRT services have shown poorer dropping rates. But this can be considered a very reasonable outcome because RT services are the higher priority services. A voice or video conference call interrupted can be far more annoying to a user than the termination of a webpage download or the failure of an e-mail delivery.

The reason for this improvement can be explained by the SDD based degradation mechanism and the NRT overload scheme. With the application of the degradation mechanism, most calls that are considered would be content with lower bandwidth and power allocations. This scheme would particularly influence the services with high bandwidth requirements such as class 1 and 2. The NRT overload scheme is also designed to prioritise RT services over NRT services. That explains the observation of high dropping rates for the NRT services and contributes to reducing RT losses even more.

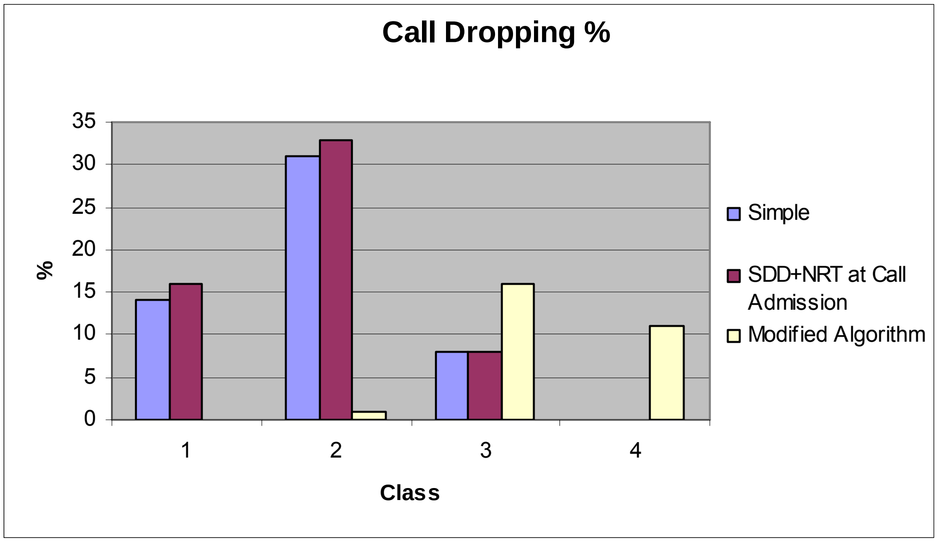


Figure 5.4-1: Call dropping percentages for the three algorithms (nearest 1%).

The handoff blocking percentages and new call blocking percentages are shown in Figure 5.4-2 and 5.4-3. No changes are observed in terms of handoff blocking. But a higher new call blocking rate is observed. That can be explained by the removal of the NRT overload mechanism from the New Call admission scheme. Overall the modified algorithm produced low RT call dropping rates at the expense of higher NRT dropping and New Call Blocking. That is the expected rule of thumb and it can be concluded that the performance of modification is far superior to that of the original version.

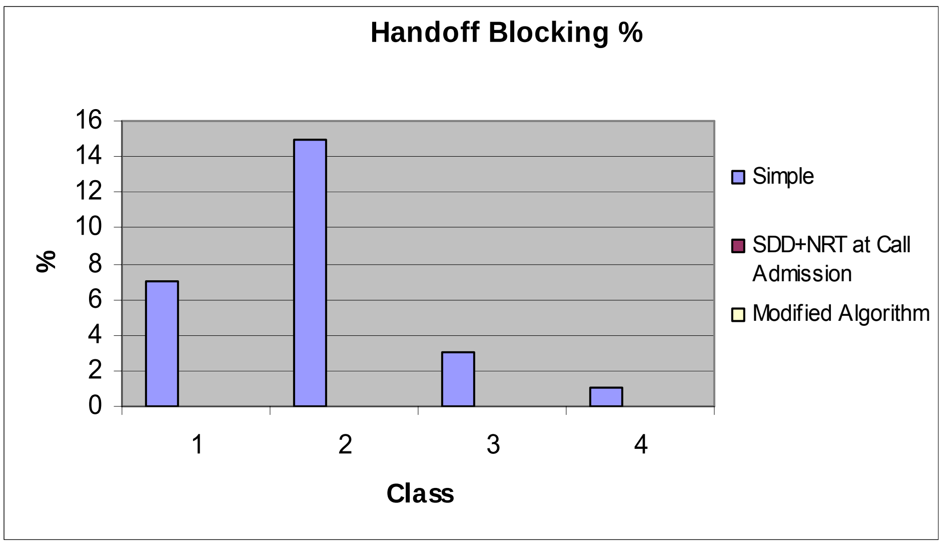


Figure 5.4-2: The Handoff blocking percentages for the three algorithms (nearest 1%).

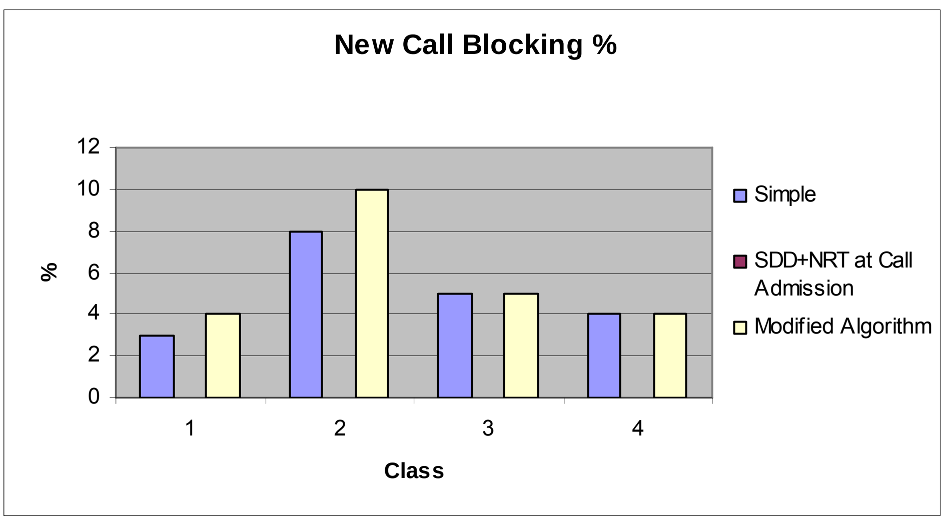


Figure 5.4-3: The New Call Blocking Percentages for the three algorithms (nearest 1%).

5.5 Reducing NRT Drop Rate

The NRT overload admission mechanism works by backing off ongoing NRT calls in order to accept new handoff RT calls and by accepting new NRT handoff calls in a backed off state. The mechanism is designed to drop each backed off NRT calls after a certain time period, if it has been unable to obtain sufficient resources to resume the call during that time. Therefore during heavy congestion more NRT timeouts are expected. The amount of NRT overload the system is willing accept, is limited by Equations 3.3.2-1 and 3.3.2-2. Therefore the amount NRT timeouts are limited by those equations. The two equations are reintroduced as follows:

$$\sum P_i(t) \leq (1+\alpha)P_{\max}$$ **Equation 5.5-1**

$$\sum P_{i/RT}(t) \leq \beta P_{\max}$$ **Equation 5.5-2**

Equation 5.5-1 explains that the system would accept users until the Base Station power requirements are up to a certain factor above the maximum power available by using the NRT overload mechanism. The additional factor β is introduced in Equation 5.5-2 to limit the number of NRT users backed off to accommodate RT users. β varies from 0 to 1. At β =1, the system is willing to allocate all the power available to RT users. At β =0, the system is not willing to allocate any extra resources to RT users at the expense of NRT users. This section studies the effect of changing α and β on the dropping and blocking rates.

5.5.1 Influence of α

This set of experiments was conducted using the modified algorithms introduced earlier in this chapter. The value of β was held constant at 1 and the α value was varied for each experiment (β has to be held at a non-zero value because it determines how much priority is given to RT services over NRT services. If not, varying α also does not have an effect on the process. Refer to Section 3.3.2 for more detail.) The results of varying α are shown in Figures 5.5.1-1, 5.5.1-2, and 5.5.1-3.

Figure 5.5.1-1 shows increasing dropping rates for NRT services with increasing α while the dropping rates for RT services (particularly class 2) reduce very slightly. Figure 5.5.1-2 shows a small difference in NRT handoff blocking between α values 0 and 0.1. No major improvements were observed in the New Call blocking rates. As seen from the graphs, increased overload has reduced RT dropping rates very slightly (combined improvement of less than 1%) by significantly increasing the NRT dropping rate (combined NRT dropping rate of about 10%) after backing off too many calls and timing out. Also an α value of at least 0.1 is required to obtain a handoff blocking rate of 0% for NRT services when β is set to 1.

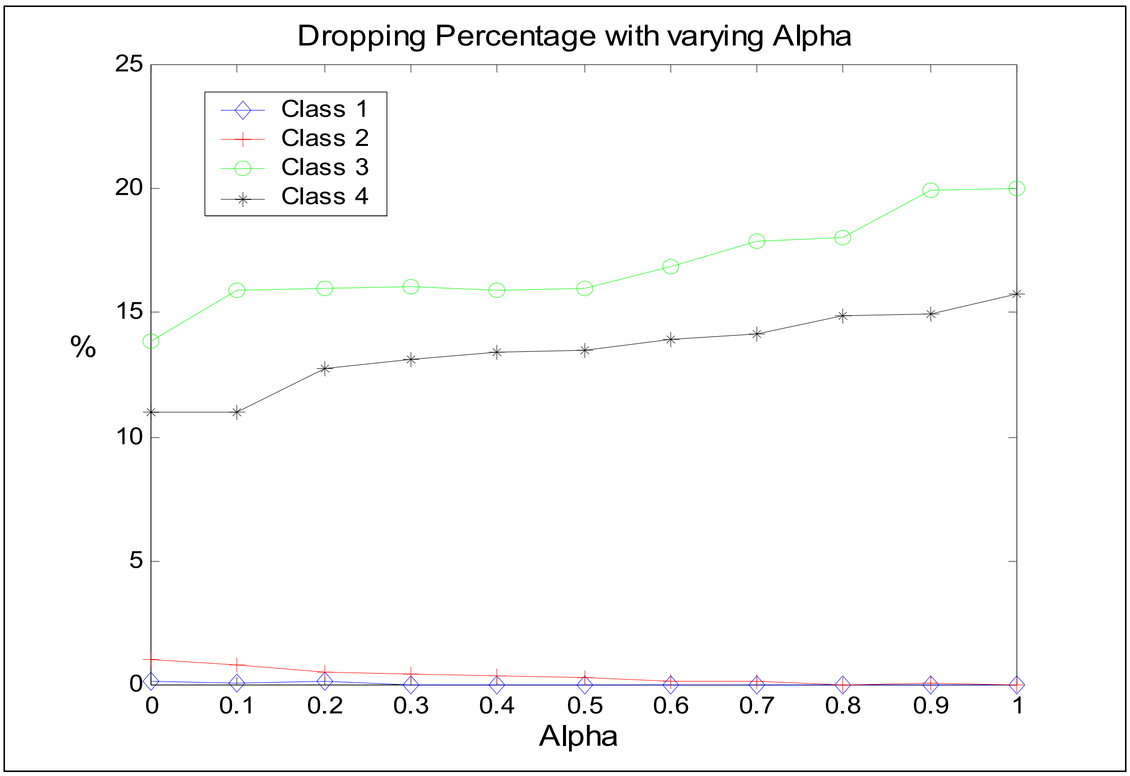


Figure 5.5.1-1: Variation of the Dropping percentage with varying Alpha

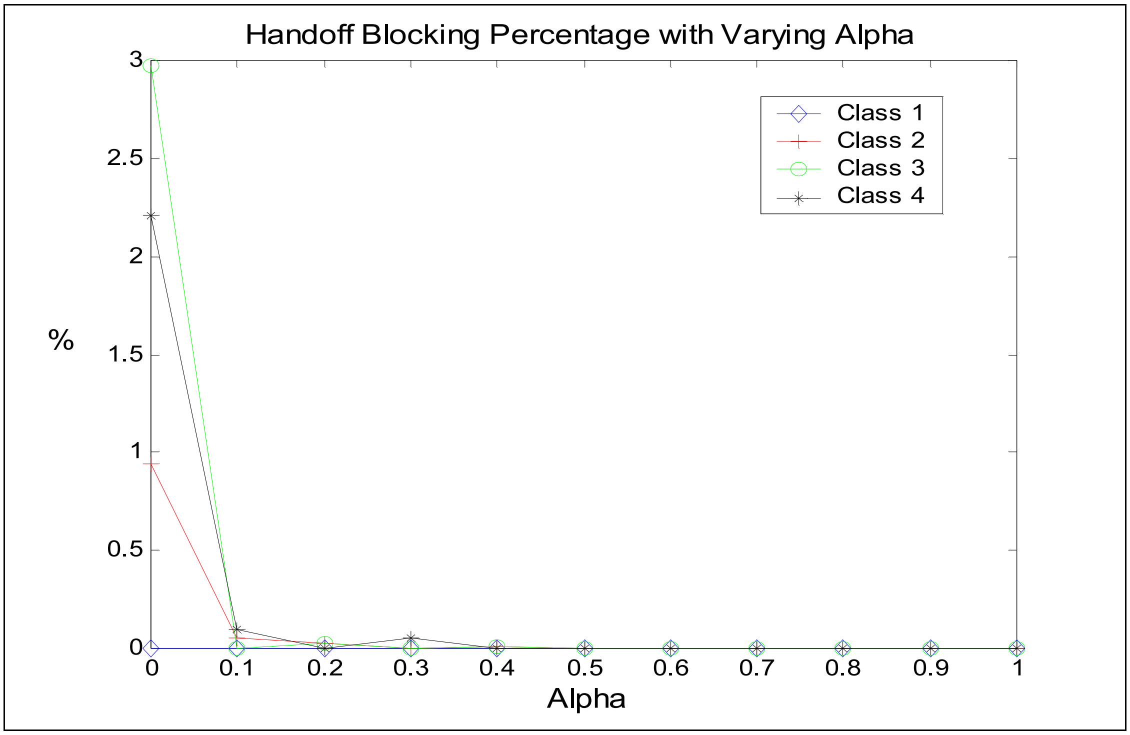


Figure 5.5.1-2: Variation of the Handoff Call Blocking percentage with varying Alpha

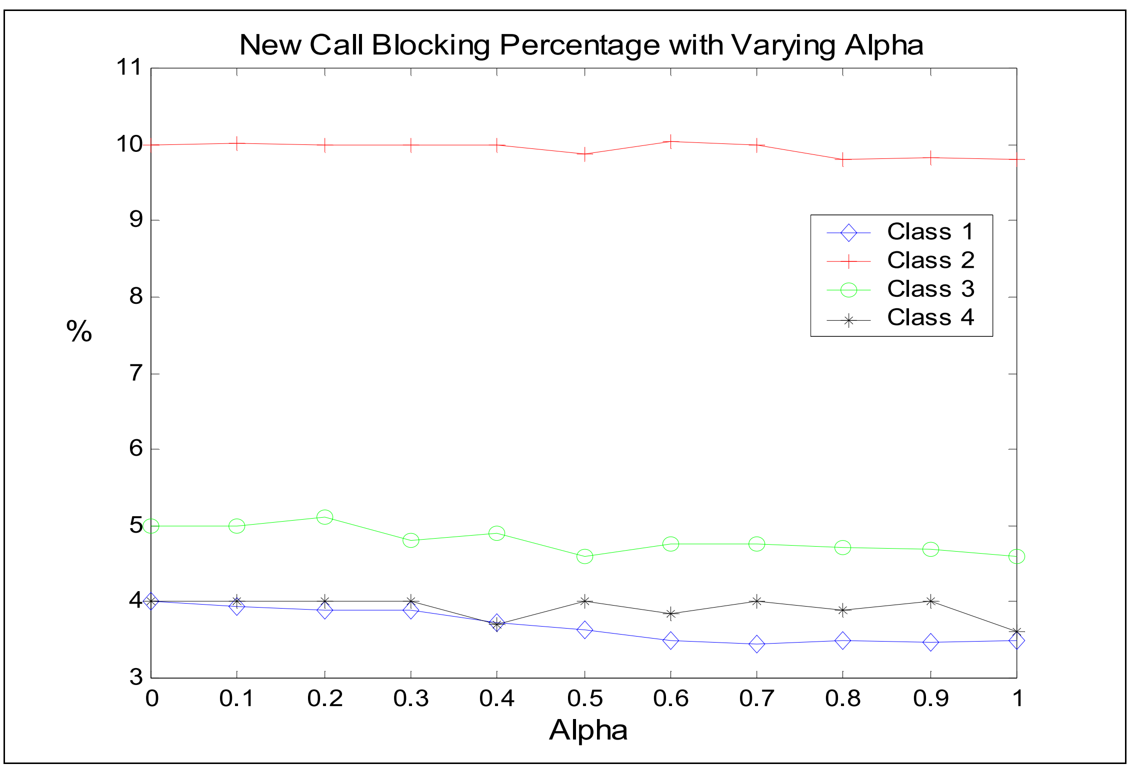


Figure 5.5.1-3: Variation of the New Call Blocking percentage with varying Alpha

5.5.2 Influence of β

For this set of experiments the value of α was held constant at 0.1 and the β value was varied for each experiment. β, by definition, limits the number of RT users the system is willing to accept at the expense of NRT users. As expected initially the number of dropped RT calls decreases with increasing β. But after β = 0.5, no significant improvement is observed. The number of dropped NRT calls increases with increasing β. This phenomenon is observed in Figure 5.5.2-1.

The number of blocked NRT handoff calls in Figure 5.5.2-2 has remained relatively unchanged with varying β. This shows that NRT handoff calls acceptance is more significantly influenced by α. But Figure 5.5.2-2 shows a decrease in RT handoff call rejection with increasing β. The New call blocking mechanism (Figure 5.5.2-3), which does not utilise NRT overload, has remained relatively unchanged for β > 0.5. The slight increase with increasing β is because when β is high more RT calls that consume a large quantity of resources can be maintained. When β is low more RT calls will be dropped and there is a better chance of accepting new requests with more resources being available.

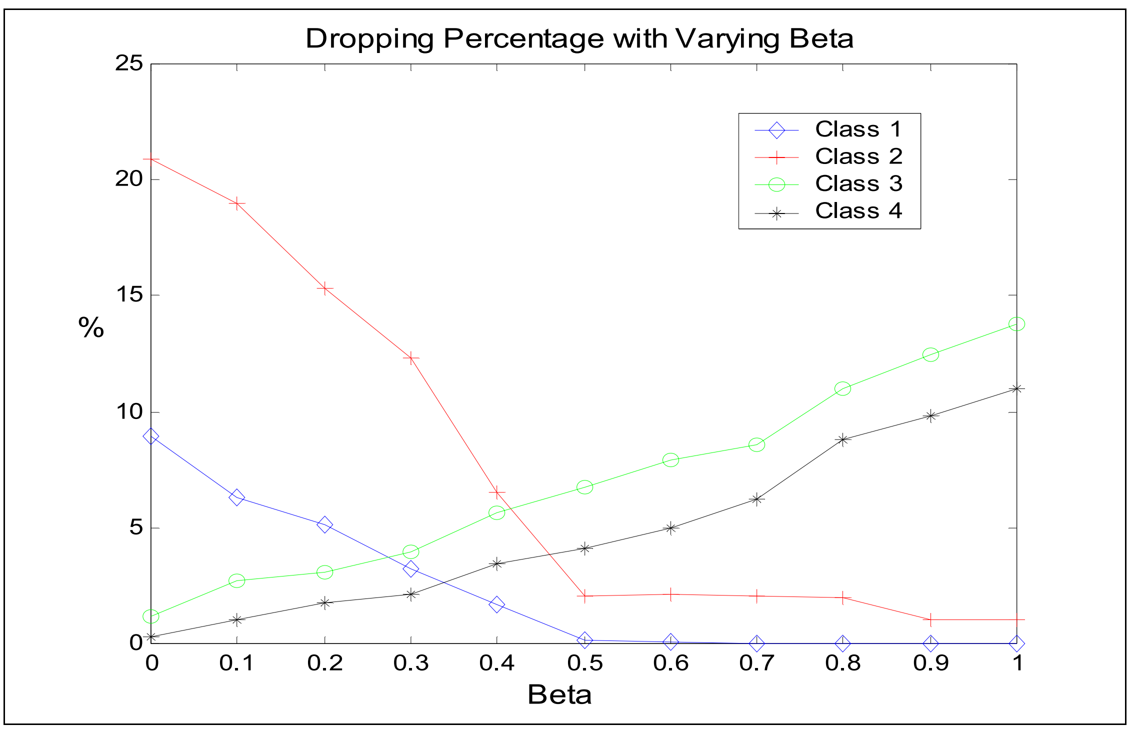


Figure 5.5.2-1: Variation of the Dropping percentage with varying Beta

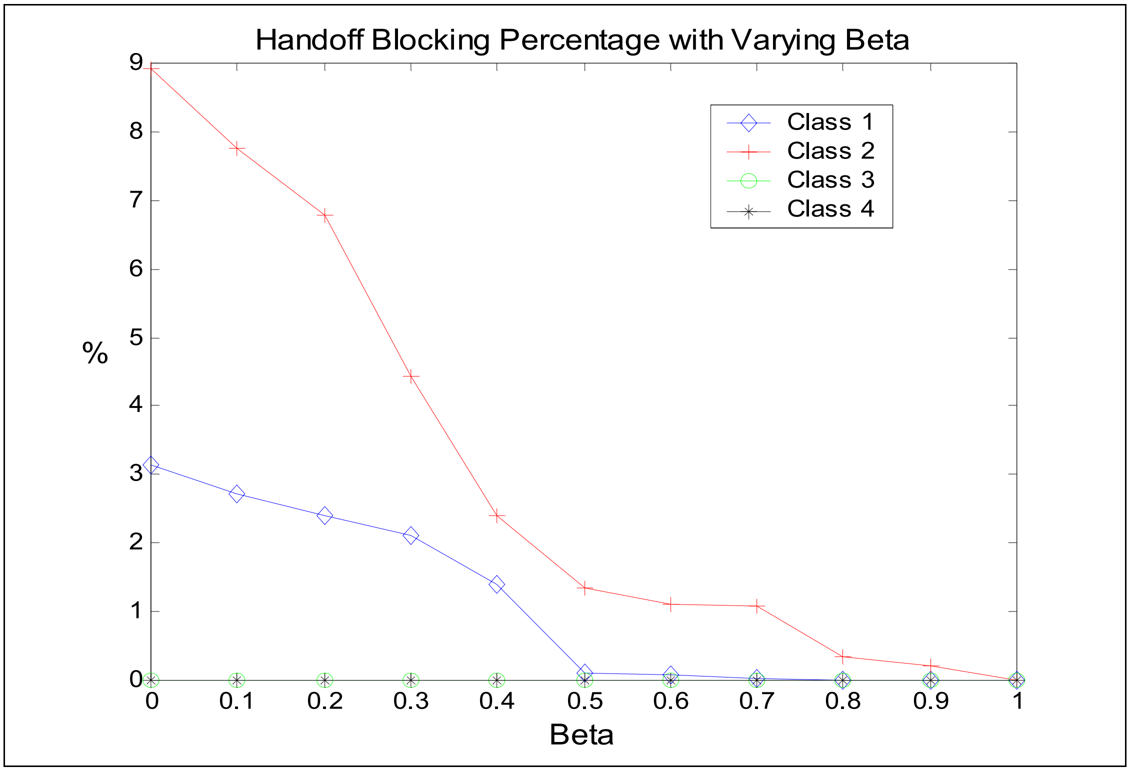


Figure 5.5.2-2: Variation of the Handoff Call Blocking percentage with varying Beta

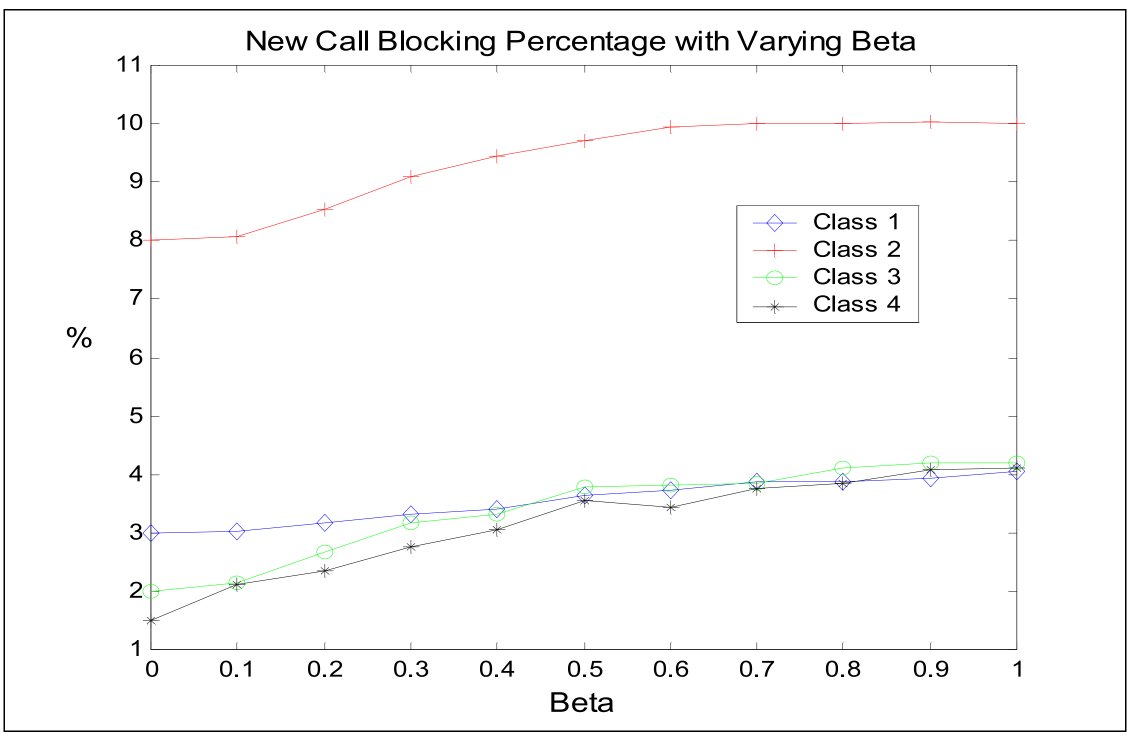


Figure 5.5.2-3: Variation of the New Call Blocking percentage with varying Beta

5.5.3 Optimum Solution and Analysis

Blocking a handoff call is identical to dropping a call from the user perspective. Ideally both proportions should be equally low. Therefore it seems unnecessary to tolerate an α value of 0.1 to obtain a NRT handoff blocking rate of 0% if it produces a high NRT dropping rate. Also Figure 5.5.2 shows that β values greater than 0.5 do not significantly improve the RT dropping rate or the RT handoff blocking rate. Performance evaluation of the optimum α and β solution compared to the solution with α and β values of 0.1 and 1, respectively, is presented in the following diagrams.

Figure 5.5.3-1 shows the algorithm with the configuration at α = 0 and β = 0.5 (Selection based on optimum results from Sections 5.5.1 and 5.5.2) graphed against the configuration α = 0.1 and β = 1 that was considered earlier in the Chapter and used by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004]. It is seen that α = 0 and β = 0.5 performs better in terms of dropping percentages (0% for class 1, 2% for class 2, 6% for class 3 and 3% for class 4). Handoff blocking percentage was higher for RT calls with α = 0 and β = 0.5 compared to the other cases as seen in Figure 5.5.3-2. But that is expected because the Overload factors are lower, which allows less calls to be accepted. Low call acceptance reduces the burden on the power control mechanism and fewer RT calls need to be dropped. This phenomenon balances the additional capabilities provided to the Power control mechanism by high β values and explains the behaviour of RT dropping rates for β > 0.5 in Figure 5.5.2-1. Low overload factors have a more direct impact on dropping NRT calls because they force fewer NRT calls to be backed off and timed out. Since Handoff blocks are seen to be identical to dropped calls from a user's perspective, it is suitable if the Dropping percentages and Handoff blocking percentages are at similar values. Overload factors of α = 0 and β = 0.5 achieve just that (Hand-off blocking at 0% for class 1, 1% for class 2, 6% for class 3 and 4% for class 4).

Higher overload factors achieve lower handoff blocks but higher call drops. Conversely lower overload factors would achieve a slightly lower dropping rate and a higher Handoff blocking rate, before the NRT overload technique's usefulness to the Power Control and SIR maintenance mechanism reduces and the dropping rate as well as the Blocking rates increase. New call blocking percentages in Figure 5.5.3-3 are not influenced in this instance**.**

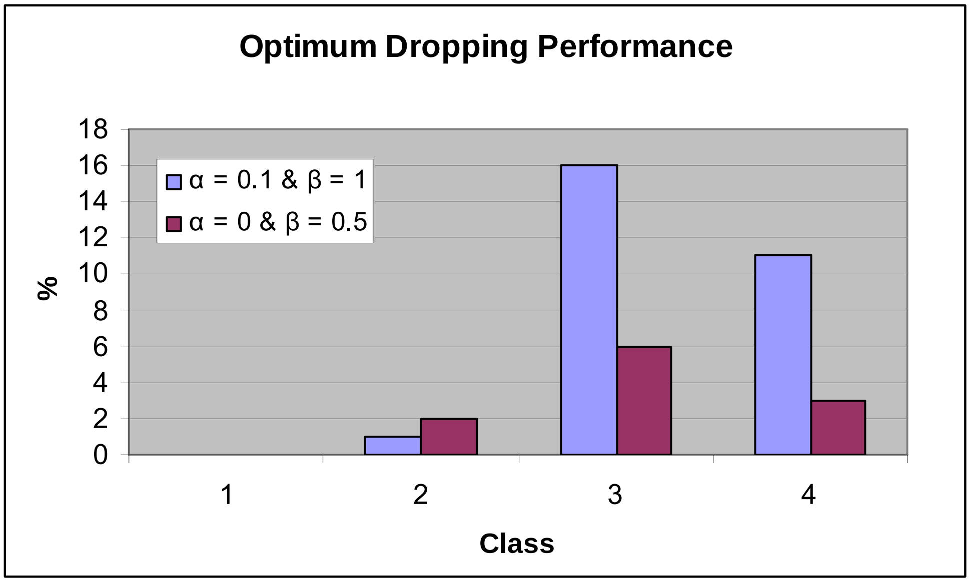


Figure 5.5.3-1: Optimum Dropping Performance (nearest 1%).

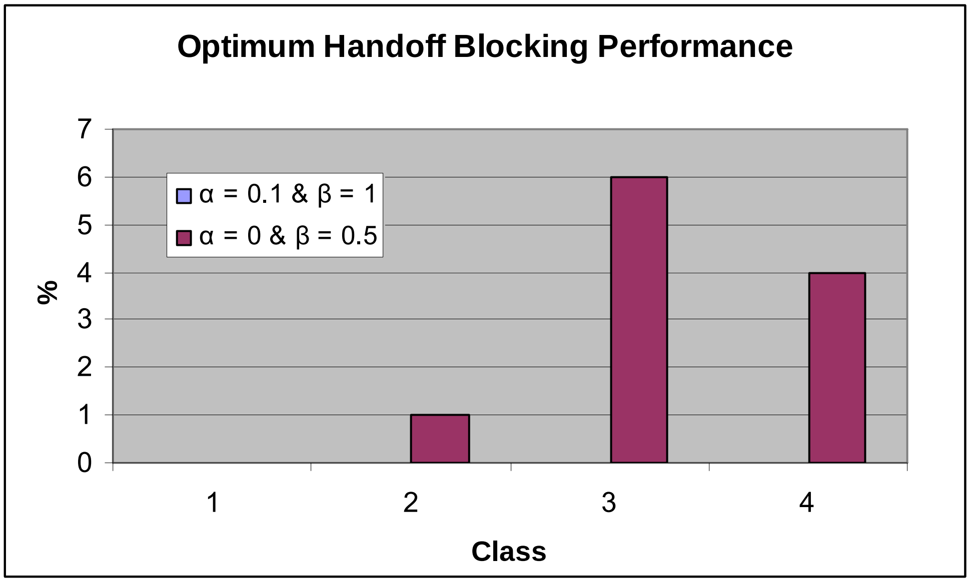


Figure 5.5.3-2: Optimum Handoff Blocking Performance (nearest 1%).

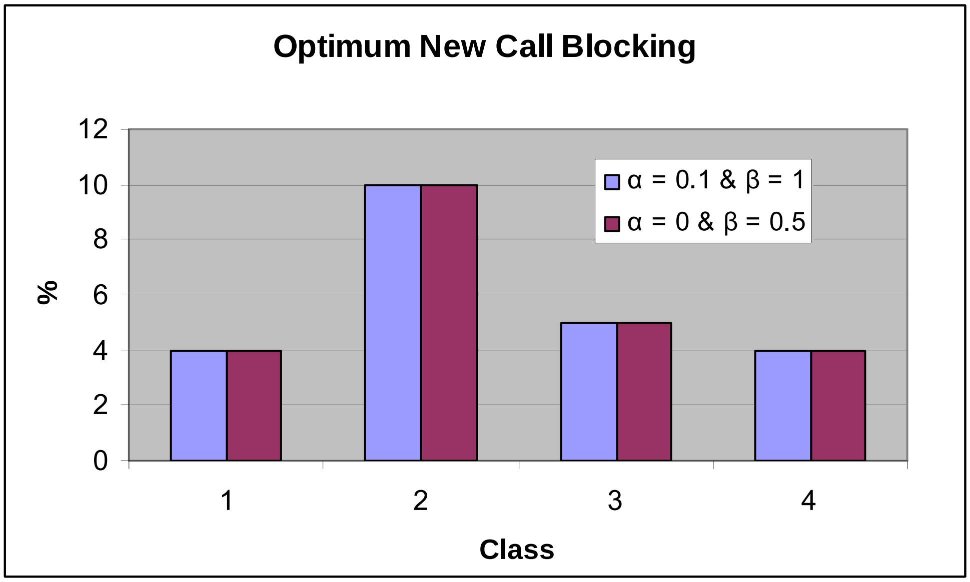


Figure 5.5.3-3: Optimum New Call Blocking Performance (nearest 1%).

5.6 Summary

This chapter introduced improvements to the algorithms discussed in Chapter 4. Initially the problem lay in the high dropping percentage of the Degradation and Overload algorithm. That problem was solved to an extent by introducing the SDD based degradation and NRT overload schemes to the Power Control Mechanism.

Further investigations into the α and β parameters of the NRT overload mechanism revealed that the NRT dropping percentage could be further reduced by selecting the parameters appropriately. Investigations revealed that high α values were not necessary to obtain good performance at the traffic conditions that were investigated. Higher β achieved lower handoff blocks but higher call drops. Conversely lower overload factors achieved a lower dropping rate and a higher Handoff blocking rate, before the NRT overload technique's usefulness to the Power Control and SIR maintenance mechanism reduces and the dropping rate as well as the Blocking rates increase. The optimum performance of the algorithm was produced when the α and β parameters were set to 0 and 0.5 respectively.

Chapter 6. Feedback Control for NRT Overload

6.1 Introduction

As discussed in earlier chapters the popularity of cellular communications has expanded wireless technology from just a voice service to a host of other multimedia applications. But challenges arise in providing multiple services from limited resources in the network, because some modern services require large amounts of bandwidth. Resource provisioning has also become a complex mechanism. Handoff failure and call dropping are seen as severe QoS errors compared to call blocking, and the importance of prioritizing RT services was also discussed in earlier chapters. We discussed SDD based degradation and NRT overload as possible solutions to provisioning. In this chapter, we describe classical feedback control theory to adaptively control the NRT overload mechanism in the Call admission and Power Control algorithm. We introduce a discrete time dynamic feedback control system that aims to keep the dropping and handoff loss rates for RT services below target values regardless of the traffic dynamics or the bandwidth requirements.

6.2 Feedback Control

The purpose of a controller is to keep a controlled variable at its desired value in the presence of disturbances from various sources and to cause it to follow changes in said desired value as closely as possible [F. G. Shinskey 1994]. Feedback control was first introduced in Greece around 1000 B.C for water level regulation and is in widespread use today in Electrical and Mechanical systems of all kinds and also in process quality control [G. Venkatesan 2002]. Examples of feedback control theory applications in communications engineering include web server enhancements [T. F. Abdelzaher and N. Bhatti 1999, T. F. Abdelzaher and C. Lu 2000], congestion control in IP routers [C. V.

Hollot, V. Misra, D. Towsley, and W.-B. Gong 2001], CPU resource management [[A. Goel, M. H. Shor, J. Walpole, D. Steere and C. Pu 2001] and resource management in media servers [Yu Chen and Qionghai Dai 2003]. Classical feedback control is illustrated in Figure 6.2-1.

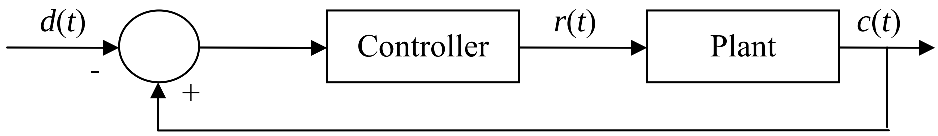


Figure 6.2-1: Classical Feedback Controller

The process that requires controlling is called the plant. The process controlling the plant is called the controller. The variable $c(t)$ represents the output of the system and the variable $d(t)$ represents the desired output of the system. The controller applies a control function to the difference between the desired and the measured outputs to produce variable $r(t)$ that works as an input to the plant to control the output. The control function is designed to regulate the output so that it is maintained at the desired value.

The classical feedback control mechanism described above is assumed to be a linear system with a linear relationship between the plant input $r(t)$ and output $c(t)$. But in the real world most systems exhibit non-linear characteristics. Non-linear characteristics include unmodelled uncertainties of the plant, noise and system parameter variations that can be expected during the operations of the plant. It is possible to design complex models to represent non-linear systems very closely. Traditionally linear systems are used as the first approximation for the ease of modelling, before fine tuning the initial approximation with computer simulations that include non-linear characteristics to obtain the closest possible approximation. Such simulations can confirm system characteristics such as stability and robustness and allows engineers to predict the true performance of the system. Such simulations are also used to tune free parameters of the control function in order to obtain optimum performance.

The model introduced in this chapter to control the NRT overload mechanism was designed following the guidelines described above. The combined proportion of RT handoff losses and RT drops was modelled as the system output we wish to control. The NRT overload parameter β was modelled as the input. The current RT dropping and Handoff failure rate is measured and compared with the desired rate in order to adjust the β parameter and allocate the overload required after a certain interval T for the digital control function. As discussed earlier, the real world communication systems are influenced by a host of other system variables such as arrival rate, call duration and QoS requirements. Our Linear Control System model will be included with such non-linear parameters later in the chapter to confirm the usefulness of our design.

6.3 Types of Feedback

The bulk of the control work in the process industry is done by members of the proportional-derivative-integral (PID) control family and this will continue to be true in to the future owing to their combination of simplicity, familiarity and robustness [F. G. Shinskey 1994]. This section will concentrate on the functionality of these controllers and their suitability for the required application.

6.3.1 Proportional Feedback

The simplest but not necessarily the best understood of the PID controllers is the proportional controller. Here the output is directly proportional to the input. The general form of the controller is given by Equation 6.3.1-1.

$$r(t) = Ke(t) \qquad \textbf{Equation 6.3.1-1}$$

Where $e(t)$ is the error which is the difference between the observed output and the desired output. K is the proportional gain of the controller. The proportional controller is considered the simplest because only one control parameter requires adjusting in order to obtain the desired performance. The stability analysis for this controller is also very simple. On the down side however, the output produces non-zero steady state error.

6.3.2 Integral Feedback

Integral action is introduced to eliminate the steady-state error. The idea is that control action is taken even if the error is very small provided that the average of the error has the same sign over a long period. This enables the controller to correct for accumulation of small errors over time (cumulative error). Integral control is often referred to as the 'memory' of a controller. The general form of the controller is shown below.

$$r(t) = r(t-1) + K_1 e(t) \qquad \textbf{Equation 6.3.2-1}$$

Where $r(t\text{-}1)$ is the previous output of the controller and K_1 is the gain of the controller. The advantage of this controller is that it is capable of reducing steady state error. On the negative side, it introduces more overhead to control desired system performance. It also reduces system stability.

6.3.3 Derivative Feedback

One of the biggest drawbacks with proportional control alone is that new desired outputs must be reached quickly while avoiding overshoot and minimizing ripple once the target is reached. Responding quickly suggests a high proportional gain; minimizing overshoot and oscillation suggests a small proportional gain. Achieving both at the same time may not be possible in all systems. But generally information about the rate of change of the plant's output is available or could be derived. If the output is changing rapidly, overshoot or undershoot may lie ahead. In such a case, the magnitude of change suggested by the proportional controller must be reduced. The rate of change of a signal is also known as its derivative. The derivative at the current time is simply the change in value from the previous error sample to the current one. The general form of the feedback controller is shown below.

$$r(t) = K_2[e(t) - e(t-1)] \qquad \textbf{Equation 6.3.3-1}$$

Where K_2 is the gain of this controller and $e(t\text{-}1)$ is the previous error. Derivative feedback improves stability and is commonly used in conjunction with proportional and/or integral feedback.

6.3.4 PID Controller

When derivative and integral terms are added to the proportional controllers to improve qualitative properties of a particular plant's response, the acronym used to describe the controller is PID. The general form of the PID controller is shown below:

$$r(t) = r(t-1) + Ke(t) + K_1 e(t) + K_2[e(t) - e(t-1)] \quad \textbf{Equation 6.3.4-1}$$

The advantage of the PID controller is that it reduces steady state error and improves stability of the system. The main drawback is that it introduces greater complexity in systems by having three control parameters that require tuning to obtain optimum performance.

6.3.5 Appropriate Feedback for the NRT Overload Mechanism

The main objective of this controller is to maintain the RT dropping and Handoff loss rates at a target value. In order to achieve this, we require an almost zero steady state error. The Integral Feedback controller has such qualities and also contains few control parameters.

6.4 NRT Overload Controller

The NRT overload controller proposed produces the single input multi output control system shown in Figure 6.4-1.

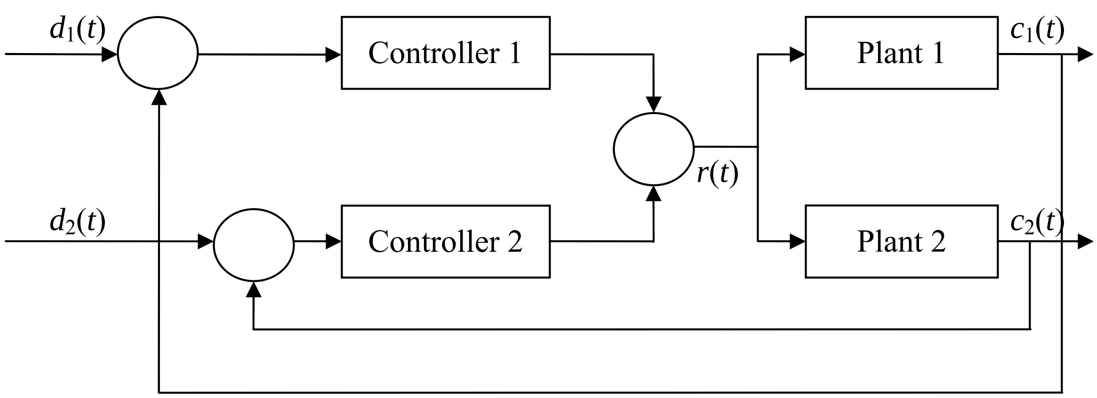


Figure 6.4-1: Integral Feedback Control System for NRT Overload

The system has separate controllers to monitor class 1 and class 2 traffic and to achieve the target loss rates. The plant input $r(t)$ is calculated using the following control function.

$$r(t) = r(t-1) + K_1 \sum_{i=1}^{2} (c_i(t) - d_i(t)) \qquad \textbf{Equation 6.4-1}$$

Where $r(t)$ represents the current value of the overload factor β at time t and $r(t\text{-}1)$ represents the value of the overload factor β at time t-1. The target and the observed loss rates at t are represented by $c_i(t)$ and $d_i(t)$ respectively for the RT class i. K_1 is the control parameter that requires tuning to obtain optimum performance.

The performance of this controller depends on two control parameters, namely the gain constant K_1 and the update interval T. K_1 is chosen on the standard theoretic trade off between speed of response (faster for increasing K_1) and the stability (poorer for increasing K_1). The update interval T should be large enough for the system to settle down and small enough for the controller to respond effectively. Tuning of these parameters will be discussed in the following section.

6.5 Controller Parameter Tuning and Performance Evaluation

The best performing Call Admission and Power Control model developed at the end of Chapter 5 was used to evaluate the controller. The NRT overload mechanism of the Call admission and Power Control model was reprogrammed to manipulate β according to the controller model discussed above. Note: α = 0 will be set for the controller.

This simulation model was used to tune the control parameters T and K_1 and to predict the true performance with varying nonlinearities expected to be observed during operation. The goal of the controller is to keep the Class 1 and Class 2 handoff and drop rates below a certain target value and to achieve the best effort performance for NRT services.

6.5.1 Common Nonlinearities

The session arrivals considered for the call admission and power control models discussed in Chapters 4 and 5 were Poisson processes with a worse case λ = 1. Here the Poisson arrival rate will be set to λ = 1 during controller tuning. The session durations were set according to the service class and the variety of durations can be expected to test the controller extensively. The other important nonlinearity of interest is the traffic ratio. Here, a constant ratio of 15% class 1, 5% class 2 and 80% for NRT traffic is set during tuning.

6.5.2 Performance Metrics

Dropping and Handoff blocking are equivalent from the user perspective and are considered the worst form of wireless transmission errors. This is particularly true for Real-Time services because a voice call disconnecting is far more annoying to a user than the interruption of a webpage download or an e-mail download.

Therefore most call admission and resource allocation schemes in existence function towards guaranteeing the performance of Real-Time traffic and reducing dropping and Handoff blocking. This controller aims at keeping the combined RT Dropping rate and the Handoff blocking rate for class 1 and class 2 services within 1.5%. The NRT dropping and Handoff blocking is considered to be of secondary importance and the best performance after reaching the target of RT traffic will be considered. New Call blocking (regardless of the service class) is considered of least concern and is treated likewise.

6.5.3 Tuning Control Parameters

Control Parameters T and K_1 were tuned using simulations at a high Poisson arrival rate of $\lambda = 1$. Figure 6.5.3-1 shows the plots for the variation of the combined RT dropping and handoff loss rates with varying feedback time intervals T. Figure 6.5.3-1(a) shows the RT loss rates increasing considerably for feedback intervals greater than 10s. Figure 6.5.3-1(b) with a shorter graphing interval further clarifies the picture and shows consistently good performances between the intervals of 0.1s to 7.5s.

Large intervals make the controller ineffective by not being able to identify instances of congestion quickly enough to allocate a larger overload. This produces larger error rates for RT traffic. Figure 6.5.3-2 confirms this phenomenon with its decreasing trend for NRT error with high feedback intervals. Large intervals tend to miss the instances of high congestion and the required NRT overload is not allocated in time. The overload increase comes too late (a large number calls in need of resources are already dropped) and may never be used, which means the burden on the NRT error rate becomes less. If the Feedback interval is too low the system does not have sufficient time to settle with the new NRT overload allocation and more than a required overload maybe allocated. This is another reason for high NRT error rates at low feedback intervals.

Because the NRT error rate has a decreasing trend for high Feedback intervals, the critical variable is the RT error rate. The highest Feedback time interval that produces the best RT error performance needs to be identified. We have chosen an interval of 7s because it holds the RT error rate at 1.5% comfortably and it produces the smallest NRT error rate possible before the RT error performance starts degrading.

Once the optimum Feedback interval was identified, the next goal is to identify the optimum controller gain K_1. The following experiments for varying K_1 were conducted with a feedback interval of 7s. Figure 6.5.3-3 shows that the RT loss rates can be maintained for almost any arbitrary K_1 value within the interval 0.5 – 10. But Figure 6.5.3-4 shows the performance of NRT loss rates degrade with the increasing K_1. High K_1 values produce NRT overload allocations that are greater than or less than required. Such overshoot causes the NRT error performance to suffer (Still larger K_1 values are capable of also degrading the performance of RT error rates due to overshoot). Extremely small K_1 values make the controller ineffective with smaller than required overload allocations making the RT error performance suffer.

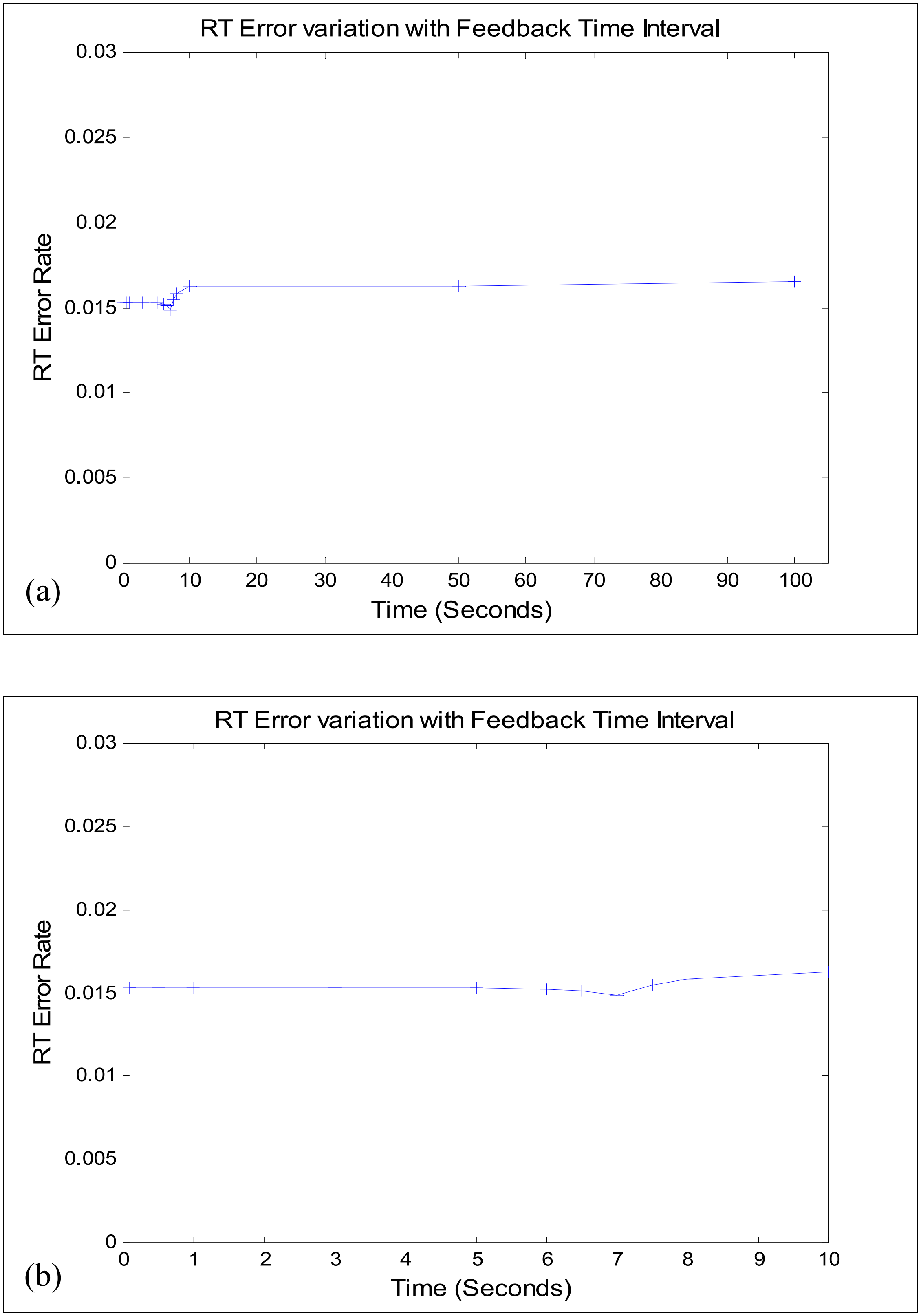


Figure 6.5.3-1: Parameter Tuning – RT Error Variation with Varying Feedback Intervals.

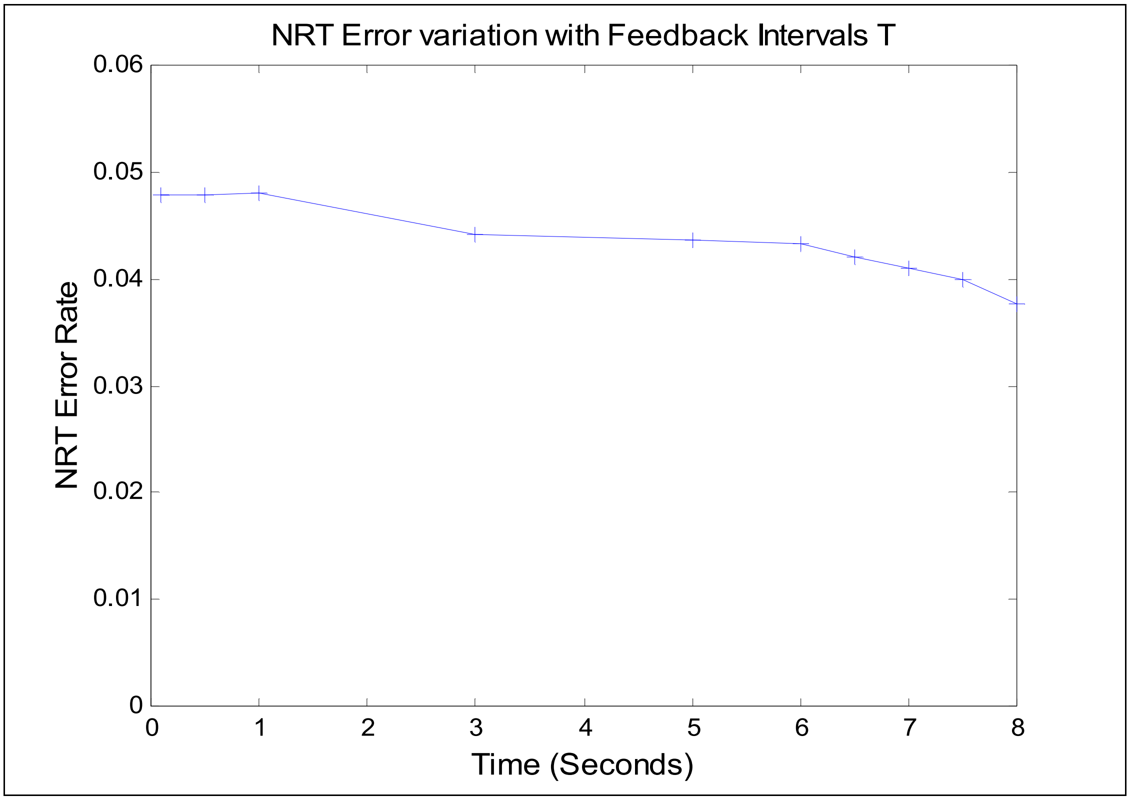


Figure 6.5.3-2: Parameter Tuning - NRT Error with Varying Feedback Intervals.

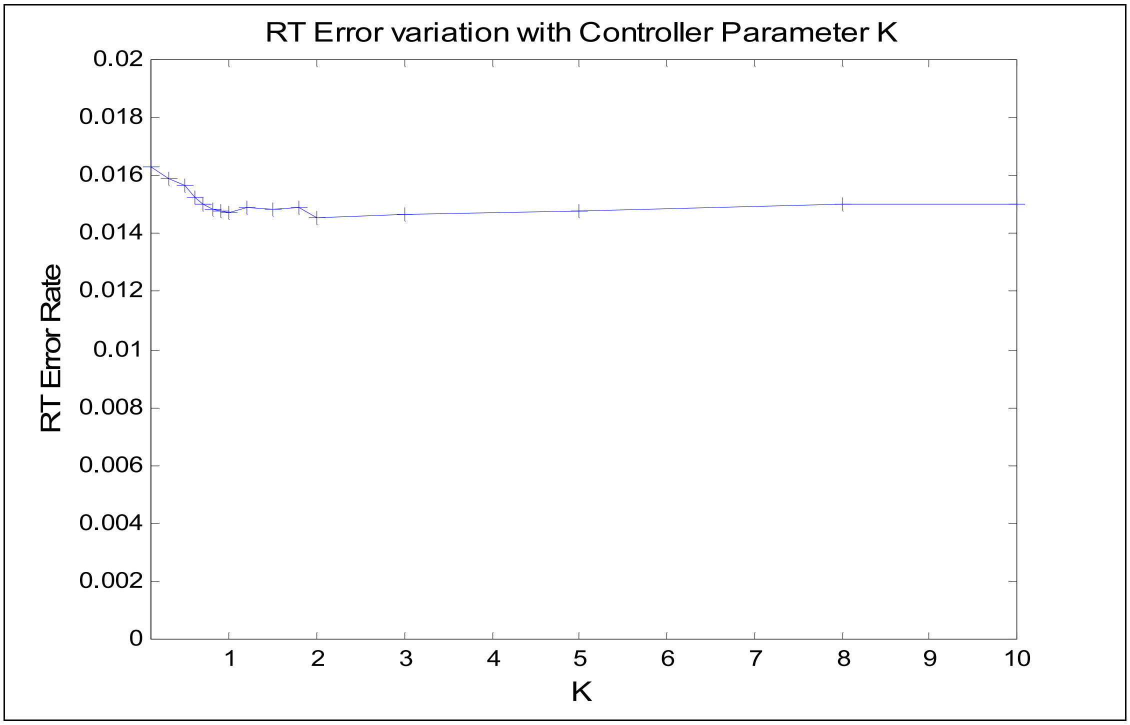


Figure 6.5.3-3: Parameter Tuning – RT Error Variation with Varying K_1.

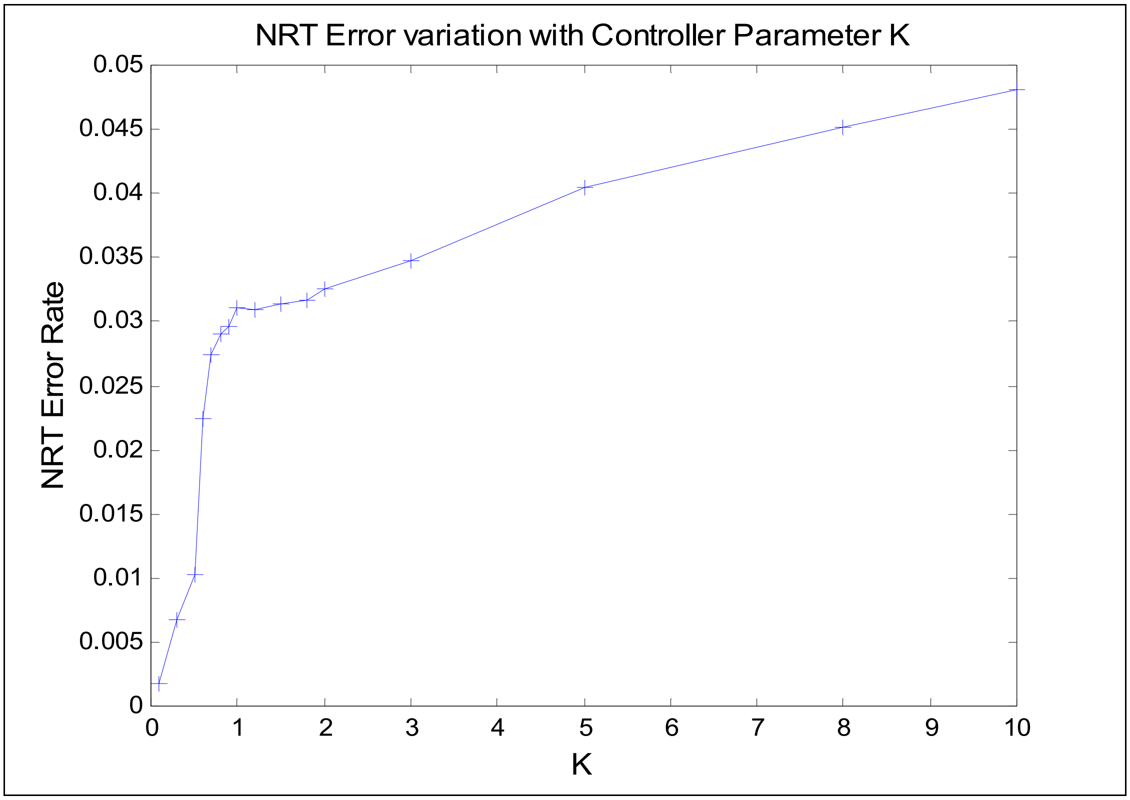


Figure 6.5.3-4: Parameter Tuning – NRT Error Variation with Varying K_1.

Therefore the required K_1 value has to be selected from an interval of around 0.5 – 2. The large dip of NRT error rates for low K_1 values in Figure 6.5.3-4 coincides with the degradation in performance of RT error rates at low K_1 values. The reason is the controller is designed to prioritize RT performance at the expense of NRT calls and therefore the degradation in performance for RT error rates results in desirable NRT error rates. The graphs were redrawn in Figure 6.5.3-5 and Figure 6.5.3-6 to obtain a clearer picture of the controller performance within the interval 0.5 – 2.

In this case the lowest controller parameter K_1 value that produces the best RT error performance needs to be identified. We have chosen a K_1 value of 0.7 because it is capable of holding the RT error rate at 1.5% while producing the smallest NRT error rate possible before the RT error performance starts degrading.

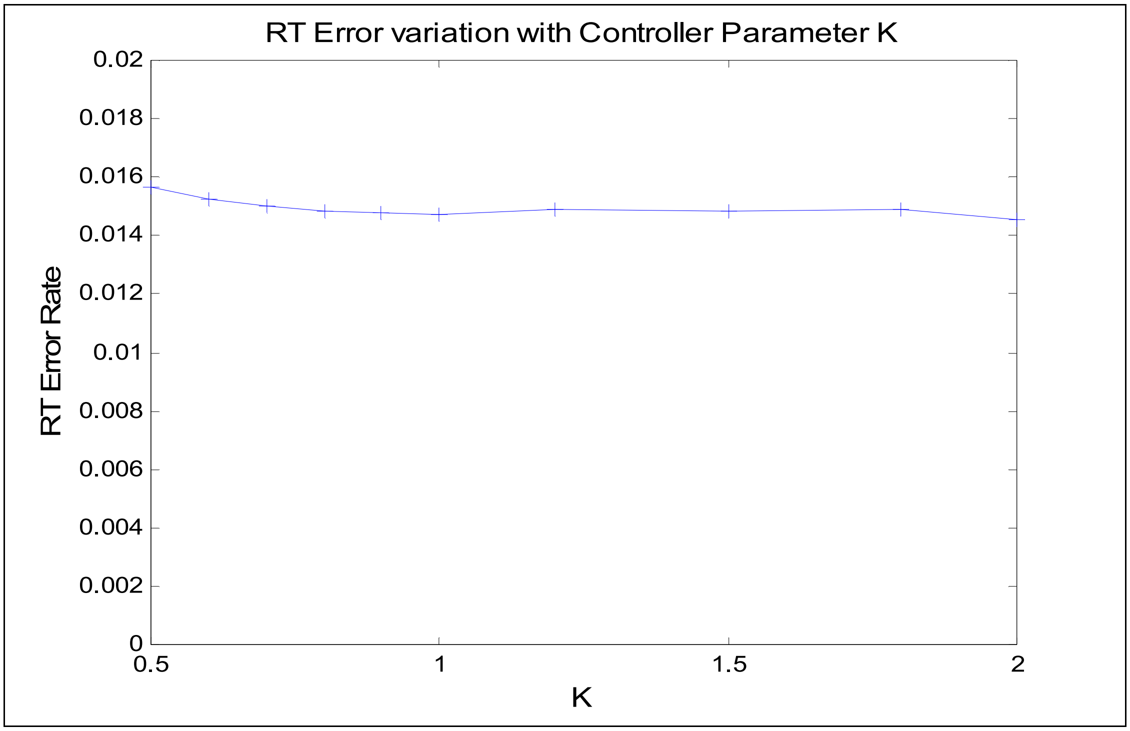


Figure 6.5.3-5: Parameter Tuning - RT Error with Varying K_1 (shorter interval).

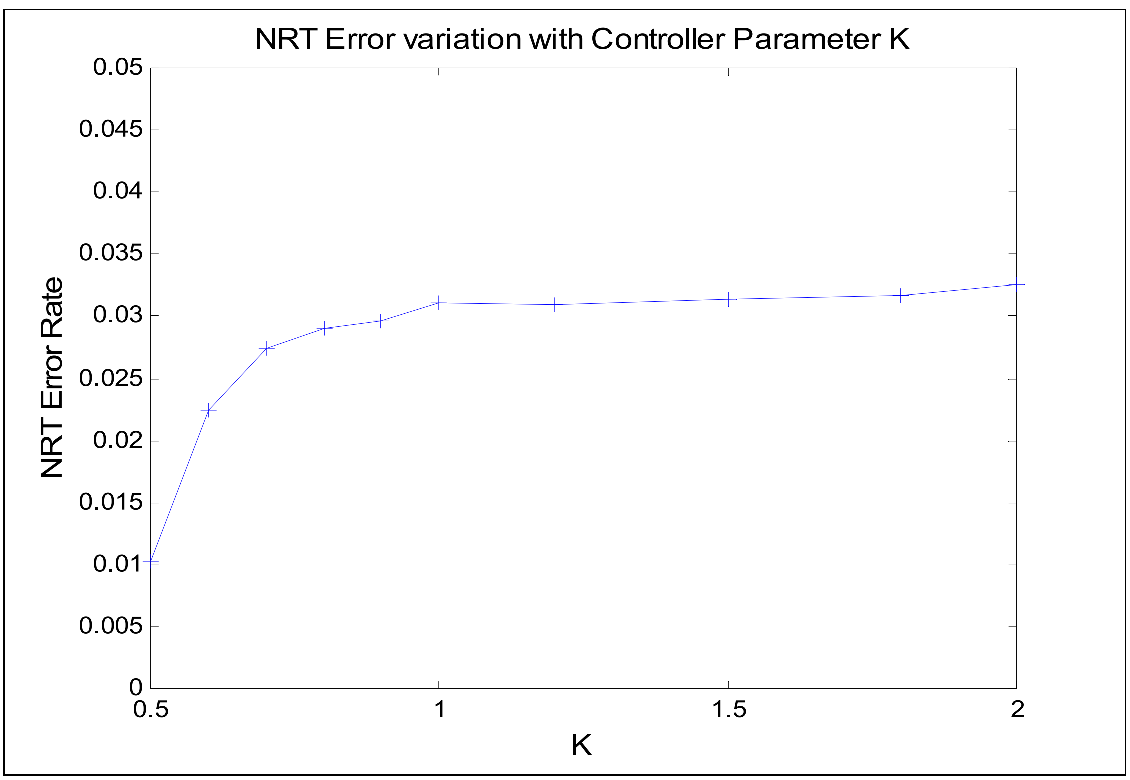


Figure 6.5.3-6: Parameter Tuning - NRT Error with Varying K_1 (shorter interval).

6.6 Performance Evaluation

During operation of a cellular base station, it may suffer from congestion and may become temporarily overloaded due to the variability of traffic conditions and the diverse requirements of different users and services. The system may face difficulty in maintaining low RT failure rate and a reasonable NRT failure rate during such circumstances. This is precisely the reason we chose a high arrival rate and a heavy load requirement when designing Call admission and Power control algorithms and when tuning the controller.

Further experimentation was conducted to compare the Call Admission and Power Control algorithm with the controller against Call Admission and Power Control without a controller. Experiments were performed with various traffic arrival rates and traffic loads to understand the usefulness of the controller. What we have defined as varying traffic load is the varying ratios of Class 1, 2, 3 and 4 services offered to the system (A higher percentage of Class 1 and 2 services place a greater burden on the system as they have higher bandwidth requirements.). The arrival rate is varied by varying Poisson arrival rate λ. Figure 6.6-1 investigates RT Error rate behaviour at various arrival rates of the Call Admission scheme with a controller to allocate NRT overload factors dynamically against Admission schemes that have fixed α and β parameters. Note: the controller keeps $\alpha = 0$ and only varies β. The traffic load was fixed at 15% class 1, 5% class 2 and 80% NRT calls.

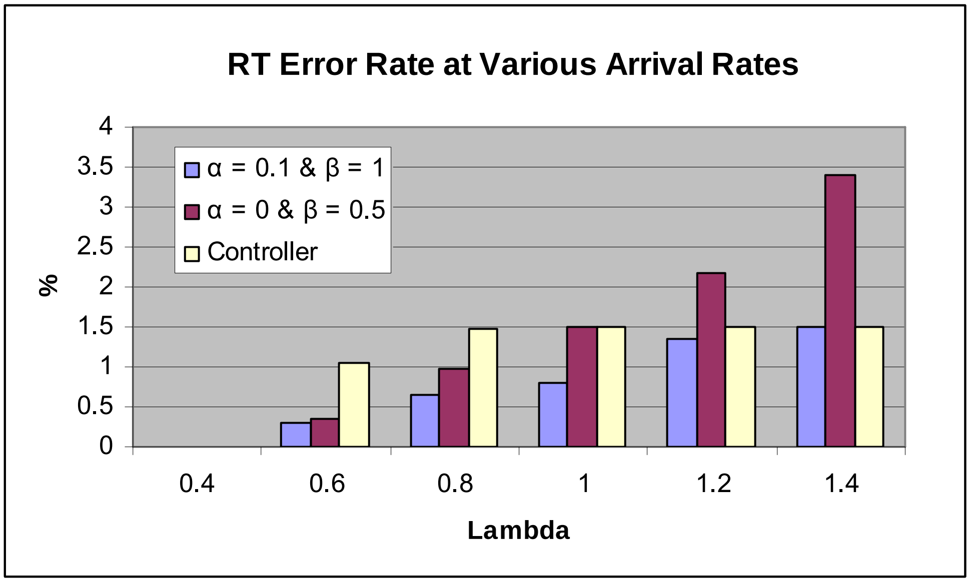


Figure 6.6-1: RT Error Rate Variation for Various Arrival rates (Nearest 0.1%).

The graph above shows that the controller is capable of holding the error rate at 1.5% for the high λ values. For the lower λ values the controller produces an error rate higher than the mechanisms with fixed α and β parameters. This is not a shortcoming of the Feedback controller; the reason for the higher percentage being the target we have specified for the controller. Because the target RT error rate is set to 1.5% the controller allows RT errors to occur until the rate reaches 1.5%. But the target rate can be adjusted depending on the traffic conditions expected. If the type of traffic expected for a certain system is low the target error rate can be set to a lower value. Figure 6.6-2 shows the NRT error rates for the three mechanisms.

The primary concern of the controller is to keep the RT error rate below 1.5%. The NRT error performance is best effort. The Controller produced the lowest NRT error rates for the majority of the arrival rates tested and held the RT error rates below 1.5%. The mechanism with NRT overload factors set to $\alpha = 0$ and $\beta = 0.5$ produced lower NRT error rates when the traffic arrival rate was set to λ = 1.4. But this NRT error rate has come at the expense of a higher RT error rate. Therefore it can be concluded that the controller is very effective in obtaining the optimum results from this Admission Control and Power Control Mechanism. Figure 6.6-3 shows the RT Error rate variation with various traffic loads and Figure 6.6-4 shows NRT Error rate variation with various traffic loads. The arrival rate here is kept at λ = 1.

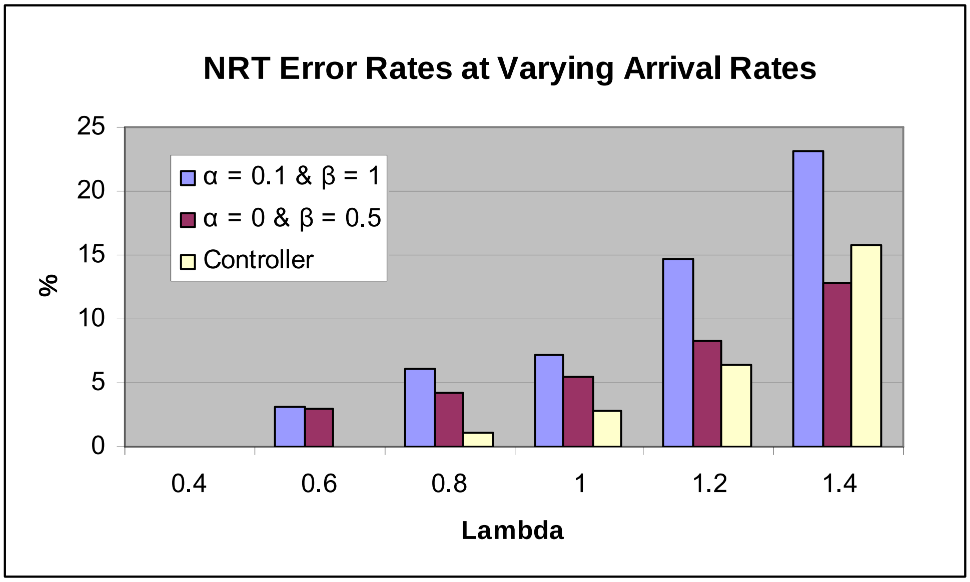


Figure 6.6-2: NRT Error Rate Variation for Various Arrival rates (Nearest 0.1%).

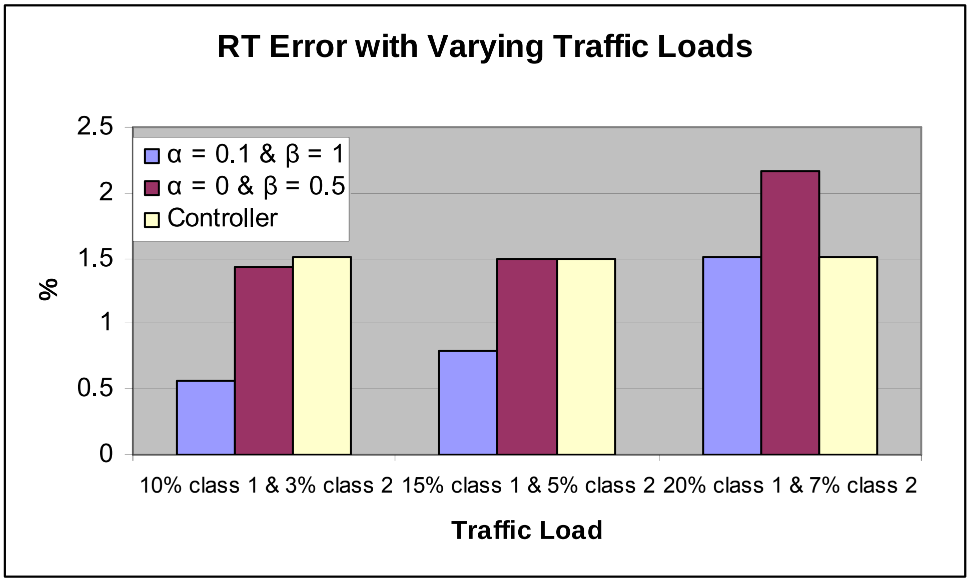


Figure 6.6-3: NRT Error Rate Variation for Various Traffic Loads (Nearest 0.1%).

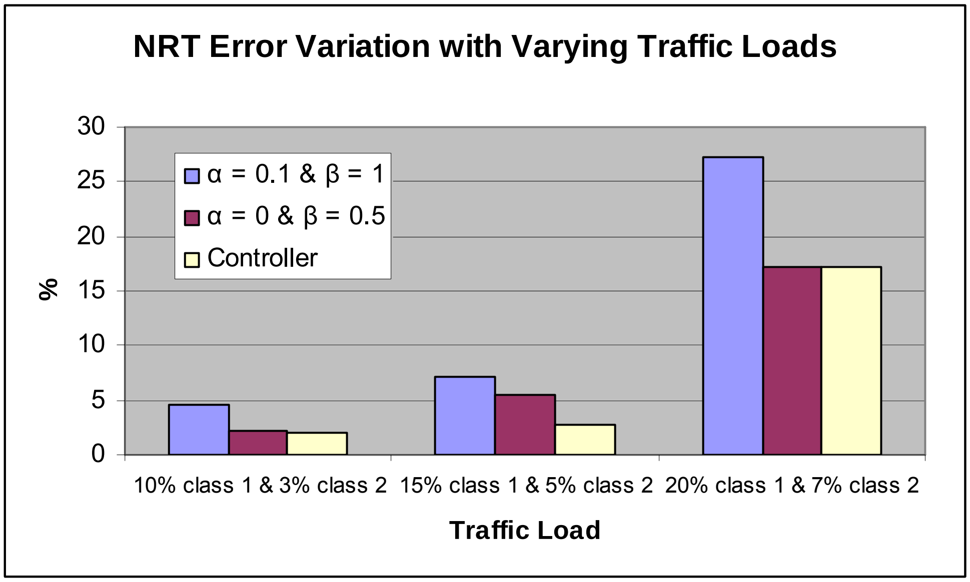


Figure 6.6-4: NRT Error Rate Variation for Various Traffic Loads (Nearest 0.1%).

Figure 6.6-3 shows that the controller is capable of maintaining the RT Error rate below 1.5%. It has also maintained a comparatively low NRT error rate. The controller will continue to perform well and maintain a low RT error rate under all circumstances apart from the extreme hypothetical situations where only an extremely low NRT user percentage is available with a high arrival rate which forces the system into congestion.

Figure 6.6-5 shows some insight into the operation of the controller. Figure 6.6-5 (a) shows the fluctuation of the RT Error rate over time during a simulation run. Figure 6.6-5 (b) shows how the controller changes the NRT overload factor β according to the fluctuating RT Error rate.

An interesting observation is the spiky waveforms in Figure 6.6-5 (a). In fact these are not as spiky as they appear. The reader must keep in mind that the Power Control and Call admission processes take place over millisecond intervals during simulations. This means the waveforms would not be spiky over millisecond intervals as shown in Figure 6.6-6. But when graphed over the entire duration of the simulation all that can be observed are vertical spikes that indicate comparatively high dropping rates observed over small intervals of congestion. Such a graph is not indicative of the Controller performance.

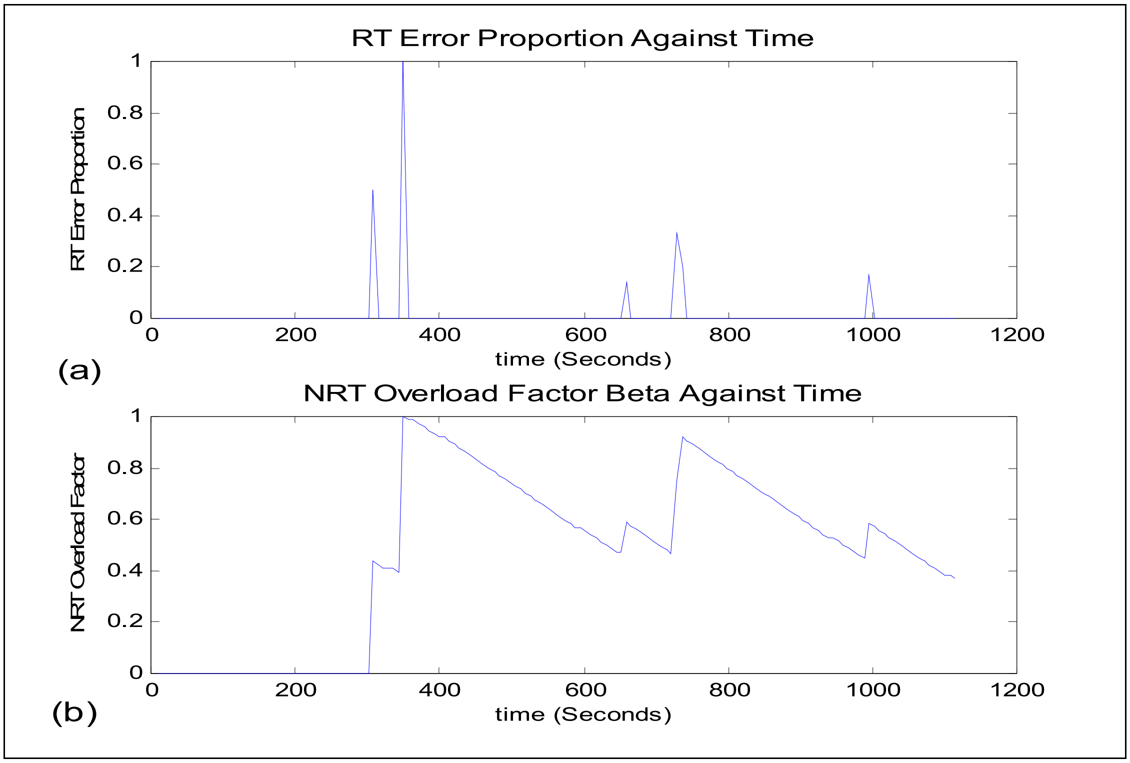


Figure 6.6-5: Example of the operation of the Controller during a trial run.

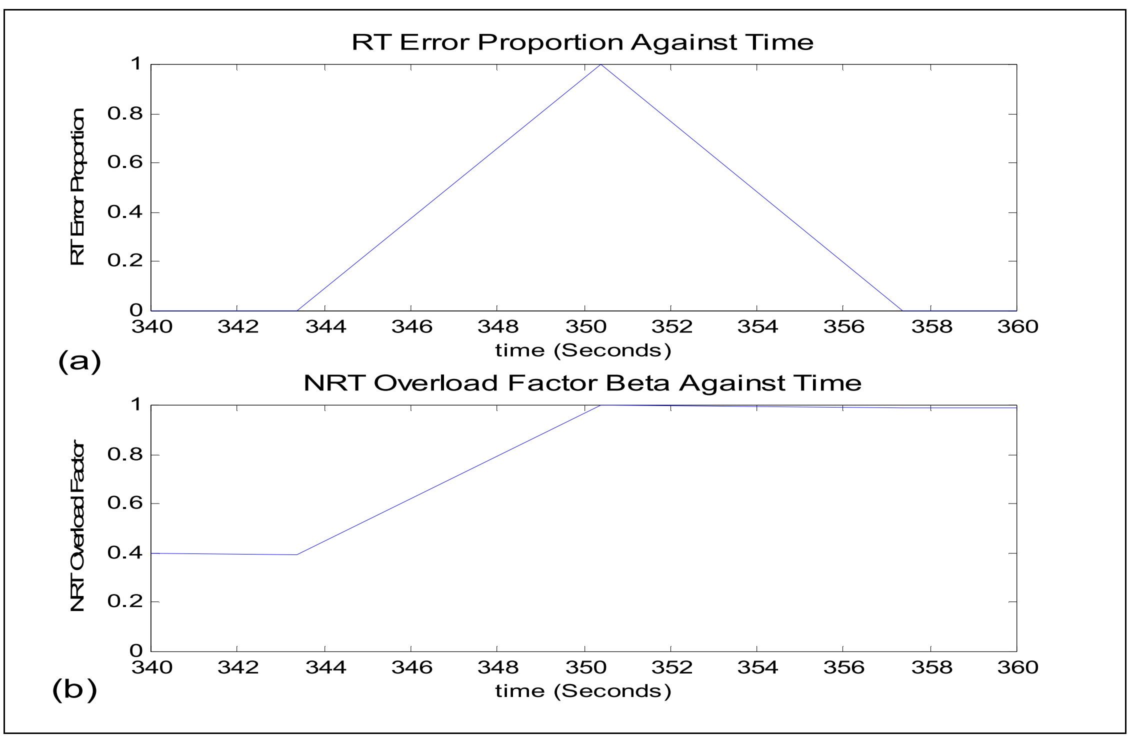


Figure 6.6-6: Example of the operation of the Controller during over a Shorter Interval.

Our intention here is to clearly show the controller's ability to follow the RT Error Rate and therefore a graph of RT Error Proportions with a running average is shown in Figure 6.6-7 along with the NRT overload factor variation.

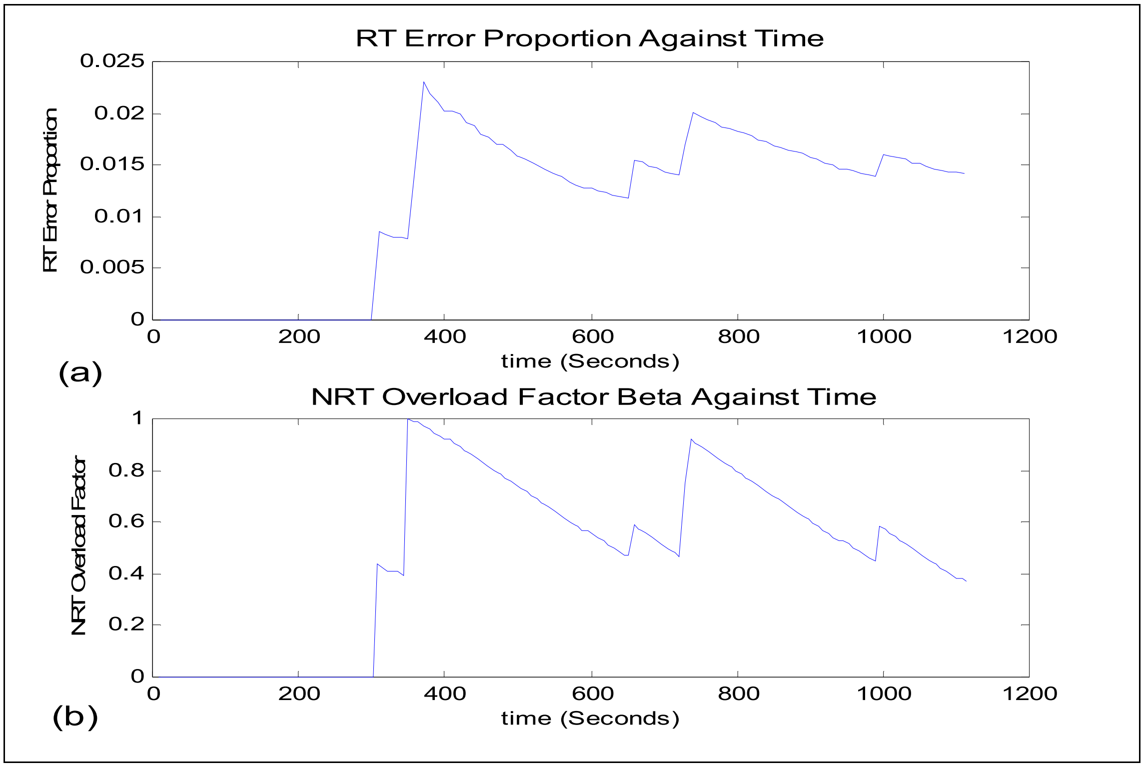


Figure 6.6-7: Operation of the Controller with running Average for RT Error

In integral controllers, past differences between the observed RT Error rate and the Target Error rate can charge up the integration to some value that will remain in memory even if the difference becomes zero for a certain time period because an Integral controller has the ability 'remember' past differences. This phenomenon can be observed in Figure 6.6-5 where the NRT overload factor β does not respond immediately to RT Error fluctuations. Figure 6.6-7 shows very similar waveforms for both the RT Error proportion and NRT overload factor. This is another indication of an Integral controller's ability to remember past fluctuations because the running average also takes in to account the past fluctuations of the waveform. This shows that the NRT overload factor β follows the RT Error rate very closely.

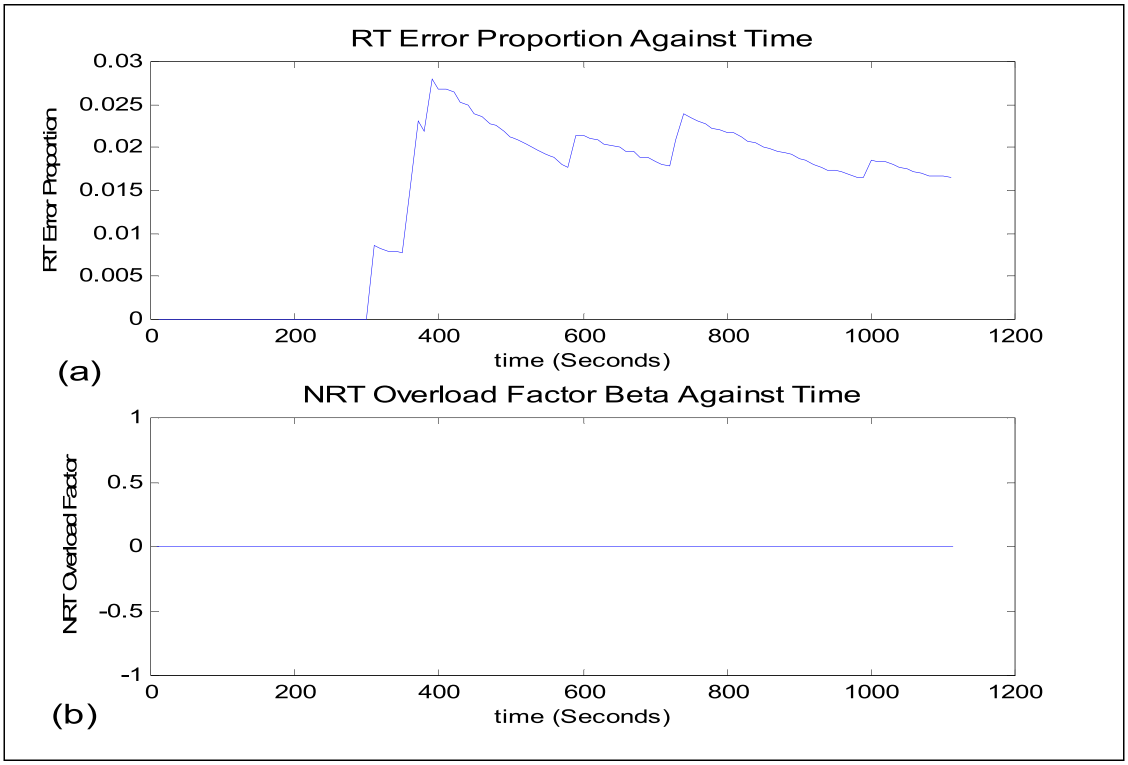


Figure 6.6-8: Example of Algorithm with Beta set to 0 (Running average).

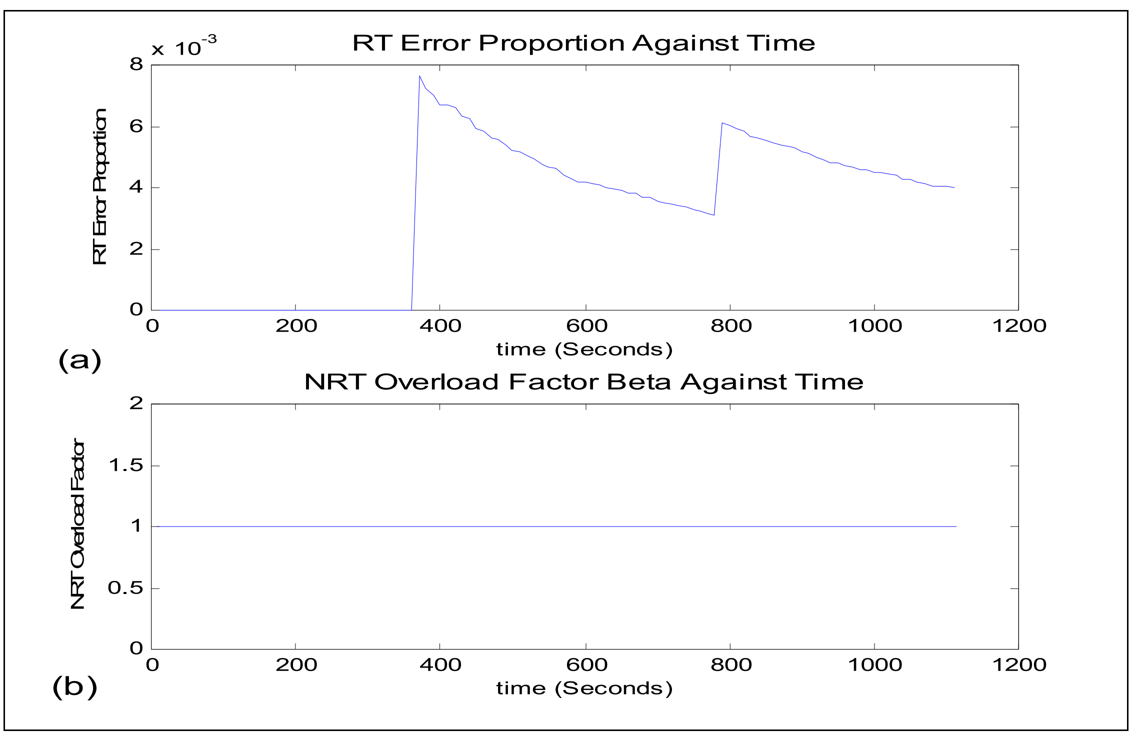


Figure 6.6-9: Example of Algorithm with Beta set to 1 (Running average).

Examples of algorithms with constant NRT overload settings simulated under the same conditions as the controller example above are shown in Figures 6.6-8 and 6.6-9. Figure 6.6-8 has the NRT overload parameter β set to 0 for the duration of the simulation and Figure 6.6-9 has NRT overload parameter β set to 1 for the duration of the simulation. The NRT Error results of all three algorithms under the same conditions are given in Figure 6.6-10.

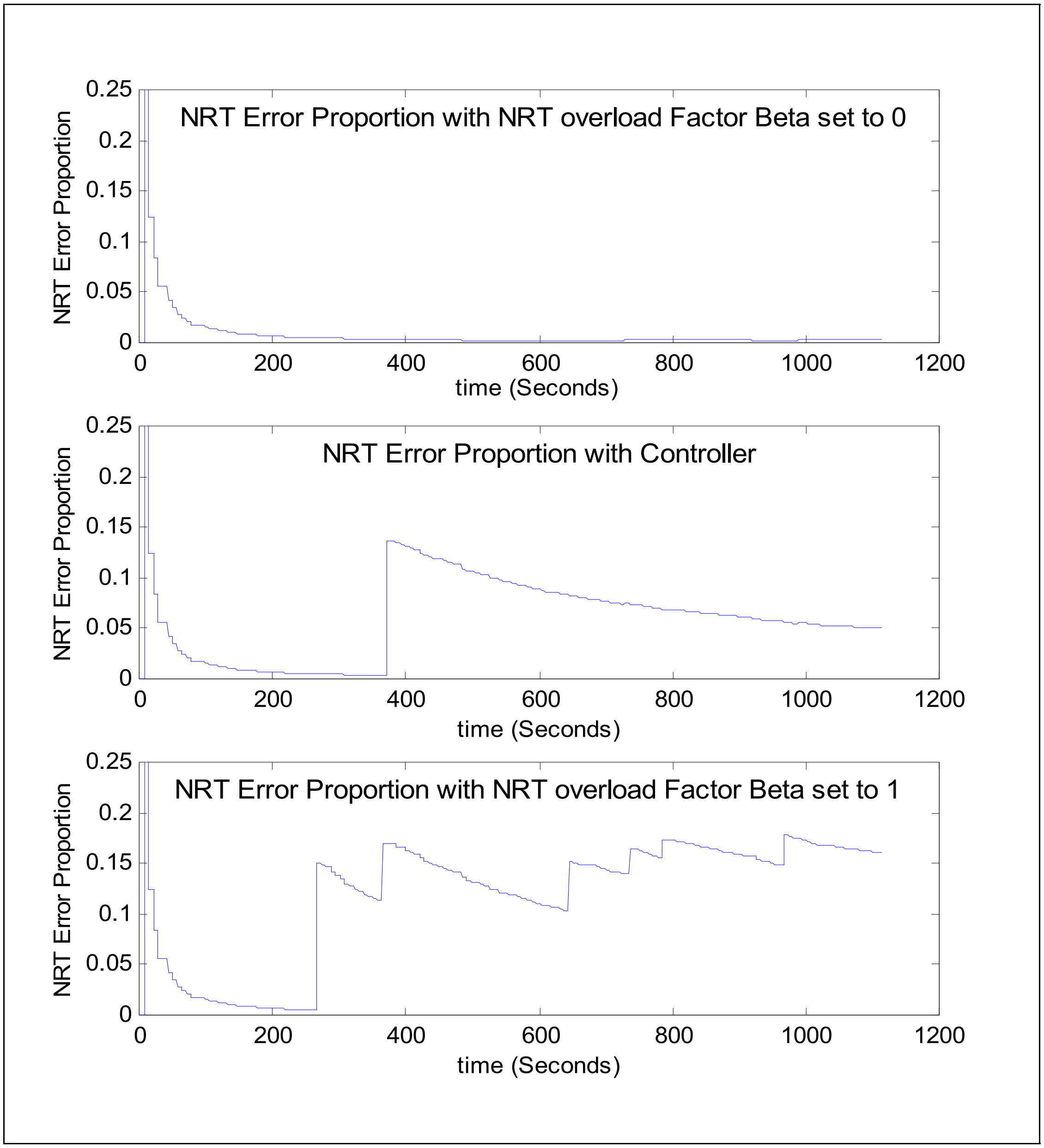


Figure 6.6-10: NRT Error for the Algorithms during the simulation run (Running average).

The algorithm with NRT overload parameter β set to 0 produces a final RT Error rate greater than 1.5% (The controller has settled down to this figure faster.) while the algorithm with NRT overload parameter β set to 1 produced a very low RT Error performance. The NRT Error rate is very low for the algorithm with NRT overload parameter β set to 0 and it is high for the algorithm with NRT overload parameter β set to 1 (The algorithm with our controller is the required medium.). The algorithm with a controller to the NRT overload mechanism displays a clear ability to maintain the RT Error rate at a target value over long periods while maintaining a relatively low NRT error rate. The high NRT error proportion shown early in all three graphs of Figure 6.6-10 are due to an NRT error that is not caused by the NRT overload mechanism (Which is the reason why it is present in the graph with the NRT overload factor β set to 0.). This could be due to the lost user / users requesting more bandwidth than the channels are capable of offering.

6.7 Summary

In this chapter we have shown that the RT Error rates in a multi-service WCDMA network can be regulated accurately using feedback control regardless of the uncertainties and traffic dynamics. The performance of the feedback model was studied through simulation experiments. Experimental results showed much superior performance of the proposed feedback models when compared to techniques that did not utilise feedback control for the same problem. This indicates the usefulness of feedback controllers for resource management problems in multi-service networks.

However it is important to realise that we used a simple linear feedback controller and the controller parameters were tuned using simulations. This works in many practical situations but may not provide a general solution. The proposed controller model is not restricted to the considered network scenarios and can be applied to other scenarios with little modification.

Chapter 7. Conclusions and Future Work

7.1 Conclusions

Unlike in an FDMA or TDMA system the number of users in the system does not have a fixed upper bound as no “channels” are present. Instead the system is interference limited and has a soft capcity which changes depending on the interference felt at the base station at a given time. If interference increases beyond an acceptable level the system becomes unstable and may experience call dropping and call blocking. Admitting a new call always increases the interference level in the system. Hence a robust method of accepting or blocking potential users i.e. a call admission control (CAC) technique is required.

Even with efficient Admission Control, congestion could still be caused, mainly by users moving from one area within the cell to another area. When affected by congestion the output powers are rapidly increased by the fast closed-loop power control until one or several transmitters are operating at their maximum power. The connections unable to achieve their required *SIR* levels are considered to be useless and are only adding interference to the system. Therefore a procedure to reduce congestion by removing such users is required.

In chapter 3 the main approaches mentioned in the literature for Call Admission Control and Power Control were presented. The main advantages and drawbacks of the schemes were also highlighted. The QoS aware Power Control and Handoff Prioritization scheme introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] and the Received Power based simulation model discussed in [A. Capone and S. Redana 2001] were considered two of the best modern solutions to Call Admission and Congestion Control and hence were used as the base of this thesis.

The mechanism introduced by [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] was found to be every effective in minimizing handoff failures and the work introduced by [A. Capone and S. Redana 2001] included a very good simulation model. But the criticisms of the two algorithms were that they led to higher dropping percentages when simulated with modern High Bandwidth service requirements. Particularly the performance of RT services was very poor. It was observed that the algorithm introduced in [T. Rachidi, A. Y. Elbatji, M. Sebbane, and H. Bouzekri 2004] produced a higher dropping rate because the degradation and overload scheme accepts more users than the simple scheme would in [A. Capone and S. Redana 2001]. The main improvements had to be made to the Power Control mechanism.

The first suggested improvement was to include the degradation and overload techniques to the Power Control Mechanism. Also since Call dropping is considered to be far worse than call blocking, the NRT overload consideration at New Call admission was removed and was only utilized in the Power Control process and the Handoff Admission Process. The suggested improvements were instantly successful with the Real-Time dropping percentages of class 1 services reducing by almost 16% and class 2 reducing by a mammoth 30%. Only the NRT services have shown poorer dropping rates. This was considered a reasonable outcome because Real-Time services are the higher priority services. The reason for improvement was the success of the SDD based degradation mechanism and the NRT overload scheme. The Degradation mechanism lowered the Bandwidth requirements of users already admitted to the system and reduced the load for the system and the NRT overload mechanism prioritized High Bandwidth RT users who may otherwise be dropped.

Two parameters of the NRT overload mechanism, α and β, were identified as being capable of reducing the NRT dropping percentage even further. We redefined the system of equations belonging to the NRT overload mechanism and investigated the effect of α and β on the performance of the Call Admission and Power Control process by varying α and β over a series of simulation experiments. Increasing α produced high dropping rates for NRT services while the RT dropping rates were improved very little. The only improvements for the Handoff Blocking rates were observed when α was varied between 0 and 0.1. No major improvements were observed in the New Call blocking rates. β, by definition, limits the number of RT users the system is willing to accept at the expense of NRT users. As expected initially the number of dropped RT calls decreased with increasing β. But after β = 0.5, no significant improvement is observed.

The number of dropped NRT calls increased with increasing β. The number of blocked NRT handoff calls remained relatively unchanged with varying β. It was concluded that NRT handoff call acceptance is more significantly influenced by α. RT handoff call rejection improved with increasing β. The New call blocking rate showed a slight increase with increasing β because when β is high, more RT calls that consume a large quantity of resources can be maintained and less resources are available to accept new calls.

The optimum solution in terms of α and β were produced when they were set to 0 and 0.5, respectively. It was concluded that higher overload factors achieve lower handoff blocks but higher call drops. Also lower overload factors would achieve a slightly lower dropping rate and a higher Handoff blocking rate, before the NRT overload technique's usefulness to the Power Control and *SIR* maintenance mechanism reduces and the dropping rate as well as the Blocking rates increase.

As the next task a discrete time dynamic feedback control system that aims to keep the dropping and handoff loss rates for RT services below a target value regardless of the traffic dynamics or the bandwidth requirements was designed. A simple Integral Feedback controller was chosen for this task because a controller that is capable of reducing steady state error was required. The controller was proposed for the NRT overload mechanism by controlling β according to traffic dynamics and the NRT error rate was left as best effort. The controller parameters were tuned using simulations and the final result was benchmarked against two algorithms that had fixed α and β values by simulating in environments under various Poisson call arrival rats and traffic loads. The NRT overload mechanism with a controller performed best by holding the RT error rate at the required target value and by producing comparatively lower NRT error rates. But it is important to realize that we used a simple linear feedback controller and the controller parameters were tuned using simulations. This works in many practical situations but may not provide a general solution. The proposed controller model is not restricted to the considered network scenarios and can be applied to other scenarios with little modification.

7.2 Future Work

The ideas in this thesis were proposed for the types of Services and Service requirements that are expected today just like the algorithms introduced in [A. Capone and S. Redana 2001] were for that time period. But as shown in this thesis the algorithms described in [A. Capone and S. Redana 2001] failed in the face of traffic requirements of today. Similarly the work introduced in this thesis would also become obsolete in the face of traffic requirements that will be generated with the evolution of time. Therefore continuing research is essential for subjects such as Call Admission Control and Power Control.

Also the discussion and proposals in this thesis were targeted at the Downlink of WCDMA networks and further efficiency should be obtained through further research in to the Uplink path of the network. This thesis made assumptions about the amount intercellular interference the base station and mobiles are likely to suffer. Better and much more accurate models for intercellular interference can be developed as a future project.

The Admission schemes discussed in this project considered techniques to be utilized once the decision is made to admit calls in to the network. Further research can be conducted on predictive schemes to utilize assistance from neighboring base stations during periods of congestion. The SDD based degradation scheme and the NRT overload mechanism could become even more useful if they are utilized by a team of base stations rather than being implemented on just one base station.

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Appendix A: Confidence Interval Calculation

The rule of thumb according to [J. T. McClave and T. Sincich 2000] is to calculate small sample confidence intervals for sample sizes less than 30. We have chosen to calculate small-sample confidence intervals (as opposed to large sample confidence intervals) because we conducted just 20 trials for each experiment due to time constraints. We are 95% confident that all result variations obtained in this research are within 0.0001 from the mean. The procedure for obtaining a small-sample confidence interval for population mean μ is summarized below:

$$\overline{x} \pm t_{\alpha/2}\left(\frac{s}{\sqrt{n}}\right)$$

Where:

- $\overline{x}$ is the sample mean.
- $t_{\alpha/2}$ is the critical t value based on $(n-1)$ degrees of freedom.
- s is the sample standard deviation.
- n is the number of trials conducted (sample size).

Appendix B: Statistics of All Results Presented

The table below shows the statistical data of the analysis conducted on the two existing algorithms found in literature (presented in Chapter 4).

	Simple		SDD + NRT at Call Admission	
Class	Mean	STD	Mean	STD
1	0.067	7.2×10^{-4}	0.000	0
2	0.148	6.3×10^{-4}	0.000	0
3	0.031	8.6×10^{-4}	0.000	0
4	0.012	5.5×10^{-4}	0.000	0

Table B-1: Handoff Blocking Rate

	Simple		SDD + NRT at Call Admission	
Class	Mean	STD	Mean	STD
1	0.029	8.1×10^{-4}	0.000	0
2	0.080	8.7×10^{-4}	0.000	0
3	0.049	9.1×10^{-4}	0.000	0
4	0.040	8.9×10^{-4}	0.000	0

Table B-2: New Call Blocking Rate

	Simple		SDD + NRT at Call Admission	
Class	Mean	STD	Mean	STD
1	0.147	6.1×10^{-4}	0.159	7.3×10^{-4}
2	0.315	8.3×10^{-4}	0.334	5.3×10^{-4}
3	0.083	3.5×10^{-4}	0.082	6.8×10^{-4}
4	0	4.3×10^{-5}	0.001	9.3×10^{-5}

Table B-3: Dropping Rate

The table below shows the statistical data of the simulations conducted on our modified algorithm presented in Chapter 5.

	Drooping Rate		Handoff Blocking		New Call Blocking	
Class	Mean	STD	Mean	STD	Mean	STD
1	0	10.1×10^{-5}	0.000	1.3×10^{-5}	0.039	6.3×10^{-4}
2	0.011	11.3×10^{-4}	0.000	0.3×10^{-5}	0.101	4.2×10^{-4}
3	0.158	9.2×10^{-4}	0.000	0.9×10^{-5}	0.050	3.1×10^{-4}
4	0.113	8.4×10^{-4}	0.000	0.1×10^{-5}	0.039	4.9×10^{-4}

Table B-4: Statistical Data of Algorithm Modifications Analysis

The table below shows the statistical data of the Analysis with varying Alpha (presented in Chapter 5).

	Class 1		Class 2		Class 3		Class 4	
α	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.002	6.3×10^{-5}	0.010	3.8×10^{-5}	0.139	8.5×10^{-4}	0.110	2.5×10^{-4}
0.1	0.001	2.7×10^{-5}	0.008	3.7×10^{-5}	0.159	7.3×10^{-4}	0.110	7.5×10^{-4}
0.2	0.002	3.2×10^{-5}	0.005	2.9×10^{-5}	0.160	1.4×10^{-3}	0.128	8.4×10^{-4}
0.3	0.000	1.4×10^{-5}	0.005	4.1×10^{-5}	0.160	6.1×10^{-4}	0.131	9.3×10^{-4}
0.4	0.000	1.0×10^{-5}	0.003	1.1×10^{-5}	0.159	8.3×10^{-4}	0.134	1.1×10^{-3}
0.5	0.000	1.1×10^{-5}	0.003	0.7×10^{-5}	0.160	3.5×10^{-4}	0.135	2.1×10^{-4}
0.6	0.000	0.3×10^{-5}	0.002	1.2×10^{-5}	0.168	6.9×10^{-4}	0.139	3.5×10^{-4}
0.7	0.000	0.4×10^{-6}	0.001	0.4×10^{-6}	0.179	12×10^{-3}	0.142	5.7×10^{-4}
0.8	0.000	0.9×10^{-7}	0.000	1.1×10^{-7}	0.180	9.5×10^{-4}	0.149	3.5×10^{-4}
0.9	0.000	0.5×10^{-7}	0.000	0.9×10^{-7}	0.197	1.5×10^{-3}	0.149	3.7×10^{-4}
1	0.000	0.6×10^{-7}	0.000	1.7×10^{-7}	0.200	8.7×10^{-4}	0.158	8.9×10^{-4}

Table B-5: Statistical Data for the Dropping Rate

	Class 1		Class 2		Class 3		Class 4	
α	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.000	0	0.009	0.6×10^{-7}	0.003	0.4×10^{-7}	0.002	0.8×10^{-8}
0.1	0.000	0	0.000	0.3×10^{-7}	0.000	0.1×10^{-8}	0.000	0.4×10^{-8}
0.2	0.000	0	0.000	0.1×10^{-7}	0.000	0.2×10^{-8}	0.000	0.7×10^{-8}
0.3	0.000	0	0.000	0.2×10^{-8}	0.000	0.4×10^{-8}	0.000	0.1×10^{-8}
0.4	0.000	0	0.000	0.6×10^{-8}	0.000	0.7×10^{-8}	0.000	0.2×10^{-8}
0.5	0.000	0	0.000	0.3×10^{-8}	0.000	0.9×10^{-8}	0.000	0
0.6	0.000	0	0.000	0.6×10^{-8}	0.000	0.5×10^{-8}	0.000	0
0.7	0.000	0	0.000	0	0.000	1.1×10^{-8}	0.000	0
0.8	0.000	0	0.000	0	0.000	0	0.000	0
0.9	0.000	0	0.000	0	0.000	0	0.000	0
1	0.000	0	0.000	0	0.000	0	0.000	0

Table B-6: Statistical Data for the Handoff Blocking Rate

	Class 1		Class 2		Class 3		Class 4	
α	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.040	6.1×10^{-4}	0.100	0.9×10^{-3}	0.050	4.4×10^{-4}	0.040	2.1×10^{-4}
0.1	0.039	5.2×10^{-4}	0.100	1.1×10^{-3}	0.050	4.8×10^{-4}	0.040	3.5×10^{-4}
0.2	0.039	7.2×10^{-4}	0.100	8.9×10^{-4}	0.051	5.2×10^{-4}	0.040	5.7×10^{-4}
0.3	0.039	8.6×10^{-4}	0.099	1.3×10^{-3}	0.048	7.7×10^{-4}	0.040	3.5×10^{-4}
0.4	0.037	4.4×10^{-4}	0.099	1.1×10^{-3}	0.049	8.4×10^{-4}	0.037	3.7×10^{-4}
0.5	0.036	4.8×10^{-4}	0.100	9.2×10^{-4}	0.046	5.4×10^{-4}	0.040	8.9×10^{-4}
0.6	0.035	5.2×10^{-4}	0.100	8.4×10^{-4}	0.048	7.7×10^{-4}	0.039	6.3×10^{-4}
0.7	0.034	7.7×10^{-4}	0.098	5.4×10^{-4}	0.048	6.4×10^{-4}	0.040	4.2×10^{-4}
0.8	0.035	9.4×10^{-4}	0.098	7.7×10^{-4}	0.047	6.1×10^{-4}	0.039	3.1×10^{-4}
0.9	0.035	5.6×10^{-4}	0.098	6.4×10^{-4}	0.047	8.3×10^{-4}	0.040	4.9×10^{-4}
1	0.035	5.2×10^{-4}	0.098	8.4×10^{-4}	0.046	3.5×10^{-4}	0.036	8.4×10^{-4}

Table B-7: Statistical Data for the New Call Blocking Rate

The table below shows the statistical data of the Analysis with varying Beta (presented in Chapter 5).

	Class 1		Class 2		Class 3		Class 4	
β	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.090	4.4×10^{-4}	0.209	8.9×10^{-4}	0.012	5.2×10^{-4}	0.003	7.2×10^{-4}
0.1	0.063	4.8×10^{-4}	0.190	1.3×10^{-3}	0.027	7.7×10^{-4}	0.011	8.6×10^{-4}
0.2	0.051	5.4×10^{-4}	0.153	1.1×10^{-3}	0.031	6.4×10^{-4}	0.018	4.4×10^{-4}
0.3	0.033	7.7×10^{-4}	0.123	9.2×10^{-4}	0.040	6.1×10^{-4}	0.021	4.8×10^{-4}
0.4	0.017	6.4×10^{-4}	0.065	7.7×10^{-4}	0.057	8.9×10^{-4}	0.035	5.2×10^{-4}
0.5	0.001	6.1×10^{-7}	0.021	6.4×10^{-4}	0.068	4.2×10^{-4}	0.041	4.8×10^{-4}
0.6	0.001	1.1×10^{-6}	0.021	6.1×10^{-4}	0.079	3.1×10^{-4}	0.050	5.2×10^{-4}
0.7	0.000	8.9×10^{-7}	0.020	2.7×10^{-4}	0.086	4.9×10^{-4}	0.062	7.7×10^{-4}
0.8	0.000	0	0.020	1.4×10^{-4}	0.110	8.4×10^{-4}	0.088	8.4×10^{-4}
0.9	0.000	0	0.010	1.1×10^{-5}	0.125	9.2×10^{-4}	0.098	5.4×10^{-4}
1	0.000	0	0.010	7.7×10^{-6}	0.138	8.4×10^{-4}	0.110	3.5×10^{-4}

Table B-8: Statistical Data for the Dropping Rate

	Class 1		Class 2		Class 3		Class 4	
β	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.032	7.7×10^{-4}	0.089	8.1×10^{-4}	0.000	0	0.000	0
0.1	0.027	8.4×10^{-4}	0.078	8.7×10^{-4}	0.000	0	0.000	0
0.2	0.024	5.4×10^{-4}	0.068	9.1×10^{-4}	0.000	0	0.000	0
0.3	0.021	7.7×10^{-4}	0.044	8.9×10^{-4}	0.000	0	0.000	0
0.4	0.014	6.4×10^{-4}	0.024	2.8×10^{-4}	0.000	0	0.000	0
0.5	0.001	6.1×10^{-7}	0.013	3.2×10^{-5}	0.000	0	0.000	0
0.6	0.001	8.3×10^{-6}	0.011	1.4×10^{-5}	0.000	0	0.000	0
0.7	0.000	3.5×10^{-6}	0.011	1.0×10^{-5}	0.000	0	0.000	0
0.8	0.000	0	0.003	1.1×10^{-5}	0.000	0	0.000	0
0.9	0.000	0	0.002	0.3×10^{-5}	0.000	0	0.000	0
1	0.000	0	0.000	0.9×10^{-6}	0.000	0	0.000	0

Table B-9: Statistical Data for the Handoff Blocking Rate

	Class 1		Class 2		Class 3		Class 4	
β	Mean	STD	Mean	STD	Mean	STD	Mean	STD
0	0.030	3.3×10^{-4}	0.080	6.4×10^{-4}	0.020	7.2×10^{-4}	0.015	4.8×10^{-4}
0.1	0.030	4.7×10^{-4}	0.081	6.1×10^{-4}	0.021	8.6×10^{-4}	0.019	5.2×10^{-4}
0.2	0.032	9.1×10^{-4}	0.085	8.3×10^{-4}	0.027	4.4×10^{-4}	0.023	4.8×10^{-4}
0.3	0.033	8.9×10^{-4}	0.091	3.2×10^{-4}	0.032	4.8×10^{-4}	0.025	7.7×10^{-4}
0.4	0.035	2.8×10^{-4}	0.093	1.4×10^{-4}	0.033	5.2×10^{-4}	0.027	8.4×10^{-4}
0.5	0.034	7.7×10^{-4}	0.096	1.0×10^{-4}	0.036	4.8×10^{-4}	0.028	5.4×10^{-4}
0.6	0.036	6.4×10^{-4}	0.097	1.1×10^{-4}	0.038	5.2×10^{-4}	0.034	7.7×10^{-4}
0.7	0.037	6.1×10^{-4}	0.100	0.3×10^{-4}	0.042	7.7×10^{-4}	0.038	1.3×10^{-3}
0.8	0.039	8.9×10^{-4}	0.099	7.7×10^{-4}	0.044	8.4×10^{-4}	0.039	1.1×10^{-3}
0.9	0.039	6.8×10^{-4}	0.100	6.4×10^{-4}	0.048	5.4×10^{-4}	0.041	9.2×10^{-4}
1	0.039	7.2×10^{-4}	0.101	6.1×10^{-4}	0.050	3.5×10^{-4}	0.041	8.4×10^{-4}

Table B-10: Statistical Data for the New Call Blocking Rate

The table below shows the statistical data of the Optimum Alpha and Beta Configuration (presented in Chapter 5).

	Dropping		Handoff Blocking		New Call Blocking	
Class	Mean	STD	Mean	STD	Mean	STD
1	0.000	0.4×10^{-6}	0.000	0.2×10^{-7}	0.041	8.9×10^{-4}
2	0.019	6.4×10^{-4}	0.010	1.7×10^{-5}	0.101	2.8×10^{-4}
3	0.060	4.4×10^{-4}	0.060	3.4×10^{-4}	0.050	9.2×10^{-5}
4	0.028	4.8×10^{-4}	0.049	5.5×10^{-4}	0.040	1.4×10^{-4}

Table B-11: Statistical Data for the Optimum Alpha and Beta Configuration

The table below shows a selection of the statistical data for varying Feedback Intervals during controller tuning (presented in Chapter 6).

	RT Data		NRT Data	
Interval	Mean	STD	Mean	STD
0.1	0.0153	3.1×10^{-4}	0.0479	8.9×10^{-4}
0.5	0.0153	6.8×10^{-4}	0.0478	2.8×10^{-4}
1	0.0153	4.8×10^{-4}	0.0480	7.7×10^{-4}
3	0.0153	5.2×10^{-4}	0.0440	6.4×10^{-4}
5	0.0153	4.8×10^{-4}	0.0435	6.1×10^{-4}
6	0.0152	5.2×10^{-4}	0.0432	3.1×10^{-4}
6.5	0.0151	7.7×10^{-4}	0.0421	6.8×10^{-4}
7	0.0149	6.1×10^{-4}	0.0410	5.3×10^{-4}
7.5	0.0155	2.1×10^{-4}	0.0399	4.6×10^{-4}
8	0.0158	8.9×10^{-4}	0.0376	5.9×10^{-4}
10	0.0160	3.8×10^{-4}	0.0188	2.7×10^{-4}
50	0.0162	7.1×10^{-4}	0.0018	1.6×10^{-5}
100	0.0165	6.4×10^{-4}	0.0018	0.9×10^{-5}

Table B-12: Statistical Data for Controller Tuning by Feedback Interval Variation

The table below shows a selection of the statistical data for varying Feedback constant K_1 during controller tuning (presented in Chapter 6).

	RT Data		NRT Data	
K_1	Mean	STD	Mean	STD
0.1	0.0163	5.2×10^{-4}	0.0017	5.2×10^{-4}
0.3	0.0159	2.2×10^{-4}	0.0067	4.8×10^{-4}
0.5	0.0156	9.4×10^{-4}	0.0102	3.2×10^{-4}
0.6	0.0152	4.0×10^{-4}	0.0224	1.1×10^{-4}
0.7	0.0150	8.1×10^{-4}	0.0273	1.2×10^{-4}
0.8	0.0148	4.0×10^{-4}	0.0289	5.1×10^{-4}
0.9	0.0147	3.2×10^{-4}	0.0296	2.8×10^{-4}
1	0.0147	1.4×10^{-4}	0.0311	7.7×10^{-4}
1.2	0.0149	1.0×10^{-4}	0.0310	6.4×10^{-4}
1.5	0.0148	1.1×10^{-4}	0.0313	6.1×10^{-4}
1.8	0.0149	3.2×10^{-4}	0.0316	9.4×10^{-4}
2	0.0145	7.2×10^{-4}	0.0325	4.0×10^{-4}
3	0.0146	8.4×10^{-4}	0.0347	8.1×10^{-4}
5	0.0148	4.0×10^{-4}	0.0404	4.0×10^{-4}
8	0.0150	0.1×10^{-4}	0.0451	6.1×10^{-4}
10	0.0150	4.0×10^{-4}	0.0480	2.8×10^{-4}

Table B-13: Statistical Data for Controller Tuning by Controller Constant K_1 Variation