Dual Forward Packet Traffic Channel and Optimized Data Rate Selection for IS-856 (cdma2000 1xEV-DO)

by

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Abstract

The IS-856 (cdma2000 1xEV-DO) standard is designed with the intention of only providing delay tolerant high rate data services for cdma2000 systems. It uses packet switched transmission by transmitting only one packet at a time using the entire transmission bandwidth. It uses adaptive data rate transmission by transmitting to users at the highest data rate supportable by the channel. It takes advantage of multiple user environment by selecting users based on the user's channel conditions at the time of transmission.

In this thesis, improvements to the performance of the IS-856 system in terms of average system data rates and packet delays are considered. A modification to the downlink to include a second downlink packet channel is proposed . To take advantage of the two user transmission, existing packet scheduling algorithms are modified to schedule two users simultaneously. Simulations including physical layer and system level are used to determine the performance of the proposed system. The modification gives an increase in average data rates and decrease in packet delays.

The repetition coding used by the system allows for the creation of variable data rates. In the thesis, a set of optimized data rates are generated for a selected channel model. The data rates are optimized for single user transmission and for two user transmission. These optimized data rates show a performance gain for the system.

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Glossary of Terms

1xEV-DO	cdma2000 Evolution Data Only
1xEV-DV	cdma2000 Evolution Data and Voice
AWGN	additive white gaussian noise
BER	bit error rate
C/I	carrier to interference ratio
DRC	Data Rate Control
EDGE	Enhanced Data GSM Environment
EXPR	Exponential Rule algorithm
FIFO	First In First Out algorithm
FIR	finite impulse response filter
GPRS	General Packet Radio Service
HSDPA	High Speed Downlink Packet Access
MAXCI	maximum carrier-to-interference (C/I) algorithm
PER	packet error rate
PF	proportionally fair algorithm
SNR	signal-to-noise-ratio
RF	Radio Frequency
RR	Round Robin algorithm
TDM	Time division multiplexing
UMTS	Universal Mobile Telecommunications System
WCDMA	Wide Band Code Division Multiple Access

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Chapter 1

Introduction

The third generation of wireless cellular systems, such as cdma2000 and UMTS, were introduced a decade after the introduction of the second generation wireless systems. In the initial introduction of these systems, the system designs were primarily focused on providing voice communication services. The data services provided by these systems were performed using the voice infrastructure.

The requirements of voice communication services and data communications services are different. Voice service requirements are based on carrying a continuous digitized voice data. The digitized voice data often consists of constant bit rate traffic that requires low latency transmissions. Therefore, systems that provide voice services channelize the link resources to provide each user the required constant data rates, hence a circuit-switched system.

The requirements of data services varies based on the requirements of the specific application. In general, however, data services require higher data rates and are tolerant to modest packet delays. The bursty nature of data traffic creates intermittent usage of the data links. Therefore, it is preferable to use a single link that multiplexes all the terminals for systems that provide data services.

The need for more efficient and higher rate data services created the evolutionary standards to the second generation systems that support data service. These system enhancements include GPRS, EDGE, and IS-95B system standards. The enhancements provide modest increase in spectral efficiency, however, the backwards compatibility requirements on the systems force the standards to support data services less than optimally.

The third generation of wireless systems provide higher data rates and system enhancements when compared to the second generation of systems. The initially introduced systems, however, are optimal for providing voice services. Evolutions to the standards are intended to provide higher spectrally efficient data services. These evolutionary systems include three proposed standards: IS-856 (1xEV-DO), 1xEV-DV, and HSDPA.

In general, the systems use adaptive coding and modulation to provide transmissions that adapt with the channel conditions. Further adaptation is provided with the use of Hybrid-ARQ to give higher system performance when packet transmission errors occur. Furthermore, the systems take advantage of multi-user diversity gains by allowing for fast user selections using short transmission frames.

The IS-856 (3G-1xEV-DO) [3] [1] is introduced as data only evolution to the cdma2000 standard. It maintains most of the components of cdma2000 but supports only a single packet data traffic channel in the downlink. Transmissions are sent to a single user at a time with the selected data rate based on the channel conditions. Downlink data rates ranging from 38.4 to 2457.6 kbps are supported by the standard with the use of 1/3 rate and 1/5 rate Turbo codes and QPSK, 8PSK, and 16 QAM modulated transmissions. Repetition and puncturing of the modulated symbols provide finer grain coding.

The 3G-1xEV-DV [4] is introduced as voice and data evolution to cdma2000. It supports a new packet data channel labeled as Radio Configuration 10 (RC10) in addition to the existing cdma2000 channel configurations. Up to two downlink packet data channels are definable that are separated with orthogonal Walsh Codes.

The data traffic packets are encoded with 1/5 rate Turbo coding and divided into four sub-packets with data rates ranging from 43.2 kbps to 3.0912Mbps [4]. The sub-packets are created from the coded bits from the Turbo coder and are modulated with one of QPSK, 8PSK, and 16 QAM modulated transmissions. The sub-packets are transmitted one at a time as required based on the success of the packet transmissions and the sub-packets may be regenerated at any time.

The High Speed Downlink Packet Access (HSDPA) [5] [6] is a proposed configuration to support high rate data services on UMTS (WCDMA). The system provides data rates up to 10.2Mbps on a dedicated packet channel using 1/3 rate Turbo coding with rate matching [6]. The system can adaptively transmit using QPSK or 16QAM modulation. The system provides a form of incremental redundancy by transmitting additional coding information sourced from the original data packet in the event of a transmission failure.

1.1 Thesis Outline

The purpose of the research is to improve the performance of the IS-856 system in terms of average system data rates and packet delays. Modifying the downlink to include a second downlink packet channel is proposed as a method to enhance the performance of the IS-856 system. This allows for two users to simultaneously receive a packet transmission. The modification to the downlink is done such as to minimize the changes to the system in general. The data rates used in the system is maintained with the same physical layer packet sizes to the physical layer.

The small number of discrete data rates selected for IS-856 leads to loss in performance compared to the continuous data rate selection. Allowing for two users to transmit provides for better utilization of the channel by reducing the effect of the quantization loss. The two user transmission gives higher average data rates compared to the single user case. The two user transmission also has the benefit of reducing the packet transmission delay as more packets are transmitted.

Continuous data rate selection gives optimum results for system transmissions, however, for real systems such as IS-856, a finite set of data rates are used. Optimized selection of the discrete data rates is studied. The data rates are chosen as to optimize the system performance in terms of average data rate transmission for the given channel statistics. The optimized data rate selection show improved performance compared to the original data rates. A set of optimized data rates are chosen for the two user transmission. As with the single user data rate optimization, the two user optimized data rates give higher performing data rates for the two user transmission mode.

Chapter 2 provides a background on the technologies behind the enhancements to the existing cellular wireless systems to provide more spectrally efficient data services. A detailed overview of the IS-856 is provided to give the background needed for the later chapters. The proposed packet scheduling algorithms that are used to allocate physical layer resources to users is described.

In Chapter 3 a system design that provides for two users to transmit simultaneously for IS-856 is proposed. The system incorporates two downlink packet channels that allow for dynamic selection of single or two user transmission. The performance of the system is analyzed with system level simulations. The results show improved average system data rates and lower packet delays with two user transmission.

In Chapter 4 the effect of discrete data rates is analyzed. The effect of increasing the number of data rates in the system for single and two user transmission is analyzed. It is shown that increasing the number of data rates improves performance, with the two user transmission improving at a faster rate. A set of optimized data rates is selected for the system and the performance compared. The improved performance of the optimized data rates are shown in the chapter.

The conclusions and suggested work for the future are provided in Chapter 5. The performance gain from two user transmission and from optimized data rates are summarized in the conclusions.

Chapter 2

Background

The initial second and third generation systems use physical layer designs that provide voice in an optimized fashion. The designs satisfy voice service requirement of providing a constant bit rate traffic with a specific voice quality. The voice quality experienced by the terminals is dependent on the voice frame error rates. This forces the systems to maintain a minimum frame error rate, using strong error correctional coding, at the receiver of all the terminals in the system. When voice systems are used to provide data services, the systems do not provide spectrally efficient transmissions due to the lack of support for adaptation of the transmission data rate and the ability to rapidly change the destination terminal.

2.0.1 Adaptive Coding and Modulation

It is shown in [7] that maximum spectral efficiency for fading channels is achieved by adapting the transmission coding, modulation and power based on accurate knowledge of the transmission channel conditions at each instance of time. Furthermore, [7] shows that by setting the transmission power proportional to the channel gain, while maintaining a fixed average system transmission power, provides maximum spectral efficiency. These results lead to the significance of adaptive coding and modulation for wireless packet data systems.

The adaptation of transmission power, data rate and system bit-error-rate (BER) for spectral efficiency in frequency flat fading is extensively analyzed in [8] and summarized here using the same notations. In the study, the derivation of the optimum run time adaptation of the parameters to maximize spectral efficiency for both continuous data rate and discrete data rate adaptation are considered. For continuous data rate adaptation, the data rate may take any value, whereas for discrete data rate adaptation, the data rate may only take a value from a set of finite data rates. The study considers the optimum adaptation for both the instantaneous and the average BER constraint. The instantaneous BER constraint requires that a minimum BER value be maintained during the fades and the average BER constraint requires that only the average BER value be maintained. Furthermore, the effects of limiting adaptation of some of the parameters are considered in [8].

The spectral efficiency is defined as the transmission data rate for the given bandwidth R/B (bits/s/Hz), where R (bits/s) is the average transmission data rate and B (Hz) is the transmission bandwidth defined as $B = 1/T_s$, where T_s (s) is the symbol duration [8]. The instantaneous received SNR at time i is defined as $\gamma[i]$ and is a function of the channel gain and average transmission power ($\bar{S}$). With the power adaptation, the instantaneous transmission power (S) at time i is a function of $\gamma[i]$. It is assumed that $\gamma[i]$ is independent of time i with the distribution $p(\gamma)$. With these definitions (time reference i omitted), the spectral efficiency for the continuous data rate adaptation is given by [8]

$$\frac{R}{B} = \int_0^\infty k(\gamma)p(\gamma)d\gamma \tag{2.1}$$

where $k(\gamma)$ is the data rate in (bits/symbol) with the instantaneous data rate determined as $k(\gamma)/T_s$ (bits/s). The $k(\gamma)$ is a function of the transmission power and BER. In [8], the adaptive transmission power is limited by the average transmission power constraint as in:

$$\int_0^\infty S(\gamma)p(\gamma)d\gamma \leq \bar{S} \tag{2.2}$$

where $S(\gamma)$ is the instantaneous transmission power as a function of signal-to-noise-ratio (SNR).

For the discrete data rate adaptation, the data rates available to the system are limited to a finite set [8]. Each of the discrete data rates (k_i) are assigned to an SNR rate region $(\gamma_i, \gamma_{i+1}]$. The system transmits at data rate k_i when the instantaneous SNR falls within the corresponding range. The spectral efficiency of discrete data rate adaptation is given by [8]

$$\frac{R}{B} = \sum_{i=0}^{N-1} k_i \int_{\gamma_i}^{\gamma_{i+1}} p(\gamma) d\gamma \tag{2.3}$$

where $\gamma_N = \infty$ and γ_0 is the minimum required SNR for transmission. Furthermore, the adaptive transmission power is limited by the average transmission power constraint as in the continuous adaptation for both instantaneous and the average BER constraint.

2.0.2 Multiuser Diversity

The previous section showed the link adaptation required to maximize spectral efficiency for a single terminal. However, a system that serves multiple terminals is required to divide the system capacity among the active terminals. With voice services, the ideal method is to evenly divide the link to provide multiple parallel channels. For data services, the ideal method is to allow multiple bursty terminals to share a single shared channel using packet switching.

The use of a single shared channel allows the system to select the terminal that is to receive a transmission, hence selecting the terminal that has access to the resource. A method of allocating system resources is presented in [9] that takes advantage of multiuser diversity to give the highest system capacity for frequency flat fading. At each transmission time, the system selects the one terminal that is able to transmit at the highest data rate (or best channel condition) assuming that accurate channel information is available [9]. [9] shows that this selection process is a form of selective channel diversity that is normally used with multiple receiver antenna systems to reduce the effects of channel fades [10]. This form of diversity is often referred to as multi-user diversity or opportunistic packet scheduling [11]. The resulting gain from this method increases as the number of available terminals in the system are increased [9].

However, if the terminal that is experiencing the best channel conditions is repeatedly selected, it would be unfair to the other terminals that are disadvantaged [9]. A selected few terminals would dominate the transmissions. In practice, the packet scheduling algorithms are required to provide more fairness, yet still providing higher data rates associated with opportunistic packet scheduling.

2.1 IS-856 Standard

The IS-856 standard is optimized for data services, however, it still maintains compatibility to the cdma2000 and IS-95 standards using the same chip rate (bandwidth) and RF components [3] [12] [1]. This continued compatibility allows for the reuse of many of the system components from the earlier standards. Furthermore, the system provides higher data rates for packet data communications compared to the voice centric predecessor standards.

The increase in system performance is mainly provided with the use of adaptive modulation, coding and user scheduling. The fast, packet based adaptation of the transmission data rate, based on the channel conditions, provides improved system performance [3] [12] [1]. Furthermore, the system schedules packet transmissions based on the criteria of the packet scheduling algorithm that could include channel conditions and packet delays. This type of user selection allows for the system to benefit from multiuser diversity when multiple terminals are present.

2.1.1 Physical Layer Downlink

The downlink transmission is defined as the transmission from the system base station to the terminals. The IS-856 downlink is similar to cdma2000 and IS-95 using spread spectrum with a spreading chip rate of 1.2288Mcps [1]. However, the downlink only carries data packets for a single terminal at a time maintaining a constant transmission power [1]. This single traffic channel also carries system control packets intended for the entire system.

The downlink transmission is divided into small time slots of 2048 chips by the system physical layer. Each active time slot consists of the time multiplexed data segment, MAC segment and a pilot segment. Of the 2048 chips, 1600 chips are dedicated to the data segment, 256 chips are dedicated to the MAC segment and 192 chips are dedicated to the pilot [1]. The MAC segment of the time slot is used to transmit reverse power control (RPC) and reverse channel activity information to the terminals. The pilot is a known signal that the terminals use for channel estimation and synchronization. When data is not available for transmission, an idle slot is transmitted that consists only of a MAC and pilot segments. After the creation of the transmission time slots, the system performs quadrature spreading, baseband filtering and (IF or carrier) modulation.

The data segment may contain a data packet that spans several time slots. The slots of a multiple slots transmission are interleaved with slots of three other packets. This gives time for the terminal to terminate the transmission early if the partially received packet is sufficient for decoding the packet. Furthermore, the multiple slot transmissions reduces the packet latency of low data rate users [3].

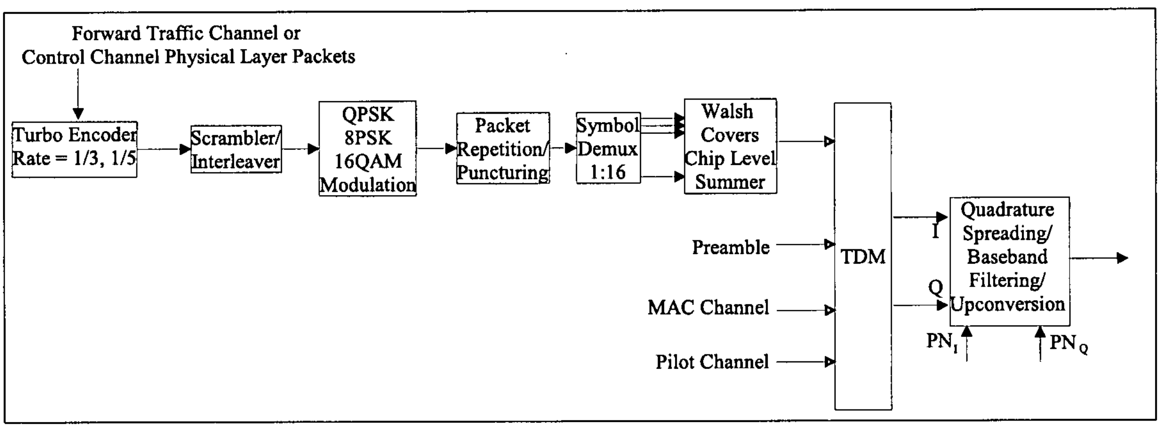

Figure 2.1: Forward Traffic Channel Structure [1]

Data Segment

The data segment of the time slot is generated from Forward Traffic Channel packets and Control Channel packets received by the physical layer of the system [1]. Furthermore, the data segment is appended by a preamble sequence to form a packet that occupies an integer number of transmission time slots. Control packets are used for system and terminal control messaging, whereas, the Forward traffic Channel packets contain the data payloads for the terminals. The same physical layer processing is performed for both types of physical layer packets. The standard defines 12 data rates for the Forward Traffic Channel as shown in Table 2.1 with the two lowest data rates available to the Control Channel [1].

The physical layer packet is processed as shown in the flow diagram of Figure 2.1 to produce the data segment of the system time slot [1]. The parameters for each component is determined by the selected transmission data rate as shown in Table 2.1. These components include the Turbo encoder, scrambler, channel interleaver, modulator, rate matching, and Walsh covers [1]. The size of the physical layer packet is either 1024, 2048, 3076 or 4098 bits, as determined by the selected data rate.

The standard uses Turbo error correctional codes for protecting the physical layer packets against transmission errors. The two constituent codes of the Turbo code are identical systematic convolutional codes [1]. The Turbo encoder produces a base code at rate 1/6. The bits produced from the Turbo encoder are then punctured to produce either a 1/3 rate code or 1/5 rate code as determined by the transmission data rate. The system, therefore, adapts the coding rate based on the channel conditions. At the receiver, decoder such as Log-Map decoder is used to decode the Turbo coded data.

The packet bits from the Turbo encoder are scrambled and reordered before modulation. The bits produced from the Turbo encoder are XOR'd with a scrambling sequence that is generated from a linear feedback shift register. After the scrambler, the bits are then reordered. The decoder at the receiver reverses this process by using the scrambling sequence and reordering method to reproduce original bit order.

After the scrambling and reordering, the bits of the packet are modulated with one of three modulation methods defined in the standard: QPSK, 8PSK and 16QAM modulation. The modulation method is selected based on the transmission data rate as shown in Table 2.1. This allows the system to adapt the modulation method to best match the channel conditions.

The data rate of the modulated packet is finely adjusted with the use of packet repetition and truncation to form the transmission packet. The system increases the data rate by truncating the packet, removing the last portion of the packet, to create a smaller packet. The original packet is recovered with the Turbo Decoder. Similarly, the system lowers the data rate by repeating the packet a selected number of times to form a larger packet. The packets are repeated in their entirety as required, with a possible additional partial packet. The first portion of the packet is used in the case of a partial packet.

The repetition of the packet reduces the data rate by adding redundant symbols to the transmission packet. This increases the error correction capabilities of the system and allows for a form of Hybrid ARQ. At the receiver, the packet repetitions are combined to form a more accurate packet before Turbo Decoding. The reception of the entire transmission packet is not required in order to decode the original packets due to the repetitions. This allows the system to terminate the transmission early, for multiple slot packets, if possible providing a form of Hybrid ARQ to the system. The terminal attempts to decode the packet after each slot reporting a positive acknowledgment if successful.

The transmission packet is bit-by-bit de-multiplexed into 16 blocks that are then multiplied (spread) using one of the orthogonal 16-ary Walsh codes. The 16 blocks are then summed together on a chip basis to form the spread transmission packet. Therefore, the resulting packet is the same length as the original transmission packet. The orthogonality of Walsh codes allows the receiver to recreate the original transmission packet by multiplying (de-spreading) the received packet with the Walsh codes. The de-spreading gives the 16 blocks that are then multiplexed, by the receiver, to form the original transmission packet.

The spread transmission packet is appended with a preamble sequence that is used for synchronization and terminal identification. The preamble sequence is a 32-ary bi-orthogonal Walsh code sequence that gives 64 possible codes. The Walsh code that is selected among the 64 codes for the preamble sequence matches the assigned identification of the terminal that is to receive the data packet. The appended packet is transmitted in the data segment of the transmission time slot. The size of the preamble is determined by the transmission data rate as shown in Table 2.1. The time slot creation for a two slot packet is shown in Figure 2.2; packets of other sizes are created similarly.

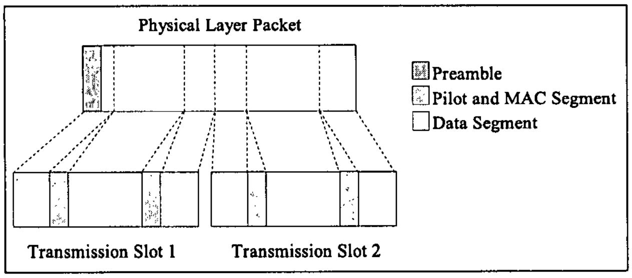

Figure 2.2: Time Slot Creation for IS-856 [1]

2.1.2 Physical Layer Uplink

The uplink channel provides the communications from the terminals to the system base station. The uplink, in the IS-856 standard, uses CDMA with terminal specific short PN code and a base station specific long PN code to separate the terminals [1]. Each terminal is assigned a short PN code that allows the system to distinguish the terminal. Each terminal supports several orthogonal channels that are separated with Walsh codes. The system uplink supports data rates ranging from 9.6kbs to 153.6kbs on the Reverse Traffic Channel as shown in Table 2.2.

DRC Index	Data Rate	Number of Slots	Packet Size (bits)	Code Rate	Modulation Type	Packet Repetition	Preamble Size (chips)
1	38.4	16	1024	1/5	QPSK	9.6	1024
2	76.8	8	1024	1/5	QPSK	4.8	512
3	153.6	4	1024	1/5	QPSK	2.4	256
4	307.2	2	1024	1/5	QPSK	1.2	128
5	614.4	1	1024	1/3	QPSK	1	64
6	307.2	4	2048	1/3	QPSK	2.04	128
7	614.4	2	2048	1/3	QPSK	1.02	64
8	1228.8	1	2048	1/3	QPSK	0.5	64
9	921.6	2	3072	1/3	8PSK	1.02	64
10	1843.2	1	3072	1/3	8PSK	0.5	64
11	1228.8	2	4096	1/3	16QAM	1.02	64
12	2457.6	1	4096	1/3	16QAM	0.5	64

Table 2.1: 1xEVDO Rate Matching [1]

To assist with the adaptive data rate selection, terminals periodically report the data rate that they can support on the Data Rate Control (DRC) Channel of the uplink. The data rate is selected from Table 2.1 and encoded with a four bit data rate index representing the DRC. Furthermore, the terminal selects the base station that provides the best reception.

2.2 Packet Scheduling Algorithms

As described in Section 2.1, the IS-856 standard uses a packet scheduling algorithm to determine the terminal that is to receive the packet transmission. The proportionally fair algorithm (PF) is proposed for IS-856 standard systems [12] [13]. There are many other packet scheduling algorithms presented in literature for wireless and land systems.

Data Rate Index	Data Rate	Packet Size	Code Rate	Packet Repetitions
1	9.6 kbps	256	1/4	8
2	19.2 kbps	512	1/4	4
3	38.4 kbps	1024	1/4	2
4	76.8 kbps	2048	1/4	1
5	153.6 kbps	4096	1/2	1

Table 2.2: 1xEVDO Reverse Traffic Channel [1]

Some of these algorithms are compared in [14] and [15] for IS-856 systems. An overview of some packet scheduling algorithms are presented in this section. These algorithms are used in the following chapters for system performance analysis.

Algorithms that do not use channel information provided by the terminals are at a disadvantage for wireless systems as there is no multiuser diversity gain. These are suitable for land line systems but are not suitable for wireless systems. These algorithms are shown to provide poor results in wireless systems in [14] for FIFO and in [15] for round robin packet scheduling. In Round Robin (RR) type of algorithm, the system takes turns scheduling packets to each terminal. If a terminal is unable to receive a packet due to reported DRC, then the next terminal in the order is scheduled [15].

From Section 2.0.2, the algorithm that provides the highest system data rate is one that selects the terminal with the highest carrier-to-interference (C/I) (MAXCI). The MAXCI algorithm schedules the terminals:

$$U(s) = \arg\max_{u} DRC_u(s)$$

where $U(s)$ is the terminal number transmitted at slot s and $DRC_u(s)$ is the data rate from terminal u. When more than a single terminal reports the highest $DRC_u(s)$, a terminal among the highest is selected randomly [15]. The MAXCI discriminates against terminals that are farther from the base stations due to the lower signal mean.

Due to the unfairness in the scheduling, MAXCI is not suitable for use in practice however it does shows the maximum system data rate possible from the set of terminals in the system.

Proportionally fair algorithm (PF) proposed for IS-856 systems in [13] includes fairness in the scheduling of terminals. The algorithm selects the terminal that has the highest proportion between its current DRC and its average awarded DRC. This results in selecting the terminal that is experiencing the best short term channel when compared to its own average channel condition. The PF algorithm schedules the terminals [13]

$$U(s) = \arg\max_{u} \frac{DRC_u(s)}{\overline{DRC_u(s)}}$$

where $\overline{DRC_u(s)}$ is the average awarded DRC_u at slot s. The average is updated every slot time for each terminal with a low pass filter [13]

$$\overline{DRC_u(s+1)} = (1 - 1/t_c)DRC_u(s) + (1/t_c)(R_u)$$

for the terminal u that receives the transmission and

$$\overline{DRC_u(s+1)} = (1 - 1/t_c)DRC_u(s)$$

for terminals that do not receive a transmission; R_u is the current transmission data rate and t_c is the filter time constant in slots. The t_c determines the averaging window of the scheduling algorithm.

The PF algorithm does not consider the packet delays, hence there are no delay guarantees in PF algorithm. Exponential Rule (EXPR) algorithm is proposed in [16][14] that explicitly controls the packet delays experienced by the terminals. It has an additional term to PF scheduler that takes into account the packet delays. The EXPR algorithm schedules the terminals [16][14]

$$U(s) = \arg\max_{u} \left[a_u \frac{DRC_u(s)}{\overline{DRC_u(s)}} \exp\left(\frac{a_u W_u(s) - \overline{aW}}{1 + \sqrt{\overline{aW}}} \right) \right]$$

where $W_u(s)$ is the number of slots since the last transmission for terminal u at time slot s, a_u is the weight for terminal u and $\overline{aW}$ is the mean of all the terminals. The weight is selected with $a = -log(\gamma)/T$ where γ is the highest probability that the terminal experiences a packet delay greater than T [16]. In the simulations of the following chapters, $\overline{DRC_u(s)}$ is calculated as in PF with a filter as opposed to the true mean value. The $\overline{aW}$ is calculated as the true mean from the beginning of the simulation to the time slot s as suggested in [14].

An algorithm proposed in [17], provides fairness by allocating the system resources evenly in time (or slots) to all the terminals. The algorithm optimizes the performance factor (that can represent anything) yet still equalizing the slot allocation between users over a long time scale. The performance factor chosen here is the transmission data rate $D'_{u(s)}$. The value for $D'_{u(s)}$ is set based on the $DRC_u(s)$, as done in [15], as the number of multiples of 38.4 kbps the $DRC_u(s)$ represents. The algorithm, termed here as LCS (the authors initials), schedules the terminals

$$U(s) = \arg\max_u (D'_{u(s)} + v_u(s))$$

where $v_u(s)$ is the distribution parameter. The value of this parameter is selected to give the desired time slot allocation between terminals. The correct value requires the knowledge of the $D'_{u(s)}$ distribution, however, it can be estimated with

$$v_u(s+1) = v_u(s) - a(s)(1 - r_u)$$

for the terminal that is scheduled to transmit and

$$v_u(s+1) = v_u(s) - a(s)(0 - r_u)$$

for the terminal that is not scheduled to transmit [17]. The term r_u is the proportion of time allocated to terminal u and $a(s)$ is the step size for the adaptation. The estimate is calculated every slot time even if there is an IS-856 multi slot packet in transmission as was done in PF and EXPO. The initial value of $v_u(s)$ is set to zero. For fair sharing of slots between terminals, the r_u is set to 1/(Number of users) [17].

Chapter 3

IS-856 Two User Transmission

This chapter introduces a method of transmitting two users simultaneously for the IS-856 standard. The standard only allows for the transmission of a single user, therefore, the standard is modified to provide two user support. The method proposed in this section is intended to minimize the required changes to the standard and to take into account the associated overhead of supporting two simultaneous users. In addition to the physical layer changes, the scheduling algorithm used in the standard requires modification to support two user packet scheduling. The effects and benefits of using two user transmission are investigated. The performance of the modification is simulated in the physical layer and in the system level. System simulations of both single user and two user transmission are performed and compared in this chapter.

The purpose of two user transmission is to allow the system to better utilize the transmission medium by reducing the effect of data rate quantization without increasing the number of data rates. The quantization occurs as a result of only having 12 data rates supported by the system. The resulting performance loss is intuitive as there is a

[0]A version of this chapter has been accepted for publication. S. Dost, O. Sunay, and V. Bhargava, "A Feasibility Study of Two User Downlink Transmission for IS-856 System", in The 15th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, PIMRC, September 2004

performance loss when discrete data rate adaptation is used rather than continuous data rate adaptation [8]. The quantization of the data rates can be seen by the simulation results in Section 3.3.1. The combined transmissions of the two users allows for better utilization of the medium by allowing the second user to transmit when the single user quantization loss is high.

3.1 Two User Division

The two user transmission maintains the same set of data rates, whereby, reducing the required changes to the existing system. The existing orthogonal 16-ary Walsh code in the downlink of the standard are used to separate the transmissions of the two users. Walsh codes are an orthogonal set of codes, therefore the two users can be separated without mutual interference. However, the 16-ary Walsh codes only provides 16 additional quantization levels.

The Walsh codes can be arbitrarily divided between the two users. However, to maintain minimum impact on the first user, the first user is given the minimum allocation required to support the requested data rate. By adapting the modulation and coding of the first user, the number of Walsh codes used by the first user is reduced. The system allocates the remaining codes to the second user.

The system maintains the requested data rate of the first user by lowering the coding rate when fewer number of Walsh Codes are allocated in the transmission. Whereas, the second user receives the highest data rate that it can support given the available Walsh Codes. Therefore, the transmission of the second user adds to the system throughput. The system uses the system resource more efficiently with the transmission of the second user. This leads to an increase in the spectral efficiency of the system.

The actual transmission data rate of the second user depends on the number of available Walsh codes. The average second user date rate for a given number of Walsh codes (w) is defined as

$$\kappa(w) = \sum_{i=0}^{N(w)-1} k_i \int_{\gamma(w)_i}^{\gamma(w)_{i+1}} p(\gamma) d\gamma \tag{3.1}$$

where $N(w)$ is the maximum number of data rates supportable given the number of Walsh codes (w). It is assumed that the data rates available to the second user are the same as the first user, therefore, the same set of data rates k_i are used. However, with fewer number of Walsh codes, a different signal-to-noise-ratio (SNR) region $(\gamma(w)_i, \gamma(w)_{i+1}]$ is valid for k_i.

In determining the two user spectral efficiency, it is assumed that the second user contributes the average data rate $(\kappa(w))$ for the allocated number of Walsh codes (w). The first user transmits at k_i when the γ is within the SNR rate region $(\gamma_i, \gamma_{i+1}]$. As the first user γ increases within the rate region $(\gamma_i, \gamma_{i+1}]$, fewer number of Walsh codes are required by the first user to transmit at k_i. The second user, therefore, is allocated an increasing number of Walsh codes as the first user γ increases. This results in the subdivision of the first user SNR region $(\gamma_i, \gamma_{i+1}]$ into smaller regions $(\gamma_{i,w}, \gamma_{i,w+1}]$ based on the Walsh code allocation (w). With this criteria, the two user spectral efficiency is defined as:

$$\frac{R}{B} = \sum_{i=0}^{N-1} \left[k_i \int_{\gamma_i}^{\gamma_{i+1}} p(\gamma) d\gamma + \sum_{w=1}^{M(i)} \kappa(w) \int_{\gamma_{i,w}}^{\gamma_{i,w+1}} p(\gamma) d\gamma \right] \tag{3.2}$$

where $M(i)$ is the maximum number of Walsh codes available for the second user within the first user rate region of k_i. The average data rate of the second user is $\kappa(w)$ within the first user SNR region $(\gamma_{i,w}, \gamma_{i,w+1}]$ for w Walsh Codes.

Dividing the SNR region into smaller regions reduces the effect of data rate quantization without increasing the number of data rates. The system better utilizes the available resources and increases system performance. However, greater number of terminals are required to be available to support two user transmission. Furthermore, the terminals are required to report a more granular DRC.

3.2 IS-856 Standard Modifications

A new downlink forward traffic channel is defined in order to provide two user transmission for IS-856 as described in the previous section. In this system, the traffic channel architecture defined in IS-856 standard is used for the single user transmission and the new traffic channel architecture is used for two user transmission. The system is able to switch between signal and two user transmission within packet boundaries. Furthermore, the users in the two user transmission can receive packets at different at different data rates.

The block diagram of the new downlink channel is shown in Figure 3.1. As shown in the diagram, each user is assigned a number of Walsh codes whereby segregating the two users. However, the two user transmission requires the additional overhead of a control channel. The control channel is required to identify the users that are to receive the packets and to indicate the number of Walsh codes that each user is assigned. One of the Walsh codes is, therefore, assigned as the control channel for the two user transmission. This leaves 15 Walsh codes for the two users. The different allocations of Walsh codes requires modifications to the rate matching and the symbol demuxing sections of the IS-856 standard. Furthermore, the preamble transmission is required to be modified to indicate the two user transmission. The sections preceding the rate matching are not modified, therefore the two user modifications have minimal effect on the layers above the physical layer.

The symbol demultiplexer divides the bit stream from each channel into a number of parallel blocks that are then covered by one of the 16-ary Walsh codes. The number of parallel blocks that are created for each channel is equal to the number of Walsh codes assigned to the channel. A total of 16 parallel blocks are created from the three channels spanning the 16-ary Walsh codes. The resulting 16 Walsh encoded streams are then summed together to create the transmission data frame.

With different number of Walsh codes assigned to a user in two user transmission, the rate matching section of the traffic channel is required to perform varying amount of

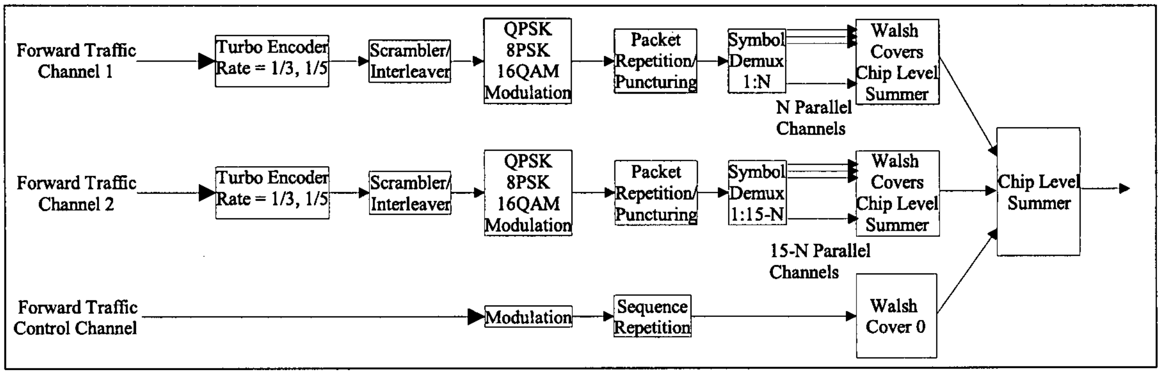

Figure 3.1: Modified Forward Traffic Channel Structure

repetition and puncturing. A user can be assigned 1 to 15 Walsh codes for the different data rates. Each possible combination of data rate and assigned Walsh code requires a different rate match. However, combinations that require too much puncturing may not be feasible.

The repetition and the puncturing is performed such that the system maintains the physical layer frame sizes and transmission data rates as that of the single user transmission for any Walsh codes assignment. The frame sizes before and after rate matching are shown in Table 3.1. The number of symbols in the packet before rate matching is the same as that of the single user mode. The packets are repeated or punctured by the rate matching section to provide the number of symbols needed after the rate matching as indicated in the table. The table shows the number of symbols needed based on the number of Walsh codes that have been assigned to the user.

As with IS-856, the coding rate of the transmission varies with the repetition and the puncturing in the rate matching. The higher amount of puncturing or lower amount of repetition in the rate matching section lowers the coding rate of the packet. The resulting repetition factor of the rate matching is shown in Table 3.2. There is, however, overlapping of the coding rates that can be seen from the table. For example, the repetition rate of 38.4kpbs with 8 Walsh codes is the same as the single user 76.8kpbs. The two data rates do require different number of slot for transmission, however the performance of the two would be similar in an AWGN channel.

Data Rate (kbps)	Number of Modulation Symbols Provided	Number of Modulation Symbols Needed for Single User	Number of Modulation Symbols Needed for the given Number of Assigned Walsh Codes													
			14	13	12	11	10	9	8	7	6	5	4	3	2	1
38.4	2560	24576	21504	19968	18432	16896	15360	13824	12288	10752	9216	7680	6144	4608	3072	1536
76.8	2560	12288	10752	9984	9216	8448	7680	6912	6144	5376	4608	3840	3072	2304	1536	768
153.6	2560	6144	5376	4992	4608	4224	3840	3456	3072	2688	2304	1920	1536	1152	768	384
307.2	2560	3072	2688	2496	2304	2112	1920	1728	1536	1344	1152	960	768	576	384	192
614.4	1536	1536	1344	1248	1152	1056	960	864	768	672	576	480	384	288	192	96
307.2	3072	6144	5376	4992	4608	4224	3840	3456	3072	2688	2304	1920	1536	1152	768	384
614.4	3072	3072	2688	2496	2304	2112	1920	1728	1536	1344	1152	960	768	576	384	192
1228.8	3072	1536	1344	1248	1152	1056	960	864	768	672	576	480	384	288	192	96
921.6	3072	3072	2688	2496	2304	2112	1920	1728	1536	1344	1152	960	768	576	384	192
1843.2	3072	1536	1344	1248	1152	1056	960	864	768	672	576	480	384	288	192	96
1228.8	3072	3072	2688	2496	2304	2112	1920	1728	1536	1344	1152	960	768	576	384	192
2457.6	3072	1536	1344	1248	1152	1056	960	864	768	672	576	480	384	288	192	96

Table 3.1: Number of Symbols Before and After Rate Matching

The number of Walsh codes assigned to a user is fixed for the entire duration of a packet transmission. This is not relevant for single slot packets but this assumption reduces the complexity of the two user transmission for multi-slot packets. Consequently, the allocation may only be changed when there are no packets in transmission. Multiple sequential packets may, however, be transmitted on one channel during the transmission of a long packet on the other channel. These multiple sequential packets may be intended to different terminals that are indicated on the control channel. Furthermore, early packet transmission (hybrid ARQ), as defined in IS-856, is still possible.

In IS-856, the physical layer packet is appended with a preamble sequence that identifies the destination terminal. Each active terminal is assigned one of the 64 unique sequence based on its MAC index. The transmitted preamble sequence is generated with repetitions of the 32 chip length bi-orthogonal Walsh sequence. The number of repetitions (the length) used is determined by the data rate of the transmission. The length of the preamble, in general, is proportional to the length of the physical layer packet.

Data Rate (kbps)	Repetition Factor For Single User	Repetition Factor for the given Number of Assigned Walsh Codes													
		14	13	12	11	10	9	8	7	6	5	4	3	2	1
38.4	9.6	8.4	7.8	7.2	6.6	6	5.4	4.8	4.2	3.6	3	2.4	1.8	1.2	0.6
76.8	4.8	4.2	3.9	3.6	3.3	3	2.7	2.4	2.1	1.8	1.5	1.2	0.9	0.6	0.3
153.6	2.4	2.1	1.95	1.8	1.65	1.5	1.35	1.2	1.05	0.9	0.75	0.6	0.45	0.3	0.15
307.2	1.2	1.05	0.98	0.9	0.83	0.75	0.68	0.6	0.53	0.45	0.38	0.3	0.23	0.15	0.08
614.4	1	0.88	0.81	0.75	0.69	0.63	0.56	0.5	0.44	0.38	0.31	0.25	0.19	0.13	0.06
307.2	2	1.75	1.63	1.5	1.38	1.25	1.13	1	0.88	0.75	0.63	0.5	0.38	0.25	0.13
614.4	1	0.88	0.81	0.75	0.69	0.63	0.56	0.5	0.44	0.38	0.31	0.25	0.19	0.13	0.06
1228.8	0.5	0.44	0.41	0.38	0.34	0.31	0.28	0.25	0.22	0.19	0.16	0.13	0.09	0.06	0.03
921.6	1	0.88	0.81	0.75	0.69	0.63	0.56	0.5	0.44	0.38	0.31	0.25	0.19	0.13	0.06
1843.2	0.5	0.44	0.41	0.38	0.34	0.31	0.28	0.25	0.22	0.19	0.16	0.13	0.09	0.06	0.03
1228.8	1	0.88	0.81	0.75	0.69	0.63	0.56	0.5	0.44	0.38	0.31	0.25	0.19	0.13	0.06
2457.6	0.5	0.44	0.41	0.38	0.34	0.31	0.28	0.25	0.22	0.19	0.16	0.13	0.09	0.06	0.03

Table 3.2: Resulting Repetition Factor by the Rate Matching

For two user transmission, the general slot structure of IS-856 is maintained, however, the preamble structure is modified. The two users may have different data rates, hence require different preamble lengths. Therefore, instead of appending the preamble to the beginning of each physical layer packet, a preamble sequence of 64 chips is appended to the beginning of each slot. This length of preamble is chosen as it provides the same or longer preamble length for multiple slot packets when compared to IS-856. The preamble procedure is illustrated in Figure 3.2 for two user transmission.

To identify that a two user transmission is in progress, a single dedicated preamble sequence is assigned to all two user transmissions. The MAC index of the terminals (the identities) are instead transmitted on the control channel. The terminals, therefore, attempt to decode this dedicated preamble sequence for two user transmission as well as their own unique sequence for single user transmission. The preamble structure defined in IS-856 is used for single user transmission insuring backward compatibility.

The terminal receives the MAC index and Walsh code from the proposed Forward Traffic Control Channel. The channel uses one dedicated Walsh code that allows the base station to transmit 96 modulated symbols. In order to maintain compatibility with the existing system, 16 bits of additional control information must be transmitted

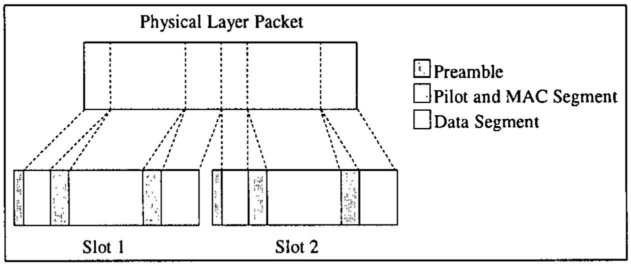

Figure 3.2: Example of Slot Creation For the Two User Transmission Mode

every slot: 6-bits is required for the MAC index of each channel (therefore 12-bits total) and 4-bits for the Walsh code allocation. The 6-bits for the MAC index can represent 64 terminal addresses. The 4-bit Walsh code allocation indicates the number of Walsh Codes that are assigned to the Forward Traffic Channel 1 with the remaining Walsh codes (from 15 codes) allocated to channel 2. The control frame can be constructed with modulation, coding and repetition of the 16-bits to form the 96 symbol frame. With multi-slot packets, the control frame is repeated over each slot providing additional error correction capability. The detail design of the control channel coding is not presented here.

In order to allocate terminals different number of Walsh Codes, the terminals are required to provide more granular representation of the received SNR than that is provided by the DRC. With backward compatibility in consideration, the terminals report the minimum number of Walsh Codes that it requires to support the DRC in addition to the DRC value normally reported. This extended DRC value gives an indication of the minimum SNR that the terminal requires and is used to determine the combinations of data rates and Walsh code allocations that the terminal is able to support. More that a single combination is possible as different combinations of DRC and Walsh code assignments have overlapping SNR requirements. The overlapping requirements are organized in to a table based on the physical layer simulations in Section 3.3.

Two user transmission uses the overlapping table to designate the data rate of the transmission to the terminal. The terminal infers the transmission data rate based on the number of Walsh codes that it has been assigned and the extended DRC that it had originally sent to the base station. The table is then used to identify the exact data rate. This is similar to the IS-865, where the base station transmits to the terminals using the data rate indicated by the DRC from the terminal. Therefore, if the terminal is receiving a transmission in two user mode then the data rate is determined by the table and if it is receiving a single user transmission then the data rate is determined by the DRC.

Terminals report the DRC on the uplink channel with a compact 4-bit number, however, additional bits are required for the extended DRC. A 4-bit Walsh Code separation number can be reported in addition to the normal DRC. Alternatively, the terminal can transmit a new compact number from a lookup table containing critical DRC and Walsh code combinations. Many DRC and Walsh code combinations are never used, therefore 2 additional bits would be sufficient. With either method, the base station has an extended DRC value that contains the original DRC value and a minimum Walsh code separation value.

3.3 IS-856 Two User Simulation

The performance of the two user transmission method, as described in Section 3.2, is determined with simulations. In order to generate a table of overlapping C/I values as described in the previous section, physical layer simulations of each combination of DRC and Walsh code allocation are performed. The results of the simulations provide the C/I required by the receiver to achieve 1% packet error rates (PER).

It is assumed that the relative difference in C/I in the table of overlapping C/I values is valid for real terminals. The reason for the importance of this assumption is that it allows for the two user mode to modify the second user data rate based on the DRC and the Walsh code allocation. The absolute C/I values are not critical for the system design in general, as terminals report the DRC values to the base station rather than the absolute C/I.

The performance of the IS-856 system depends on packet scheduling algorithms, therefore the system level simulations include the packet scheduling. The system simulations require channel simulations to generate sample C/I values seen at the terminals. The C/I values are then used by the packet scheduling algorithm simulations to estimate the system performance.

3.3.1 Physical Layer Simulation

The simulation environment package, Advanced Design System (ADS) from Agilent Technologies, is used to perform the physical layer simulations. The package provides a sample IS-856 (cdma2000 1xEV-DO) downlink physical layer simulation with all of the signal processing components required for simulating the physical layer.

The sample transmitter section of the standard is implemented with processing components as described in Section 2.1. The receiver section is implemented to decode the transmission packets of the standard. The receiver contains a down-converter and RAKE receiver to provide the time slots frames. The time slot frames are demultiplexed into the three segments: data segment, MAC segment and the pilot segment. The data segment is then de-spread with the 16 ary-Walsh codes to reproduce the 16 blocks. The blocks are rate de-matched using soft equal gain combining to give the modulated physical layer packet. The modulated packets are demodulated with QPSK, 8PSK or 16QAM to produce a soft output and deinterleaved/descrambled. The Turbo Decoder using 12 iteration Log-MAP decoder decodes the packets to produce the original physical layer packet. Only the physical layer section of the Traffic Channel is simulated.

The simulation of the physical layer is performed by processing 1000 randomly generated packets through the transmitter, AWGN (additive white gaussian noise) channel and receiver section of the downlink physical layer as shown in Figure 3.3.

The packets represent the physical layer packets that are given to the physical layer from the MAC layer. The packets resulting from the receiver are then compared to the transmitted packets for the calculation of the PER. In the simulation, the channel noise power is swept in order to determine the C/I value that results in a 1% PER.

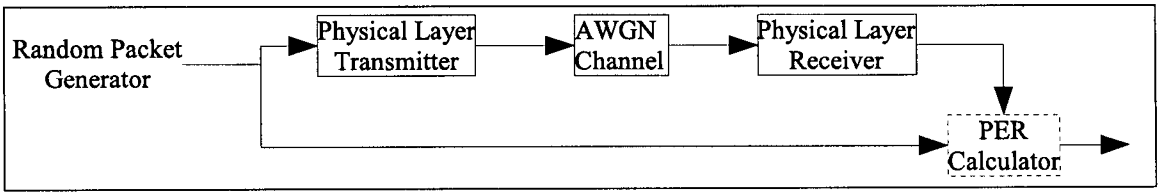

Figure 3.3: Physical Layer Simulation with AWGN Channel

The two user transmission system design is created by modifying the downlink physical layer design provided by the simulation package. The rate matching, Walsh spreading, and the time slot TDM sections of the transmitter, and the corresponding sections of the receiver are modified, as described in Section 3.2. The transmitter section is modified to allow for two user packet sources, whereas, the receiver is modified to correctly decode one of the user packet streams.

For the simulation of the two user transmission, the single user simulation is modified to include a second user and control channel. Three random packet sources are used to provide the forward traffic channel sources and control channel sources. The first forward traffic channel is simulated with the entire processing for the transmitter. The second forward traffic channel and the control channel are only simulated from the modulator onward in the transmitter to reduce the simulation time. The resulting system transmission is then processed through an AWGN channel. The receiver decodes the packets from the first user, as determined from the Walsh code allocations. The second user and the control channel are ignored by the receiver. Figure 3.4 shows the simulation of the two user transmission. The PER calculation is performed in the first user packet source.

As with the single user simulations, the two user transmission simulations are per-

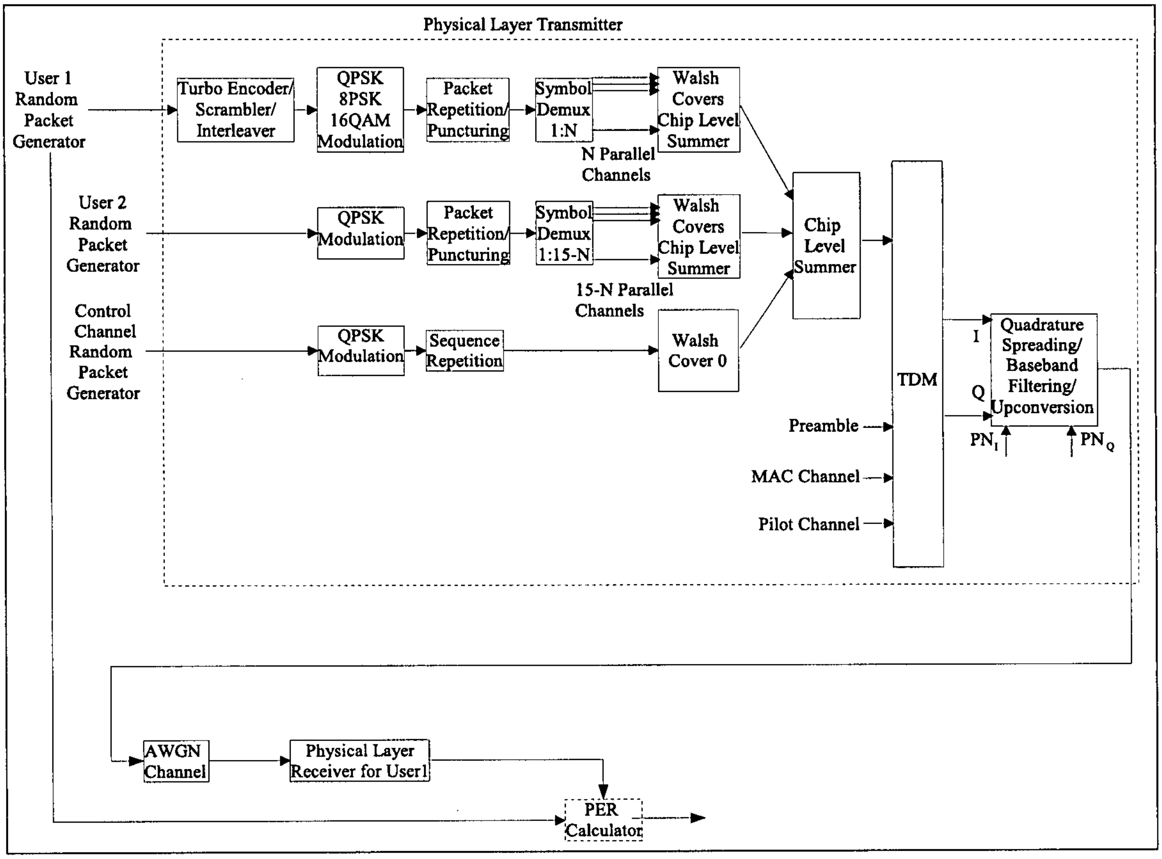

Figure 3.4: Physical Layer Simulation of Two User Transmission

formed to determine the C/I value that results in a 1% PER in AWGN channel. The simulation is performed for each combination of data rate and Walsh code separation. Table 3.3 shows the summary of the results for both the two user and single user modes. The blank combinations and the combinations that require C/I values above 13dB cannot be used. The 13dB limit is based on the assumption as described in Section 3.3.2. Consequently, the highest data is only possible with the use of all 16 Walsh codes.

The physical layer results are used to produce the table of two user data rate mapping values as shown in Table 3.4. The terminal reports the DRC and the Required Number of Walsh Codes. This infers the list of data rates that the terminal can support

			Number of Walsh Codes					C/I(dB)									
		Data	16	14	13	12	11	10	9	8	7	6	5	4	3	2	1
	Slots	Rate	*One User*	Two User													
0	16	38.4	-11.68	-11.31	-11.11	-10.83	-10.48	-10.26	-9.78	-9.29	-8.72	-7.97	-7.48	-6.11	-4.8	-3	-0.68
1	8	76.8	-9.3	-8.73	-8.51	-8.01	-7.64	-7.48	-6.72	-6.2	-5.82	-4.72	-3.68	-2.96	-2.38	-0.63	3.79
2	4	153.6	-6.16	-5.8	-5.48	-4.8	-4.15	-3.74	-3.3	-2.95	-2.85	-2.35	-1.69	-0.71	0.88	3.86	
3	2	307.2	-2.97	-2.88	-2.72	-2.35	-2.02	-1.67	-1.23	-0.8	0	0.86	1.99	3.78	7.55		
4	1	614.4	-0.72	-0.03	0.48	0.82	1.39	1.91	2.78	3.72	5.23	7.31					
5	4	307.2	-3.91	-2.82	-2.31	-1.88	-1.51	-1.25	-1.1	-0.87	-0.16	0.66	1.79	3.59	7.18		
6	2	614.4	-0.91	-0.21	0.27	0.62	1.18	1.8	2.59	3.6	4.99	7.14					
7	1	1228.8	3.6	5.03	6.1	7.27	10.1										
8	2	921.6	1.65	2.7	3.19	3.65	4.29	5.1	6.2	7.79	10.65	18.23					
9	1	1843.2	7.8	10.5	12.63	18.21											
10	2	1228.8	3.79	4.69	5.28	5.88	6.63	7.56	8.95	11.25	15.11						
11	1	2457.6	11.3	15	20.8												

Table 3.3: Physical Layer Results for AWGN Channel

for each of the possible number of transmission Walsh code assignments. The table of two user data rates is generated by selecting the best data rate possible given the DRC and the Required Number of Walsh Codes, and the number of transmission Walsh codes from the results shown in Table 3.3.

IS-856 DRC Index	Number of Walsh Codes	SINR (dB) For 1% PER	One User Data Rate (kbps)	One User	Transmitted Num. Walsh Codes (two user)													
					14	13	12	11	10	9	8	7	6	5	4	3	2	1
					Data Rate - DRC Index (-1 = N/A)													
0	16 (1 User)	-11.68	38.4	0	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	14	-11.31	38.4	0	0	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	13	-11.11	38.4	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	12	-10.83	38.4	0	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	11	-10.48	38.4	0	0	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	10	-10.26	38.4	0	0	0	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1	-1
0	9	-9.78	38.4	0	0	0	0	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1
1	16 (1 User)	-9.3	76.8	1	0	0	0	0	0	0	-1	-1	-1	-1	-1	-1	-1	-1
1	14	-8.73	76.8	1	1	0	0	0	0	0	0	-1	-1	-1	-1	-1	-1	-1
1	13	-8.51	76.8	1	1	1	0	0	0	0	0	0	-1	-1	-1	-1	-1	-1
1	12	-8.01	76.8	1	1	1	1	0	0	0	0	0	-1	-1	-1	-1	-1	-1
1	11	-7.64	76.8	1	1	1	1	1	0	0	0	0	0	-1	-1	-1	-1	-1
1	10	-7.48	76.8	1	1	1	1	1	1	0	0	0	0	-1	-1	-1	-1	-1
1	9	-6.72	76.8	1	1	1	1	1	1	1	0	0	0	0	-1	-1	-1	-1
2	16 (1 User)	-6.16	153.6	2	1	1	1	1	1	1	1	0	0	0	-1	-1	-1	-1
2	14	-5.8	153.6	2	2	1	1	1	1	1	1	1	0	0	0	-1	-1	-1
2	13	-5.48	153.6	2	2	2	1	1	1	1	1	1	0	0	0	-1	-1	-1
2	12	-4.8	153.6	2	2	2	2	1	1	1	1	1	0	0	0	0	-1	-1
2	11	-4.15	153.6	2	2	2	2	2	1	1	1	1	1	0	0	0	-1	-1
5	16 (1 User)	-3.91	307.2	5	2	2	2	2	1	1	1	1	1	0	0	0	-1	-1
3	16 (1 User)	-2.97	307.2	3	2	2	2	2	2	2	1	1	1	1	0	0	0	-1
3	14	-2.88	307.2	3	3	2	2	2	2	2	2	1	1	1	1	0	0	-1
3	13	-2.72	307.2	3	3	3	2	2	2	2	2	2	1	1	1	0	0	-1
3	12	-2.35	307.2	3	3	3	3	2	2	2	2	2	2	1	1	1	0	-1
3	11	-2.02	307.2	3	3	3	3	3	2	2	2	2	2	1	1	1	0	-1
3	10	-1.67	307.2	3	3	3	3	3	3	2	2	2	2	2	1	1	0	-1
3	9	-1.23	307.2	3	3	3	3	3	3	3	2	2	2	2	1	1	0	-1
6	16 (1 User)	-0.91	614.4	6	3	3	3	3	3	3	2	2	2	2	1	1	0	-1
4	16 (1 User)	-0.72	614.4	4	3	3	3	3	3	3	3	2	2	2	1	1	0	-1
4	14	-0.03	614.4	4	4	3	3	3	3	3	3	5	2	2	2	1	1	0
4	13	0.48	614.4	4	4	4	3	3	3	3	3	3	2	2	2	1	1	0
4	12	0.82	614.4	4	4	4	4	3	3	3	3	3	5	2	2	1	1	0
4	11	1.39	614.4	4	4	4	4	4	3	3	3	3	3	2	2	2	1	0
8	16 (1 User)	1.65	921.6	8	4	4	4	4	3	3	3	3	3	2	2	2	1	0
8	14	2.7	921.6	8	8	4	4	4	4	6	3	3	3	3	2	2	1	0
8	13	3.19	921.6	8	8	8	4	4	4	4	3	3	3	3	2	2	1	0
7	16 (1 User)	3.6	1228.8	7	8	8	4	4	4	4	6	3	3	3	5	2	1	0
10	16 (1 User)	3.79	1228.8	10	8	8	8	4	4	4	4	3	3	3	3	2	1	0
10	14	4.69	1228.8	10	10	8	8	8	4	4	4	3	3	3	3	2	2	1
10	13	5.28	1228.8	10	10	10	8	8	8	4	4	4	3	3	3	2	2	1
10	12	5.88	1228.8	10	10	10	10	8	8	4	4	4	3	3	3	2	2	1
10	11	6.63	1228.8	10	10	10	10	10	8	8	4	4	3	3	3	2	2	1
9	16 (1 User)	7.8	1843.2	9	7	7	7	10	10	8	8	4	4	3	3	3	2	1
9	14	10.5	1843.2	9	9	7	7	7	10	10	8	4	4	3	3	3	2	1
11	16 (1 User)	11.30	2457.6	11	9	7	7	7	10	10	10	8	4	3	3	3	2	1

Table 3.4: C/I to Data Rate Mapping Results in AWGN Channel

3.3.2 Radio Environment Simulation

The radio environment simulation is conducted to provide a simulation of the time varying channel for the users in the system. The simulation is to provide time varying SINR data for each of the terminals for use in the packet scheduling simulations. Accurately simulating cellular radio environments can be complex [18], however, the intent of these simulations is to provide a reasonable simulation of a cellular environment for analyzing the proposed system. Path loss, shadow fading, fast multi-path fading and the wide band multi-path propagation of the transmitted downlink signal are taken into account in the simulations. Similar modeling is performed in [19] [15] for IS-856 system simulations.

Hexagonal cells with maximum cell radius of 1000m, as shown in Figure 3.5, are used for the simulation cellular system. Furthermore, for the simulations a minimum distance between a base station and a mobile is set to 35m. The base station transmission level is set to 40dBm and each base station in the system is assumed to use the same frequency (2000Mhz). It is assumed that this transmission level encompasses all other components of a link budget such as antenna gain, etc. The simulation only considers the users in the center cell.

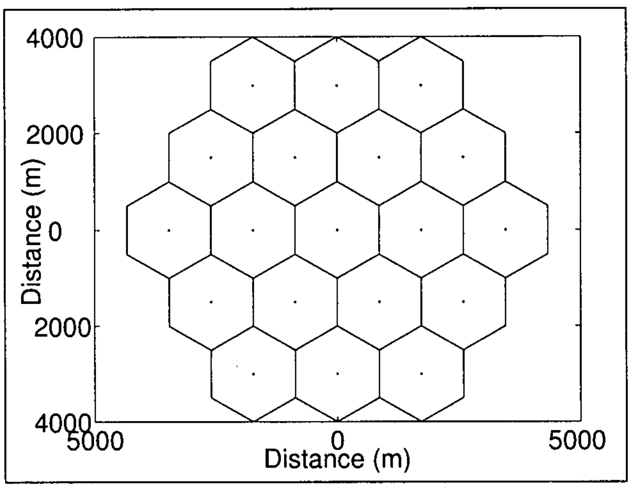

Figure 3.5: Simulation Cell Layout

The simulations are divided into 60 second simulation time blocks. Each mobile is randomly repositioned every time block simulating movement within a cell. The resulting position of the mobile is used in the path loss calculations of the signals from each of the base stations in the system. Fifty time blocks are used in the simulations resulting in 3000 seconds of simulation time. The same velocity is used for all the mobiles and is assumed a constant for the entire simulation.

For the simulations, the system is assumed to operate in an urban environment with path loss that can be modeled with the COST231-Walfish-Ikegami as described in [18]. Many of the urban environmental factors such as building and base station height are included in the loss calculations in the model. A set of parameters given to the model describes these environment features. For the simulations, a set of parameters are selected based on the suggested default values in [18]. The model parameters are: 13m base station height, 2m mobile height, 12m roof height, 20m building separation, roads 90 degree rotated from the propagation path, and carrier frequency of 2000Hz. The resulting path loss for the given distance is shown in Figure 3.6.

A path loss value is calculated for each terminal and base station pair. This gives a set of path loss values for each user with respect to the center cell base station as well as to each of the outer cell base stations. A calculated mean received signal power from each base station is used for the time block.

Shadow fading (or slow fading) is a result of attenuation of the received signal due to large obstructions between the transmitter and receiver. These small variations in the received signal can be modeled with a Log-Normal distribution [18]. For the system simulations, the shadow fading is modeled as described in [18] using Gudmundson model [20] by filtering a white noise source by a first order low pass filter

$$L_{k+1(\mathrm{dBm})} = \zeta \Omega_{k\mathrm{dBm}} + (1-\zeta)\nu_k$$

where $\Omega_{k+1(\mathrm{dBm})}$ is the mean envelope at location k, and ν_k is a zero-mean Gaussian random variable with variance σ^2. The resulting mean envelope is a time varying attenuation of the mean signal power at the receiver. The correlation parameter (ζ) is

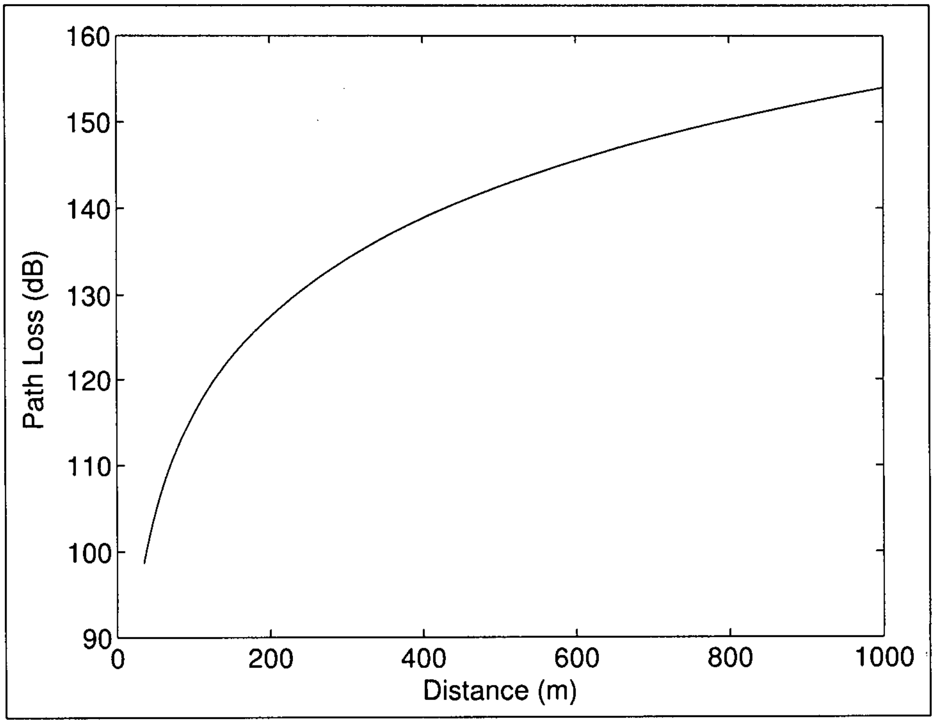

Figure 3.6: Simulation Path Loss

determined using [18][20]

$$\zeta = \zeta_D^{(vT/D)}$$

where D is the geographical distance in (m) between the transmitter and the receiver that gives a correlation of ζ_D. The v is the velocity in (m/s) of the mobile and T represents the sampling time in (s). Parameters estimated by [20] for microcellular systems are used for the simulations: ζ_D of 0.3 for a distance $D = 10$ m, zero-mean Gaussian variable with $\sigma = 4.3$ dB and a sample rate $T = 0.5$ sec. It is assumed that the resulting shadow fading value is constant between the sampling periods.

Multi-path fading (or fast fading) is the result of constructive and destructive combination of multiple reflected copies of the transmitted signal [18]. Objects (scatterers) in the local area of the mobile reflect the transmitted signal that combine at the mobile. Each reflected signal propagates through a slightly different path that results in a difference in the phase of the signal. Furthermore, the movements of the mobile introduces Doppler that effects the time variation of the phases. The envelope of the received signal can be modeled with a Rayleigh distribution if there are no line-of-sight components [18]. For the system simulations, the time variations in the multi-path fading is modeled as in [18] using Clark's scattering model. In this model, the Doppler power spectrum of the mobile channel is

$$S(f) = \frac{1.5}{\pi f_m} \frac{1}{\sqrt{1-(f/f_m)^2}}$$

where f_m is the maximum Doppler frequency. The f_m is determined by

$$f_m = v\frac{f_c}{c}$$

where v is the mobile velocity (m), f_c is the carrier frequency (2000Mhz), and c is the speed of light (3×10^8).

The Rayleigh faded envelope is generated by filtering two independent white Gaussian noise sources as shown in Figure 3.7 [18]. The filters are implemented with a 600-tap FIR filter designed to reproduce Clark's Doppler power spectrum with frequency response

$$H(f) = \sqrt{\frac{1.5}{\pi f_m} \frac{1}{\sqrt{1-(f/f_m)^2}}}$$

sampled at the frame rate (1/600 seconds). The filter is designed using Matlab FIR filter design package. The resulting fading envelope value is used in the calculation of the signal power at the mobile.

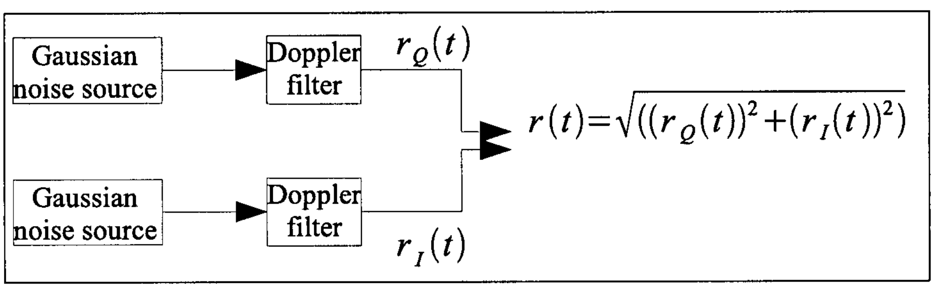

Figure 3.7: Rayleigh Fading Simulation

The effects of the path loss, shadow fading and Rayleigh fading are combined to give the received signal from a single propagation path as

$$\Omega_{r(\mathrm{dB})} = \Omega_{t(\mathrm{dB})} + L_{\mathrm{p(dB)}} + L_{\mathrm{s(dB)}} + L_{\mathrm{r(dB)}}$$

where Ω_r is the received signal level, $\Omega_{t(\mathrm{dB})}$ is the transmit signal level, $L_{\mathrm{p(dB)}}$ is the path loss, $L_{\mathrm{s(dB)}}$ is the loss due to shadowing and $L_{\mathrm{r(dB)}}$ is the loss due to Rayleigh fading.

The effects of a wide-band channel is simulated, as shown in [18], by combining discrete propagation paths to determine the received signal with

$$R(t) = \sum b_i \Omega_{r,i} u(t - \tau)$$

where i is the path index, b_i is the gain of the path and τ is the path delay. The ITU Pedestrian A multipath model as shown in Table 3.5 with user velocity of 3 km/h and Vehicular B multipath model as shown in Table 3.6 with user velocity of 100 km/h are used in the simulations.

Tap	Relative delay (ns)	Average power (dB)
1	0	0
2	110	-9.7
3	190	-19.2
4	410	-22.8

Table 3.5: Pedestrian A Multipath Model [2]

In the simulations, it is assumed that the mobile has a RAKE receiver with three fingers that lock on to the three strongest resolvable paths. The paths that are unresolvable with any of the three paths are assumed to contribute with signal ratio based on the delay difference. The remaining resolvable paths contribute to the interference

Tap	Relative delay (ns)	Average power (dB)
1	0	-2.5
2	300	0
3	8900	-12.8
4	12900	-10
5	17100	-25.2
6	20000	-16.0

Table 3.6: Vehicular B Multipath Model [2]

at the receiver. The transmissions from the other base stations add to the interference energy. The receiver is assumed to be able to resolve between two paths that have a delay more than a chip time. The chip time for IS-865 operating at 1.2288Mcps is approximately 814ns. The final SINR at the receiver is calculated by combining signals from all the base stations

$$(C/I)' = \frac{\left[\sum_{i\in k} b_i\Omega^0_{r,i} + \sum_{i\in m} b_i\Omega^0_{r,i}(1-\tau_i/\lambda)\right]}{\left[\sum_{j\neq 0}\sum_{i\in n} b_i\Omega^j_{r,i} + \sum_{i\notin k,m} b_i\Omega^0_{r,i} + \sum_{i\in m} b_i\Omega^0_{r,i}(\tau_i/\lambda)\right]}$$

where $\Omega^0_{r,i}$ is the received signal level from i^{th} path of the center cell base station (0), $\Omega^j_{r,i}$ is the received signal level from i^{th} path of the j^{th} base station and λ is the chip duration 1.2288×10^{-3} ns. The set k contains the three strongest paths from the center base station (0), the set m contains the unresolvable paths from the center base station (0), and the set n contains the set of all paths. τ_i is the time offset of the unresolvable path to the resolvable path.

As suggested by the system standard contribution [21] and paper [19], a maximum possible C/I value of 13dB is assumed due to receiver capabilities. This normalization of the C/I is performed as:

$$\frac{C}{I} = \frac{(C/I)'}{1 + \frac{(C/I)'}{10^{13dB/10}}}$$

For every physical layer frame time, a C/I value is generated for each mobile representing the received signal level. The resulting C/I value is mapped to a DRC value based on Table 3.4. The CDF of the system C/I of Pedestrian A and Vehicular B channel model are shown in Figure 3.8. The CDF is generated with 60 second simulations that have 1500 uniformly distributed mobiles.

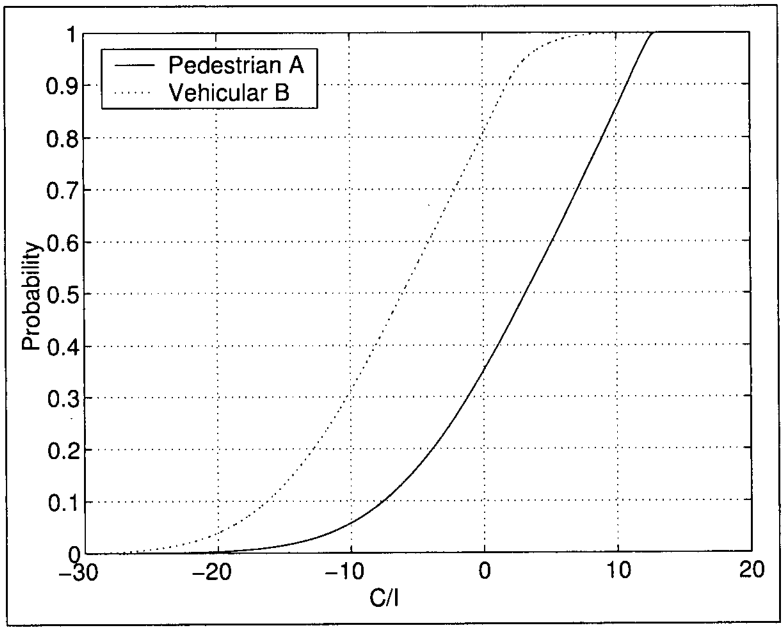

Figure 3.8: CDF of the system C/I

3.4 Two User Packet Scheduling

The packet scheduling algorithms described in Section 2.2 are intended for the selection of a single user for transmission. Allowing a second user to transmit simultaneously with the first user requires a different type of packet scheduling algorithm that takes advantage of the second user. To minimize the effect of the second user on the first user, the first user is selected as in the single user algorithms. The second user is selected as to provide the system with the highest throughput.

The selection of the second user is limited by the conditions described in Section 3.1 including the number of Walsh codes available (α) and the transmission duration of the first user. The second user is assigned the Walsh codes that are not required by the first user, hence, only users that are able to support transmission with α codes can be selected. Furthermore, the second user transmission duration is limited to a duration shorter than the first user. Forcing the second user to terminate before or at the same time as the first user allows for the alignment of the two users at the end of the first user transmission. If the first user still has slots available after the termination of the second user, then scheduling another (second) user is attempted.

To gain the maximum throughput from the transmission of the second user, the terminal with the highest data rate given the available Walsh codes (α) with a packet size that takes the same or fewer number of slots as the first user is selected as the second user. The second user is selected with

$$U'(s) = \arg\max_{u} DRC_u(s)$$

with the conditions on u:

$$u \neq U(s)$$
$$\Gamma_u(15 - \alpha_U(s)) \geq 0$$
$$\Lambda_u(15 - \alpha_U(s)) \leq \rho_U(s))$$

where $U'(s)$ is the terminal selected as the second user. The term $\alpha_U(s)$ is the minimum number of Walsh codes required to support the DRC reported by $U(s)$ and $\rho_U(s)$ is the number of slots required by $U(s)$. The function $\Gamma_u(\mu)$ gives the data rate index (DRC) that can be used by terminal u to transmit with μ Walsh codes. The function $\Lambda_u(\mu)$ gives the number of slots that is required for μ Walsh codes by terminal u. The two functions are determined by a lookup to Table 3.4 using the $DRC_u(s)$. Furthermore, the data rate of the second user is determined by the table.

The two user transmission is applied to the MAXCI algorithm, the PF algorithm and the EXPO algorithm, as described in Section 2.2, to measure the performance of the two user transmission mode. The two user version of the algorithms are termed MAXCI2, PF2, and EXPO2 respectively.

The algorithms in Section 2.2 are simulated with selected parameters. For the PF and PF2 simulations the value of 1000 slots is used for t_c, as suggested in [13]. For the EXPO and EXPO2 simulations: the same weight a_u is used for all the terminals, the value of t_c is set to 1000 as in the PF algorithm, γ is set to 10^{-2} and T is set to 300 slots. The EXPO algorithm defines the packet delay as the number of slots since the last transmission, therefore, the algorithm would limit this delay to 300 slots for majority of the packets.

For comparison, the round robin (RR) and the LCS algorithm, as described in Section 2.2, are simulated. The RR algorithm does seem fair except that in IS-856 the packets are contained in a varying number of slots. Therefore, this algorithm equalizes the number of packets transmitted rather than slots. The LCS algorithm uses channel information with the intension of optimizing throughput and equalizing the number of slots assigned to the terminals. For the LCS algorithm, a constant value of 0.3 for $a(s)$ is used for the simulations. The adaptation rate changes with $a(s)$, therefore, the performance of the algorithm changes with different values of a.

3.5 System Simulations and Results

The system performance of the two user transmission is determined using system simulations consisting of radio environment simulations as described in Section 3.3.2 and packet scheduling algorithms as described in Section 3.4. The packet scheduling algorithms use the channel simulation data for all the terminals to schedule the packets. The packet scheduling algorithms are given the DRC and minimum Walsh code as in a real system. The same set of feedback information is given to all the algorithms to make a fair comparison.

Simulations are performed with different number of terminals in the system for Pedestrian A, and Vehicular B channel environments. The performance of the scheduling algorithms are compared based on the average system date rate, the packet delay and system access fairness. It is assumed that the DRC estimate provided by the terminal is accurate at the packet transmission time. Errors in the DRC due to feedback delay or terminal measurement inaccuracy is not considered. Furthermore, it is assumed that the DRC is valid for the entire duration of the packet transmission with no packet loss. The simulations do not include the effects of early packet termination (H-ARQ). It is assumed that packets are always available for transmission (i.e. the terminals are backlogged).

3.5.1 Average System Data Rates

The average system data rates for Pedestrian A and Vehicular B channel models are shown in Figure 3.9 and Figure 3.10 respectively. The average system data rate is calculated as the sum of the system transmission data rates divided by the duration of the simulation time. For the two user transmissions, the transmissions from the second user is added to the calculations.

The RR scheduling algorithm does not utilize the channel information, therefore produces poor average system data rates. Furthermore, the RR uses packet based round robin scheduling that results in the lower data rate users gaining more time slots. The LCS algorithm gives average system bit rates similar to that of PF and EXPO for Pedestrian A and slightly lower for Vehicular B. In particular, it performs poorly when there are fewer number of terminals in the system.

The MAXCI has the best system average bit rate and shows the maximum achievable system bit rate by the system. As would be expected, the algorithm does not balance the system access between the terminal. During the simulations, only a few of the terminals get access to the system. These terminals provide all the system throughput. For Pedestrian A model the system gets close to the system capacity given enough users. The MAXCI2 algorithm provide only a slight addition to the system bit rate compared to the MAXCI. The MAXCI2 algorithm is intended to show the maximum achievable system data rate for two user transmission in the simulations.

The PF produces higher average system rates than the EXPO algorithm. This is expected as the EXPO algorithm is also attempting to limit the packet delays. The gain from two user transmission can be seen from Figure 3.9 and Figure 3.10. The PF2 algorithm provides about 1.5% to 3% gain when compared to the PF algorithm for Pedestrian A model and about 4% to 7% gain for the Vehicular B model. Similarly, the EXPO2 algorithms provides about 1.5% to 3% gain for Pedestrian A model and provides about 4% to 9.5% gain for Vehicular B model.

The two user transmission provides more gains when the system has more users in the system. The two user transmission is more beneficial when the second user has a high C/I. This can be seen in the results of PF2, EXPO2, and MAXCI2 when compared to their single user versions in particular for the Vehicular B model. The highest data rate (2457.6 kbps) is not used in two user transmission mode, therefore, when the system transmission to the first user at the highest data rate, there can be no transmissions to a second user. Hence, when the system average data rates are high, the gains from the two user transmission is limited. This can be seen the in the results for Pedestrian A models.

3.5.2 Average System Packet Delays

The packet delay results are intended to show the amount of delay incurred by the packets. For the packet delay calculation, the time (in terms of slots) between packet scheduling (or the head of line delay) and the packet transmission time after scheduling are added together. To compare the different sized packets of the data rates, the delay value is divided by the size of the packet as the number of multiples of 1024 bits to give an approximate packet delay in terms of 1024 bit packets. The MAXCI and MAXCI2 results are omitted from the packet delay results due to the large packet delays created by the algorithm.

The mean packet delay results are shown in Figure 3.11, and Figure 3.12 for the Pedestrian A, and Vehicular B channels respectively. The results can be correlated to the data rates results. The lower data rates require more number of slots to transmit and each packet takes more of the system transmission time. Furthermore, the average packet delay experienced by the terminals increases as the number of users is increased. The RR algorithm experiences a large packet delay due to the lower average system data rate. The mean packet delay for the two user algorithms shows a small decrease when compared to their respective one user transmission mode scheduling algorithm. This is expected as the two user mode does not alter the single user mode but provides more packet transmissions.

The maximum packet delay is considered here to show the extreme values of packet delay. The maximum delay results are shown in Figure 3.13, and Figure 3.14 for the Pedestrian A, Vehicular B channels respectively. The results show the worst case packet delay for the duration of the simulation. The EXPO and EXPO2 algorithms produce the best results as the algorithms specifically consider the packet delay. The algorithms for one user transmission and two user transmission show very little difference in terms of maximum delay.

Figure 3.15, and Figure 3.16 show the 99-percentile of the packet delay distributions for the Pedestrian A, and Vehicular B channels respectively. These can be interpreted as the maximum delay that 99% of packets experience. The EXPO and EXPO2 algorithms are able to maintain the lowest delay results. The two user transmission shows slightly lower delays for the Vehicular B channel and about the same for the Pedestrian A model.

To determine the fairness in the system transmission resource allocations for the terminals, the difference in the number of slots given to the terminals is considered. The difference between the terminal that receives the maximum number of transmission slots and the terminal with the minimum are compared. As in [22] [17], for any two terminals i and j, the maximum value of $|S_i/r_i - S_j/r_j|$ is used to indicate the fairness of the algorithm. S_i is the number of slots allocated to user i and the weight terms r_i and r_j are set to 1/(Number of Users). The comparison is shown in Figure 3.17, and Figure 3.18 for the Pedestrian A, and Vehicular B channels respectively. The LCS algorithm perform the best in fairness as the algorithm is primarily meant to fairly equate the number of slots allocated to the terminals. The two user transmission mode algorithms show less fairness compared to their respective one user algorithms. By selecting the second user based on data rate, the two user transmission has benefited the terminals with higher C/I.

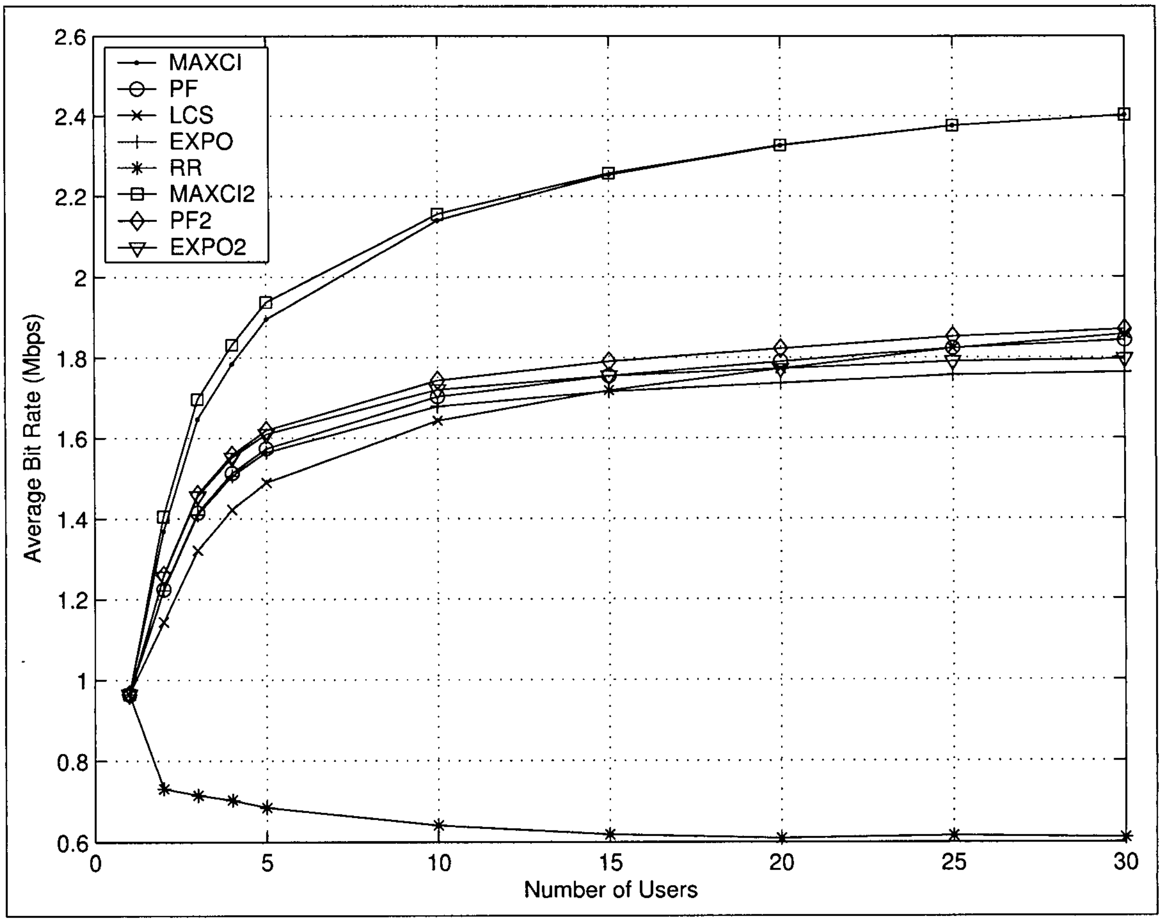

Figure 3.9: Pedestrian A: Average System Data Rate

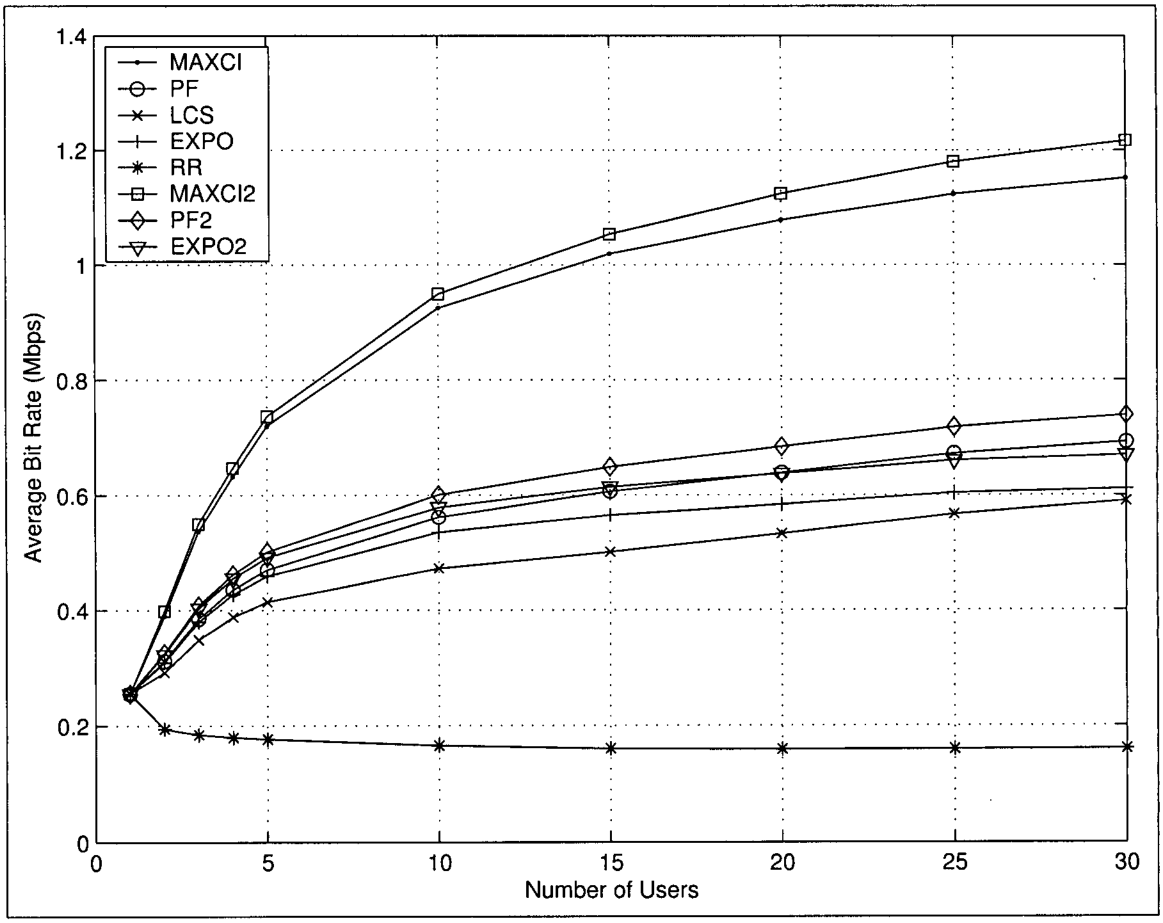

Figure 3.10: Vehicular B: Average System Data Rate

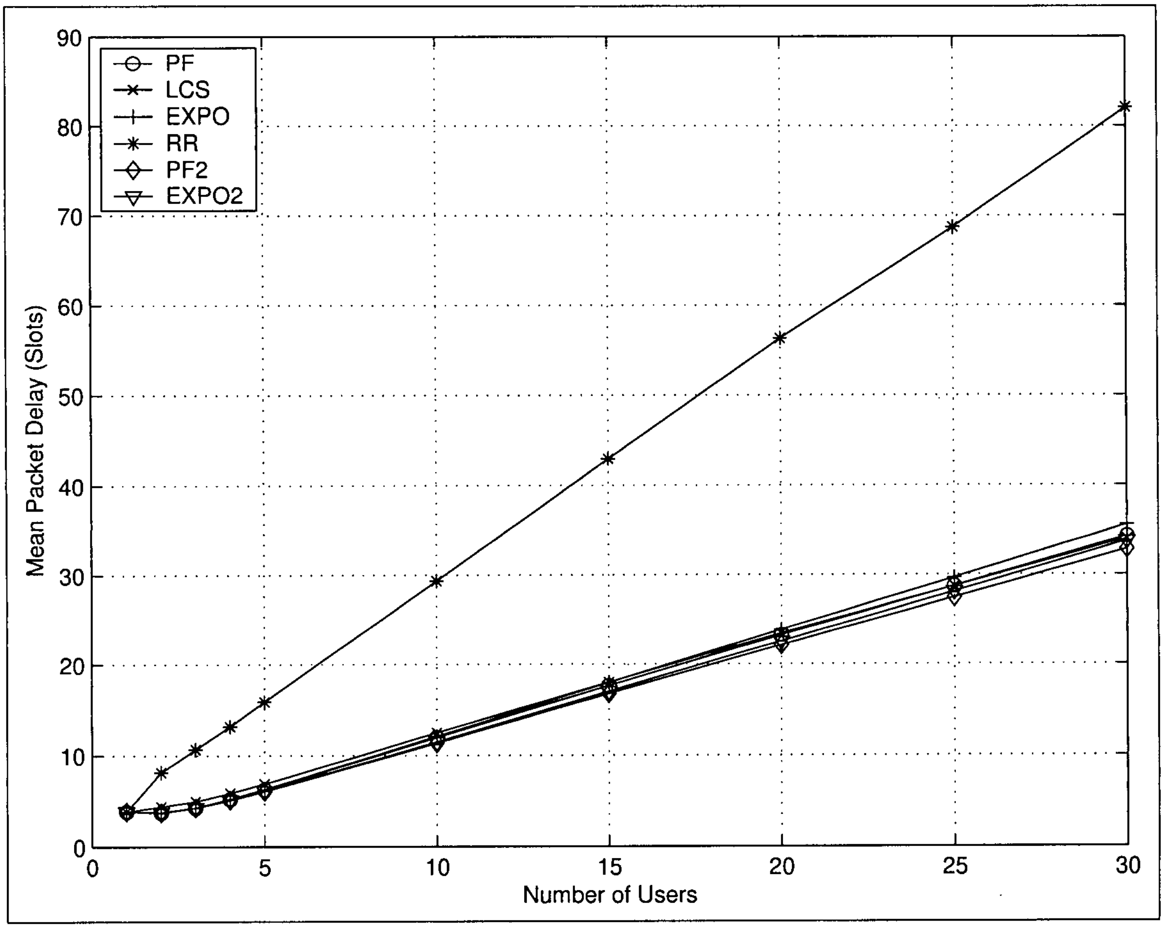

Figure 3.11: Pedestrian A: Mean Packet Delay

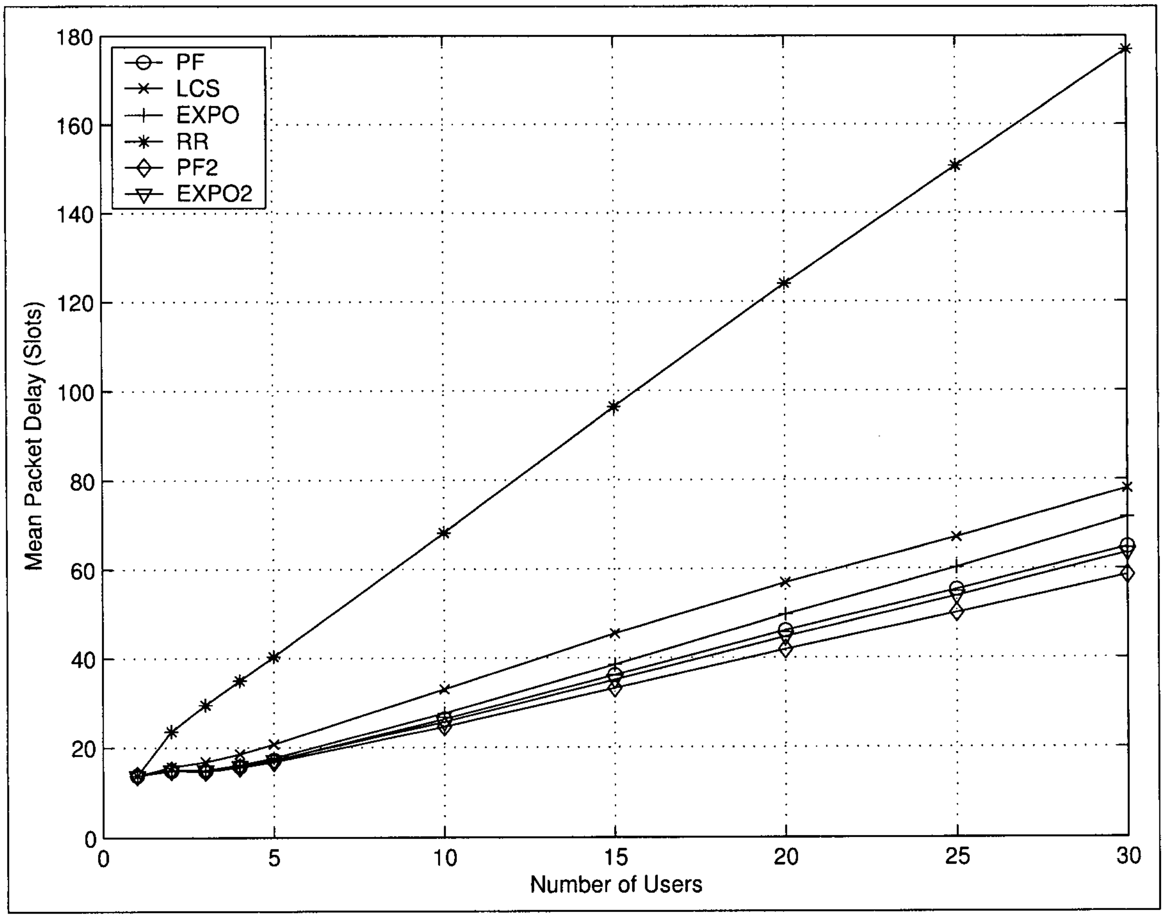

Figure 3.12: Vehicular B: Mean Packet Delay

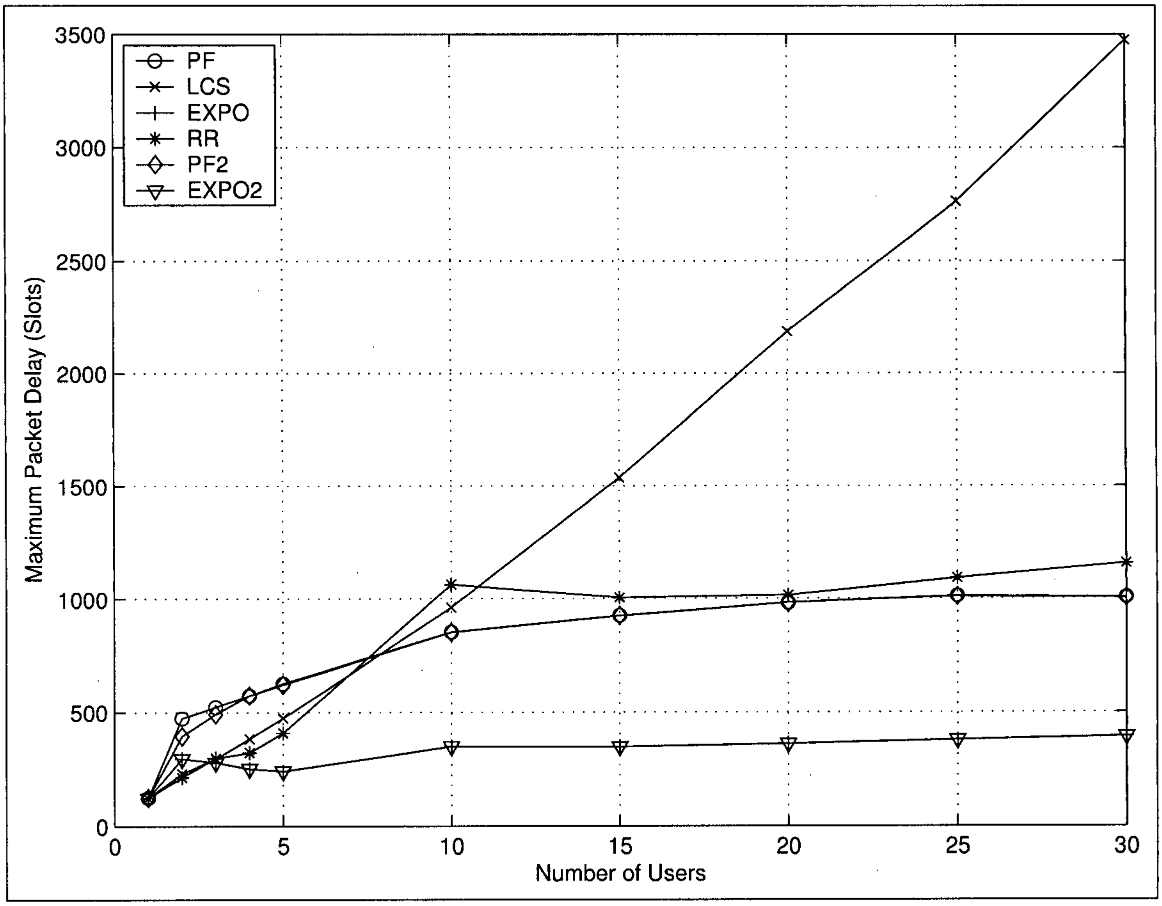

Figure 3.13: Pedestrian A: Maximum Packet Delay

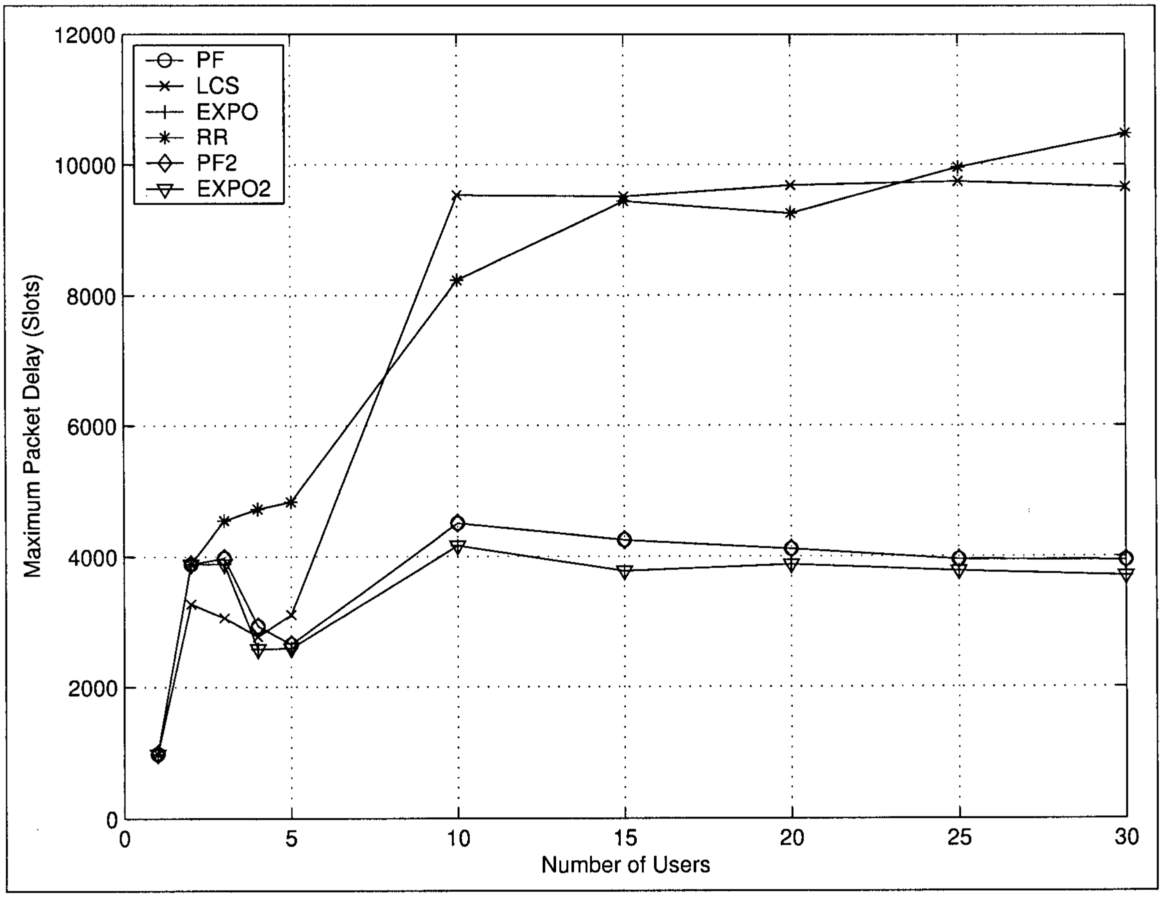

Figure 3.14: Vehicular B: Maximum Packet Delay

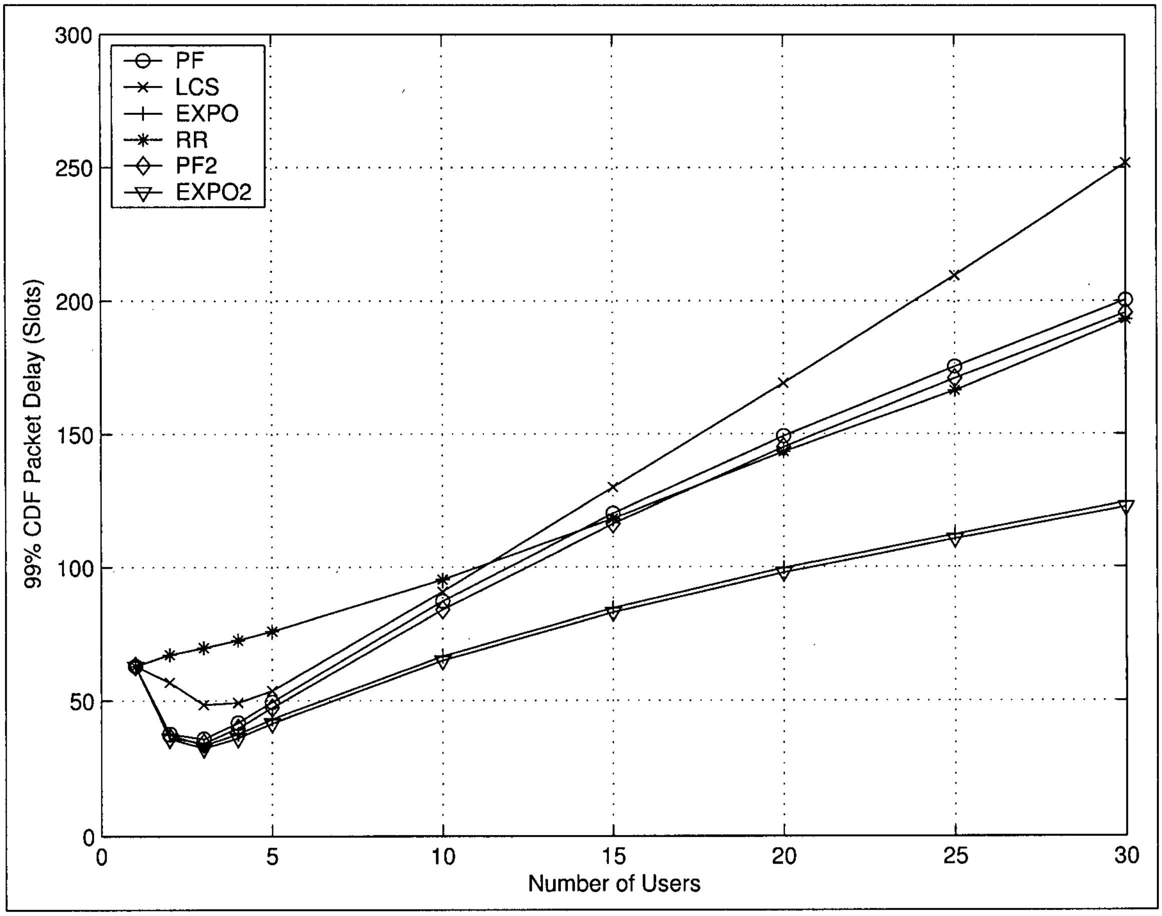

Figure 3.15: Pedestrian A: 99th Percentile of Packet Delay Distribution

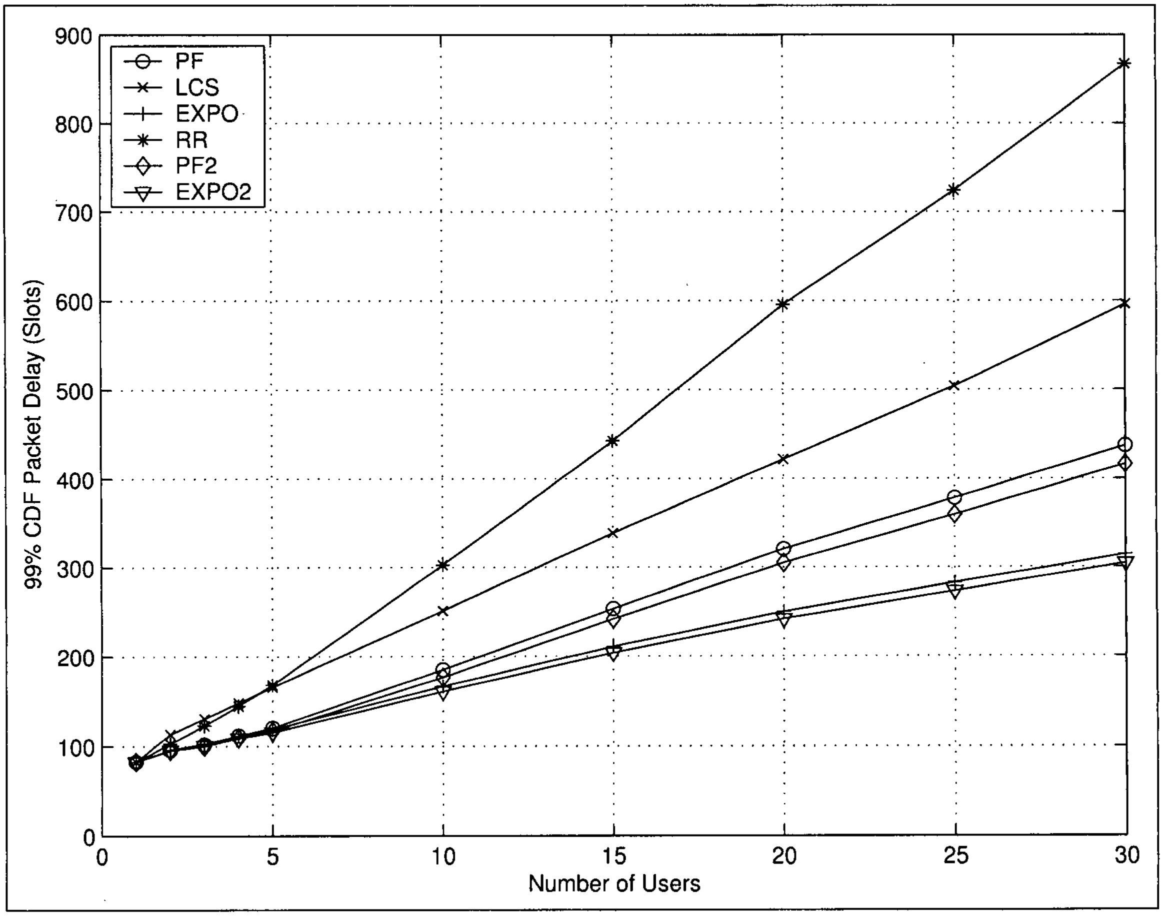

Figure 3.16: Vehicular B: 99th Percentile of Packet Delay Distribution

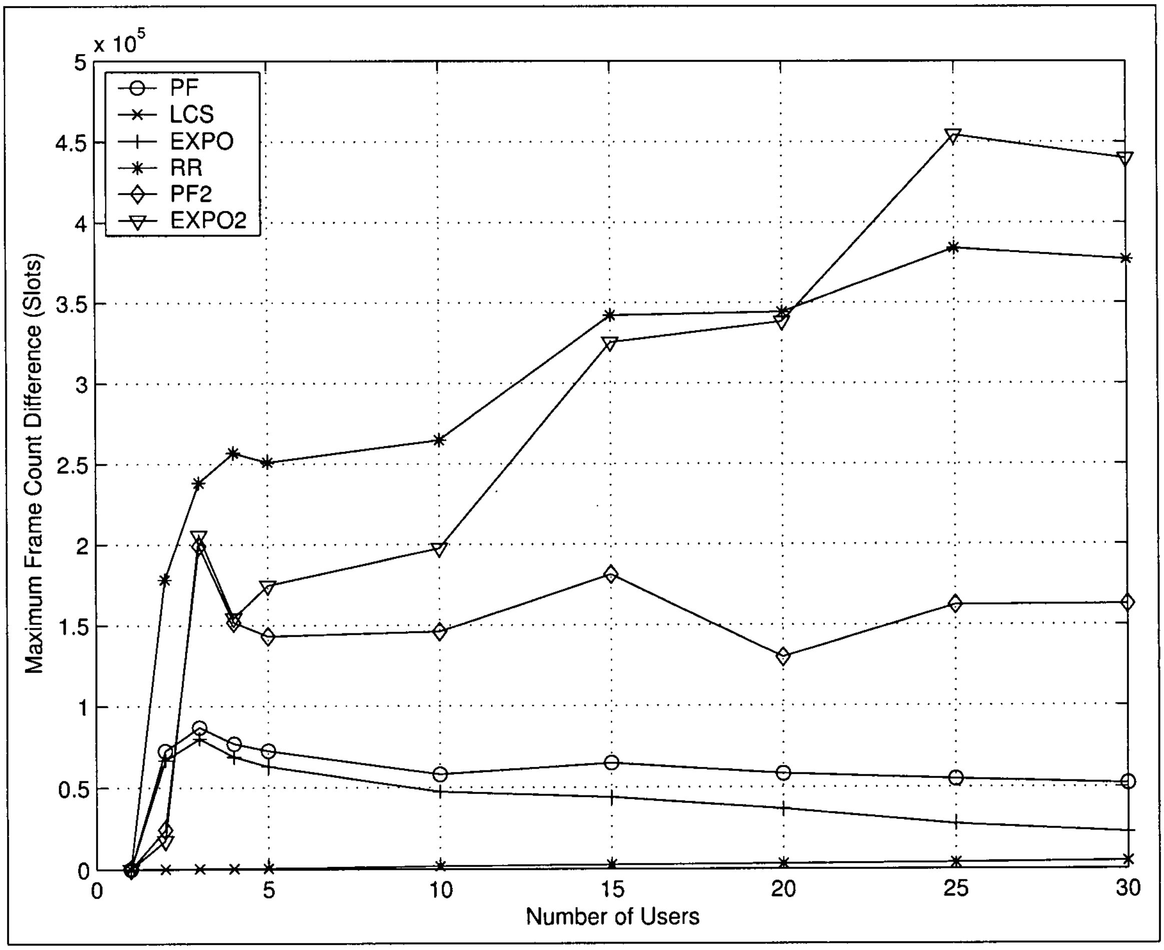

Figure 3.17: Pedestrian A: Maximum Difference in Slot Count Between Terminals

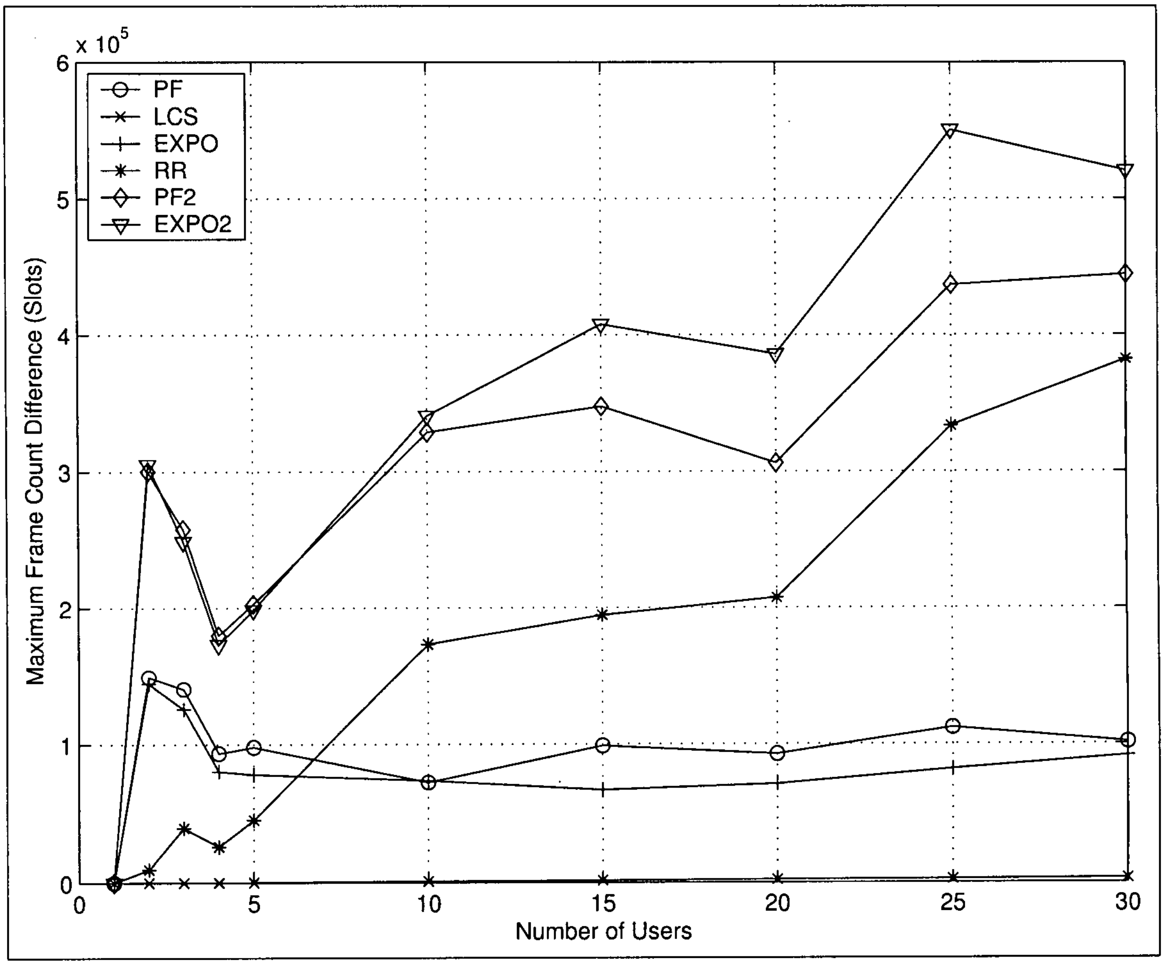

Figure 3.18: Vehicular B: Maximum Difference in Slot Count Between Terminals

Chapter 4

System Analysis

In this chapter the performance loss due to the quantization of the data rates and the performance gain of optimizing the data rates for the transmission channel are analyzed. The quantization loss is analyzed by comparing the performance of the system with various number of discrete data rates. This is extended to both the single user and the two user transmissions. A set of optimized data rates are compared to the original set of data rates using the system simulations. The optimized discrete data rates are selected to prove higher average system data rates for the specific channels.

4.1 Discrete Data Rates

In order to determine the performance loss due to the quantization of the data rates, a continuous data rate function must be available. To establish this function, data rates other than the ones defined in the IS-856 standard must be achievable. The original data rates defined in the standard are created using different combinations of modulation and coding. It is, therefore, assumed that data rates other than the defined data rates can be produced with the appropriate selection of modulation and coding. Moreover, it is assumed that the C/I required for 1% PER of such intermediary data rates can be approximated with linear interpolation of the results from the defined data rates.

The full set of data rates defined by the IS-856 standard cannot be used as some of the data rates are duplicated. The duplicate data rates have different packet sizes that require different C/I for 1% PER. The performance of the two packet sizes are different due to the different length of preamble and data segment per slot. Therefore, only the data rates that use 32 chips of preamble per slot are used. This allows for the consistent performance of the data rates in producing the continuous data rates function and eliminates the duplicate data rate definitions. Naturally, the results from the interpolation would deviate from the original IS-856 standard but would provide consistent results. The data rates selected for the interpolation are shown in Table 4.1.

Data Rate	C/I (dB) for 1% PER	Packet Size (bits)	Slots
38.4 kbps	-11.68	1024	16
76.8 kbps	-9.3	1024	8
153.6 kbps	-6.16	1024	4
307.2 kbps	-2.97	1024	2
614.4 kbps	-0.72	1024	1
1228.8 kbps	3.6	2048	1
1843.2 kbps	7.8	3072	1
2457.6 kbps	11.3	4096	1

Table 4.1: Data Rates Used for Interpolation (Results from Section 3.3)

Using these selected data rates, a function of the continuous data rates and discrete data rates for the given C/I is defined as shown in Figure 4.1 for one user transmission. These results are produced with interpolation, hence, the performance of the data rates beyond the defined data rates range are undefined. Therefore, it is conservatively assumed that the data rates below the lowest defined data rate (by the standard) are not possible and the maximum data rate is the highest defined by the standard. To this end, no transmissions are assumed when the C/I is below the required C/I of the lowest define data rate. Similarly, the maximum defined data rate is used when the C/I is above the C/I required for the highest standard data rate.

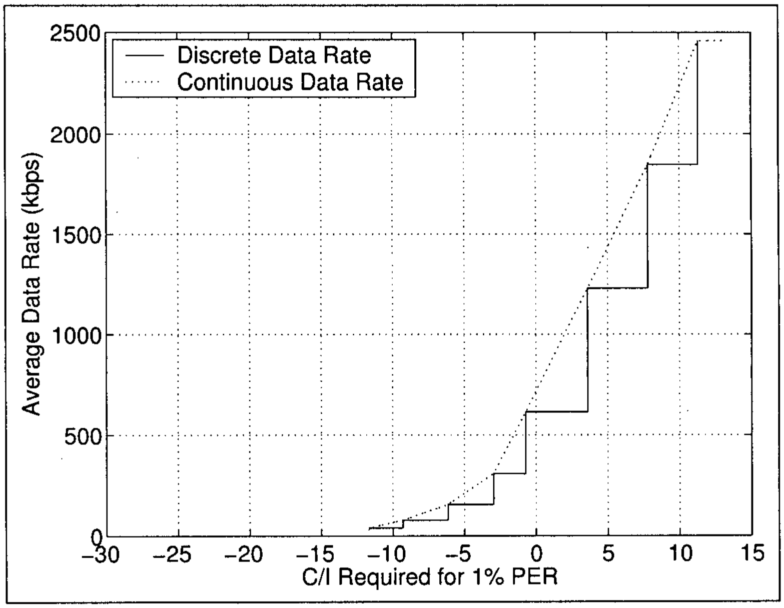

Figure 4.1: Function of the Continuous Data Rates and Discrete Data Rates

The spectral efficiency of the rate adaptation outlined in Chapter 2 and Chapter 3 are used to estimate the performance loss due to quantization. The spectral efficiency for continuous data rate adaptation is given by (2.1), discrete data rate adaptation is given by (2.3), and discrete data rate adaptation for the two user transmission mode is given by (3.2). The distribution of the channel models defined in Section 3.3.2, as shown in Figure 3.8, are used as the distributions of the instantaneous received SNR ($p(\gamma)$) in these relations. The transmission power is assumed to be constrained to a fixed value and is therefore not adapted. The adaptation is also constrained to selecting data rates that can maintain a 1% PER for the given C/I. Furthermore, all the data rates are assumed to transmit for a single slot duration. The resulting average data rates (R) are shown in Table 4.2.

Channel Model	Continuous Adaptation	Discrete Adaptation	Two Users
Pedestrian A	1182	1000	1094
Vehicular B	339	253	287

Table 4.2: Average Data Rate (R) (kbps)

4.2 Increasing Number of Data Rates

The resulting average data rates of Table 4.2 are valid for the data rates shown in Table 4.1. However, the effect of quantization is reduced by increasing the number of data rates available to the system. By increasing the number of data rates, the discrete rate adaption in (2.3) approaches the continuous data rate adaptation in (2.1). Figure 4.2 and 4.3 shows the average data rates for increasing number of discrete data rates for Pedestrian A and Vehicular B channel models.

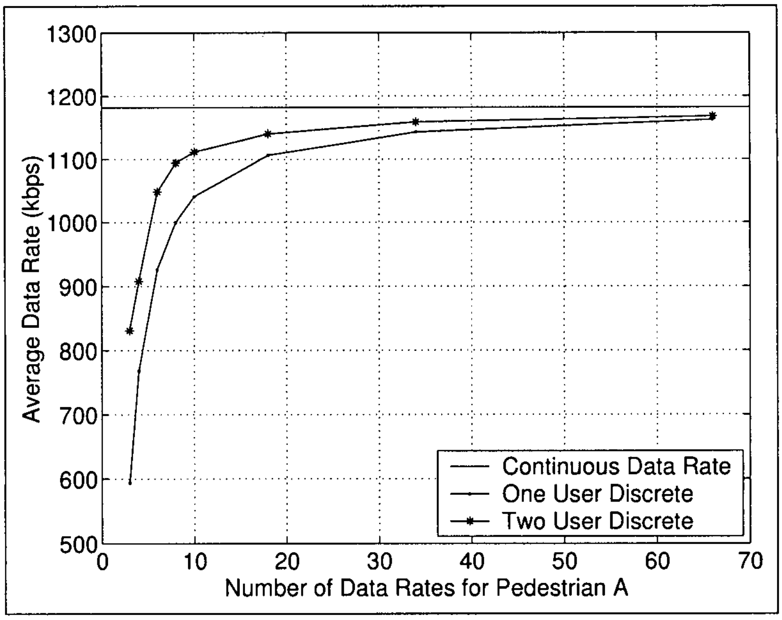

Figure 4.2: Average Data Rates for Discrete Data Rates for Pedestrian A

In the figure, the data rates are chosen by evenly distributing the data rates based

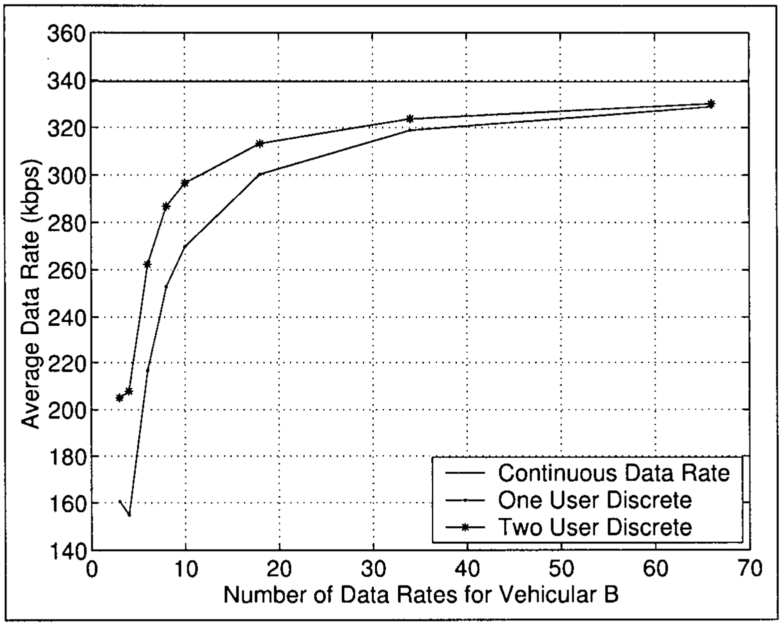

Figure 4.3: Average Data Rates for Discrete Data Rates for Vehicular B

on the required C/I for 1% PER. The minimum data rate supported by the standard (38.4 kbps) and the maximum data rate (2457.6 kbps) are always included, hence, the minimum number of data rates shown in the figure is three. The other data rates evenly cover the range of C/I available to the system between the minimum and maximum data rate. For the data rates used in the one user transmission, the required C/I for 1% PER is determined by sampling the interpolated continuous data rate function.

For two user transmission, the users are each allocated a number of Walsh codes. Depending on the number of Walsh codes (w) assigned to a user, the C/I required for 1% PER is different for a given data rate. As with the single user case, the C/I required for 1% PER, for each w, is determined with interpolation using the the data rates defined in Table 4.2.

4.3 System Simulations

The system simulations of Section 3.5 are duplicated to show the effect of changing the number of data rates available to the system. The average data rate calculated in the previous section does not take into account the multiple slot transmissions of IS-856 system. Therefore, the system simulation in this section assume that each data rate transmits for a single slot. The system simulations are conducted using MAXCI, PF, MAXCI2 and PF2 packet scheduling algorithms as described in Section 3.5.

The system simulation channel samples are independently generated from that of the samples used in the distributions of Section 3.3.2. The system simulations use fifty time blocks of 60 seconds in duration, whereas, the samples of the distribution are generated from a single terminal with 1500 time blocks of 60 seconds in duration as described in Section 3.3.2. Fewer time blocks are used in the system simulations due to the simulation time for multiple terminals.

Figure 4.6, and Figure 4.7 show the resulting MAXCI average system data rates for the Pedestrian A and Vehicular B channel models respectively. When there is only a single user in the system, the results are similar to the results from the previous section as the data rate distribution of the previous section is valid for a single user. However, as more users are available to the system, the system can find more users that can achieve the higher data rates, hence changing the effective channel distribution and providing multi-user diversity gain. The gain from increasing the number of data rates, therefore, is lower as the number of users are increased.

The two user mode average system data rates for MAXCI2 are shown in Figure 4.8, and Figure 4.9. The two user mode results achieve higher results compared to the single user mode. As the number of data rates are increased, the benefit of the two user mode is diminished. The two user mode does not have any ability to transmit the second user when there are higher number of data rates due to the decreasing quantization.

These results can be reiterated by taking a cross section of the results by isolating the results for 10 Users and 1 User as shown in Figure 4.10 for Pedestrian A with MAXCI and Figure 4.11 for Vehicular B with MAXCI. The figures show that the system simulations for a single user follows the results in Section 4.2. The 10 User case gains from the multi-user diversity but follows similarly. The two user transmission allows the system to more quickly eliminate the quantization loss by converging to the results of the continuous data rates adaptation faster. The figures also show that the two user transmission has lower gain as the number of data rates increase.

4.4 Multiple Slot Transmissions

The results from the Section 4.3 assume that each data rate occupies only a single slot. The relevance of single slot assumption is that the spectral efficiency calculations assume that the system can change the data rates at each slot instance. However, the standard defines several data rates with multiple slots to allow for terminals using a lower data rate to transmit a minimum sized packet when scheduled. Therefore, the effect of multiple slot data rates are taken into account in this section. The results from this section are used for comparison in the next section.

The standard does not allow the base station to change the data rate of a multiple slot packet transmission once the transmission has started but does allow the terminal to terminate the transmission early if it is able to successfully decode the transmission before receiving the entire packet. For the system simulations in this section, as in Section 3.5, it is assumed the DRC is valid for the duration of the packet transmission, hence, the effect of packet failures and early packet termination is not considered. As a consequence of this assumption and the standard's definition, the multiple slot transmissions create a difference between the channel distribution and the effective distribution of the data rate transmissions.

For the second user, an additional discrepancy results in (3.2) from limiting the second user to fewer or equal number of slots as that of the first user, as described in

Chapter 3. This limitation would effect the average second user data rate calculation used in (3.2). The data rates allowed by the second user are limited to those that are possible given the data rate of the first user, hence, effecting the actual average data rate of the second user.

To use the data rates from Section 4.3 in system simulation for multiple slot transmissions, the slot count for the data rates need to be determined. The number of slots that these packets occupy is not explicitly defined, therefore, these are approximated. Even though the data rates can be modified by changing the coding rate, the resulting modulated and coded packet may not fit into an integer number of time slots. For these data rates, a simple slot count rule is assumed in this section based on the slot counts shown in Table 4.2. The data rates above 614.4 kbps use a single slot. Therefore, the actual size of these packets are not multiples of 1024 bits. For data rates less than 614.4 kbps, the fewest number of slots that gives a packet size greater than 1024 bits is used. Therefore, the packet size for these data rates would vary from 1024 bits to 2048 bits.

4.5 Optimized Data Rates

The discrete data rates selected in the previous sections were chosen by evenly dividing the C/I range. This method or even the original IS-856 data rate selection may not be the optimum selection of the data rates. Therefore, discrete data rates optimized to improve the system performance are analyzed. The objective of the optimization in this section is to improve the performance of the system simulations.

Directly using the system simulations for optimization is difficult, therefore, optimization of the data rates are based on the channel models and the spectral efficiency definition with the single slot assumption. The values of the discrete data rates are optimized by maximizing (2.3) and (3.2). These definitions provide an optimization of data rates for a system with a single terminal. Optimization for multiple transmission slots or multiple terminals in the system are not considered. Furthermore, the performance gain from optimizing for multi-user diversity or a specific packet scheduling algorithm are not considered.

For the optimization, the discrete data rates (k_i) of (2.3) are variables for the optimizer. As in Section 4.2, the minimum data rate (38.4 kbps) and the maximum data rate (2457.6 kbps) are always included in the set of data rates. The other intermediary data rates are determined by the optimization. The interpolation from Section 4.1 provides the corresponding C/I rate region $(\gamma_i, \gamma_{i+1}]$ for the selected data rates. The optimization is performed as to maximize (2.3) subject to

$$\gamma_0 \geq -11.5$$

$$\gamma_N \leq 11.2$$

$$\gamma_{i+1} - \gamma_i \geq 0.1$$

for the single user adaptation. For the two user optimization, (3.2) is maximized subject to the same variable constraints. The optimization is performed with Matlab optimization package (fmincon function) with the initial values of the discrete data rates (k_i) set to the values used in previous section (evenly dividing the C/I range).

This optimization procedure can be used to optimize for any number of discrete data rates (k_i). The resulting average data rates using 2.3 and 3.2 with the discrete data rates (k_i) selected by the optimization is shown in Figure 4.4 and Figure 4.5. The optimization is performed over varying number of data rates to show the effect of the optimization. The figures show the improvement in the average data rates with optimized discrete data rates. The original discrete data rate selection shows the equally spaced selection of C/I (and not the IS-856 data rates). For the single user discrete data rates, the optimization takes advantage of the channel model to determine the best discrete data rates for the system.

For the two user mode, the optimization takes advantage of the channel models as well as the contributions of the second user. The large gain is attributed to the average data rate assumption in the two user average data rate calculation (3.2). The data rates selected by the optimizer for two user mode are higher than that of the single user. For example, the C/I rate region of the lower data rates have been significantly extended. This provides the second user with more Walsh codes when the first user C/I is lower. The second effect of the use of higher data rates is that it would improve the performance when there are many users in the system as multi-user diversity tends to favor higher data rates.

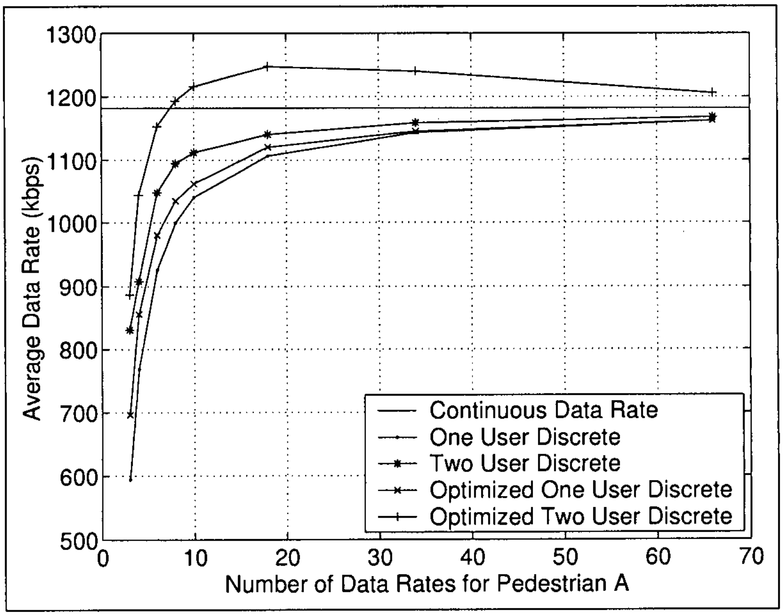

Figure 4.4: Average Data Rates for Optimized Discrete Data Rates for Pedestrian A

A comparison between the data rates given by equally spaced selection of C/I and the optimized data rates is performed using the system simulations. The simulation results are shown in Figure 4.12, Figure 4.13, Figure 4.14, and Figure 4.14 for Pedestrian A and Vehicular B models with MAXCI and PF algorithms. The figures show the average system data rate results for variable number of data rates with 10 users in the system.

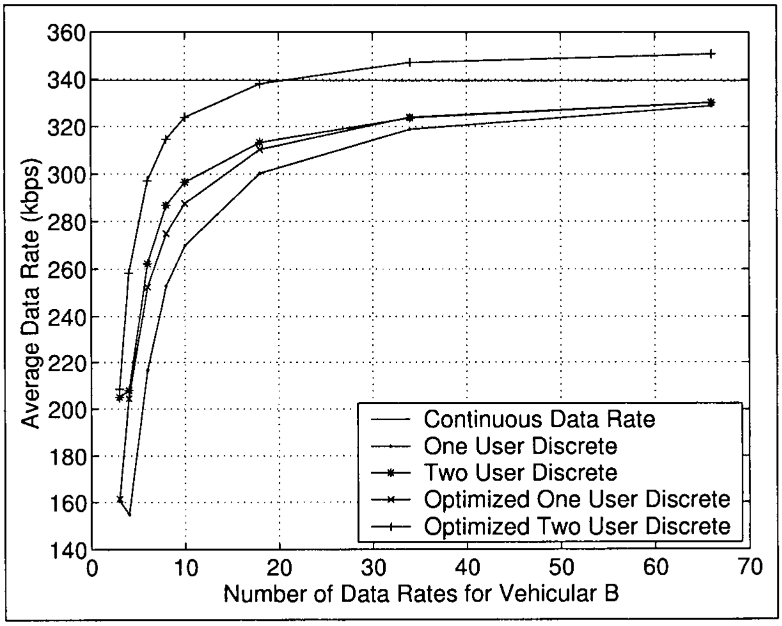

Figure 4.5: Average Data Rates for Optimized Discrete Data Rates for Vehicular B

4.5.1 Optimized Data Rates for IS-856

In this section, the optimization criteria from Section 4.5 is used to optimize the IS-856 data rates to determine the performance gain from optimization of the data rates. The original system data rates used in the standard include duplicate data rates. For the optimization in this section, duplicate data rates are not included. The performance difference between the duplicate data rates adds to the complexity of the optimization if the duplicate data rates are included as variables. Therefore, the nine unique data rates listed in Table 2.1 are used as the original IS-856 data rates for the optimization.

The optimization is performed with single slot assumption as described in the previous section but the system simulations are performed with the slot count numbers generated as described in Section 4.4. Therefore, the optimized data rates are not truly optimal for the system but provide an approximation. The system simulation performance of the IS-856 data rate set (of 9 rates) is compared to the performance of the optimized data rate set (of 9 rates). The system simulations are performed as described in Section 4.3. Therefore, the optimization uses a larger sample set than the system simulations results.

For the first case of optimized data rates, two sets of optimized data rates are generated for single user transmission: one optimized for Pedestrian A model and Vehicular B model. The same set of data rates are used for both the single user case and the two user case. The data rates optimized for Pedestrian A model are compared to the IS-856 data rates in terms of the average system data rate in Figure 4.16 for MAXCI, MAXCI2, PF and PF2 algorithms for Pedestrian A model and Figure 4.17 for the Vehicular B model. The system simulations for the optimized data rates show improvement in the performance compared to the original IS-856 data rates. Obviously, the improvement is due to the gain from the knowledge of the channel distribution, however, it shows the gain that can be achieved from optimizing to the channel. Furthermore, it shows that the two user transmission provides gain even if the system data rates are optimized for the channel.

In the second case of optimization, a different set of data rates are chosen for the single user and two user case. Figure 4.18 and Figure 4.19 show the results. The optimized data rates for two user mode shows higher gain when there are a larger number of users in the system. The two user mode requires large number of users to be able to take advantage of the second user. The second user C/I needs to be higher for the system to be able to schedule the terminal. The data rates selected by the optimization for the two users transmission are higher than the single user mode. This tends to favor the system when there are many users.

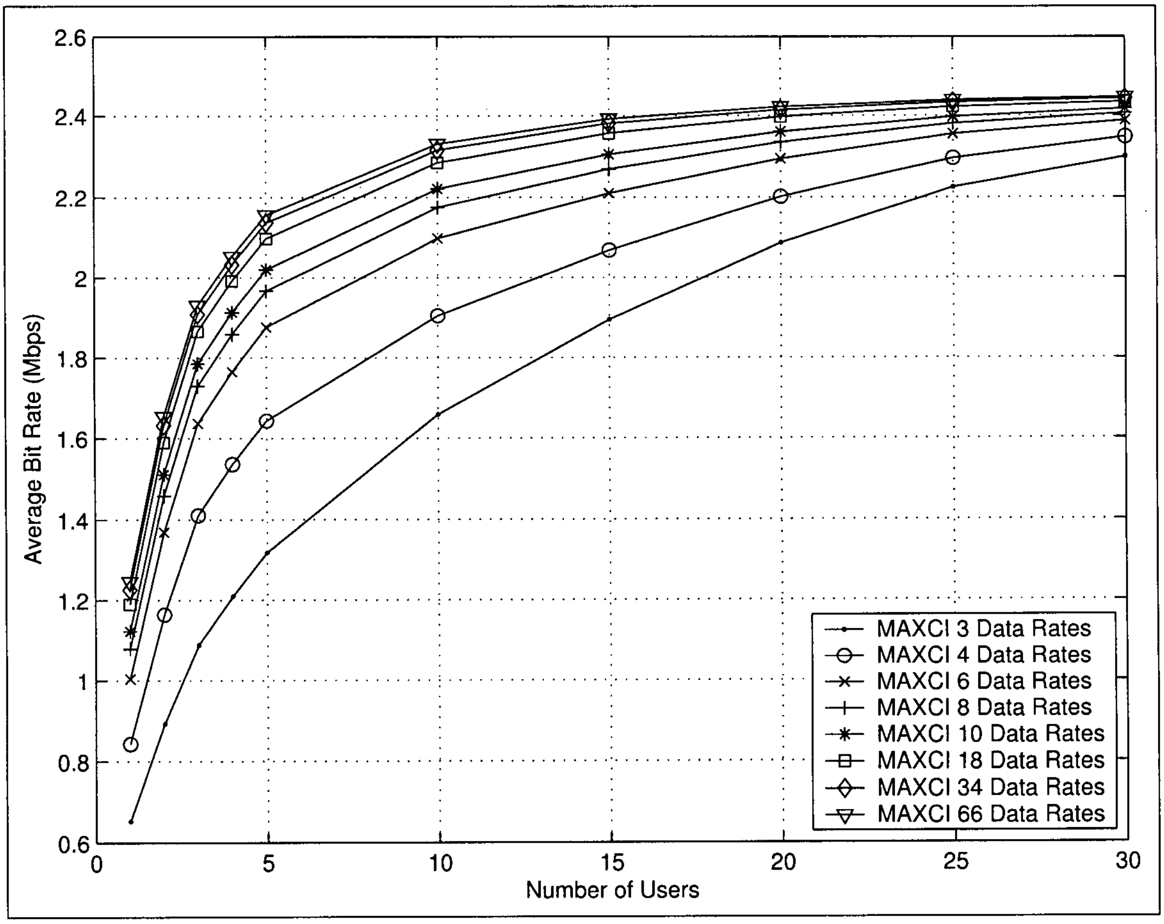

Figure 4.6: Pedestrian A: Average System Data Rate for Discrete Data Rates Using MAXCI with One Slot Transmission

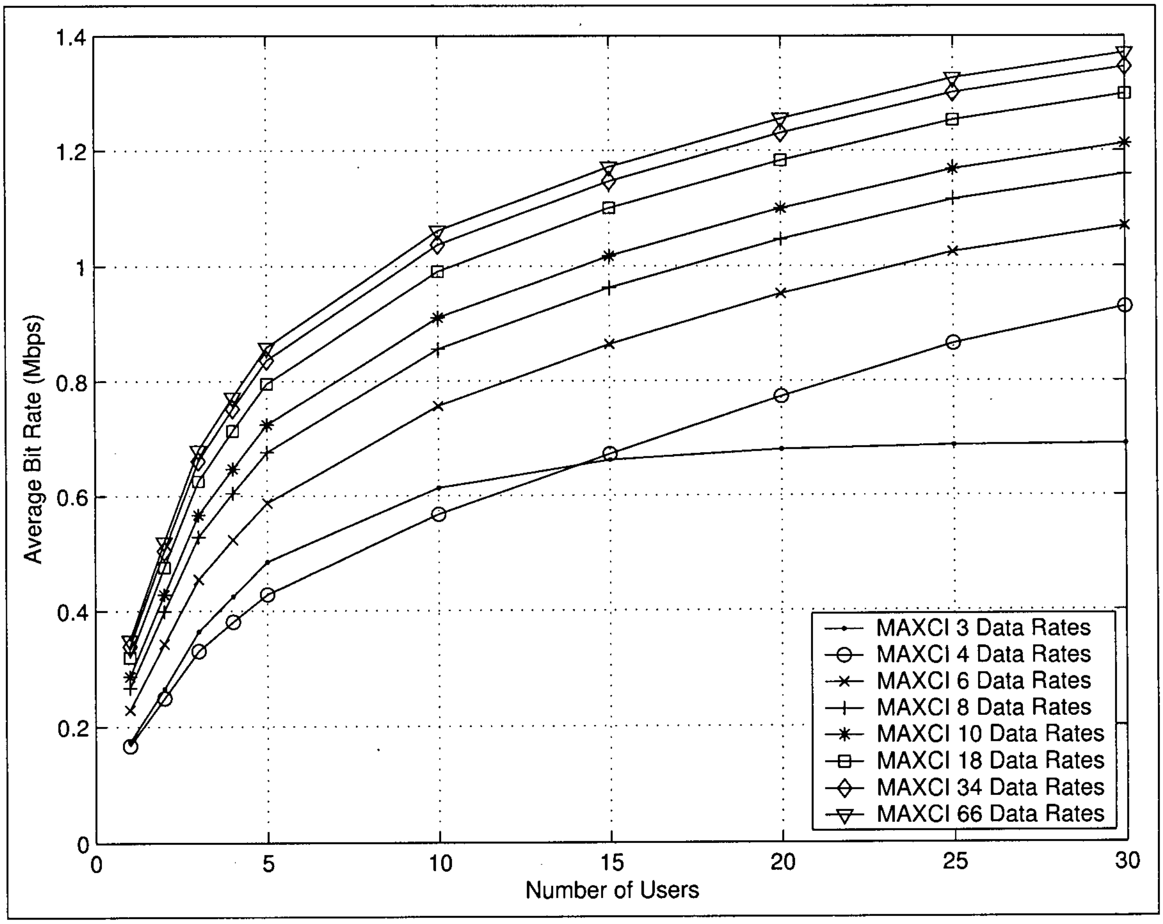

Figure 4.7: Vehicular B: Average System Data Rate for Discrete Data Rates Using MAXCI with One Slot Transmission

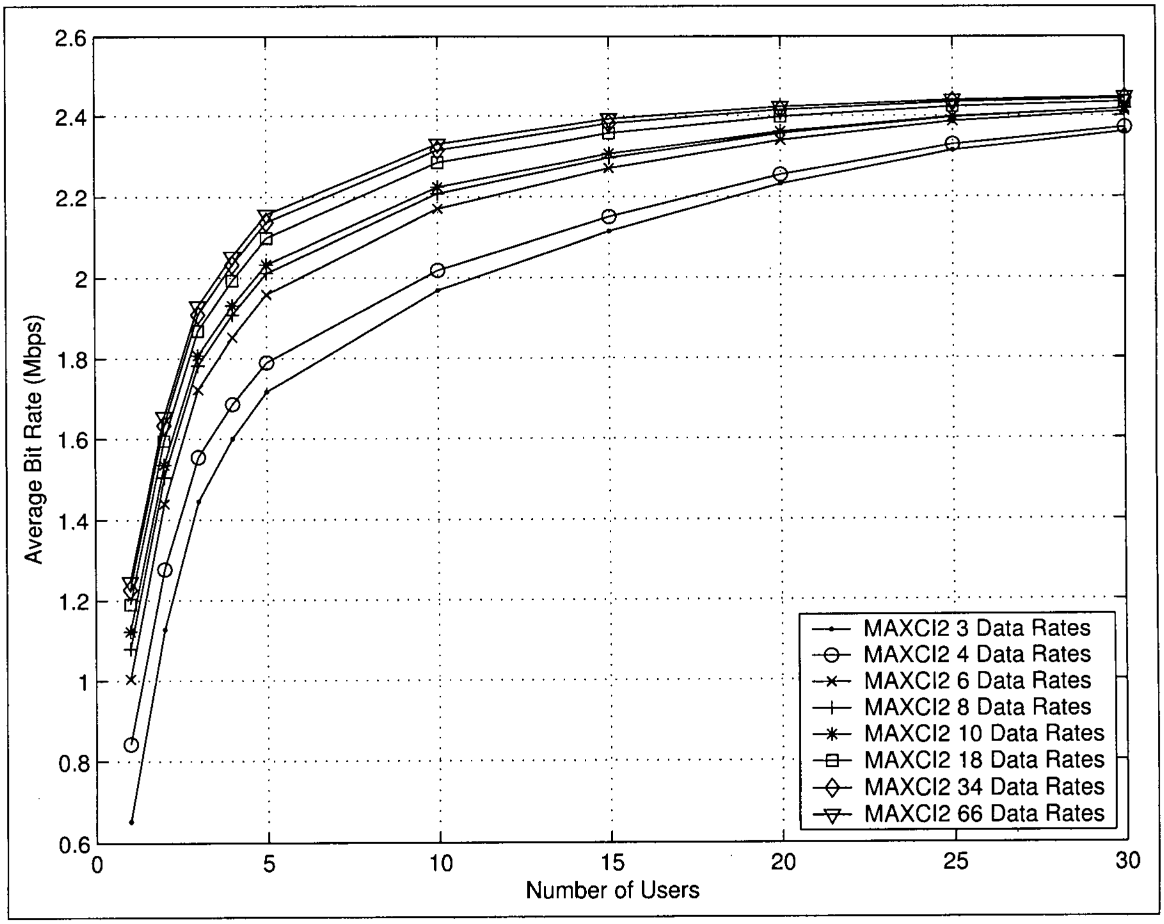

Figure 4.8: Pedestrian A: Average System Data Rate for Discrete Data Rates Using MAXCI2 with One Slot Transmission

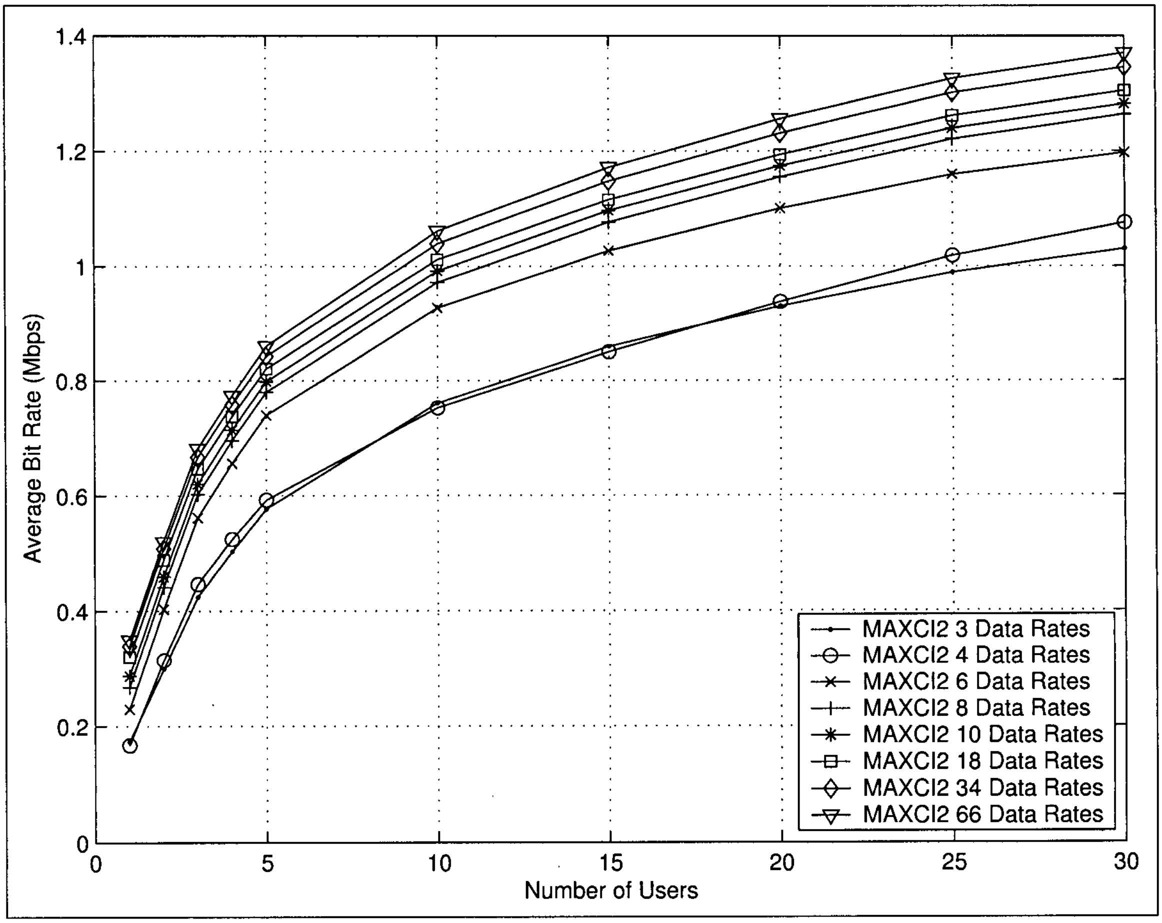

Figure 4.9: Vehicular B: Average System Data Rate for Discrete Data Rates Using MAXCI2 with One Slot Transmission

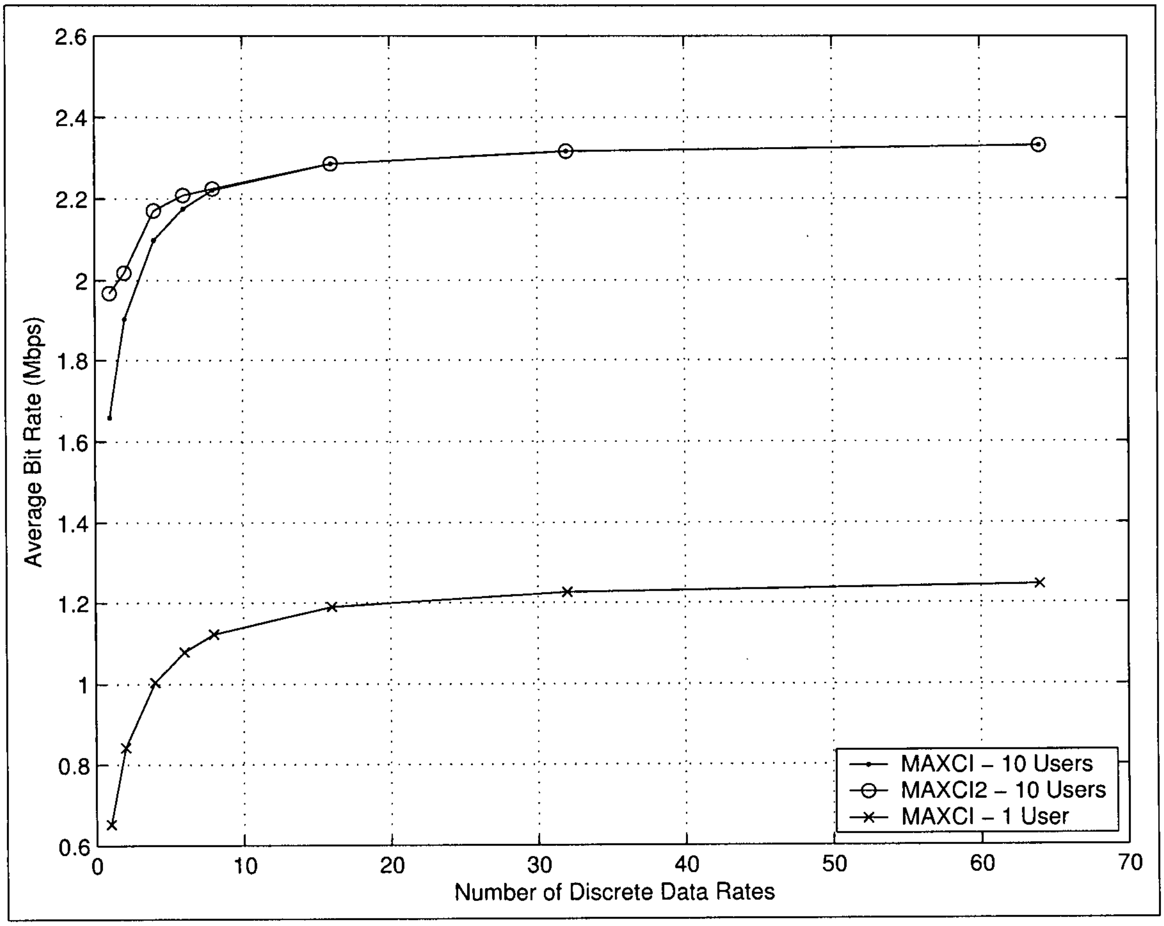

Figure 4.10: Pedestrian A: Average System Data Rate for Discrete Data Rates Using MAXCI and MAXCI2 for 10 Users and 1 User with One Slot Transmission

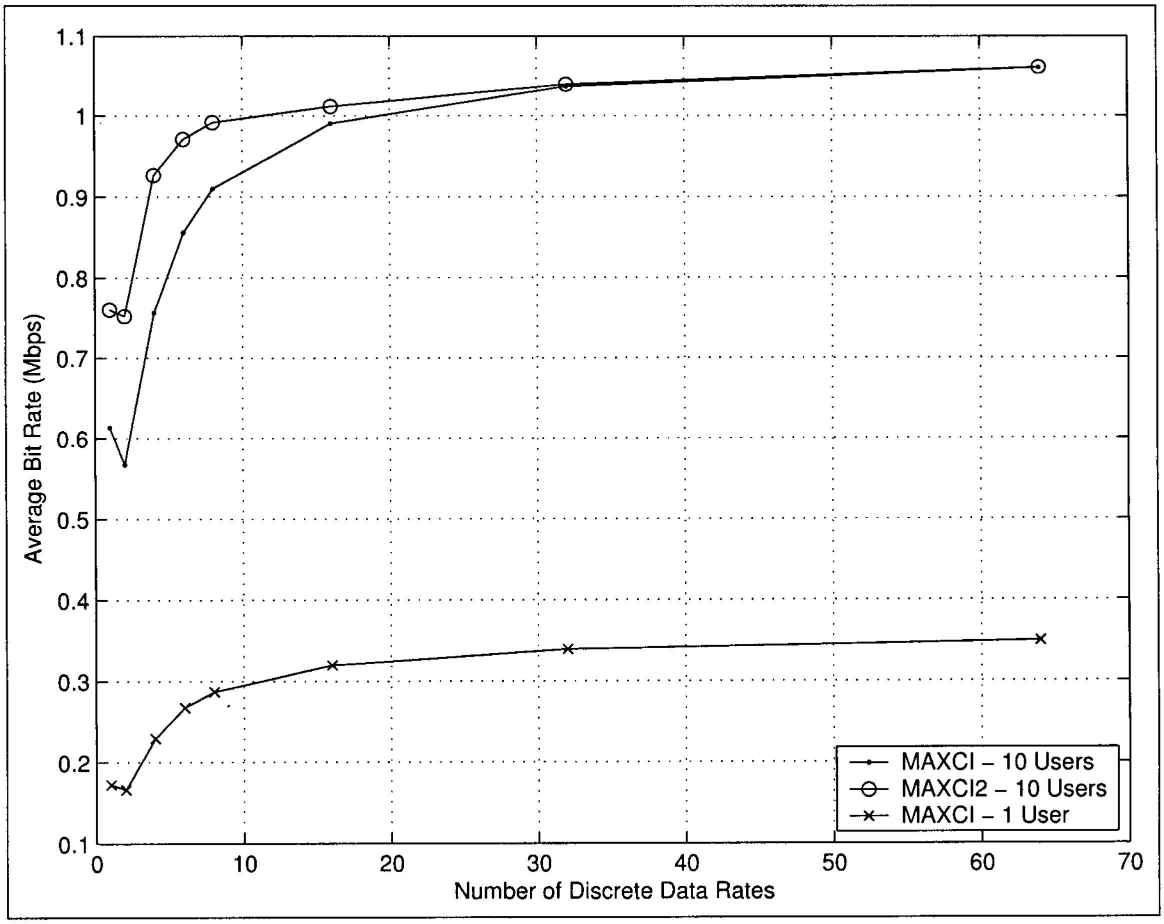

Figure 4.11: Vehicular B: Average System Data Rate for Discrete Data Rates Using MAXCI and MAXCI2 for 10 Users and 1 User with One Slot Transmission

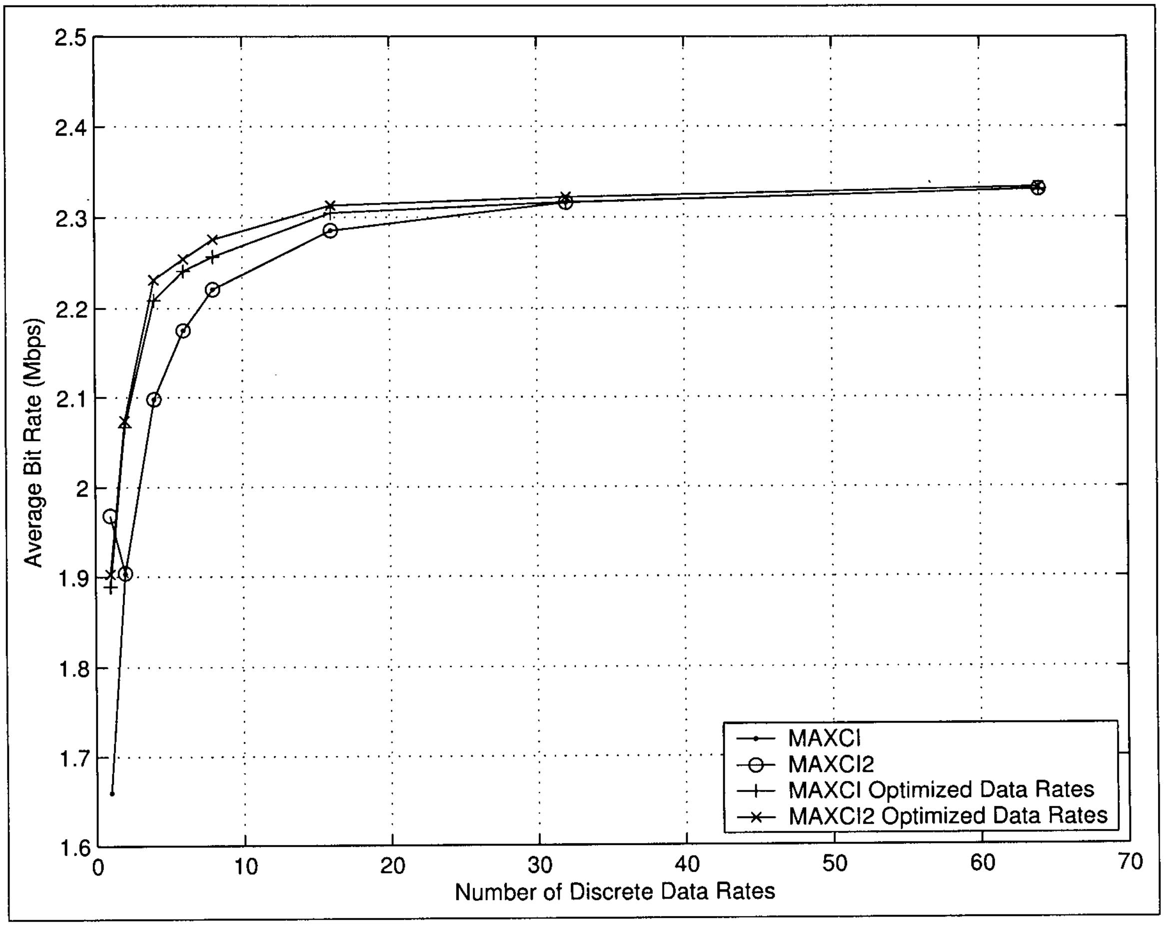

Figure 4.12: Pedestrian A: Average System Data Rate for Discrete Data Rates Using MAXCI and MAXCI2 for 10 Users

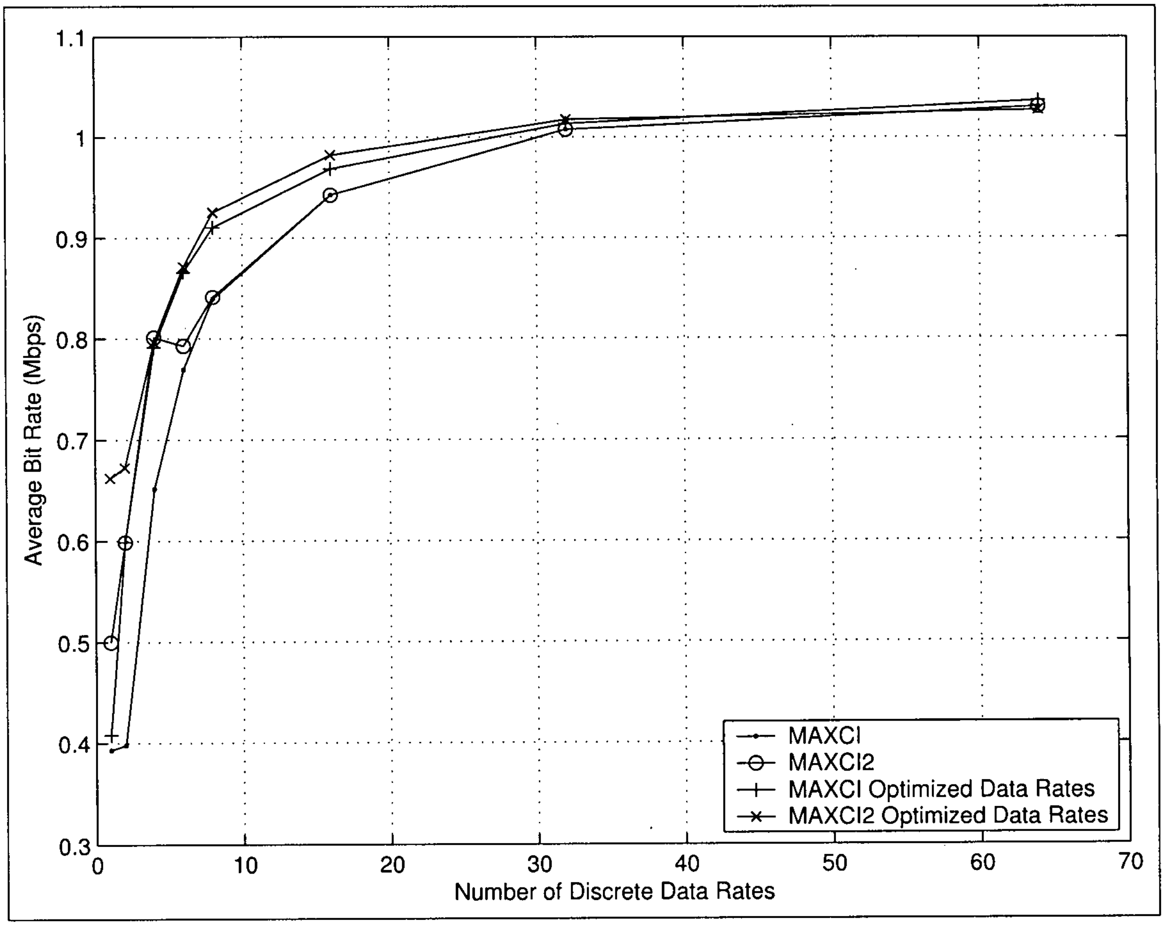

Figure 4.13: Vehicular B: Average System Data Rate for Discrete Data Rates Using MAXCI and MAXCI2 for 10 Users

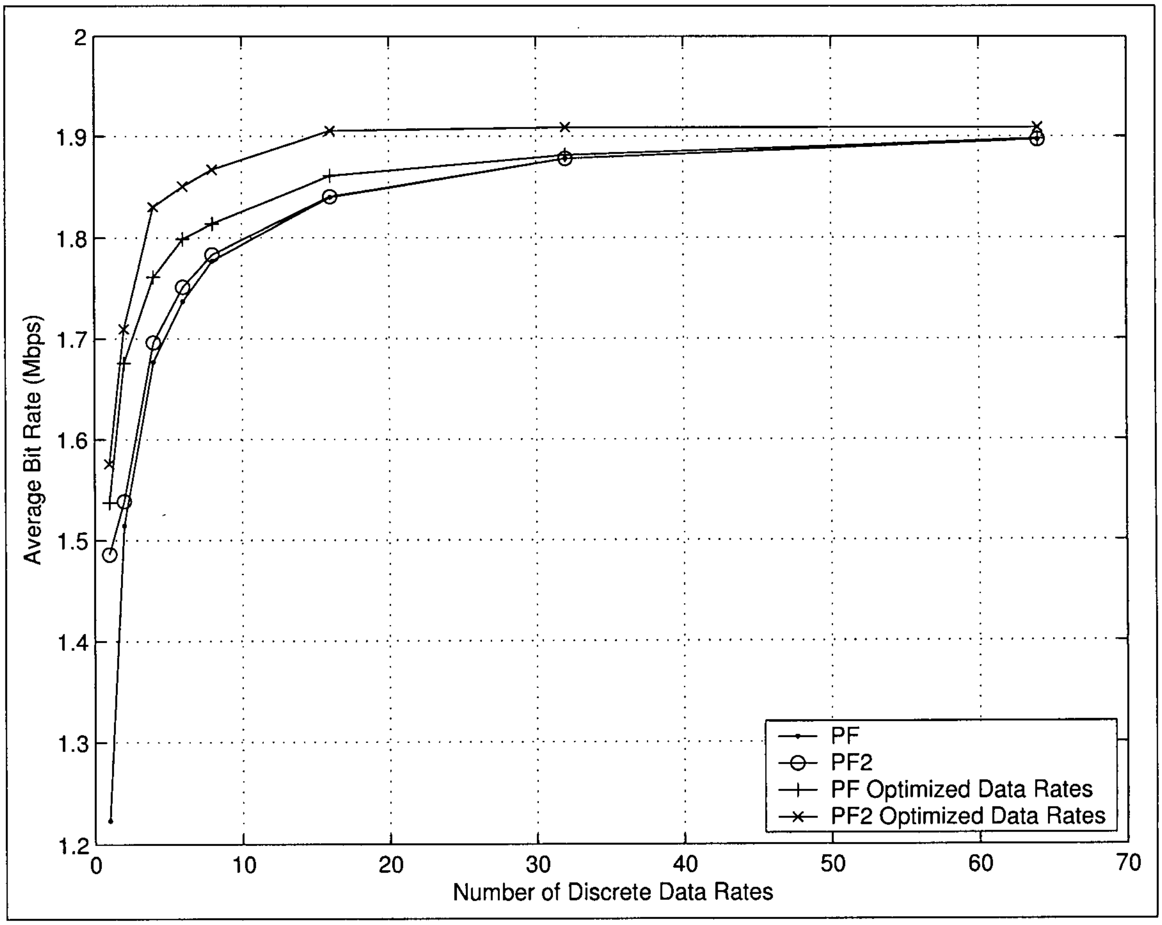

Figure 4.14: Pedestrian A: Average System Data Rate for Discrete Data Rates Using PF and PF2 for 10 Users

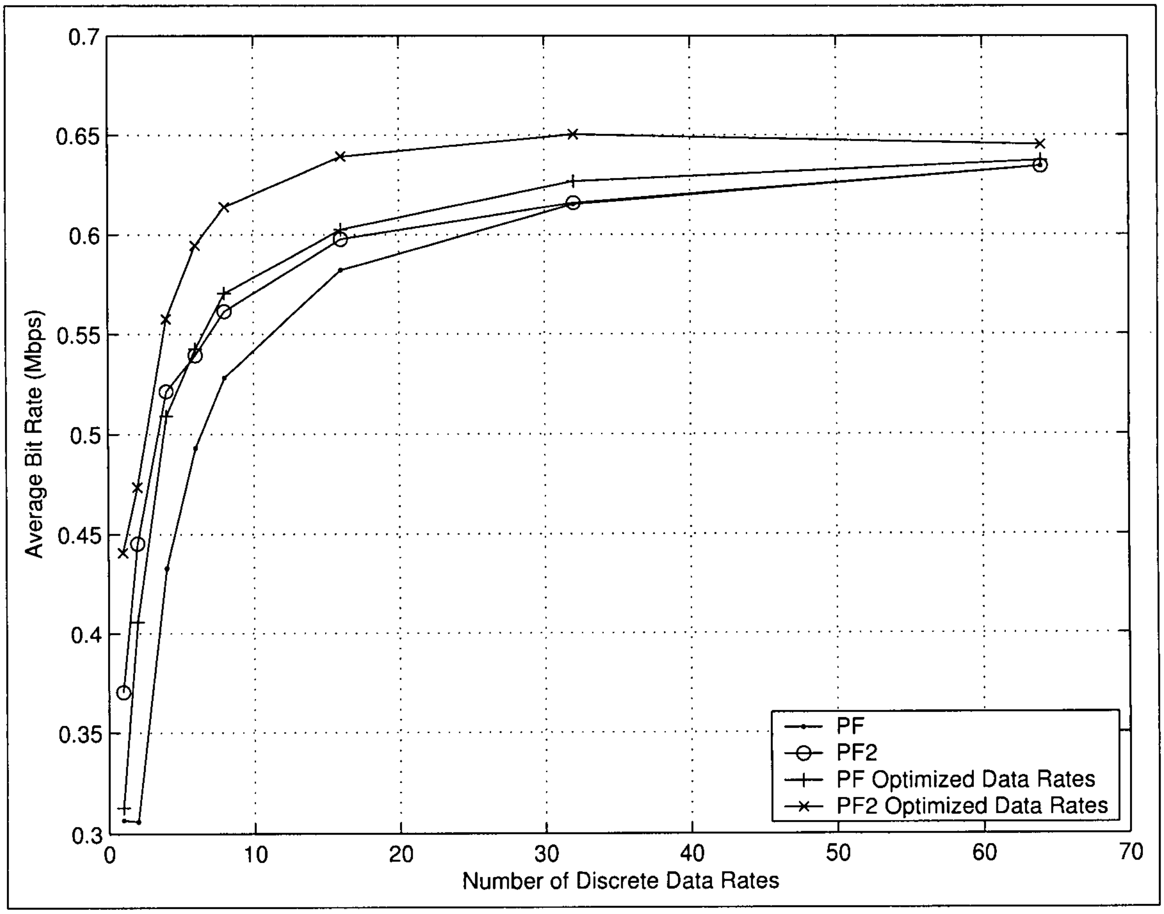

Figure 4.15: Vehicular B: Average System Data Rate for Discrete Data Rates Using PF and PF2 for 10 Users

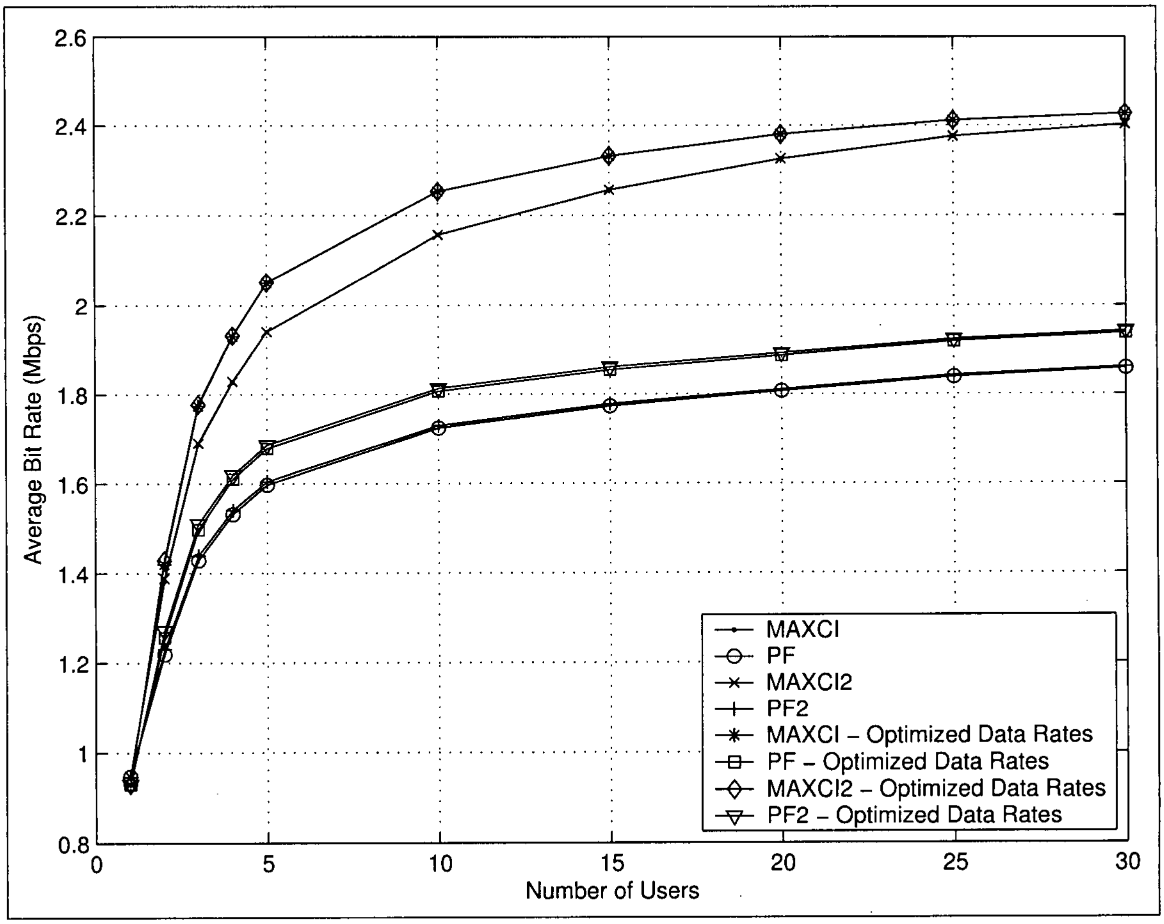

Figure 4.16: Pedestrian A: Average System Data Rate for IS-856 Define Data Rates and Optimized Data Rates for 1 User

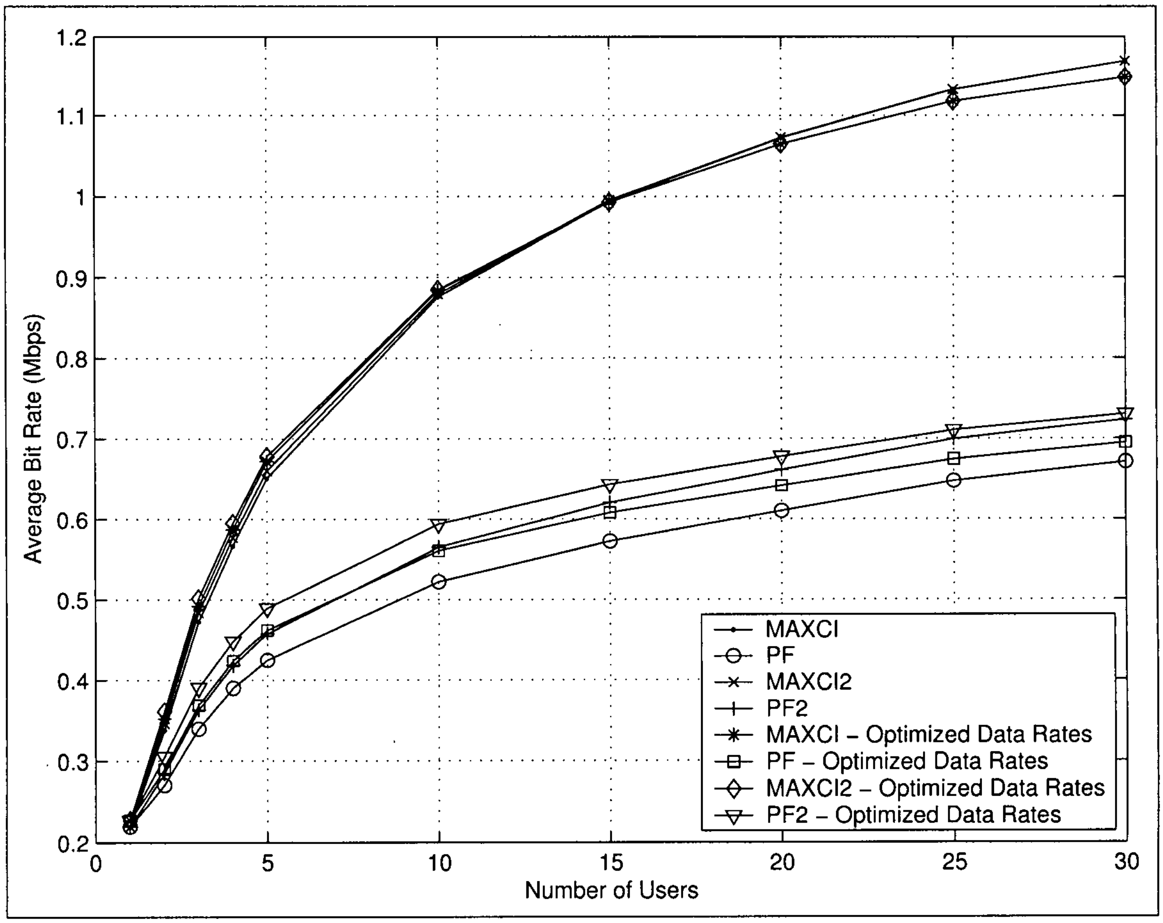

Figure 4.17: Vehicular B: Average System Data Rate for IS-856 Define Data Rates and Optimized Data Rates for 1 User

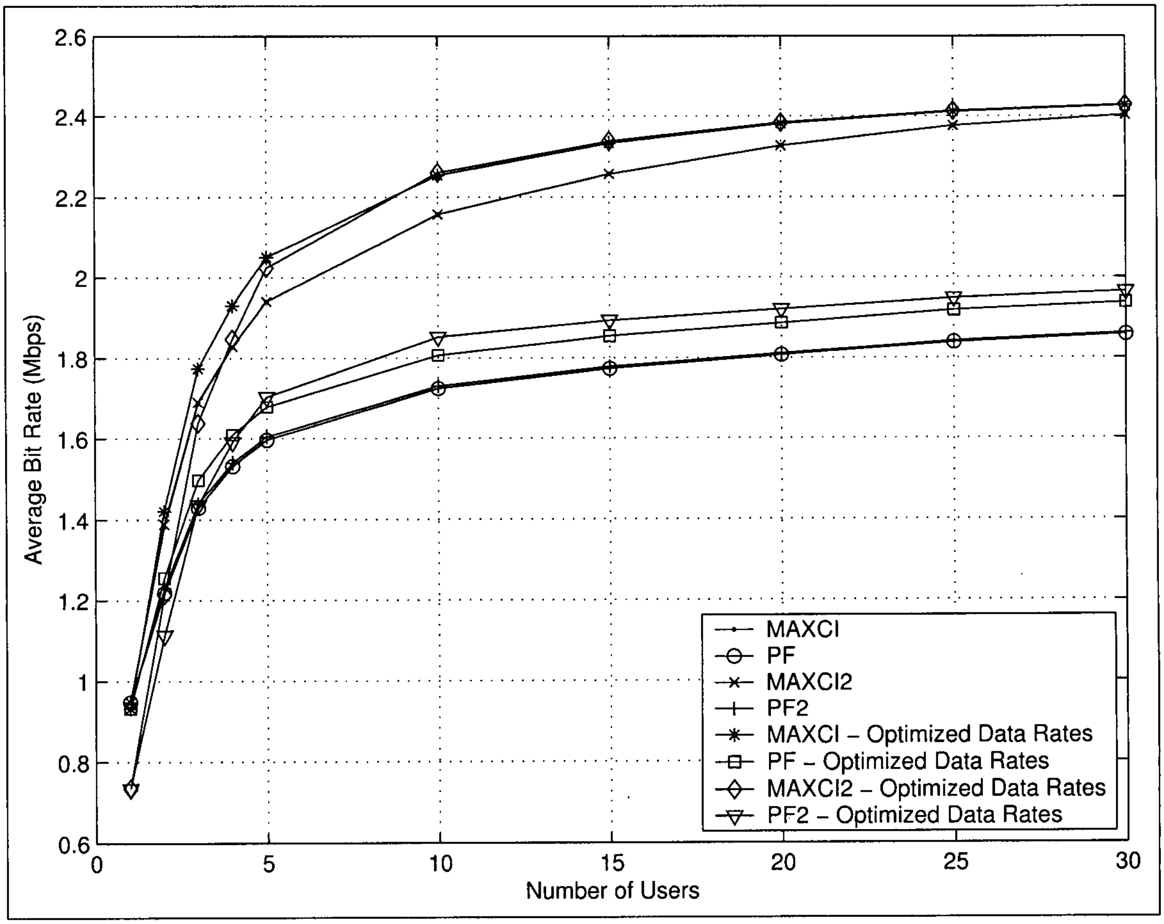

Figure 4.18: Pedestrian A: Average System Data Rate for IS-856 Define Data Rates and Optimized Data Rates

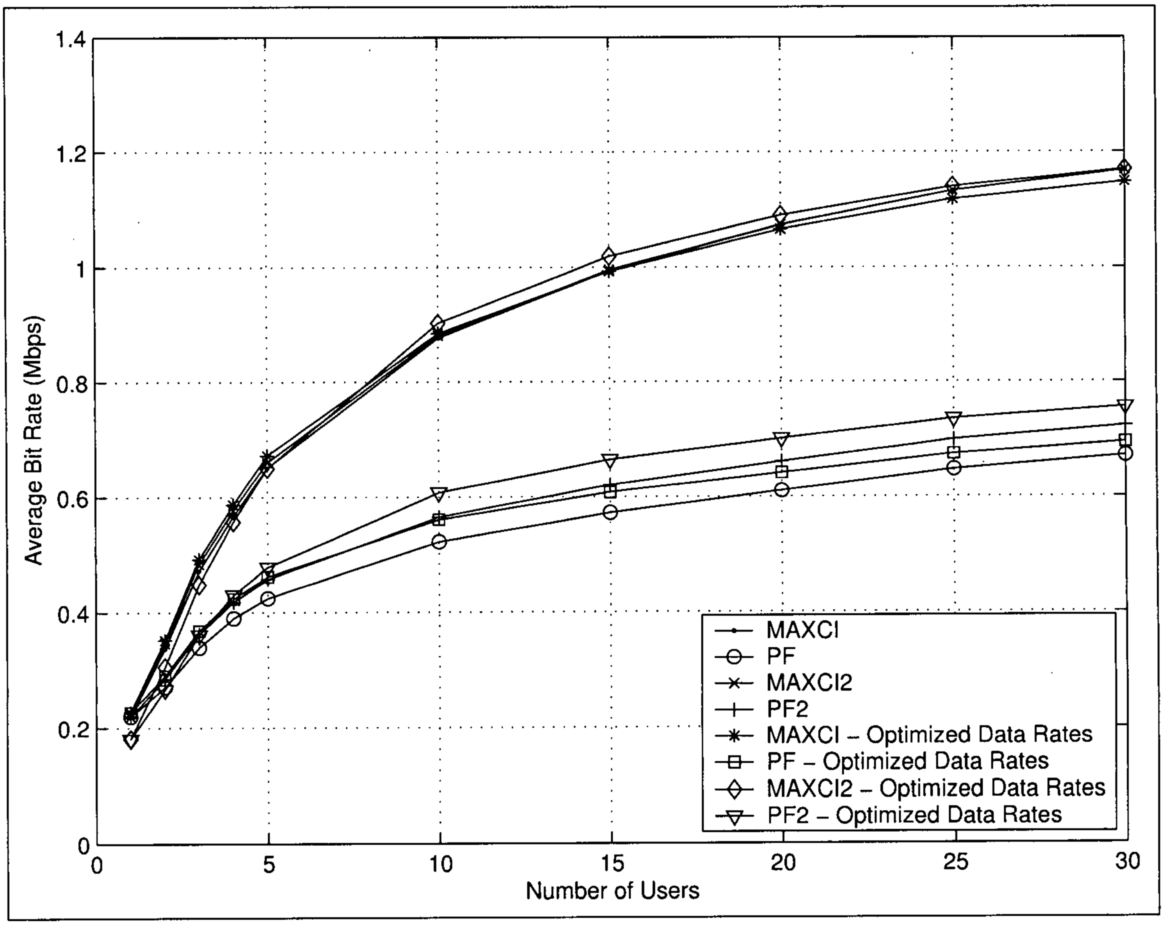

Figure 4.19: Vehicular B: Average System Data Rate for IS-856 Define Data Rates and Optimized Data Rates

Chapter 5

Conclusions

A modification is proposed to the IS-856 system downlink that allows for two user transmission. The proposed modifications to implement the two user transmission are intended to have minimal changes to the existing system design. This allows for easier implementation on to the existing systems. Furthermore, the original downlink is used for the single user transmissions that allows the system to be backwards compatible with the existing terminals. The changes for the two user transmission require the downlink be divided into two packet data traffic channels and a control channel with the use of orthogonal Walsh Codes.

This modification adds complexity to the system downlink but provides a modest performance gain in terms of average system data rate with a reduction in the packet delays. The system performance is verified with simulations. The physical layer simulations provide the signal-to-noise-ratio (SNR) required to achieve 1% PER in an AWGN channel. These results are then used in the system level simulations that provide the average system data rates and delay performances.

The performance gain in terms of average system data rates varies based on the channel model and the packet scheduling algorithm used. The two user transmissions benefit the most when the system average data rates are not high. As higher data rates are selected by the system for the first user, the number of Walsh codes available to the second user is limited. Therefore, the two user transmission is beneficial to packet scheduling algorithms that do not necessarily select the user with the best channel.

The repetition coding used by the system allows for the creation of variable data rates. The performance of these data rates can be interpolated from the results generated in the physical layer simulations. This provides a way to determine the performance of a different number of data rates and allows for the generation of optimized data rates. As the number of data rates are increased the two user transmission is shown to achieve higher data rates faster. However, the benefit of two user transmission only exists when there is enough quantization of the data rates. A set of optimized data rates are generated for a selected channel model and are shown to give performance gains. The data rates generated for two user transmission take advantage of the two user properties to give higher performance.

5.1 Suggestions for Further Work

Two user transmission is performed as to provide minimal impact on the performance first user. However, the two user mode does not necessarily require that this be maintained. More complex scheduling algorithms that assign data rates to the first user less than requested may provide for additional flexibility. More likely, this would not provide for increases in system data rates performances but could be used to lower the packet delays or to maintain specific quality of service requirements.

The effects of the early termination, allowed by the system to provide Hybrid-ARQ, is not considered in the system simulations. In addition, the effects of packet transmission failures due to the inaccuracies of the channel estimates provided by the terminals are not considered in this study. The effects of these would need to be further studied to see the effect on the two user transmission.

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