12-2004

Cross-Layer Design of a Dynamic Resource Allocation Control for the Next Multimedia Wireless Generation 1xEV-DV Systems

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CROSS-LAYER DESIGN OF A DYNAMIC RESOURCE ALLOCATION CONTROL FOR THE NEXT MULTIMEDIA WIRELESS GENERATION 1xEV-DV SYSTEMS

by

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A Thesis
Submitted to the
Faculty of The Graduate College
in partial fulfillment of the
requirements for the
Degree of Master of Science
Department of Electrical and Computer Engineering

Western Michigan University
Kalamazoo, Michigan
December 2004

CROSS-LAYER DESIGN OF A DYNAMIC RESOURCE ALLOCATION WITH QoS FOR THE NEXT MULTIMEDIA WIRELESS GENERATION 1xEV-DV SYSTEMS

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Western Michigan University, 2004

1x*EV-DV* has been proposed as one of the standards of the next Multimedia Wireless Generation (MWG) systems, which adopts new techniques to enhance wireless data Quality of Service (QoS). Despite the recent research efforts in mapping QoS parameters cross the different layers, there is still no cross layer QoS design that focuses in providing QoS assurances in the 1xEV-DV. Moreover, many possible combinations of resource parameters have been specified in the current standard. However, there are no specifications about the dynamic resource allocation. In this thesis, a suite of Cross Layer Multiuser Diversity Modules are proposed. These modules include a priority admission module, a dynamic resource allocation module, a resource scheduling module, and a Fuzzy QoS manager. All these modules are closely integrated with the recently developed 1xEV-DV technologies with a concentration on the dynamic resource allocation module that will utilize the effective capacity concept. This module will enable dynamic resource allocation of the optimal combination of resource parameters such as the number of Walsh codes, the number of time slots, and the modulation scheme. It will allocate resources not only based on the channel condition estimate but also according to the type of services requested in order to achieve promised system overall throughput gain and to meet the QoS requirement imposed by the different user's applications. Performance evaluation of the proposed design that complies with the 1xEV-DV *Evaluation Methodology* is also presented.

ACKNOWLEDGMENTS

I would like to begin by acknowledging the Department of Electrical and Computer Engineering at Western Michigan University. The program has given me the knowledge and experience to excel in my field. I would like also to greatly appreciate my thesis advisers Dr. Ikhlas Abdel-Qader, Dr. Mohsen Guizani, Dr. Norali Pernalete and Dr. Song Ci for their guidance and help throughout the thesis work. I would also like to acknowledge the help of my friend Ghassen Ben Brahim, for his support and help. Finally, I want to thank my parents Abdallah Bourouha and Zainab Al-Najjar for their moral support. Lastly, I would like to extend my thanks to my wonderful wife, Houda. She has been supportive and understanding of everything it has taken to put this thesis together.

Mohamed Bourouha

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CHAPTER 1

Introduction

1.1 Overview

With the rapidly growing demands for wireless communication systems, new types of user's applications are emerging. These applications of mixed traffic such as voice, data, and real time audio/video have challenged the current Third Generation service providers to respond with new generation of system specifications capable of providing increased data throughput. The next generation wireless communication systems need not only to provide a higher data throughput but also to support integrated applications with various Quality of Service (QoS) requirements. Providing QoS control for the emerging Multimedia Wireless Generation (MWG) is a challenging task, due to the time varying and nonstationary wireless links. Being different from wired communication networks, providing QoS in the form of absolute [2] guarantee as of the well accepted Diffserv (Differentiated Services) [4], Intserv (integrated Services) [6], and RSVP [7] protocols may not be possible. Instead, a soft QoS guarantee that is flexible to instantaneously changing channel condition is reasonable [2]. One of these MWG systems that is working toward providing soft QoS control is 1xEV-DV. 1xEV-DV is the second phase of the Code Division Multiple Access (CDMA) evolution, as will be discussed in more details in section 3.5. It is a standard that is promising a more efficient mixed traffic services delivery. In addition, to satisfying QoS requirements aiming to achieve the best possible MWG subscriber's experience. The system employs an arsenal of new technologies such as Adaptive Modulation and Coding (AMC) [3], hybrid Automatic Repeat Request (H-ARQ), Base Station Selection, and flexible TDM/CDM Multiplexing. These and the introduction of new physical channels are enhancing the system performance and enabling the delivery of forward data at a peak rate of 3.072 Mbps [8]. However, more work needs to be done in terms of providing QoS guarantees to different types of streams or traffic classes that are sensitive to QoS parameters such as priority, delay, data loss, and data rate [10]. In this thesis, a suite of Cross Layer Multiuser Diversity Modules are proposed. It is especially important to provide proper resources to the most valuable services, which make it necessary to discriminate between users according to their QoS priority profile through a Priority Admission module. There is also a need to design a Dynamic Resource Allocation module able to dynamically allocate the optimal combination of resource parameters for transmission **??**. It is also important to have a Resource Scheduling Control and QoS Fuzzy Manager that will assign queues according to different traffic classes rather than different users. All the modules proposed in this thesis are closely integrated with the previously mentioned 1xEV-DV technologies to enable explicit QoS and to achieve promised system overall throughput gain[1].

1.2 Problem Statement

The goal of 1xEV-DV standard is to provide high-quality wireless multimedia services such as mobile Internet, video and audio streaming, etc. [17]. To meet this end, the current 1xEV-DV standard [19] has employed many technologies and enabled configurable resource parameters such as Walsh codes, frame lengths, and

[1] The average data rate of the received data packets

adaptive modulation and coding schemes. However, there is no specifications on how to determine the optimal combination of resource parameters in order to support the committed QoS in terms of throughput, delay and delay violation probability. One of the solutions to this problem is to choose a fixed system configuration for a user during the whole service time. However, there are several problems in using a fixed resource allocation scheme. First, due to the existence of fast fading, slow fading, and inter- and intra-cell interferences, the channel error rate could dramatically vary during a short period of time, leading to a time-varying channel bandwidth. Second, from a user point of view, there may be multiple services running on the mobile terminal at one particular time. Different services have different requirements toward the resource allocation. For example, downloading a file has a totally different delay and loss requirements from watching a video clip. Third, even for a certain type of service, the bursty characteristics of the data traffic also causes a large variation of required bandwidth. Finally, different users experience fading independently, meaning if we use fixed resource allocation scheme, network resources will be wasted inevitably when link outages occur. Therefore, we need to design a dynamic resource allocation scheme that will help decides the optimal system configuration to meet the QoS requirements imposed by the wireless multimedia services.

1.3 Related Work

Numerouse research has been done in different aspects of resource allocation in CDMA networks. Dynamic resource allocation and management in CDMA networks have been studied in [20] and [21]. Knopp and Humblet have shown that using multiuser diversity can significantly increase the system capacity [22]. This idea has been further expanded by Srinivasan and Baras [12] and Wu and Negi [11] in scheduler design. Much work also has been done on MAC designs to support QoS through dynamic resource allocation. Akyildiz *et al.* proposed a new MAC called WISPER for CDMA-based Systems [23] that has been extended by Wang with considerations of minimum power allocation and rate scheduling in TD-CDMA systems [24]. In [25] Baiocchi *et al.* proposed and evaluated a new MAC layer for IP QoS delivery. Some dynamic resource allocation schemes are based on the source traffic pattern. Tong and Ramanathan [26], proposed an adaptive power and rate allocation scheme using service curve concepts for DS-CDMA network. Jafar and Goldsmith [27] developed an adaptive multirate scheme by maximizing the uplink throughput. Evans and Everitt [28] used effective bandwidth concepts to analyze the resource usage and multi service support. Choi and Shin [29] studied the resource assignment based on QoS guarantees for heterogeneous traffic. Moreover, more work has been done on performance evaluation and resource utilization of multimedia CDMA systems [30, 32].

1.4 Thesis Goal

Despite the recent research efforts in mapping QoS parameters across the different layers, there is still no cross layer QoS design that focuses on providing QoS assurances that satisfies new data service classes as described in the 1xEV-DV QoS standard [10]. In this thesis, we propose a cross layer design for supporting QoS in the 1xEV-DV system through dynamic resource allocation. As discussed earlier, most of current research efforts concentrate on design resource allocation schemes by utilizing the information extracted from the physical layer such as Signal-to-Noise Ratio (SNR) and Bit Error Rate (BER). On the other hand, the end-to-end QoS in terms of throughput, delay and loss rate cannot be fully reflected by only the BER. Moreover, the next MWG networks are all IP-based packet networks, meaning that the next MWG networks are using packet switching where the queuing and buffering delay

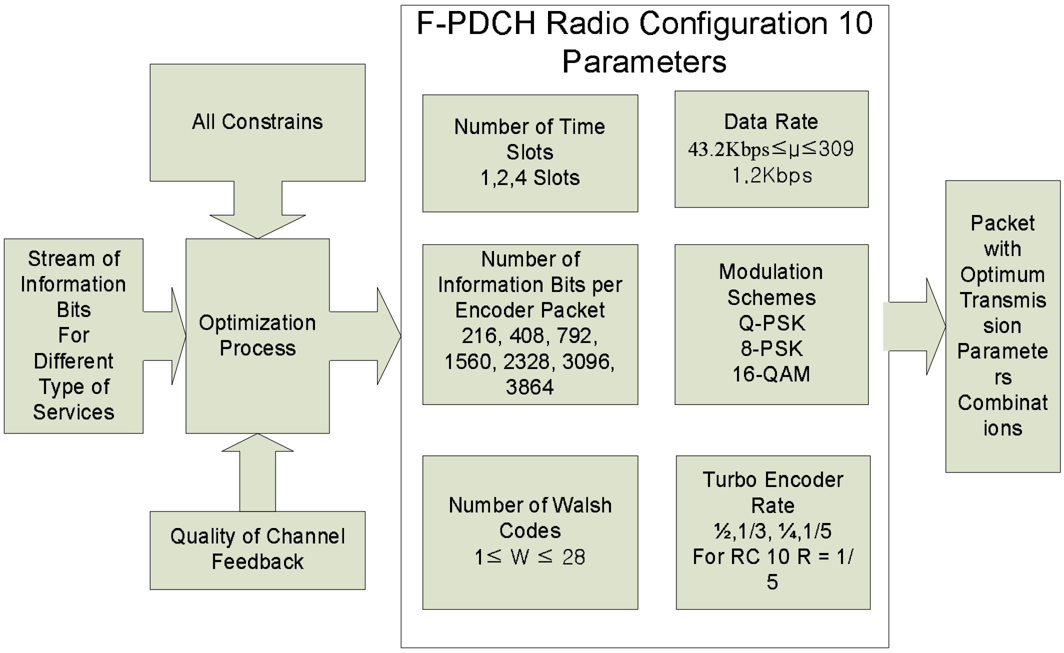

Figure 1.1: System Optimization

are very important for QoS assurance. Therefore, how to associate the delay bound with the channel quality in a systematic way is one of the most important issues for designing an effective resource allocation scheme. In this thesis, the *Effective Capacity (EC)* concepts defined in [9] is used into a dynamic resource allocation scheme to verify QoS guarantees in 1xEV-DV systems. The scheme will optimize resource parameters such as the number of Walsh codes, the number of time slots, the modulation scheme and the channel code rate. It makes the optimum selection according to the forward link channel conditions. It will also support QoS guarantees to different type of traffic classes that are sensitive to different QoS parameters such as priority, delay, data loss, and data rate in 1xEV-DV systems [10]. Figure 1.1 shows the optimization block diagram of all these resource parameters. Simulation results show that the proposed scheme can greatly improve the delay and throughput performance for various applications with different QoS requirements.

CHAPTER 2

Cellular Concepts

2.1 Cellular System Architecture

In this section, some of the major cellular concepts such as the cell and frequency reuse are introduced before the thesis problem is addressed.

2.1.1 Cells

The concept of a cellular communication system comes from the hexagonal shape of the areas into which a coverage region is divided. Though, such a shape cannot be generated in the real life due to constraints imposed by natural terrain and the man made structures. However, the hexagonal shape simplifies the design of cellular communication systems since it approaches the circular cell shape that is the ideal power coverage area. A hexagonal shape pattern provides equidistant antennas between adjacent cell sites. Furthermore, a hexagonal shaped cell fit the coverage region nicely, with no gaps and no overlapping between adjacent cells. Figure 2.1 illustrates the obvious advantage of choosing a hexagonal over a circular shaped cell.

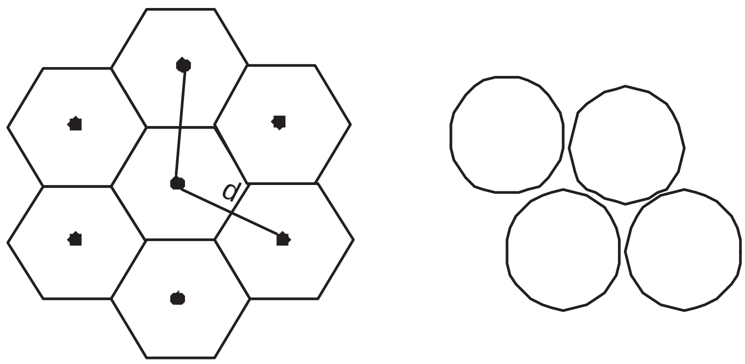

Figure 2.1: Hexagonal vs Circular Cells

2.1.2 Frequency Reuse

The second important cellular concept is Frequency Reuse. It is to determine how many cells must intervene between cells using the same channel frequencies to prevent from interfering with each other. This is due to the limited number of radio channel frequencies that are available for the cellular communication systems. Since cells are assigned a group of channels that are completely different than the neighboring cells, users located in far enough geographic areas may simultaneously use the same frequency channels. Serious interference known as the *co-channel interference* may occur if the cellular communication system is not properly designed. The frequency reuse concept can be seen in the time domain by using the same frequency in different time slots, which is called, Time Division Multiple Access (TDMA) that is covered in more details in section 2.5.2. In the space domain however, frequency reuse can be classified into two different categories. The first is the reuse of the same frequency into two geographical areas such as radio stations broadcasting on the same frequency in two different cities. Second in the cellular system where frequencies are repeatedly reused in the same system on the same geographical area by dividing the total frequency spectrum allocation into N frequency reuse patterns as shown in Figure 2.2.

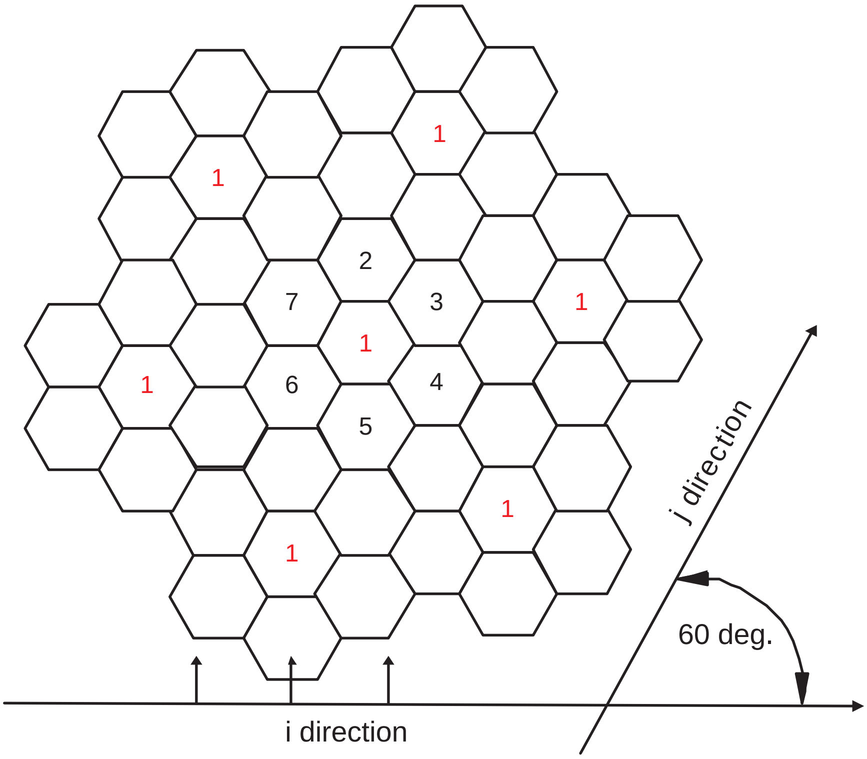

Figure 2.2: Frequency Reuse Pattern

The frequency reuse pattern N can be calculated using the following equation.

$$N = i^2 + i \times j + j^2 \tag{2.1}$$

where i and j represent the number of cells to be traversed along direction i and j, respectively as shown in Figure 2.2. By selecting $i = 1, 2, \ldots$ and $j = 1, 2, \ldots.$;N can then have the values $3, 7, \ldots$

2.2 Cellular System Components

The basic wireless communications system consists of four main components: the mobile unit, the mobile telephone switching office, the base station, and the public switched telephone network as shown in Figure 2.3. All these components are connected together in order to provide different wireless services to thousands of subscribers within the wireless system.

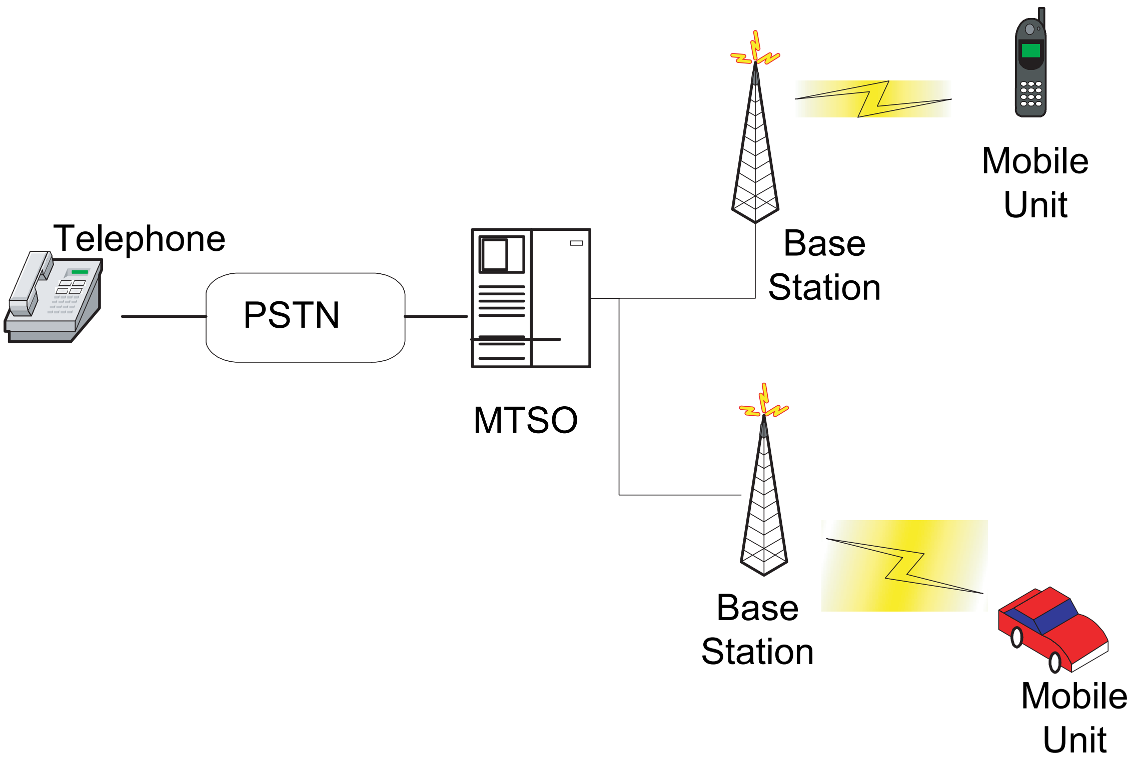

Figure 2.3: Cellular System Principal Elements

2.2.1 Mobile Unit (MU)

The mobile unit (or the user or the subscriber) is the service demanding element (the end point of the network). Unlike the traditional telephone which requires wire connections between telephones, the MU broadcasts its signals through the air. There are two types of MUs: the mobile telephone typically installed in a vehicle and transmitting a 4.0 watts of power and the portable telephones that are handy and can be used anywhere and transmit a 0.6 watts to 1.6 watts of power.
A Mobile Unit consists of three major components:

- **Control Unit:** The control unit is used to handle the call process between the MU and the rest of the network.

- **Transceiver:** The Transceiver is used to transmit and receive radio signals to and from a Base Station.

- **Antenna System:** The antenna system is the port through which radio frequency (RF) energy is coupled from the transmitter to the outside world and, in reverse, to the receiver from the outside world.

2.2.2 Mobile Telephone Switching Office (MTSO)

The Mobile telephone Switching Office is the central coordinating component of the wireless communications system for all the Base Stations. MTSO contains the cellular processor and the cellular switch. MTSO interfaces with the Public Switched Telephone Network (PSTN) to make connections between a fixed subscriber to the public network and a mobile subscriber to the cellular network. Some of other MTSO operations are: assigning the voice channel to each call, performing Handoffs[1], and monitoring subscriber's call for billing information.

2.2.3 Base Station (BS)

Base station is the component of the wireless communications system that provides the interface between the MTSO and the MUs. The BS equipments include: antenna systems, interface equipment, radio frequency transmitters and receivers, and power plants are physically located in the BS to provide the cellular coverage within a cell.

[1]Handoff is to release a signal from one cell to the neighboring cell

2.2.4 Public Switched Telephone Network (PSTN)

The Public Switched Telephone network is the network that interconnects telephones and other wire-line communication devices (MUs). The PSTN started as plug boards and analog circuit switching systems that are manually operated. Today's PSTN are almost operating completely digital except for the final connection to the MUs.

2.3 Modulation

Since digital source information cannot be directly transmitted over the air, it must be imparted onto a bandpass signal, with a carrier frequency f_c by the introduction of amplitude or phase perturbations or both, that will carry the digital information over the wireless channel. These bandpass signals are usually referred to as the *Radio Frequency* signals (RF). Generally, an RF signal may carry the digital source information under one of these two types:

- **Phase:** The digital information can be carried over the RF signal by switching (keying) the phase of the signal. A digital "0" can be represented by one phase (e.g., 0 degree) while a digital "1" can be represented by shifting the phase of the RF signal to a different phase (e.g., 180 degree).

- **Frequency:** The digital information can be carried over the RF signal by switching (keying) the frequency of the signal. A digital "0" can be represented by one frequency (e.g., 10 Hz) while a digital "1" can be represented by shifting the frequency of the RF signal to a different frequency (e.g., 20 Hz).

Some of the most common digital modulation schemes (techniques) are discussed below:

2.3.1 Frequency-Shift Keying (FSK)

FSK is a type of frequency modulation that can be generally described by the following analytical expression:

$$F_i(t) = \sqrt{\frac{2E}{T}} \cos(\omega_i t + \phi); \quad 0 \leq t \leq T; \quad i = 1, \ldots, M \tag{2.2}$$

where ϕ is any arbitrary constant angel and ω_i is the variable frequency term that has M discrete values, E is the symbol energy and T is the symbol time duration.

2.3.2 Phase-Shift Keying (PSK)

PSK is a type of phase modulation that can be generally described by the following analytical expression:

$$F_i(t) = \sqrt{\frac{2E}{T}} \cos(\omega t + \phi_i(t)); \quad 0 \leq t \leq T; \quad i = 1, \ldots, M \tag{2.3}$$

where $\phi_i(t)$ is the variable phase term. $M = 2^k$ where $k = 1, \ldots, N$ is the number of source bits the processor accepts at a time and instructs the modulator to produce one of an available set of M waveform types.

$$\phi_i(t) = \frac{2\pi_i}{M} \quad i = 1, \ldots, M \tag{2.4}$$

Binary Phase-Shift Keying (BPSK) is a special case of this type of modulation with $M = 2$ ($k = 1$), the phase of the waveform $F_i(t)$ is typically shifted to one of these two states, a 0 degree or a 180 degree.

2.3.3 Multiple Phase-Shift Keying (MPSK)

For $M = 4$, usually referred to as Quadrature Phase-Shift Keying (QPSK), the transmitter collects binary digits two at a time. For each symbol interval, the two sequential digits instruct the modulator as to which of the four signals to produce.

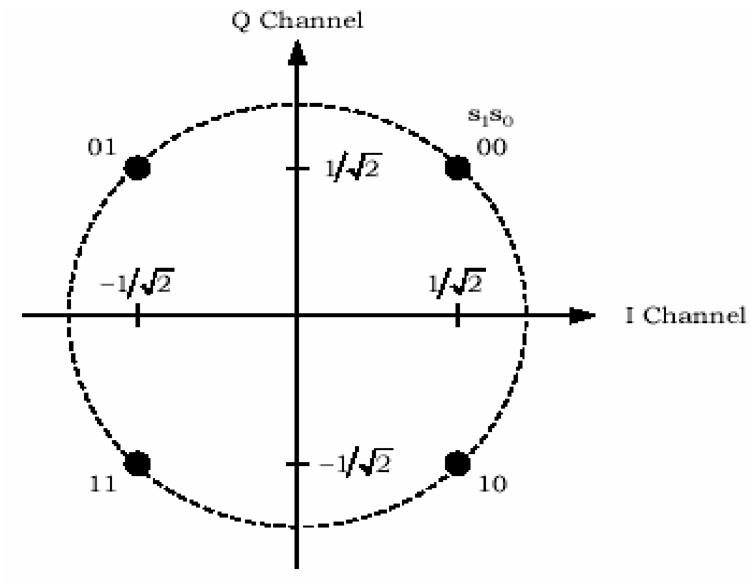

Figure 2.4: QPSK Signal Space Diagram [17].

Figure 2.4 shows the projections of the rotating signals components on the Inphase (I: real component) and Quadrature (Q: imaginary component) channels axis.

Octal Phase-Shift Keying (8-PSK) is another special case of Phase-Shift Keying, with M = 8. Figure 2.5 shows the signal space diagram of 8-PSK modulation.

2.3.4 Quadrature Amplitude Modulation (QAM)

When the set of M symbols in the two dimensional signals space are arranged in a rectangular constellation and are not restricted to having permitted signaling points on a circle as for MPSK, the signaling is referred to as QAM. QAM consists of two independently amplitude-modulated carriers in quadrature. Each block of even number of bits k can be split into two $(k/2)-bit$ blocks which uses $(k/2)-bit$ *Digital-to-Analog Converters* to provide the modulation voltages for the carriers. 16 QAM involves splitting the signal into eight different phases and two different amplitudes for a total of 16 different possible values each encoding 4 bits. Figure 2.6 shows a signal space diagram of QAM with M =16.

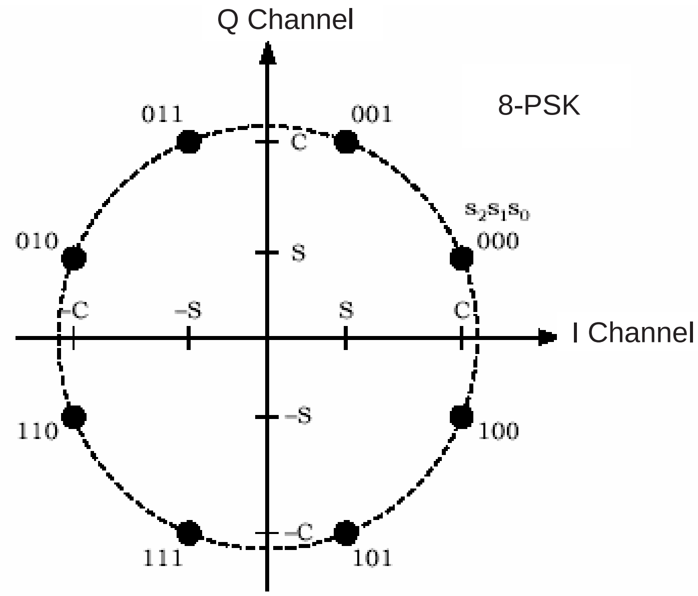

Figure 2.5: 8-PSK Signal Space Diagram [17].

2.4 Coding

Coding improves communication performance by immunizing the transmitted signal to the effect of noise, interference and fading. The data to be transmitted is encoded using a code that will separate it from other encoded data. The code used to encode and spread the transmitted data is known to the receiver. The receiver will use the code to despread/decode the signal and recover the encoded data and perform an error correction operation to recover possible data corruption during the transmission process. This process will spread the encoded symbols over the entire bandwidth of the CDMA channel.

2.4.1 Turbo Coding

Turbo coding (known also as *Parallel Concatenated Convolutional Coding (PCCC)* with interleaving) was introduced by *Claude Berrou et. al.* Turbo coding was an evolution over all other coding methods, approaching the Shannon limit by a margin

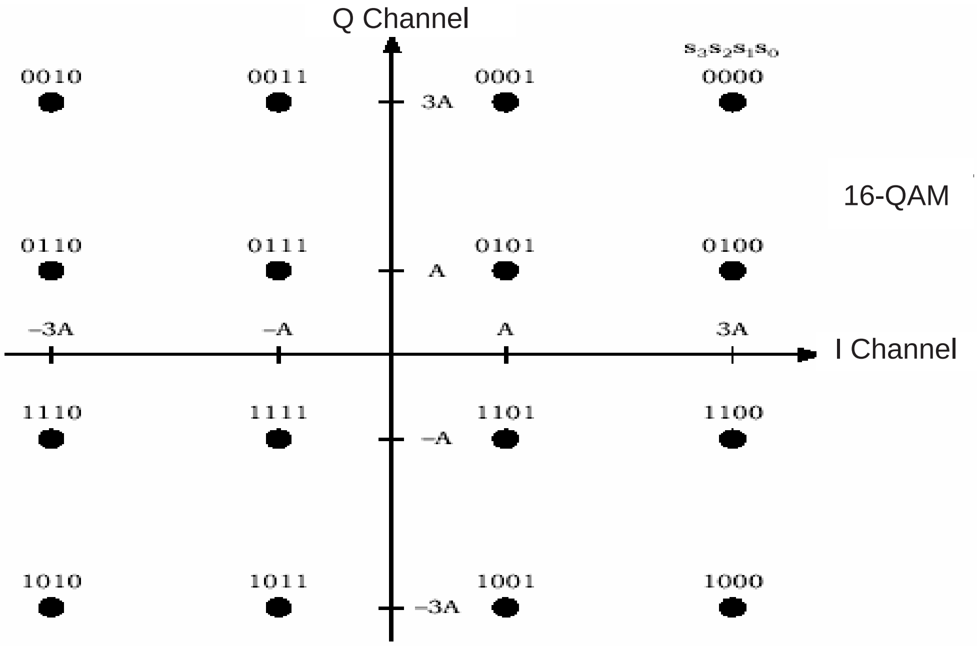

Figure 2.6: 16-QAM Signal Space Diagram [17].

of only $0.7dB$. A basic block diagram of a *Turbo Coding* system is shown in Figure 2.7. The *Turbo Encoder* (transmitter side) is composed of two parallel *Convolutional Encoders* (CE) and an *Interleaver* that precedes the second CE. In the receiver side the *Turbo Decoder* uses two decoders and an interleaver. Starting at the transmitter side, information bits k enter the first CE and get scrambled by the interleaver before entering the second CE. Each CE generates a string of parity bits, used for error-correction, by performing a series of calculations on the information bits to be transmitted. The original information bits plus the two strings of parity bits are then combined into a single block of output bits n and sent over the wireless fading channel. The rate of the turbo encoder will then be k/n which can be $1/2,\ 1/3,\ 1/4$ *or* $1/5$. At the receiver side, the received signal is sampled and assigned guessing integers. These integers will indicate the chances for a bit to be a 0 or a 1. For example, -10 means the bit is almost certainly a 0; +10 means it is almost certainly a 1. Each *Convolutional*

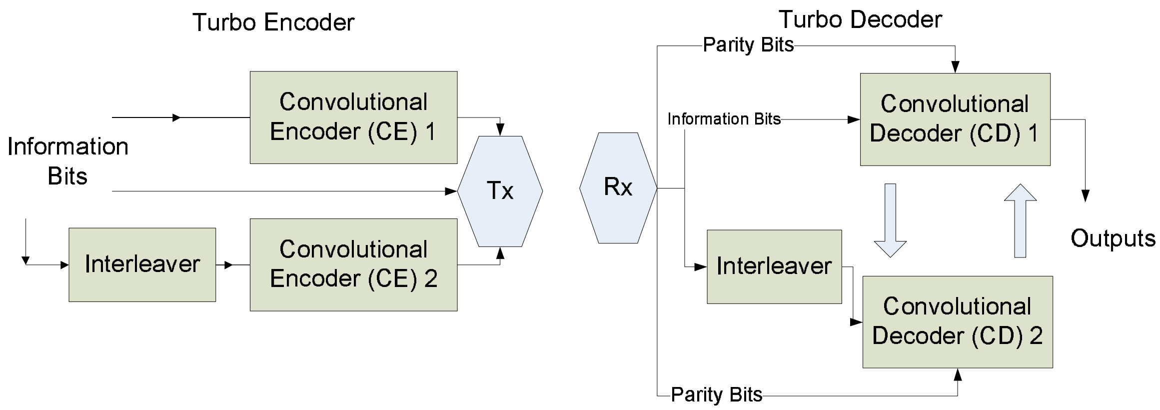

Figure 2.7: Turbo Coding System

Decoders (CD) will then take the received data plus the noise caused by the wireless channel and the respective parity information to compute how confident it is about each decoded bit. The two CDs will repetitively exchange the confidence information. After a number of exchanges, the two CDs begin to agree on all the decoded bits. The decoded stream of bits is then the sum of the received data plus noise plus the two final strings of confidence values. Finally, the error free data out of the receiver is converted back to binary digits.

2.5 Multiple Access Techniques

Wireless communication systems use air waves to transmit signals through the network. Subscribers share the same wave, creating one of the fundamental dilemmas of the wireless systems; co-channel interference (Section 2.1.2). Multiple Access Techniques come as the solution for co-channel interference. The amount of interference depends on the multiple access technique used in the wireless system. Early wireless systems generation use Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA) to avoid conflict between cells and reduce co-channel interference. FDMA and TDMA systems use a Frequency Reuse pattern

N of *seven* as shown in Figure 2.2. Whereas, the new generation wireless systems use Code Division Multiple Access (CDMA) technique to solve the co-channel interference problem. These three major multiple access techniques are the subject of the following sections.

2.5.1 Frequency Division Multiple Access (FDMA)

Using the FDMA multiple access technique, the allocated spectrum is basically split into many channels each of which can be used by one user on a wireless system as shown in Figure 2.8. In current analog cell systems, each channel is 30 KHz. While using the entire time for transmission, each user is allocated a different frequency than any other user so that receivers can discriminate among them by tuning to the desired channel. Wireless systems using FDMA are then one of the least efficient cellular systems since each channel can only be used by one user at a time and users are not taking advantage of the full bandwidth of the system. When a call is made in an FDMA system, the frequency channel is reserved for the entire duration of the call. FDMA channels are then wasted whenever there is silence during a wireless conversation. Another disadvantage of using FDMA is that knowing the nature of analog signals, frequency channels will have to be separated by guard channels in order to control co-channel interference which will in turn lower the utilization of the allocated spectrum. Given these analog disadvantages, it is obvious to see why FDMA systems are soon being replaced by digital cellular systems using more efficient techniques such as TDMA and CDMA that are covered next.

2.5.2 Time Division Multiple Access (TDMA)

Using TDMA technique on a wireless system has the potential of reducing the cost of BS and MUs, and increasing the spectrum channel efficiency. Subscribers of a

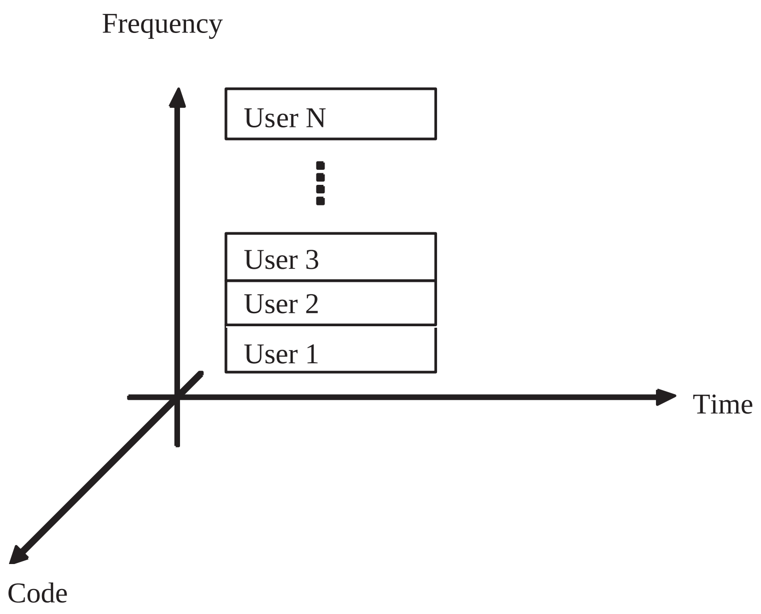

Figure 2.8: Frequency Division Multiple Access.

wireless system using TDMA share a common frequency channel. However, they use the channel only for a specific time slot as described in Figure 2.9. Each subscriber of the system is given a specific time slot and only allowed to transmit during that time slot. If all of the available time slots in a given frequency are being used the next subscriber will be assigned to a time slot on another frequency. The total number of logical channels (number of time slots in TDMA frames) is 8, allowing up to 8 subscribers to be served using the same spectrum frequency in the 800 MHz and 1900 MHz band.

2.5.3 Code Division Multiple Access (CDMA)

CDMA is another multiple access technique. However, instead of sharing available bandwidth either on time or frequency, it places all subscribers using the channel in the same bandwidth at the same time as shown in 2.10. Subscribers of a wireless

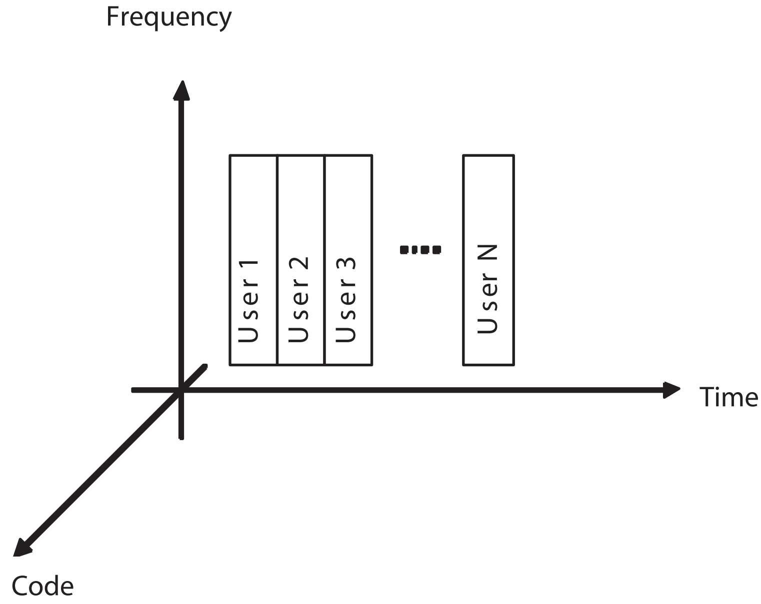

Figure 2.9: Time Division Multiple Access.

system using CDMA are isolated by a unique code sequence, an Orthogonal (Walsh)[2] code on the forward link and Pseudorandom Noise (PN)[3] code on the reverse link. This code is assigned to each subscriber, so that they can share the same carrier frequency. This in turn eliminates the frequency reuse problem encountered in FDMA and TDMA and allow for a frequency reuse pattern $N = 1$. Eliminating the need for frequency planning since each BS in the network can use the same frequency. CDMA originated from a military spread-spectrum single sideband technology. This technology has the purpose to avoid interception or jamming of narrowband military communication by the enemy. Unlike FDMA and TDMA, CDMA has a limited capacity since it uses the idea of tolerating interference by spread-spectrum modulation. Each user is a noise source on the shared channel and the noise contributed by users accumulate. This in turn limit the capacity of the system. Since certain mobiles

[2]Walsh codes provide a means to uniquely identify each user on the forward link.

[3]The PN codes are used to insure CDMA systems security since they are so unique (4.4 trillion combinations of code)

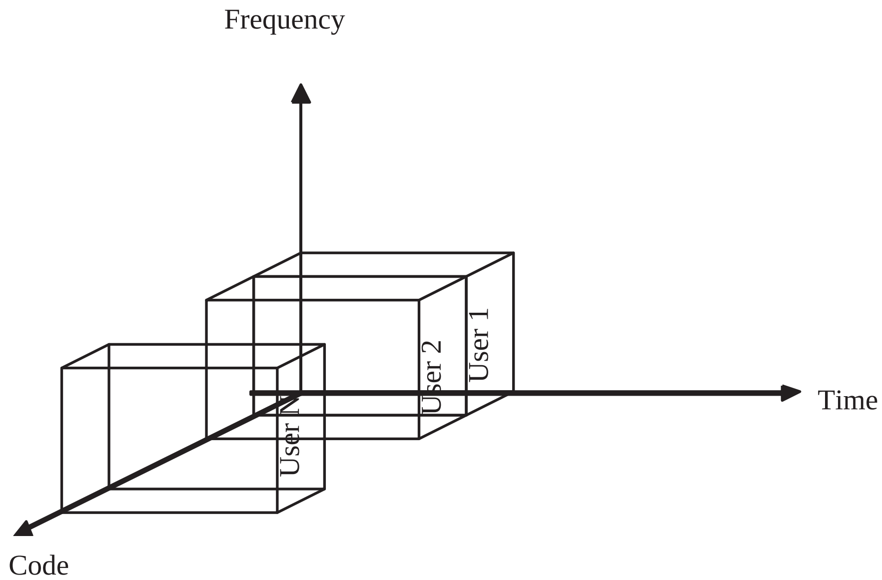

Figure 2.10: Code Division Multiple Access.

transmit excessive power that increases interference and limit capacity, the goal in CDMA is to keep each mobile in the network at the minimum possible power level.

CHAPTER 3

CDMA Evolution

3.1 cdmaOne

cdmaOne is the first 20th century system to use CDMA and is considered as a second generation wireless system. It was developed by Qualcomm and it has been standardized by the Telecommunications Industry Association (TIA) as IS-95A and further improved to IS-95B. In general, CDMA systems seem superficially simpler than those based on TDMA. They involve no slot of frame structure. Subscribers of cdmaOne share a common channel for transmission within a cell. The same frequency spectrum by any cell can also be used in the adjacent cell. cdmaOne has the advantage of being compatible with the previous analog systems such as the Advanced Mobile Phone Systems (AMPS) and use the same allocated frequency band 869-894 MHz for the uplink using offset quadrature phase shift keying (OQPSK) and 1930-1980 MHz for the downlink using quadrature (M=4) phase shift keying (QPSK) modulation [34]. IS-95A was exclusively used for circuit switched voice services using Convolutional[1] channel coding and a spreading code chip rate[2] of 1.2288 Megachips per second (Mcps)

[1] Convolutional coding is used to encode digital data before transmission through wireless channel.

[2] Chip rate is equivalent to the spreading rate of the channel. It is either 1.2288 Mcps or 3.6864 Mcps

to reach a maximum data rate of 14.4 Kbps. IS-95B on the other hand merges from IS-95A mainly by enhancing its MAC layer resulting in a maximum data rate of 115.2 Kbps. As IS-95B uses convolutional channel coding as of IS-95A. It has also an additional Walsh code and PN sequence masks that enable MUs to be assigned up to eight forward or reverse code channels simultaneously.

3.2 CDMA 2000 1x

CDMA 2000 1x also referred to as 1x Radio Transmission Technology (1xRTT) was approved as a third wireless generation standard that aims to bring high data rate capabilities to (MWG) systems. "1x" refers to the use of a single 1.25 MHz channel and a chip rate of 1.2288 Mchips/sec. It supports both data and legacy services reaching a maximum theoretical data rate of 307 Kbps. It uses the same bandwidth configuration as legacy IS-95A and IS-95B CDMA networks a 1.25 MHz bandwidth. Thus, making CDMA2000 1x technology backward compatible to IS-95A and IS-95B standards. CDMA2000 1x technology not only supports high-speed data rates, it also effectively doubles the number of voice channels available on legacy IS-95 systems. While IS-95 systems used a spreading factor of 64, CDMA2000 1x is capable of a spreading factors ranging from 2 to 64 chips per bit. A newly introduced channel Q-PCH enables the MUs to be informed about when it needs to monitor the F-CCCH[3]and Paging Channels[4], resulting in more efficient power use improving the battery life. With CDMA2000 1x, Radio Configuration (RC) was introduced. It is a set of transmission formats characterized by physical layer parameter such as spreading rate, modulation and coding schemes, and data rate. Quality and Erasure indicator bits (QIB and EIB) on the reverse power control sub-channel are also in-

[3]Forward Common Control Channel

[4]Paging Channel is a code channel in a Forward CDMA Channel used for transmission of control information and pages from a BS to a MU

troduced for BS retransmission in case of bad or lost frames. It uses convolutional, turbo channel coding and QPSK modulation technique.

3.3 CDMA 2000 3x MC

CDMA 2000 Multi-Carrier (3xMC) also referred to as 3xRTT is a third generation wireless system. Using three standard 1.25 MHz channel within a 5 MHz band, these three channels can be aggregated to reach a maximum high packet data speed of 2 Mbps. At 5 MHz, a chip rate of 3.6864 Mchips/sec rate is supported. Like CDMA 2000 1x it uses convolutional and turbo channel coding and QPSK modulation technique.

3.4 1xEV-DO

1xEV-DO also known as High Rate Packet Data (HRPD) or High Data Rate (HDR) air interface is the first evolution (EV) phase of CDMA2000. It is a third generation wireless system named by the Third Generation Partnership Project 2 (3GPP2) standards body as IS-856. It is also known as Voice and Data on separate channels. "DO" stands for Data Only as the technology was intended for data services. Like all previous CDMA systems 1xEV-DO uses a bandwidth of 1.25 MHz and a chip rate of 1.2288 Mchips/sec. 1xEV-DO has the disadvantage of not being backward compatible to CDMA2000. However, it can reach a maximum data rate of 2.4 Mbps, promising to enable new emerging multimedia services. 1xEV-DO operates under two modes: 1xEV mode optimized for a non real-time high capacity/speed data and Internet access, and a legacy 1x mode optimized for voice and medium data speeds. During a 1xEV-DO connection, each system subscriber maintains a radio link with one or more BSs (always-on operation), which constitute its active set. The active set is determined base on the Signal to Interference plus Noise Ratio (SINR) measurements of the forward link signals received by the MU. 1xEV-DO also introduces adaptive link resources techniques that allows for adaptive rate, modulation and coding with respect to channel condition, which allows for more efficient use of the air link resources.

3.5 1xEV-DV

1xEV-DV also known as *CDMA2000 Revision D* or 1x*TREME* is one of the most recent MWG considered to be a 3.5 generation system. Emerging from the need to carry both data and legacy (voice) services at the same time, 1xEV-DV allows the wireless operators to utilize their spectrum more efficiently and provide a means to balance the data and legacy loads in their system based on their specific needs. Unlike 1xEV-DO, the 1xEV-DV system was uniquely designed to maintain backward-compatibility with previous CDMA network standards. Its backward-compatibility is a feature that will make its deployment an easier task as it does not require new Base Stations (BS). As an enhancement to previous CDMA systems, data carrying capability 1xEV-DV is targeted at providing a forward peak data rate of 3.072 Mbps and a reverse peak data rate of 2 Mbps. While reusing many CDMA2000 capabilities, 1xEV-DV enhances the CDMA2000 air interface family by introducing a number of new features that are illustrated next.

3.5.1 Adaptive TDM/CDM Multiplexing

Adaptive TDM/CDM multiple access techniques is a unique and powerful feature to the 1xEV-DV system. It enables the MWG system to support all services by favoring TDM where TDM works the best (e.g., file transfer) by scheduling the forward data channel resources to users based on available data and allowing CDM when frame fill efficiency is needed (e.g., small data packets by several users). Adaptive

TDM/CDM multiplexing allows the selection of the number of time slots and Walsh codes to be allocated to a MWG subscriber [8][5].

3.5.2 New Physical Channels

New physical channels are added to the forward and reverse link to support newly introduced 1xEV-DV technologies as follows:

Forward Link Channels

Three new channels have been added to the forward link for the packet data operation, these channels F-PDCH, F-PPDCH, and F-SPDCH can not undergo a Soft Handoff[5] due to the BS selection feature and can be described as follows:

Forward Packet Data Channel (F-PDCH)

It is the data bearing traffic channel shared by the packet data users. The transmission on the channel can be of variable length 1.25 ms (1 slot), 2.5 ms (2 slots) or 5 ms (4 slots) long. F-PDCH can be time-division multiplexed between different MUs and can also be code division multiplexed to permit the BS to transmit packets to two different MUs. F-PDCH can transmit any of a set of fixed packet sizes of 216, 408, 792, 1560, 2328, 3096, and 3864 bits. [5]. The structure of the F-PDCH is shown in Figure 3.1 for Radio Configuration[6] 10 as described in the 3GPP2 standards [17].

[5]Soft handoff is a type of handoff that tries to ensures that a link is set up to the cell site in the new cell before the old one is torn down when a user is moving from a cell to another

[6]A set of Forward Traffic Channel and Reverse Traffic Channel transmission formats that are characterized by physical layer parameters such as data rates, modulation characteristics, and spreading rate.

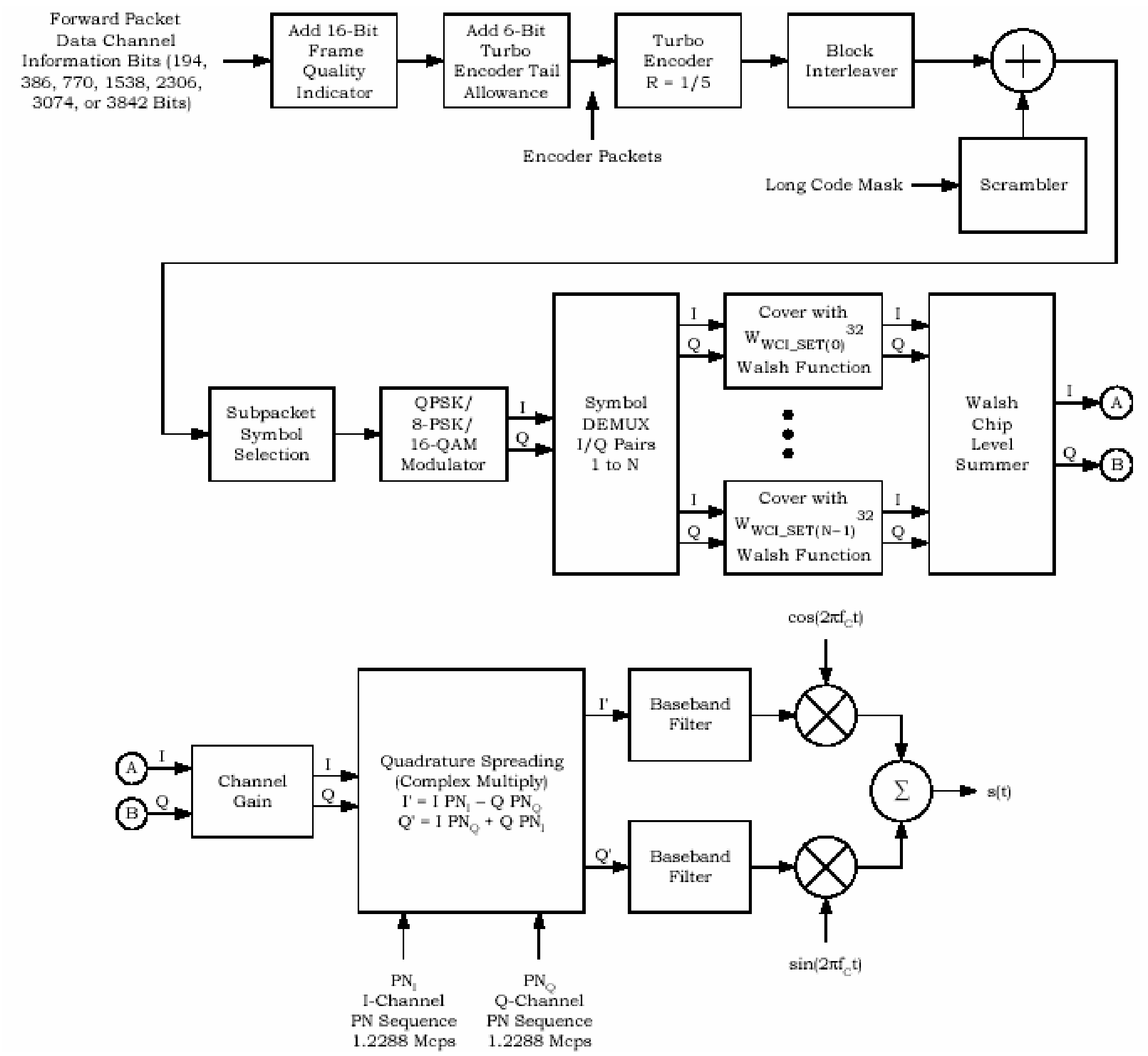

Figure 3.1: F-PDCH Structure for Radio Configuration 10 [17]

Forward Primary Packet Data Control Channel (F-PPDCCH)

F-PPDCCH is monitored by the MU when the packet data channel is assigned to correctly identify its data on the F-PDCH. The MU uses the F-PPDCCH also to learn about the duration of the F-PDCH packets.

Forward Secondary Packet Data Control Channel (F-SPDCCH)

F-SPDCCH is also monitored by the MU to correctly identify its data on the F-PDCH. The information transmitted on the F-SPDCCH are: the scheduled user's MAC ID, an 8-bit identifier known by the BS and the MU identifying the data on the shared F-PDCH, the F-PDCH packet size, number of slots per sub-packet, and the last Walsh code index that will be used by the MU to determine the Walsh cover used in the data transmission.

Two new channels have been added to the reverse link to support the operation of F-PDCH. These two channels are the R-CQICH and the R-ACKCH and are described as follows:

Reverse Channel Quality Indicator Channel (R-CQICH)

R-CQICH is one of the main new channels enabling a dynamic interaction between the MU and the BS. The R-CQICH carries feedback information to the BS from the MU about the channel quality Signal to Interference plus Noise Ratio (SINR) of the F-PDCH. This is to achieve a more efficient use of transmission resources such as data rate, modulations, and coding. In addition, the Mobile Station uses the R-CQICH to indicate its selection of serving sector via the Walsh cover applied to the R-CQICH signal. The SINR describing the channel quality is mapped into an encoded four-bit *Channel Quality Indicator (CQI)* value and then sent to the BS after every 1 slot period (1.25 ms) [14]. More details for R-CQICH are given in section 4.3.1

Reverse Acknowledgment Channel (R-ACKCH)

The R-ACKCH carries feedback information to the BS (H-ARQ) about the success of transmission of data packets after decoding of the received subpackets on F-PDCH is performed. This information can be an "ACK" for a successful transmission or a

"NAK" for an unsuccessful transmission.

3.5.3 Adaptive Modulation and Coding (AMC)

AMC is a dynamic adaptation method that will allocate channel resources of the forward link such as modulation coding scheme according to the channel condition. The channel conditions will in turn depend on few factors such as distance, path loss exponent, log-normal shading, short term Rayleigh fading and noise. A user close to the BS will then have a good radio link and will be allocated a higher modulation and higher code rates. As the user distance from BS increases, the modulation and code rate decreases.

3.5.4 Hybrid Automatic Repeat reQuest (H-ARQ)

To provide a retransmission mechanism for the lost or erroneous transmitted frames and to maintain a high bandwidth, 1xEV-DV introduces H-ARQ. H-ARQ also enables fast AMC by making the initial modulation and code rate selection process tolerant to selection errors. H-ARQ creates a more powerful code and improves system throughput by combining failed transmission attempts with current attempts rather than discarding them. Two known schemes for implementing H-ARQ: Chase Combining[7] and Incremental Redundancy (IR)[8]. The latter used in 1xEV-DV offers the potential for better performance with high initial code rates.

[7]Chase Combining is the simplest form of Hybrid ARQ. The decoder combines multiple received copies of the coded packet weighted by the SNR prior to decoding.

[8]In Incremental Redundancy instead of sending simple repeats of the entire coded packet, additional redundant information is incrementally transmitted if the decoding fails on the previous attempt.

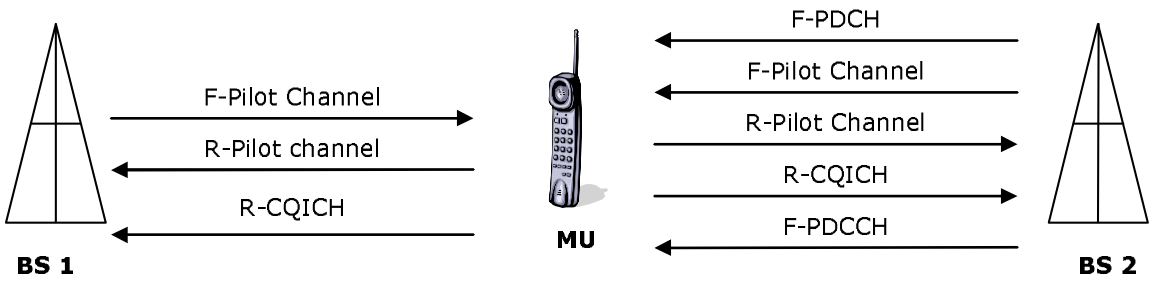

Figure 3.2: BS Selection Example

3.5.5 Base Station Selection

For a higher system throughput and more efficient use of the air link, 1xEV-DV lifted the burden of soft handoff from the system since the MU selects one BS from its active set to serve its forward link based on the channel quality (SINR) of each sector. The MU monitors the pilot channel of each BS in its active set as shown in Figure 3.2. It determines which BS has the best channel quality. It selects the BS and sends its selection via the R-CQICH channel. The system then transmits the forward link channel data through the selected BS. Figure 3.2 shows this selection process in a system of two BSs. If BS 2 is selected, the selection and the channel quality information are sent via R-CQICH. Thus, only BS 2 transmits F-PDCCH and F-PDCH to MU.

CHAPTER 4

Proposed QoS Design

4.1 Quality of Service

New MWG systems are required to support a wide range of new services and applications. This is achieved not only by allocating optimum transmission resources for a dynamically changing wireless channel but also by assuring certain QoS guarantees according to the type of traffic requested. This promises a higher system overall throughput gain. The proposed dynamic resource allocation scheme will enable service providers the ability to provide differentiated levels of service based upon the user application needs. It is especially important to provide proper transmission resources (modulation, coding rate, etc.) to the most valuable type of traffic requests. Some of these types of traffic are classified according to their sensitivity to QoS requirements (bandwidth, latency, jitter, and packet loss) into four classes [10]:

- **Conversational:** it requires low delay and low data loss rate. It is also sensitive to delay variations. An example of this class is Video conferencing.

- **Streaming:** it is less sensitive to delay and may require high bandwidth. An example of this class is Audio and Video traffic such as live TV coverages.

- **Interactive:** it has variable bandwidth requirements, moderate delay, and moderate data loss rate. An example of this class is Web browsing (such as HTTP applications, e-mails or telnet).

- **Background:** it is highly tolerant to delay and data loss rate. It has variable bandwidth. An example of this class are best effort traffic such as image downloads.

4.2 QoS Modes and Parameters

Following the 3GPP2 1xEV-DV version D specifications [5, 16], the suite of cross layer modules proposed provide two QoS modes the *Assured* Mode and the *Non-Assured* Mode. The QoS mode will depend on the MU requesting services and will be set for each user by the Priority Admission Control module described in section 4.4. The Non-Assured mode is the default mode and is set when either the MU does not provide a QoS profile to the serving BS, when the MUs QoS demands are higher than its QoS profile, or when there are limited resources. When the MU QoS profile matches its QoS demands than the Assured QoS Mode is selected and the transmission resources will be allocated to satisfy the QoS requested. Some of the main QoS parameters that will be considered in this thesis are: priority, minimum requested and accepted data rate, maximum requested and acceptable delay, and requested and acceptable data loss rate for the forward data links.

4.3 Cross Layer Design Architecture

To achieve a higher overall throughput in the emerging MWG systems yet satisfy a vast range of new applications that requires a minimum level of QoS guarantees, transmission resources need to be dynamically allocated. This dynamic allocation is not just based on the time-variant non stationary wireless channel conditions but also based on the QoS requirements set by the different user's priority and the type of traffic requested. Before we address and explain the suite of cross layer design modules proposed in this thesis, we need to discuss the general design architecture. Figure 4.1 shows the feedback loop between the MUs (the users), and the BS.

4.3.1 Mobile Unit Feedback

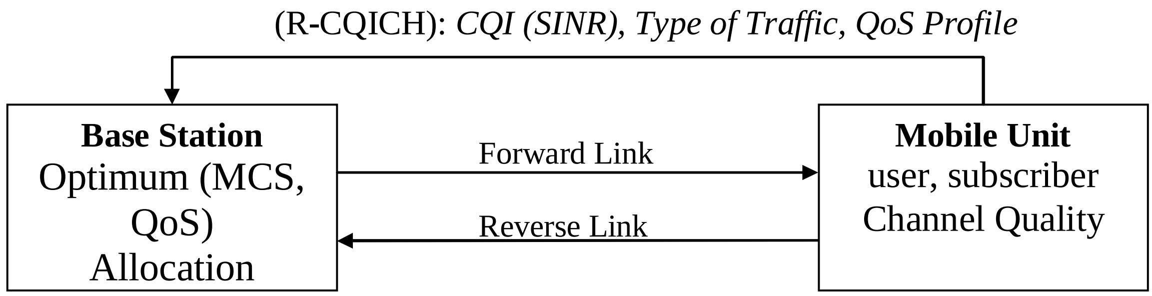

Figure 4.1: BS and MU Feedback Loop

One of the Forward Packet Data sub-channels is the Forward Packet Data Pilot Sub-channel [13]. It provides the MUs with the capability of predicting the received channel quality (SINR) from each BS's transmitter to accurately decide on the initial resource parameters of the data channel transmission. By constantly monitoring the quality of the pilot channels received from the different BSs, the MU can also identify the BS from which it would receive further signals. This is achieved by selecting the strongest pilot it receives, thereby enabling what is called *Macro Diversity* [18]. The channel quality SINR is predicted using both the Forward Packet Data Sub-channels and the continuous Pilot Channel, at a rate of 800 Hz. The CQI as indicated in [17] (the SINR, describing the instantaneous channel conditions) is continuously sent back using the reverse link F-PDCH through the R-CQICH at a rate of 800 Hz [10]. The CQI can be either a *full CQI* value which is the SINR estimate of the forward link pilot channel, or a *differential CQI* value, which is a positive or negative increment to the most recently transmitted SINR value. At the BS, Differential CQI values are interpreted in a cumulative manner. Therefore, the best current CQI estimate at the BS is the most recently received Full CQI value plus the sum of all the Differential CQI values that were subsequently transmitted. The BS can then use the information received in its reverse link to determine the best transmission resources suited for the wireless channel condition, the latter will be discussed in more details in the next section.

4.3.2 Base Station Processing

The BS is where the suite of cross layer design modules takes place. As it was described in the previous section that the MUs send on its reverse link channel (R-CQICH) the parameter (CQI) describing the quality of the wireless channel of the forward link channel F-PDCH. With this parameter and for the purpose of our design, a set of QoS parameters [16], describing the user *QoS profile* and the *Type of Traffic* requested, are also passed to the BS at service connection, negotiation/re-negotiation during a call and during a handoff [10]. As described in [10], a subscriber QoS profile shall also be maintained at the BS. These parameters could contain at least the subscriber QoS class, requested QoS class, maximum subscriber data rate, maximum allowable delay, maximum allowable latency, and maximum allowable jitter. Using all these parameters sent by the MU, the BS using the suite of cross layer design modules proposed will dynamically allocate the best transmission resources and schedule data transmission in the most efficient way that guarantees the best QoS level based upon the subscriber's QoS profile and application needs. In this section, a detailed description of all the modules in the cross layer design proposed will be presented. Figure 4.2 shows a block diagram of the proposed design architecture describing the BS to users interface and all the different modules of the design referring to appropriate layer they will be handled at the BS.

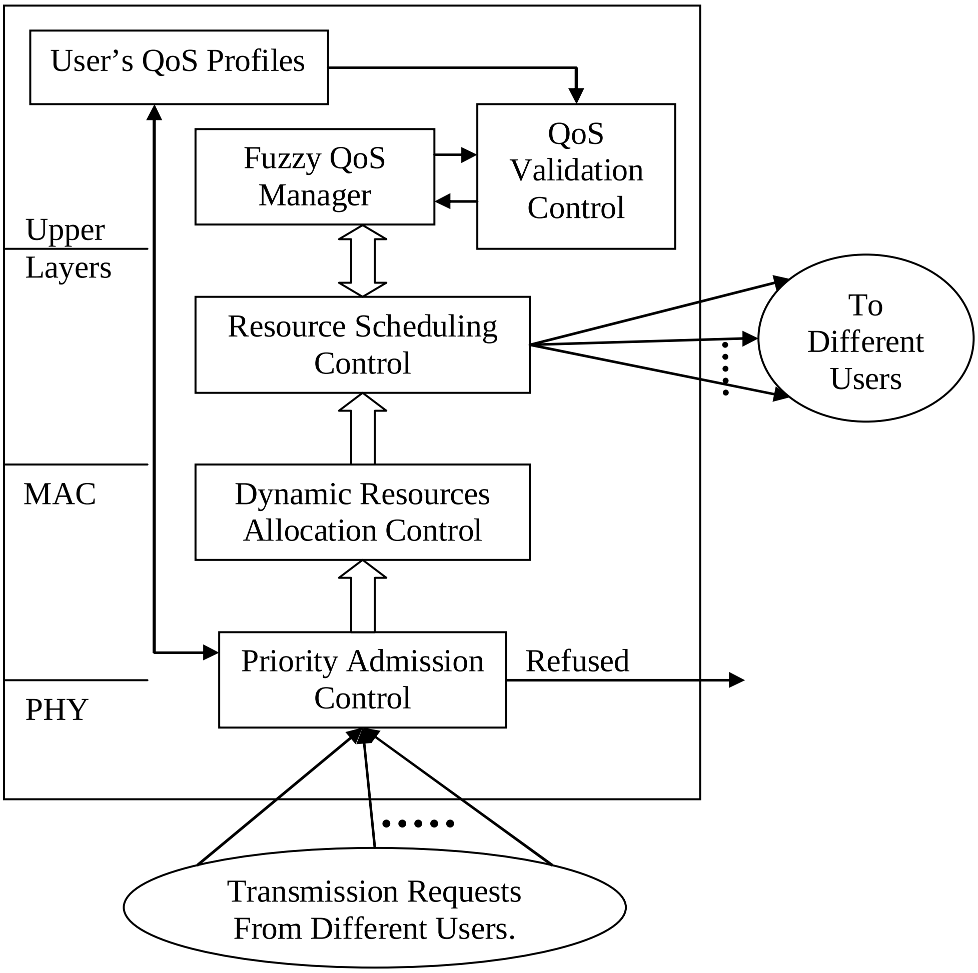

Figure 4.2: QoS Provisioning Architecture

4.4 Priority Admission Control

In the proposed Cross Layers Multiuser Diversity Modules suite, the word *multiuser diversity* comes from the fact that different users can usually have independent channel conditions for the same shared medium [8][11]. Earlier wireless generation systems allocated significant amount of transmission resources to users with poor channel conditions, which did not just wasted valuable and limit resources but also done very poorly enhancing the user experience or the overall throughput of the system. This module, Priority Admission Control, will enable mutiuser diversity: First, by prioritizing the transmission requesting users according to their channel conditions (CQI) allowing later efficient resource allocation by the Dynamic Resources Allocation Control module to the different users. Second, by admitting users based on their QoS profiles and the amount of resources requested to be allocated based on the Type of Traffic. Two possible admission scenarios are described as follows:

Scenario 1

In this scenario, a particular user competing for transmission resources has a poor QoS profile (low data rate, high allowable delay ,*etc.*) or do not have a QoS profile at all and is requesting packet data services that are not sensitive to strict QoS requirements. In this case, the QoS will be in a Non-Assured QoS mode [8] and such a user could be allowed *Best Effort* resources according to its channel condition and its priority or could be denied service. This is complying with the 3GPP2 QoS Requirements document [10]. If on the other hand the user is requesting packet data services that are sensitive to minimum data rate, delay, or error rate, such a user will be immediately denied.

Scenario 2

In this scenario, a user competing for transmission resources has a good QoS profile (high data rate, low allowable delay, *etc.*) and is requesting services that are not sensitive to strict QoS requirements. This user will be allocated the best transmission resources its channel conditions allows without going through further processing with the other modules in the suite proposed. If in the other hand, the user is requesting services that are sensitive to strict QoS requirements then this user will be admitted, transmission resources will be allocated, and QoS will be in the Assured QoS mode [8]. Such a user QoS can be guaranteed according to the channel conditions, and the QoS profile of the user by further processing with the cross layer suite proposed. The set of users admitted will be passed onto the next module, Dynamic Resources Allocation Control, with their classification parameters (*Diversity Priority* according to their (CQI), *Type of Traffic* requested, *QoS mode*, and *QoS profile*).

4.5 Dynamic Resource Allocation Control

The goal of the proposed resource allocation module is to make the optimal selection of the optimal combination of resource parameters. Such parameters are the number of Walsh codes, the number of time slots, the modulation scheme and the channel code rate. It makes the optimum selection according to the forward link channel conditions and to meet the QoS requirements imposed by the different user's applications. Figure 1.1 shows the optimization block diagram of all resource parameters. If Radio Configuration 10 is used, the resource parameters are as listed in Table 4.1 (refer to Appendix A.2 for the complete Table). This module will then allocate the optimum transmission resources to the different requesting users using their classifications parameters such as: Diversity Priority, Type of Traffic, QoS mode and QoS profile noted respectively (D_p, TT, Q_m, Q_p)as follows:

Step 1

According to user's Diversity Priority value D_p (channel conditions), a minimum Modulation and Coding Scheme (MCS) level of resource parameters listed in Table 4.1 will be allocated to each user. The actual mapping technique of the SINR values of each user to the initial and minimum MCS level is out of the scope of this thesis. One of these techniques is the Threshold Method mapping technique detailed in reference [15]. Therefore, this module will only use Table 4.1 as a look up table to decide on the minimum MCS level based on its D_p value as defined by the previous module.

Users with high D_p (have a good wireless channel conditions) are typically assigned a higher MCS level (higher order modulation (16 QAM), and higher data rate). The MCS level assigned will decrease as the D_p value of the user decreases. This is due being located at farther distance from the BS.

Step 2

When the minimum MCS level of each user has been selected based on their CQI values, the module will check the QoS mode of each user. If the QoS mode Q_m of a user is the Non-Assured QoS mode, then it allocates to the user the minimum MCS level of resources based on its D_p value and the availability of resources without going through any further processing. This will cut the processing overhead at the BS for each user and will eventually improve the overall throughput and efficiency of the wireless system. If on the other hand, the QoS mode of a user is the Assured QoS mode then it sets the minimum MCS level selected based on the user D_p value and the TT requested for further resource optimization using the dynamic resource allocation scheme. At this stage, the BS has a set of users with their MCS level intervals and their type of traffic as follows:

$$Users_{MCS} = \left\{ \begin{array}{llll} MCS_{min,k}, & MCS_{min+1,k}, & \cdots, & MCS_{max,k} \\ MCS_{min,k+1}, & MCS_{min+1,k+1}, & \cdots, & MCS_{max,k+1} \\ \cdots, & \cdots, & \cdots, & \cdots \\ MCS_{min,K}, & MCS_{min+1,K}, & \cdots, & MCS_{max,K} \end{array} \right\} \tag{4.1}$$

$$Users_{TT} = \left\{ \; User_{TT,k}, \;\; User_{TT,k+1}, \;\; \cdots, \;\; User_{TT,K} \; \right\} \tag{4.2}$$

4.5.1 System Resource Parameters

In this thesis, it is assumed that *Radio Configuration*[1] *10* is used by the BS [17]. Therefore, encoder packets are to be transmitted as one or more subpackets. The F-PDCH Encoder Packet is composed of *Information Bits* (194, 386, 1538, 2306, 3074,or 3842), a 16 bit *Frame Quality Indicator*, and a 6 bit *Turbo Encoder Tails allowance*. The Encoder Packet format is shown in Figure 4.3. The total number of bits in the encoder packet N_p as listed in Table 4.1 can then be one of these values: 216, 408,792, 1560, 2328, 3096, and 3864 Bits. The rest of the table values are:

Information Bits 194, 386, 1538, 2306, 3074, or 3842 Bits	Frame Quality Indicator (16 Bits)	Turbo Encoder Tail allowance (6 Bits)

Figure 4.3: F-PDCH Encoder Packet Format.

- $N_{w,k}$ is the number of 32-chip Walsh channels for the k^{th} subpacket.
- $N_{s,k}$ is the number of 1.25 ms slots for the k^{th} subpacket.
- M_k is the modulation order for the k^{th} subpacket of an encoder packet and can be decided on using the following modulation order product code rate equation:

$$M_{PCR,k} = \frac{N_p}{48 N_{w,k} N_{s,k}} \tag{4.3}$$

The modulation order is then decided on using the following rules known that $M = 2^{M_k}$:

  - If $0 \leq M_{PCR,k} \leq \frac{3}{2}$, then a modulation order M_k of 2 is used corresponding to QPSK modulation scheme.

[1]It is one of the radio configuration defined in new 1xEV-DV standard. It is a set of forward and reverse traffic channel transmission resources that is defined by physical layer parameters such as modulation scheme, coding, and data rates.

- If $\frac{3}{2} \leq M_{PCR,k} \leq 2$, then a modulation order M_k of 3 is used corresponding to 8-PSK modulation scheme.
- If $2 \leq M_{PCR,k} \leq 3.2$, then a modulation order of M_k 4 is used corresponding to 16-QAM modulation scheme.

- μ_k is the subpacket data rate and can be calculated using the following equation:

$$\mu_k = \frac{N_p}{D_k} \tag{4.4}$$

where $D_k = 1.25\ N_s$ ms is the subpacket duration in term of time.

- $R^c_{effective,k}$ which is the effective subpacket code rate of the k^{th} subpacket of an encoder packet and can be calculated using the following equation:

$$R^c_{effective,k} = \frac{N_p}{L_k} \tag{4.5}$$

where L_k is the number of subpacket binary code symbols in all the slots and Walsh channels of F-PDCH subpacket. L_k can be calculated using the following equation:

$$L_k = 48 N_{w,k} N_{s,k} M_k \tag{4.6}$$

4.5.2 Dynamic Resource Allocation Scheme

To get the optimal combination of resource parameters that will satisfy a specific QoS request and counts for the instantaneous changes of wireless capacity, our resource allocation scheme will go through four major steps.

Step 1

With Radio Configuration 10, a BS of 1xEV-DV system can only support one or two F-PDCH channels.

MCS Level	N_p	$N_{w,k}$	μ_k	$N_{s,k}$	$R^c_{effective,k}$	M_k	Modulation Type
0	2,328	28	1,862.4	1	0.5774	3	8-PSK
1	3,864	27	1,545.6	2	0.7454	2	QPSK
2	3,096	26	2,476.8	1	0.6202	4	16-QAM
3	3,864	26	3,091.2	1	0.7740	4	16-QAM
4	1,560	25	1,248.0	1	0.6500	2	QPSK
5	2,328	25	1,862.4	1	0.6467	3	8-PSK
6	3,096	25	1,238.4	2	0.6450	2	QPSK
7	3,864	25	1,545.6	2	0.5367	3	8-PSK
8	2,328	23	931.2	2	0.5272	2	QPSK
9	2,328	23	1,862.4	1	0.5272	4	16-QAM
10	3,096	23	2,476.8	1	0.7011	4	16-QAM
11	3,864	23	1,545.6	2	0.5833	3	8-PSK
12	1,560	22	1,248.0	1	0.7386	2	QPSK
13	3,096	22	1,238.4	2	0.7330	2	QPSK
14	1,560	21	1,248.0	1	0.5159	3	8-PSK
15	3,096	21	1,238.4	2	0.5119	3	8-PSK
16	3,096	21	2,476.8	1	0.7679	4	16-QAM
17	3,864	21	1,545.6	2	0.6389	3	8-PSK
18	1,560	20	624.0	2	0.4063	2	QPSK
19	2,328	20	465.6	4	0.3031	2	QPSK
20	2,328	20	931.2	2	0.6063	2	QPSK

Table 4.1: System Resource Parameters

- If one F-PDCH is supported, the BS could only support one to 28 32-chip Walsh channels on the F-PDCH.
- If two F-PDCH are supported, the BS could only support one to 28 32-chip Walsh channels on each F-PDCH and could not support more than 28 32-chip Walsh channels on both F-PDCH.

Therefore, the 1xEV-DV system only supports packets from one user in one of the two possible F-PDCH channels during each time slot. Also, those two channels have to be assigned to different users, meaning that only one user can transmit its packets in a time slot, even though different services can transmit at the same time in different Walsh code channels.

For a user having S_l service levels, the system allocates W_l Walsh codes for each service level without exceeding a total number of 28 Walsh codes. Thus, the channel capacity in each time slot of the F-PDCH packet is:

$$\eta = \sum_{s_l=1}^{S_l} X_{s_l} \quad S_l = 1, 2, \cdots, N \ service \ levels \tag{4.7}$$

$$Maximize \ \ X_{S_l} = \frac{P_{S_l}}{P_{S_l} + O(CRC_{S_l}, \phi_{S_l})N_s} \tag{4.8}$$

where P_{S_l} is the packet length of service level S_l, $O(.)$ is the total system overhead from the physical layer, which is the function of the Code Redundancy Check CRC_{S_l}, and the channel turbo coding rate ϕ_{S_l}, and

N_s is the number of time slots per F-PDCH packet.

Equation 4.8 has to be maximized while satisfying the following constraints:

$$Subject \ \ to \ \ P_{min} \leq \sum_{s_l=1}^{S_l} P_{s_l} \leq P_{max}, \tag{4.9}$$

$$\sum_{s_l=1}^{S_l} P_{s_l} \in \{216, 408, 792, 1560, 2328, 3096, 3864\} \quad Bits, \tag{4.10}$$

$$N_s = 2^n; \ \ 0 \leq n \leq 2; \ \ n \neq 3, \tag{4.11}$$

$$X_{S_l} \leq r_{n,l}^{S_l}, \ and \tag{4.12}$$

$$X_{S_l}^{min} \leq \sum_{s_l=1}^{S_l} X_{s_l} \leq X_{S_l}^{max} \tag{4.13}$$

Where $r_{n,l}^{S_l}$ is the instantaneous channel capacity of service level s_l, which is also affected by the wireless channel model, the modulation scheme, and channel code rate and is given by:

$$r_{n,l}^{S_l} = W \log_2(1 + |x_n|^2) \tag{4.14}$$

where W is the channel Bandwidth and x_n is the received channel SNR at time t. This step should give the optimum packet length $P_{l,OP} = \sum_{s_l=1}^{S_l} P_{s_l}$ of all service levels S_l in one time slot, and the number of time slots in an encoder packet N_s.

Step 2

Using the optimized results optimum packet length $P_{l,OP}$ and number of time slots N_s found in Step 1, we can allocate the minimum MCS level (resource parameters combinations) from the look up table given in Table 4.1 by satisfying the following legacy services load constraint:

$$Subject\ \ W_l = \sum_{s_l=1}^{S_l} N_{w,S_l} = L_{max} - \Delta \tag{4.15}$$

Where W_l is the total possible allocated number of Walsh codes, N_{w,S_l} is the number of Walsh codes of each service level S_l, L_{max} is the maximum allowable number of Walsh codes per F-PDCH channel is defined in reference [17] to be equal to 28 32-Walsh codes, and $1 \leq \Delta \leq 27$ is a random number of Walsh codes needed to support legacy services. In this stage, there are two possible outcomes:

- If the $P_{l,OP}$ value found in step 1 does not satisfy the legacy services load constraint (equation 4.15), repeat step 1 and select a different, noting that the legacy services load constraint could also change in the next iteration.

- If the $P_{l,OP}$ value found in step 1 satisfies the legacy services load constraint (equation 4.15), jump to step 3.

In this step, a minimum and initial MCS level is selected for the fixed number of Walsh codes and the maximum channel capacity of the service time slot for each user.

Step 3

Using the F-PDCH packet's data rate value μ corresponding to the optimum packet length $P_{l,OP}$ and the number of Walsh codes W_l found through a look up table operation, in this step the scheme will assure users' QoS requirements according to the type of traffic requested, and to their QoS profiles. In particular, those applications that are delay and loss bounded. For a selected data rate value μ, and assuming that user application has a delay bound of D_{max} and data loss probability not greater than a fixed value ξ, then we can define a statistical QoS guarantee equation as:

$$sup_t \Pr\{D(t)_k \geq D_{max}\} \leq \xi \tag{4.16}$$

Where $D(t)_k$ is the queuing delay of the k^{th} subpacket of an encoder packet at time t.

Users QoS requirements $\{\mu, D_{max}, \xi\}$ guarantees for the requested type of traffic and the instantaneous channel conditions using the combination of resource parameters can be verified using the effective capacity concept [9]. For the instantaneous channel capacity $r_n(t)$ at time t, the effective capacity function can be written as:

$$\alpha(\mu) = -\frac{\Lambda(\mu)}{\mu}, \quad \forall\mu \geq 0. \tag{4.17}$$

$$\alpha(\mu) = -\lim_{t\to\infty}\frac{1}{\mu t}\log E[e^{-\mu S(t)}], \quad \forall\mu \geq 0. \tag{4.18}$$

Where $\Lambda(\mu)$ is the asymptotic log-moment generation function of the accumulated channel service provided from time 0 to time t and defined as:

$$S(t) = \int_0^t r(\tau)d\tau \tag{4.19}$$

Where $r(\tau)$ is the channel service rate at time τ. Thus, for a fixed data rate μ, and a variable channel capacity r_n, the statistical QoS guarantee equation 4.16 can be re-written in term of $\alpha(\mu)$ as:

$$sup_t \Pr\{D(t)_k \geq D_{max}\} = e^{-\theta(\mu)D_{max}} \leq \varsigma \tag{4.20}$$

Where the QoS exponent function $\theta(\mu)$ is defined in terms of the effective capacity function $\alpha(\mu)$ as:

$$\theta(\mu) = \mu\alpha^{-1}(\mu) \tag{4.21}$$

Where $\theta(\mu)$ can be estimated and translated to service curve parameters used by the resource scheduling module using the following equation:

$$\theta(\mu)_{avg} = \frac{\gamma(\mu)\mu}{\mu\tau(\mu) + q} \tag{4.22}$$

Which can be estimated by simply averaging a number of samples N the following service curve triplet $\{\tau, q, \gamma\}$over a sampling period T:

$$\tau = \frac{1}{N}\sum_{0}^{N} T_n \tag{4.23}$$

$$q = \frac{1}{N}\sum_{0}^{N} Q_n \tag{4.24}$$

$$\gamma = \frac{1}{N}\sum_{0}^{N} S_k \tag{4.25}$$

Where T_n is the remaining service time (time for a packet to be transmitted) of the k^{th} subpacket of an encoder packet in service, Q_n is the number of bits in the queue, and $S_n \in \{0, 1\}$ is the sign bit indicating whether a subpacket is being transmitted.

Step 4

In this step and after estimating the QoS exponent function $\theta(\mu)_{avg}$, we could use it in equation 4.20 to check whether the QoS requirements $\{\mu, D_{max}, \xi\}$ are being satisfied using the selected resource parameters combination (MCS level). If the QoS requirements are not satisfied then step 1, 2, 3 and 4 are repeated. Thus, enabling an optimum selection of resource parameters satisfying a QoS requirements and accounting for an instantaneously changing wireless channel conditions.

4.6 Resource Scheduling Control

Given the set of admitted users with their initial transmission resource parameters (MCS levels), this module Resource Scheduling Control will not just schedule data transmission according to users priorities (favorable channel conditions: higher D_p), since it can result in an unfair scheduling among users, but also according to the different QoS requirements. Therefore, a balance between optimum resource utilization and fairness among users can be accomplished. Users will be classified first by their D_p values and then further divided into sub-classes according to each user's QoS requirements as shown in Figure 4.4. Generally, users with higher D_p values and requesting a higher level of QoS will be giving a higher scheduling priority than the ones with higher D_p values but are requesting a lower level of QoS. Users with lower D_p values but not lower than a threshold level T_{MCS} and are requesting a higher QoS level than the ones with better D_p values can also be given a higher scheduling priority. On the other hand, users with low D_p values (poor channel conditions) lower than a threshold level T_{MCS} will be given a lower scheduling priority. A higher scheduling priority will be possible for the latest depending on the higher priority's queues size and wait.

4.7 Fuzzy QoS Manager

This module will use the scheduling rules described in section 4.6 to help the Resource Scheduling Control to schedule user's packets with the best transmission resources through the *Queue* with the service curve that will satisfy a minimum requested QoS for each user application. It takes the minimum selected D_p (MCS

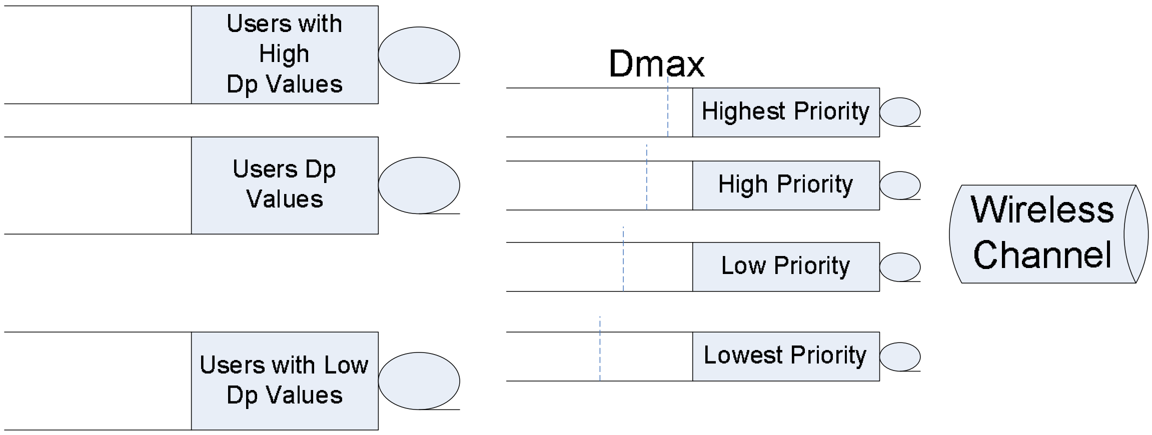

Figure 4.4: Resource Scheduling Model

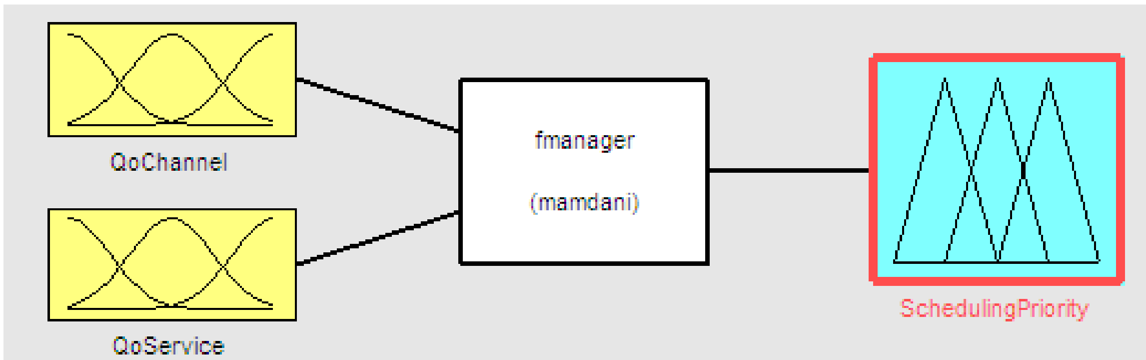

Figure 4.5: Fuzzy Manager Block Diagram

level) for each user and the user's QoS requested and pick a scheduling priority queue depending on the buffer size and waiting time. Using Matlab's Fuzzy logic toolbox we can simulate the fuzzy manager control. Taking the values of an input vector, and based on some set of rules, the fuzzy toolbox will enable the assignment of values to an output vector. In our case and as shown in Figure 4.5, the set of input vectors are the quality of the wireless channel and the QoS demanded by user's applications. The wireless channel condition can either be poor, good, or excellent. The QoS requested can either be non-sensitive, sensitive, or very sensitive. The fuzzy logic toolbox will use the input vectors and a set of scheduling rules in parallel to make a decision on the scheduling priority. It will schedule user's applications through one of the three queues, a high priority, average priority, or a low priority queue. Figure 4.6 shows the roadmap of the fuzzy logic process. The two left plots columns of the figure show how the input vectors (membership functions[2]) are used in the rules, with their input values in the top of each column. The right plots column of the figure shows how the output vectors respond to the rules and the inputs. The scheduling priority value $[0-10]$, describing the queue priority, is shown in the top of the column.

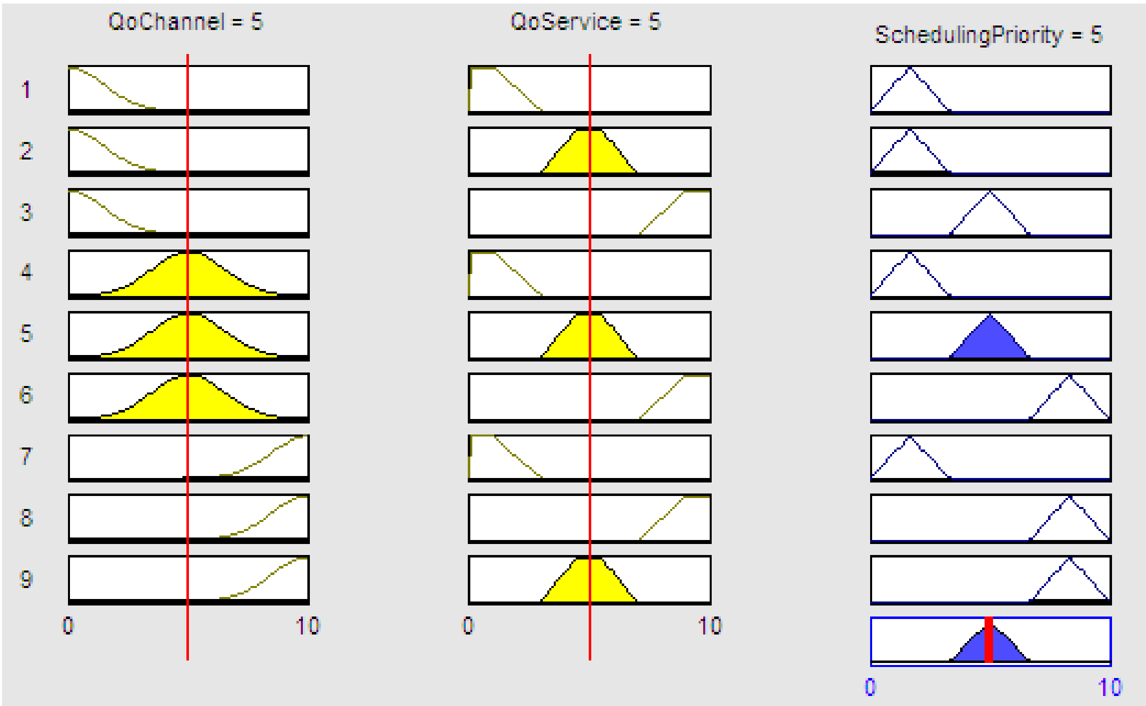

Figure 4.6: Fuzzy Manager Rules Graphics

4.8 QoS Validation Control

This module will simply compare the QoS guaranteed through the resources allocated to a particular user with its QoS profile. It will verify whether or not the minimum requested QoS level are met. If yes, then the resources will be allocated to

[2]A membership function is defined in Matlab as a curve that defines how each point in the input is mapped to a membership value between 0 and 1

the user with no further processing, if not the dynamic resource allocation scheme will take the MCS level previously selected as a new lower bound of system resources and go through the process again until QoS guarantees are at least the QoS parameters set by the user QoS profile.

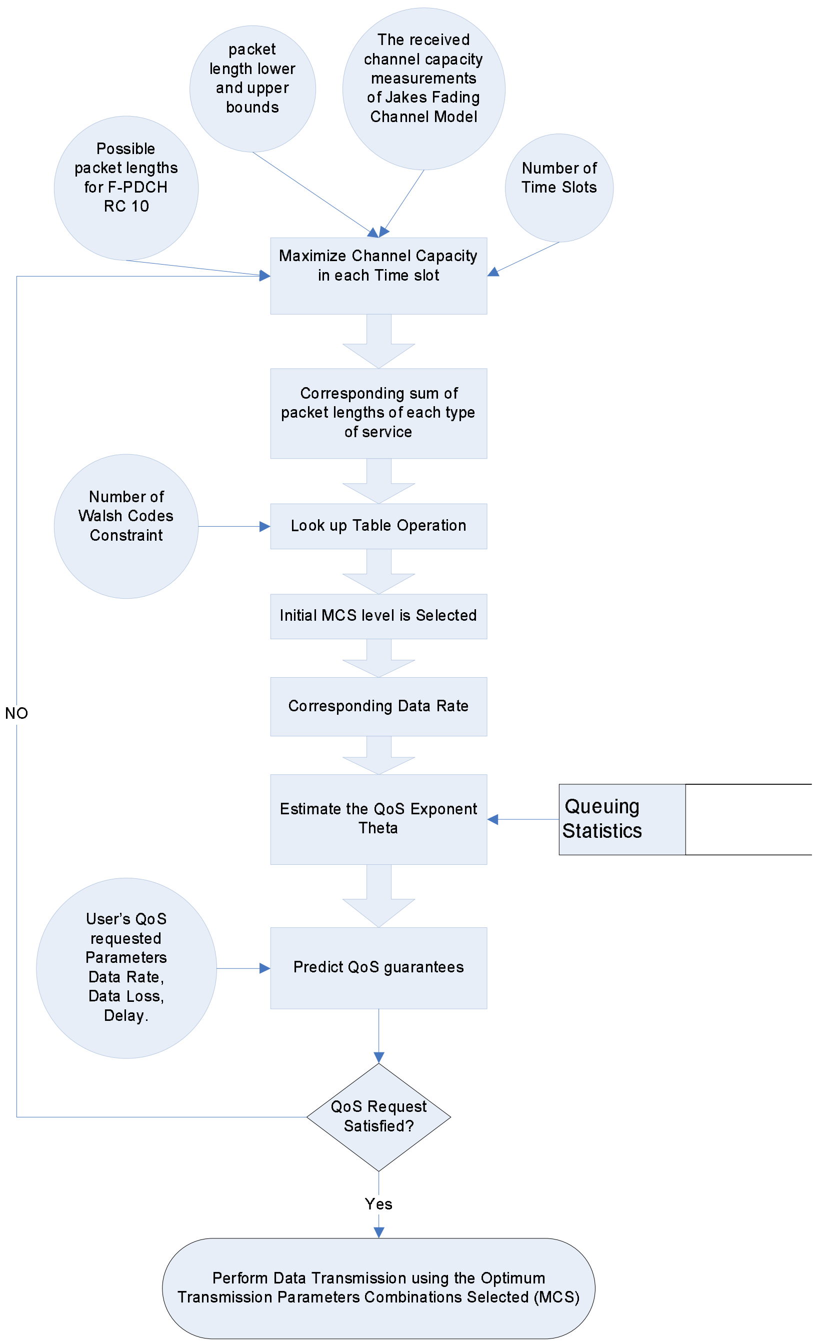

Figure 4.7: Resource Allocation Scheme

CHAPTER 5

Simulation Models and Results

The simulation in this thesis is performed on the basis of the requirements set by 3GGP2 Evaluation Methodology standard for evaluating the performance of 1xEV-DV system [18]. The simulation is carried in the link level. Link level simulation will be performed on the physical layer to determine the E_b/N_0 performance of single link. The E_b/N_0 performance from the link level simulation will be performed under Rayleigh Fading channels including the SNR, the BER, and the PER. A mix of traffic models have also been considered to analyze the throughput and the QoS guarantees of a system using the dynamic resource allocation scheme proposed. The scheme proposed has been implemented using Matlab Optimization toolbox and other simulation tools (Fuzzy logic toolbox).

5.1 Channel Models

In this section, we discuss the different channel modules that are adequate to simulate wireless systems including propagation model, shadowing model, and fading model with a concentration on the Jake's simulation model.

5.1.1 Propagation Model

The propagation model (path loss) for the two test environments *Pedestrian* and *Vehicular* are based on the *Hata Model* (COST 231) [33]. The propagation model equations are a relationship between path loss L measured in dB, the distance and the terrestrial environments of the system. The propagation model for Pedestrian (indoor or outdoor) test environment is given by:

$$L = 30.0\log_{10}(f_c) + 40.0\log_{10}(d) + 49.0 \tag{5.1}$$

where f_c is the carrier frequency in MHz and d is the distance in Km between the BS and the MU. The propagation model for the Vehicular test environment is given by:

$$L = -18.0\log_{10}(\Delta h_b) + 21.0\log_{10}(f_c) + 40.0(1.0 - 0.004\Delta h_b)\log_{10}(d) + 80.0 \tag{5.2}$$

where Δh_b is the BS antenna height measured from the average rooftop in meters.

5.1.2 Shadowing Model

Path loss is not enough to describe any particular setting or signal path since the surrounding environment may be largely different from two different locations having the same transmitter to receiver distance. This phenomena is described by the randomly distributed shadowing effect referred to as *log-normal* shadowing *Hata Model.* Log-normal shadowing can be expressed as follows:

$$L_s(x) = \begin{cases} \frac{1}{\sqrt{2\pi}x\sigma} e^{\left(\frac{-(\ln(x)-\mu)^2}{2\sigma^2}\right)} & ;(x \leq 0) \\ 0 & ;(x < 0) \end{cases} \tag{5.3}$$

where $\ln(x)$ represents the path loss measured in dB, μ is the mean path loss in dB, and S is the standard deviation of the path loss in dB, usually in the range of 5 to $12dB$. For Pedestrian testing environment, log-normal shadowing is $12dB$ indoor and $10dB$ outdoor. For Vehicular testing environment log-normal shadowing is $10dB$.

5.1.3 Fading Model

Fading is another epidemics facing data transmission through wireless channels. It is generally due to the interference caused by the multi-path nature of the propagation channel (copies of transmitted signals propagate to the receiver through different reflective paths). It is responsible for the most rapid and serious changes of the received signal strength (amplitude) and phase. For a practical system model and for realistic simulation results, it is also important to know the statistics of the received signal. One of the commonly adopted multipath fading models of the received signal strength at the MU is Rayleigh Fading. Rayleigh Fading also called small-scale fading or fast fading is described by the following PDF function as follows:

$$f_{\infty}(R) = \begin{cases} \frac{R}{\sigma^2} \exp\left[\frac{-R^2}{2\sigma^2}\right] & ; (R \geq 0) \\ 0 & ; R \ \ otherwise \end{cases} \tag{5.4}$$

where R is the envelope amplitude of the received signal, and $2\sigma^2$ is the pre-detection mean power of the multipath signal.

In this simulation we will consider different channel models as described in the 1xEV-DV evaluation methodology standard [18]. These channel models are described in Table 5.1 for a Pedestrian and Vehicular testing environments.

Channel Model	Multi-path Model	Building Penetration Loss	Number of Paths	Speed(km/hr)	Fading
Model A	Pedestrian A	8dB	1	3	Jakes
Model B	Pedestrian B	8dB	3	10	Jakes
Model C	Vehicular A	0dB	2	30	Jakes
Model D	Pedestrian A	8dB	1	120	Jakes
Model E	Single path		1	0,fD=1.5 Hz	Rician Factor K=10 dB

Table 5.1: Channel Models

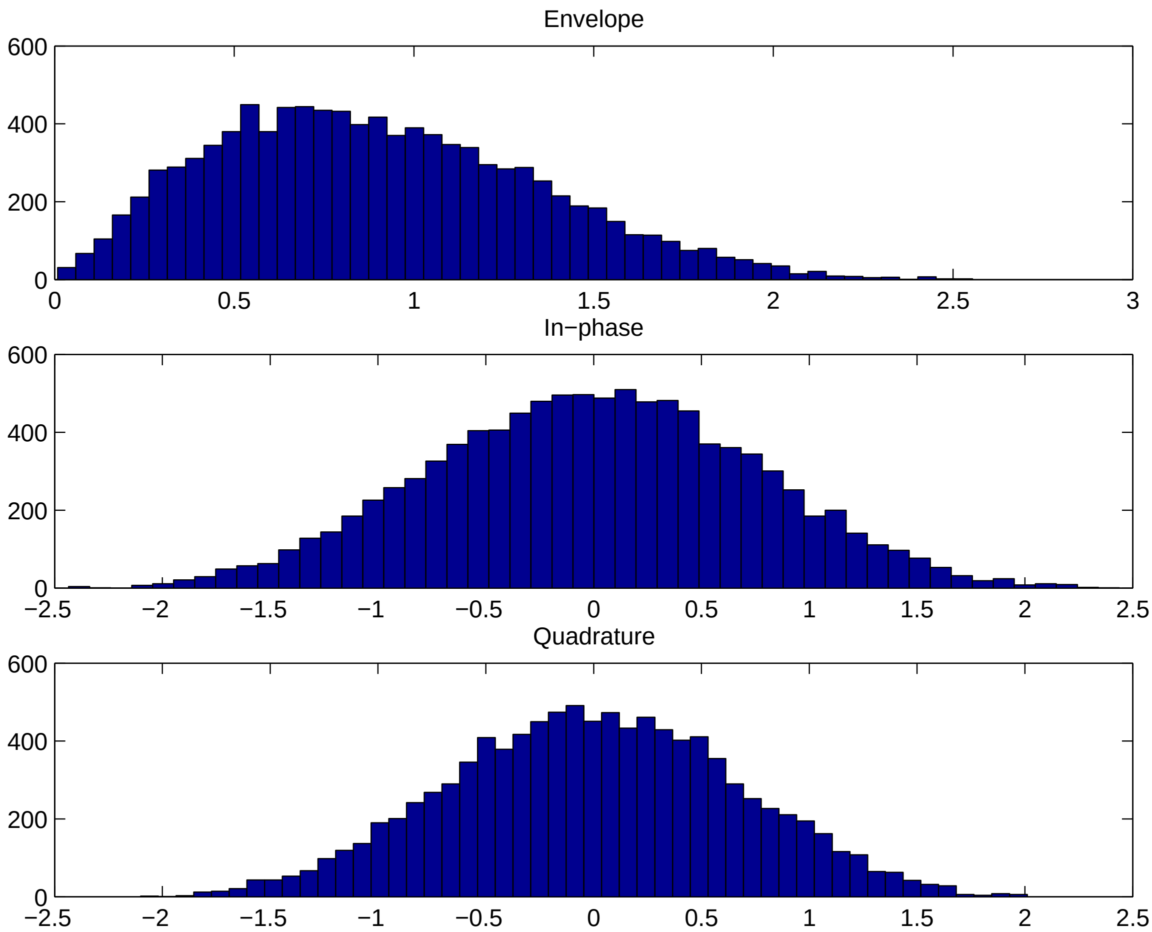

Figure 5.1: Jakes Fading Envelopes

5.1.4 Jake's Simulation Model

The simulation in this thesis will be carried out using Jake's Simulation Model for a Rayleigh Fading channel [35]. The Jake's model implements the channel as a sum of sinusoids as defined by the following equation:

$$g(t) = g_c(t) + j g_s(t) \tag{5.5}$$

where the real (In-phase) and imaginary (Quadrature) parts of the equation are shown in Figure 5.1 and defined as follows:

$$g_c(t) = \frac{2}{\sqrt{N}} \sum_{n=1}^{M+1} a_n \cos(\omega_n t) \tag{5.6}$$

$$g_s(t) = \frac{2}{\sqrt{N}} \sum_{n=1}^{M+1} b_n \cos(\omega_n t) \tag{5.7}$$

where $N = 4M + 2$, a_n ,b_n ,and ω_n are defined respectively, as follows:

$$a_n = \begin{cases} 2\cos(\beta_n) & ; n = 1, 2, \cdots, M \\ \sqrt{2}\cos(\beta_{M+1}) & ; n = M + 1 \end{cases} \tag{5.8}$$

$$b_n = \begin{cases} 2\sin(\beta_n) & ; n = 1, 2, \cdots, M \\ \sqrt{2}\sin(\beta_{M+1}) & ; n = M + 1 \end{cases} \tag{5.9}$$

$$\beta_n = \begin{cases} \frac{\pi n}{M} & ; n = 1, 2, \cdots, M \\ \frac{\pi}{4} & ; n = M + 1 \end{cases} \tag{5.10}$$

$$\omega_n = \begin{cases} \omega_d \cos\left(\frac{2\pi n}{N}\right) & ; n = 1, 2, \cdots, M \\ \omega_d & ; n = M + 1 \end{cases} \tag{5.11}$$

The maximum Doppler shift ω_d in radian/s is given by:

$$\omega_d = 2\pi f_d \tag{5.12}$$

where the maximum Doppler shift frequency f_d is given by:

$$f_d = f_c \frac{v}{c} \tag{5.13}$$

where v is the MU speed as given in Table 5.1 in Km/hr, c is a constant equals to the wavelength of the transmitted signal, and f_c is the carrier frequency.

5.2 Traffic Model

In this thesis, a mix of traffic types are also considered for the simulation of the proposed dynamic resource allocation scheme. Figure 5.2 shows the CDF plot of the different traffic types listed in Table 5.2 with their QoS triplets $\{\mu, D_{max}, \xi\}$.

Traffic Type	Application Type	Data Rate: $r^{k,l}_{AWGN}$(Kbps)	Latency: D_{max}(msec)	Loss: ξ	Distribution
Voice	Conversational Voice	4-25	<400	Tolerable (<10-2)or 3% FER	Exp(0.25)
Video	Video Phone	32-384	< 150	Tolerable (<10-2)or 1% FER	Exp(0.5)
Data	Interactive HTML	144-384	< 4000	Sensitive (<10-5)or 0% FER	Exp(0.01)

Table 5.2: Traffic Mix Parameters

5.3 Simulation Parameters and Results

5.3.1 SNR, BER, and PER Estimates

The received SNR is represented by:

$$SNR(n) = \frac{E_b}{N_0} g(n)^2 \tag{5.14}$$

where E_b is the average bit energy of the received signal, N_0 is the noise power density and $g(n)$ is the received signal gain. Figure 5.3 shows the received SNR of a Jake's fading channel. The second portion of the figure shows the probability of the SNR exceeding an attenuation of 10 %.

To see the affect of the different channel modulation schemes selected by our dynamic resource allocation scheme over the nonstationary fading channel, let us first convert SNR to Bit Error Rate (BER). For the three different modulations schemes QPSK, 8-PSK, and 16-QAM the BER is represented respectively, by the following three equations:

$$BER_{QPSK}(n) = Q\left(\sqrt{2SNR(n)}\right) \tag{5.15}$$

$$BER_{8-PSK}(n) = 2Q\left(\sqrt{2SNR(n)\log_2(M)}\sin\left(\frac{\pi}{8}\right)\right)\frac{\frac{M}{2}}{M-1} \tag{5.16}$$

$$BER_{16-QAM}(n) = \frac{2\left(1-\frac{1}{\sqrt{M}}\right)}{\log_2\left(\sqrt{M}\right)} Q\left[\sqrt{\left(\frac{3\log_2\sqrt{M}}{M-1}\right)2SNR(n)}\right] \tag{5.17}$$

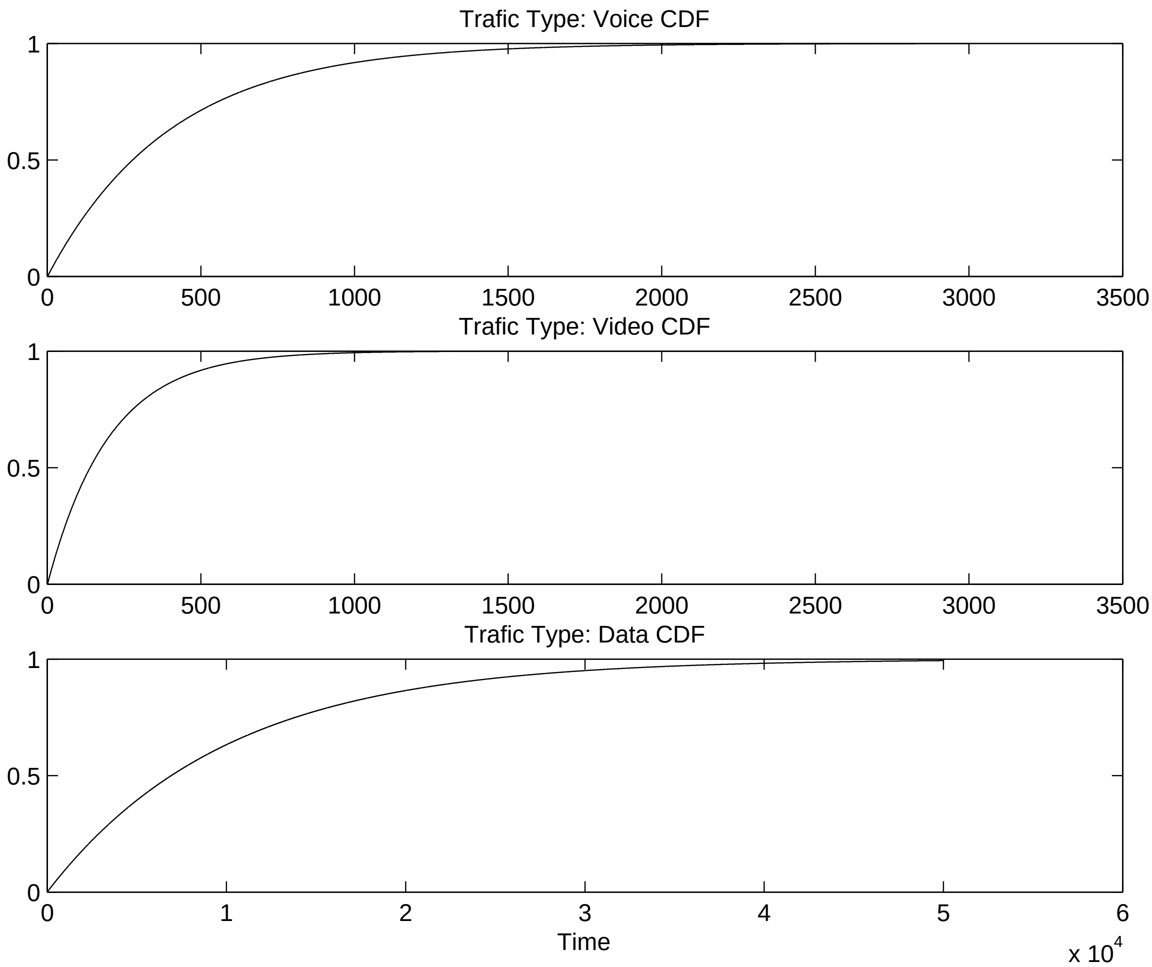

Figure 5.2: Traffic Types CDF

where $M = 2^k$ and $k = 1, \cdots, N$ is the number of source bits, the modulator accepts at a time to produce one of an available set of M waveform types, Q(.) is a complementary error function defined as:

$$Q(n) = \frac{1}{\sqrt{2\pi}} \int_u^\infty e^{-\frac{u^2}{2}} \tag{5.18}$$

To further investigate the affect of the channel coding scheme selected by our dynamic resource allocation scheme and to have an idea on the probability of packet loss at any time over the nonstationary and fading channel, let us convert BER to

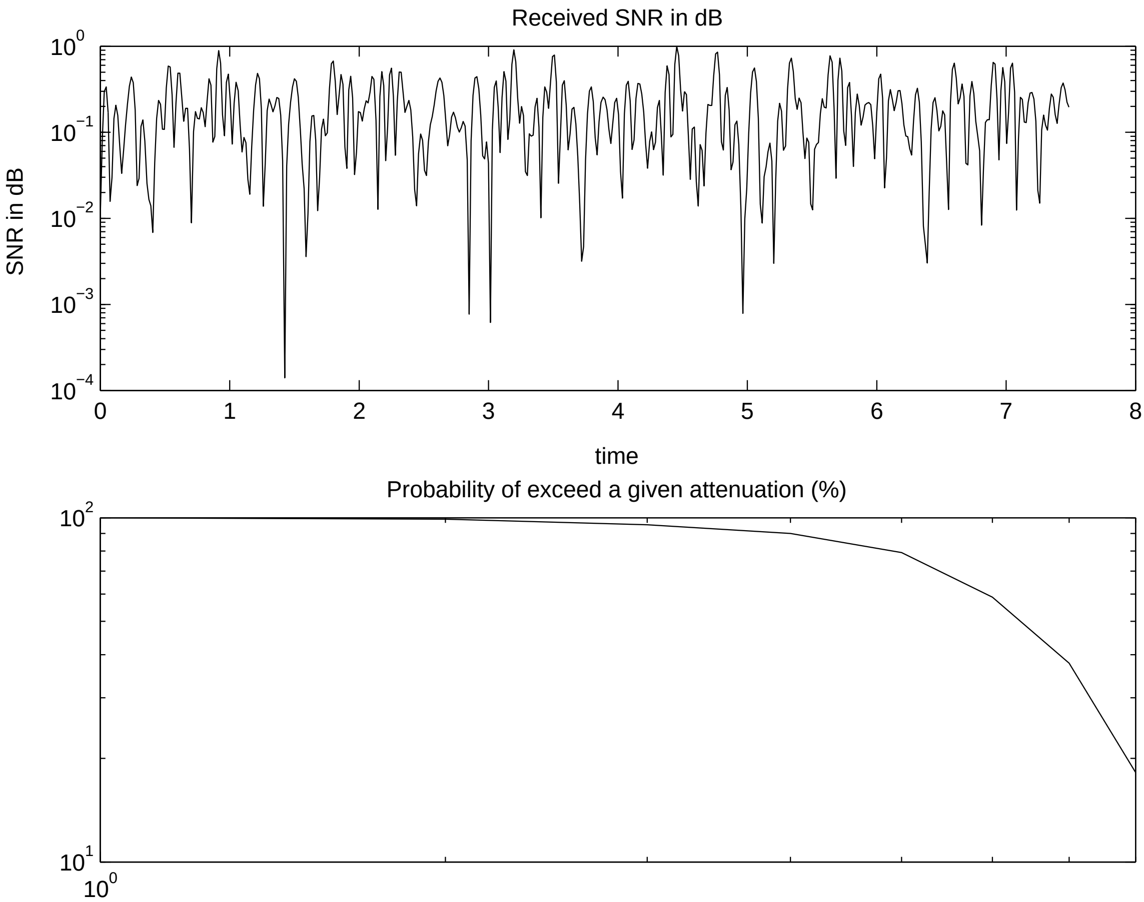

Figure 5.3: Received SNR of Fading Channel

Packet Error Rate (PER) via the following equation:

$$PER(n) = 1 - [1 - BER(n)]^n \tag{5.19}$$

where n is the possible number of bits in an encoded packet including physical layer overhead such as Frame Quality Indicator bits and Turbo Encoder tail allowance bits. The possible values of n as defined by the 3GPP2 [17] are : 216, 408, 792, 1560, 2328, 3096, and 3864 divided in 1, 2, or 4 sub packets depending on the resource allocation scheme selection.

Simulation for SNR is shown in Figure 5.3, the BER, and the PER are shown in Figure 5.4 for the received wireless fading signal was done for the following channel model and fixed parameters: channel model B, as described in Table 5.1, QPSK

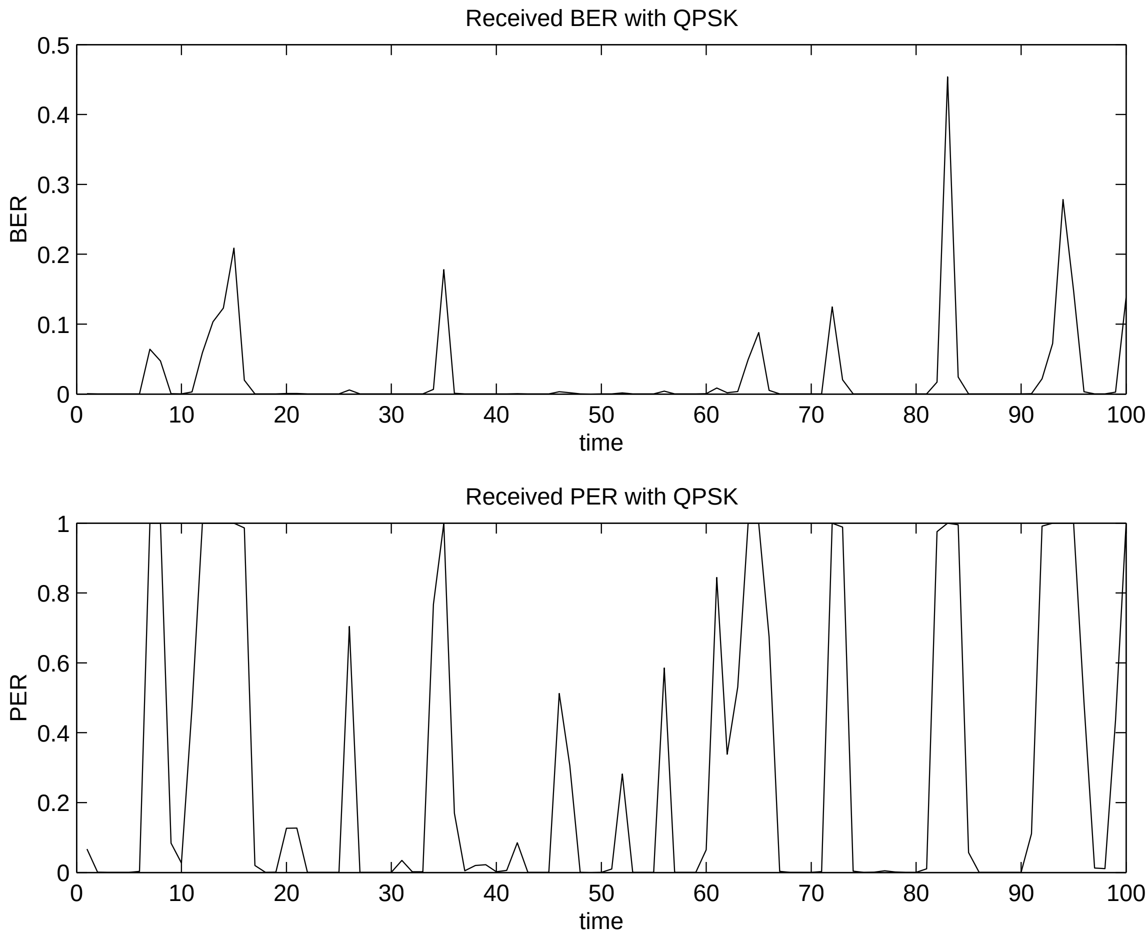

Figure 5.4: BER-PER for QPSK

modulation scheme, a carrier frequency of $900MHz$, sampling frequency $f_s = 8f_d$, a number of samples $N = 500$, a Doppler shift of $f_d = f_c\frac{v}{c}$, and a number of $n = 216$ bits in the encoded packet. The histograms of the Jake's fading envelope , the In-phase, and the Quadrature parts shown in Figure 5.1 have also been simulated using the same fixed set of parameters.

5.3.2 Estimating the Instantaneous Effective Channel Capacity

After investigating the received fading channel gain, we could use it to define one of the main constraints of equation 4.9 which is $X_{S_l,l} \leq r_{n,l}^{S_l}$ where the instantaneous effective capacity of service level S_l with the number of Walsh code channel l can be found using:

$$r_{n,l}^{S_l} = W \log_2 \left(1 + |x_n|^2\right) \tag{5.20}$$

Where $|x_n|$ is the received SNR which is dependent of the Jakes channel model and represented by equation 5.14.

To achieve the convergence of our dynamic resource allocation scheme with respect to the different constraint imposed by the nonstationary fading channel and the limitation on resource parameters, we first need to define an equivalent Additive White Gaussian Noise (AWGN) channel capacity:

$$r_{AWGN}^{S_l,l} = W \log_2 \left(1 + \frac{E_b}{N_0}\right) \tag{5.21}$$

Using $r_{AWGN}^{S_l,l}$ we can eliminate the dependency of the instantaneous effective capacity $r_{n,l}^{S_l}$ to the channel bandwidth parameters W when we substitute it into equation 5.20. With this, we can complete the setting of our simulation by selecting a set of fixed parameters: an average SNR $\frac{E_b}{N_0} = -35dB$, a sampling frequency $f_s = 8f_d$ and AWGN channel capacity $r_{AWGN}^{S_l,l}$ that depends on the type of traffic listed in Table 5.2. We start the simulation for the channel with the set of channel parameters as follows: the speed of the MU $v = 10Km/hour$ with a carrier frequency $f_c = 900Mhz$ and a Doppler shift frequency $f_d = f_c\frac{v}{c} = 30Khz$. Over a sampling period $T_s = \frac{1}{f_s} = 15\ ms$ and a number of samples $N = 1000$. The left portion of Figure 5.6 shows the instantaneous effective channel capacity estimate of three different type of traffic.

5.3.3 Estimating the Service Curve

Using the set of equations 4.23, 4.24, and 4.25 estimating the service curve of a particular queue of infinite length, we could estimate the QoS exponent $\theta(\mu)$ by averaging N samples of T_n the remaining time of packet in service, Q_n the number of bits remaining in the queue to be transmitted, and S_n the sign bit indicating whether a packet is in service. For the purpose of this study simulations, we average $N = 1000$ samples of T_n, Q_n, S_n. Figure 5.5 shows the estimate of the remaining time of packet in service, and the number of bits remaining in the queue to be transmitted for a queue of infinite length.

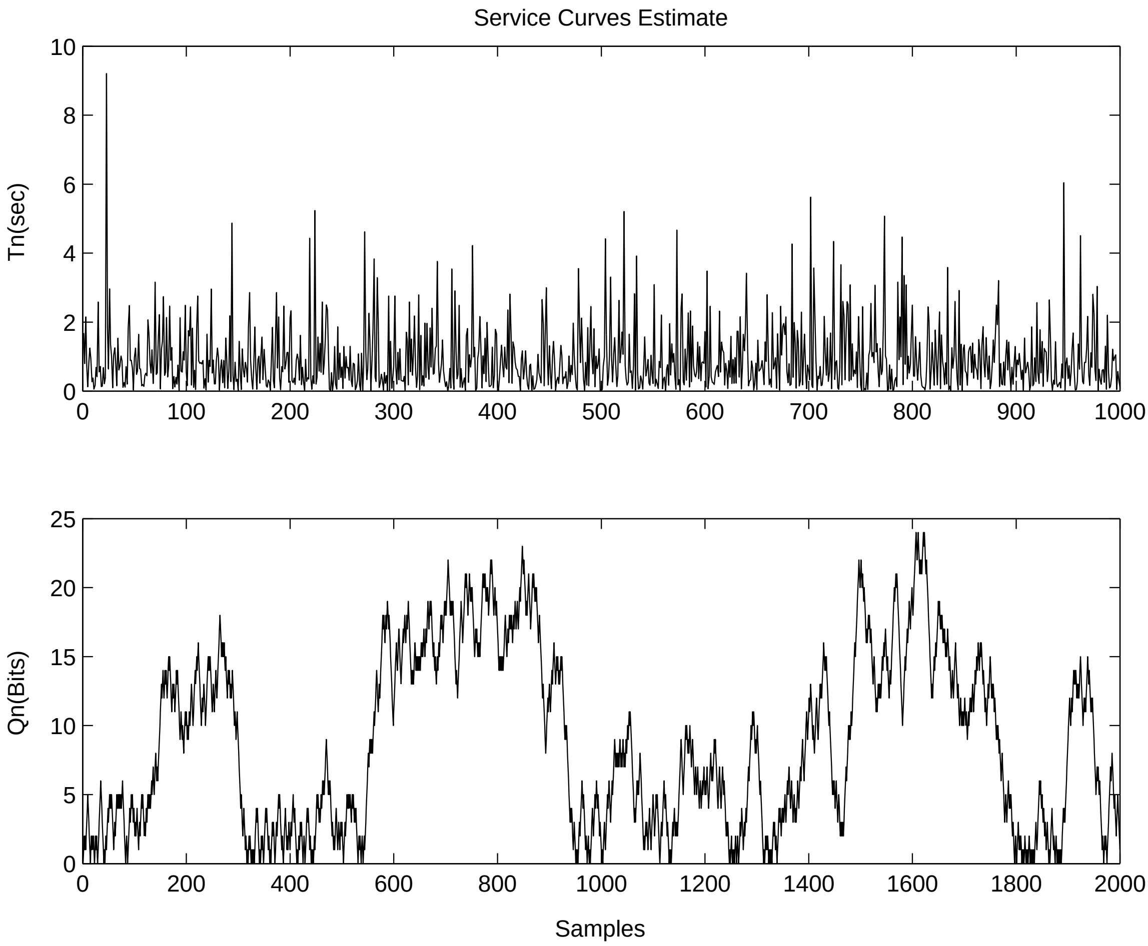

Figure 5.5: Service Curves Estimate

5.3.4 Estimating QoS Exponent

Using the service curve triplet estimate, we can estimate the QoS exponent $\theta(\mu)$ for varying values of data rate μ limited by the effective capacity of the Jakes' channel model. Figure 5.6 shows the CDF plots of the QoS exponent $\theta(\mu)$ estimates and the instant channel capacity for each type of service.

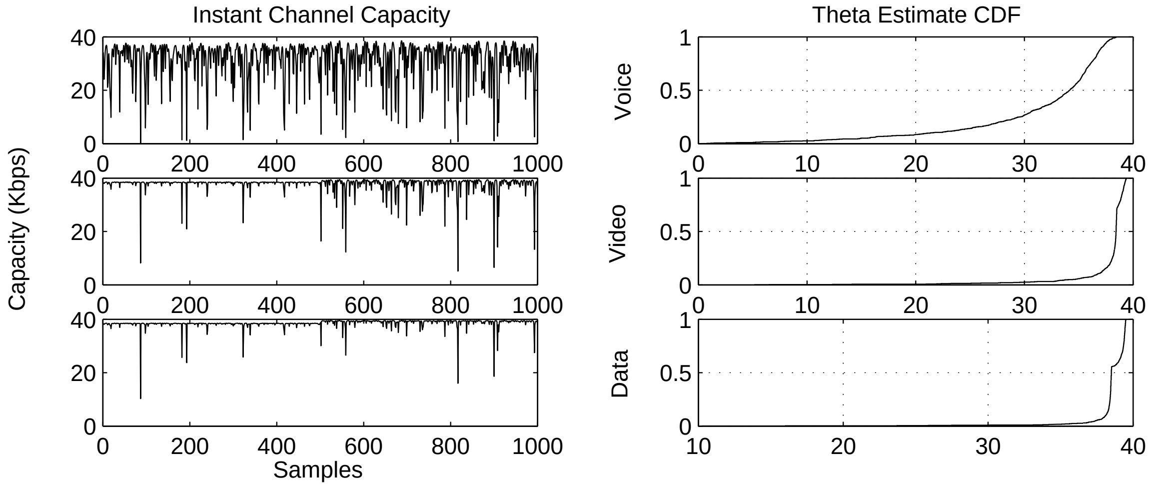

Figure 5.6: Theta and Channel Capacity Estimates

5.3.5 Estimating QoS Guarantees

After estimating the QoS exponents $\theta(\mu)$, we could use it in equation 4.20 to estimate the QoS guarantees for a specific type of user's application with a QoS requirements triplet $\{\mu, D_{max}, \xi\}$. In this simulation, the fixed requirement parameters that are listed in Table 5.2 for the mix of voice, video, and data traffics are considered. Figure A.4 shows the Encode Packet delay for a data traffic using the optimum combination of system resources selected by our dynamic resource allocation scheme. It can be seen from the figure that the delay bound D_{max} was only violated one time which makes for less than 1% data loss probability. The least times the dealy is being violated the more adequate are the MCS levels selected for the specfic type of service and the current channel condition.

5.3.6 Estimating the Throughput

The overall resulting throughput of the optimum resource parameters combination was also simulated without including the data rate when the delay bound D_{max} is being violated. Figure A.1 shows the system throughput variation over 50 sampling iterations with its CDF plot. Figure A.3 shows the system throughput variation using the proposed design verses a system using a fixed set of resource parameters. The overall throughput using dynamic resource allocation is improved over a fixed allocation of resources as shown in the second part of Figure A.3 where the data rate is fixed to 480 Kbps.

CHAPTER 6

Conclusions

This thesis presents a framework for Multimedia Wireless Generation systems. This framework is based on a cross layer design architecture. It will provide a higher data throughput in addition to the capability of supporting new integrated applications with various Quality of Service (QoS) requirements. In addition, such a design will also count for the instantaneous changes of the wireless channel condition. It first made a comprehensive review of the major cellular concepts to build all the knowledge needed for the thesis problem. Then a detailed description of latest 3GPP2 1xEV-DV standards was presented. The current standard accounts for all the possible combinations of the resource parameters. However, given a certain channel quality and application QoS requirements, how to select the optimal combination to satisfy the committed QoS has not been specified in the current standard. Therefore in this thesis, we proposed a new dynamic resource allocation scheme for the 3GPP2 network that is closely integrated with the new 1xEV-DV technologies. The core of the proposed dynamic resource allocation scheme is based on the Effective Capacity concept, where the QoS triplet, in terms of delay, throughput and delay violation probability, has been successfully integrated with the channel quality. Thus, Effective Capacity can be used in the resource allocation to meet the committed QoS. Extensive simulation results (Chapter 5 and Appendix A.1) show that the proposed dynamic resource allocation scheme significantly improves the throughput and delay performance of the various types of application traffic.

CHAPTER 7

Future Work

What have been presented in this thesis can be considered a major starting point of what seems to be a very interesting research area. Some of the possible future work that can extend from this thesis work is as follows:

- The link level simulation is not always adequate in capturing the richness and dynamics of a MWG system with multiple users. Therefore implementing the proposed design on the system level is desirable.
- As the use of one cell may not be a possible wireless system setup environment considering a multi-cell setup with possible inter-cell interference is needed.
- Simulate the F-PDCH using sumlink when different blocks are made possible.
- Study the complexity and possible data interrupts caused by repeated feedback.
- Consider possible inaccurate channel estimate feedbacks that may cause the wrong allocation of system resources

Bibliography

[1] M. Bourouha, I. Abdel-Qader "Cross-Layer Design of a Dynamic Resource Allocation Control for 3GPP2 1xEV-DV Systems", Presented in the IEEE EIT Conference, August, 26-27, 2004, Milwaukee, WI, USA.

[2] W. Kumwilaisak, Y. T. Hou, Q. Zhang, W. Zhu, and Y. Zhang, " A Cross-layer Quality-of-Service Mapping Architecture for Video Delivery in Wireless Networks," IEEE Journal on selected areas in Communications, Vol. 21, No. 10 December 2003.

[3] 3GPP, "Adaptive Modulation and Coding" TSG-RAN WG117, Ad-hoc 24, HS-DPA, Motorola, Sweden, October 2000. http://www.3gpp.org

[4] S. Blake et al, "An Architecture for Differentiated Services," RFC 2475, December 1998.

[5] Motorola, "Technical Overview of 1xEV-DV", Version: G1.4, Arlington Heights, September 2002.

[6] R. Brden, D. Clarck, S. Shenker, "Integrated services in the Internet Architecture: an Overview," Internet RFC 1633, June 1994.

[7] M. B. Srivastava and P. Mishra, " On quality of service in mobile and wireless," in Proc. 7th Int. Workshop Network and Operating Systems Support for Digital Audio and video (NOSSDAV '97), St. Louis, MO, pp. 147-158.

[8] S. Agrawal, I. Acharya, and S. Goel, "Inside 3G Wirless Systems: The 1xEV-DV Technology," TATA Consultancy Services, Technology Review 2003-01, March 2003.

[9] D. Wu and R. Negi, "Effective capacity: A Wireless Link Model for Support of Quality of Service," IEEE Transactions on Wireless Communication, Vol. 2, pp. 630-643, July 2003.

[10] 3GPP2, "Quality of Service: Stage 1 Requirements," S00-20020909-115, S.R0035, Version 1.0, September 2002.

[11] D. Wu and R. Negi, "Utilizing Multiuser Diversity for Efficient Support of Quality of Service over a Fading Channel," http://www.wu.ufl.edu/

[12] R. Srinivasan, J. Baras, "An Analysis of Delay-Constrained Opportunistic Scheduling for Cellular Wireless Systems," Technical Research Report, Center for Satellite and Hybrid Communication Networks TR 2004-2, ISR TR 2004-3, March 2004.

[13] O. Sunay, "1xEV-DV Forward Channel Structure," Lucent Contribution to 3GPP2, C502000-0918-014, September 2000.

[14] N. Joshi, and S. Kabada, "Reverse Link Structure for 1xEV-DV, " C502000-0918-015, Lucent Contribution to 3GPP2 TSG-C WG5, Seoul, Korea, September 2000.

[15] J. Yang, "Statistical Decision Making in Adaptive Modulation and Coding for 3G Wireless Systems," Thesis submitted to EE. Waterloo, Ontario, Canada 2002.

[16] 3GPP2, "Data Services Options for Spread Spectrum Systems: cdma2000 High Speed Packet Data Services Option 33," C.S0017-0.12, Version 5.0, February 2003.

[17] 3GPP2, "Physical Layer Standard for cdma2000 Spread Spectrum Systems," Revision D, C.S0002-D, Version 1.0, February 2004. http://www.3gpp2.org/

[18] 3GPP2, "1xEV-DV Evaluation Methodology," Addendum Version 6, WG5 Evaluation Ad Hoc, July 2001.

[19] 3GPP2, "Medium Access Control (MAC) Standard for cdma2000 Spread Spectrum Systems," Release D, C.S0003-D, February 2004. http://www.3gpp2.org/

[20] V. Huang and W. Zhuang, "Optimal Resource Management in Packet-Switching TDD CDMA Systems," IEEE Communication Magazine, pp. 26-31, Dec. 2000.

[21] O. Gurbuz and H. Owen, "Dynamic Resource Scheduling Schemes for W-CDMA Systems," IEEE Communication Magazine, pp. 80-84, Oct. 2000.

[22] R. Knopp and P. Humblet, "Information Capacity and Power Control in Single-Cell Multiuser Communications," in Proc. IEEE International Conference on Communications (ICC95), Seattle, USA, June 1995.

[23] I. Akyildiz, D. Levine, and I. Joe, "A Slotted CDMA Protocol with BER Scheduling for Wireless Multimedia Networks," IEEE/ACM Transactions on Networking, vol. 7, pp. 146-158, Apr. 1999.

[24] X. Wang, "Wide-Band TD-CDMA MAC With Minimum-Power Allocation and Rate- and BER-Scheduling for Wireless Multimedia Networks," IEEE/ACM Transactions on Networking, vol. 12, pp. 103-116, Feb. 2004.

[25] A. Baiocchi, F. Cuomo, and S. Bolognesi, "IP QoS Delivery in a Broadband Wireless Local Loop: MAC Protocol Definition and Performance Evaluation," IEEE Journal of Selected Areas in Communications, vol. 18, pp. 1608-1622, Sept. 2000.

[26] L. Tong and P. Ramanathan, "Adaptive Power and Rate allocation for Service Curve Assurance in DS-CDMA Network," IEEE Transactions on Wireless Communications, vol. 3, pp. 555-564, Mar. 2004.

[27] S. Jafar and A. Goldsmith, "Adaptive Multirate CDMA for Uplink Throughput Maximization," IEEE Transactions on Wireless Communications, vol. 2, pp. 218-228, Mar. 2004.

[28] J. Evans and D. Everitt, "Efffective Bandwidth-Based Admission Control for Multiservice CDMA Cellular Networks," IEEE Transactions on Vehicular Technology, vol. 48, pp. 36-46, Jan. 1999.

[29] S. Choi and K. Shin, "An Uplink CDMA System Architecture with Diverse QoS Guarantees for Heterogeneous Traffic," IEEE/ACM Transactions on Networking, vol. 7, pp. 616-628, Oct. 1999.

[30] J. Huang, R. Yao, Y. Bai, and S. Wang, "Performance of a Mixed-Traffic CDMA2000 Wireless Network With Scalable Streaming Video," IEEE Transactions on Circuits and Systems for Video Technology, vol. 13, pp. 973-981, Oct. 2003.

[31] S. Ozer and S. Papavassiliou, "Performance Analysis of CDMA Systems With Integrated Services," IEEE Transactions on Vehicular Technology, vol. 52, pp. 823-836, July 2003.

[32] C. Comaniciu, N. Mandayam, D. Famolari, and P. Agrawal, "IEEE Transactions on Wireless Communications," IEEE Transactions on Wireless Communications, vol. 2, pp. 472-483, May 2003.

[33] E. Damosso "COST 231 Final Report," Digital mobile radio towards future generations systems, European Comission, Bruxelles, 1999.

[34] M. Guizani “Computer Networks and Data Communications with the Internet” chapter ”Wireless Networks” to be published by John Wiley Publishers 2004.

[35] C. Xiao “Rayleigh Channel Fading Simulator: Problems and Solutions,” Dept of Electrical and computer Eng.,University of Missouri, Columbia, USA.

Appendix A

Results and Parameters

A.1 Simulation Results

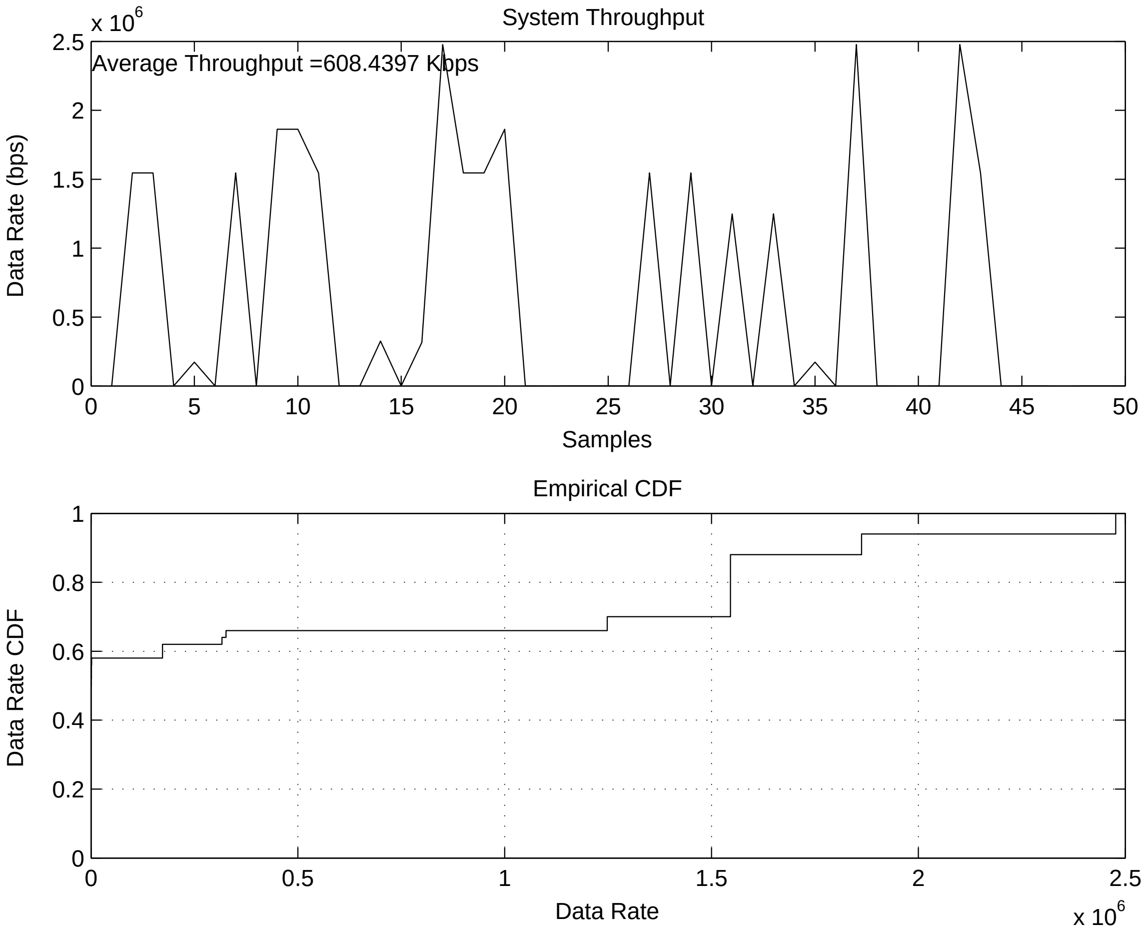

Figure A.1: System Throughput Dynamic

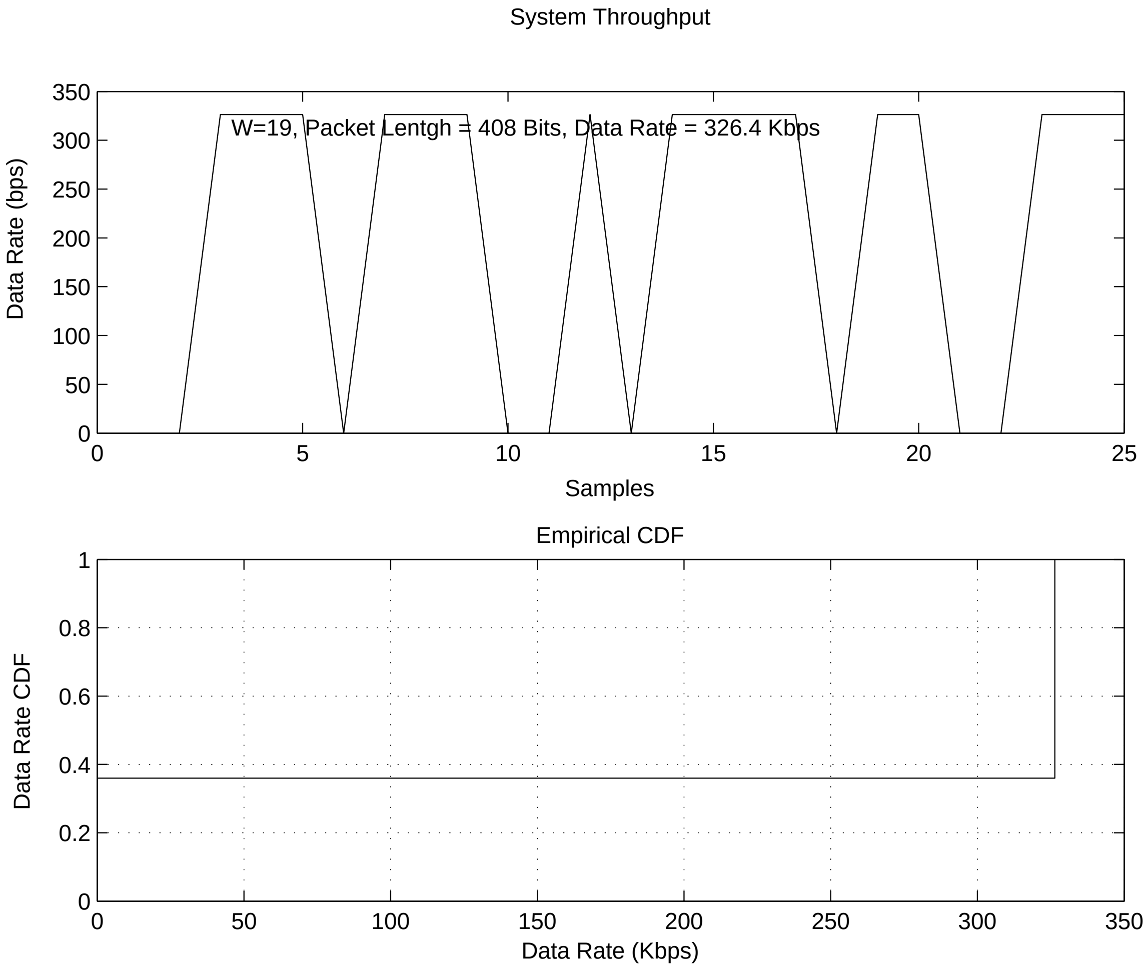

Figure A.2: Non-Dynamic System Throughput

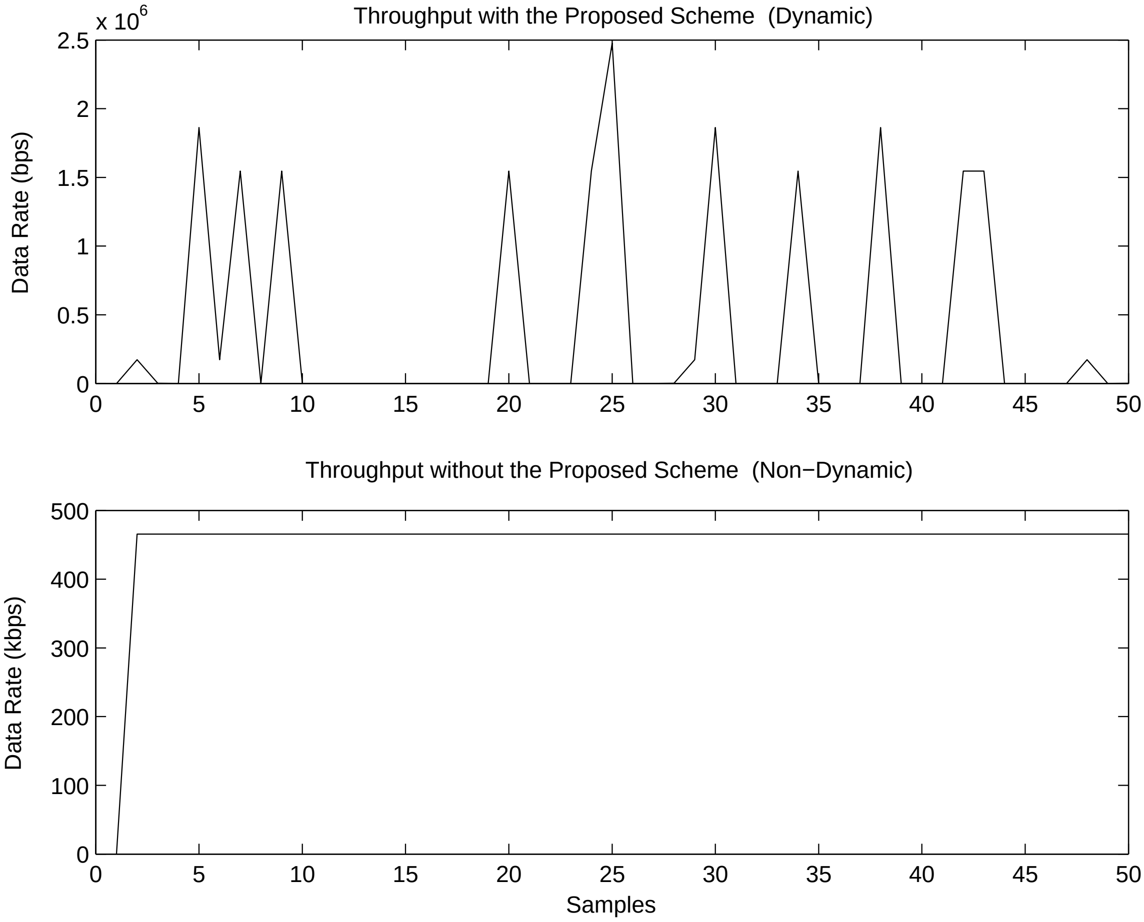

Figure A.3: Dynamic vs Non-Dynamic System Throughput

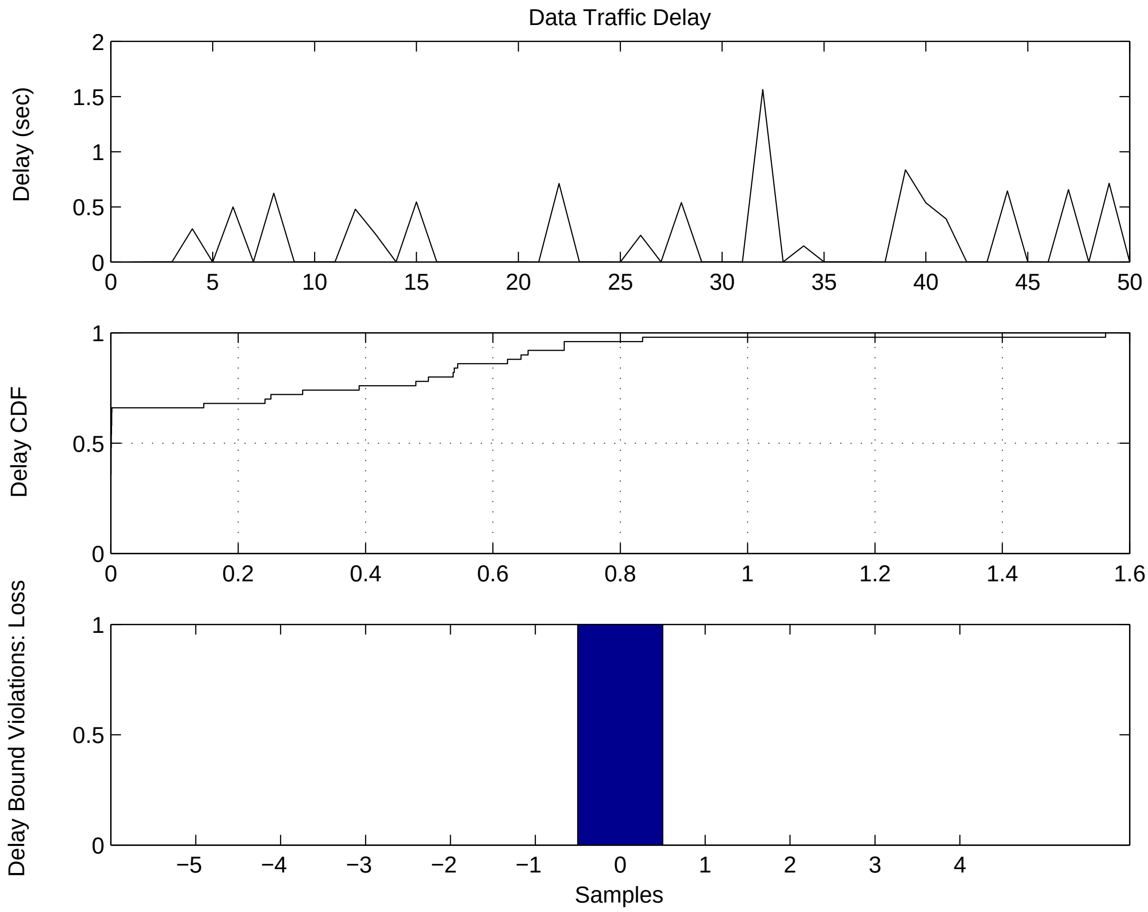

Figure A.4: Data Traffic Delay

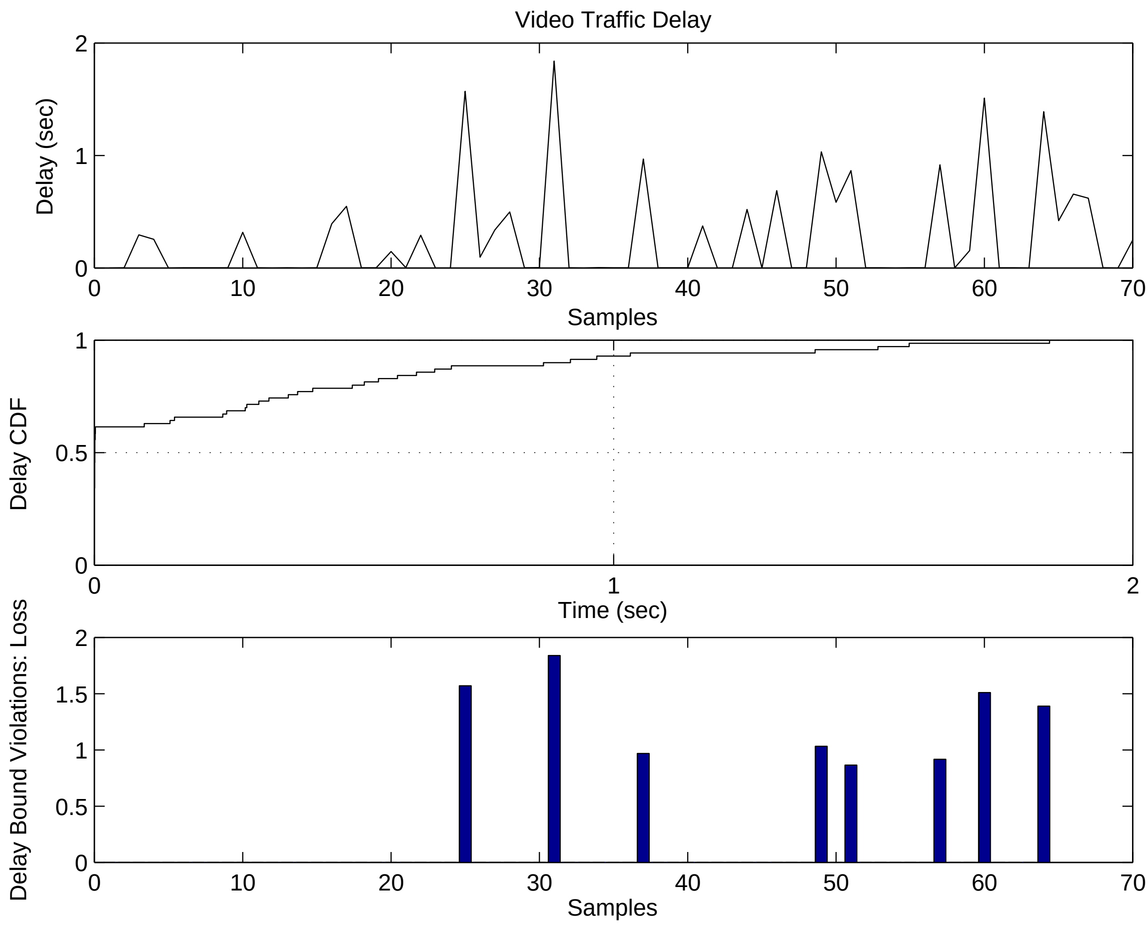

Figure A.5: Video Traffic Delay

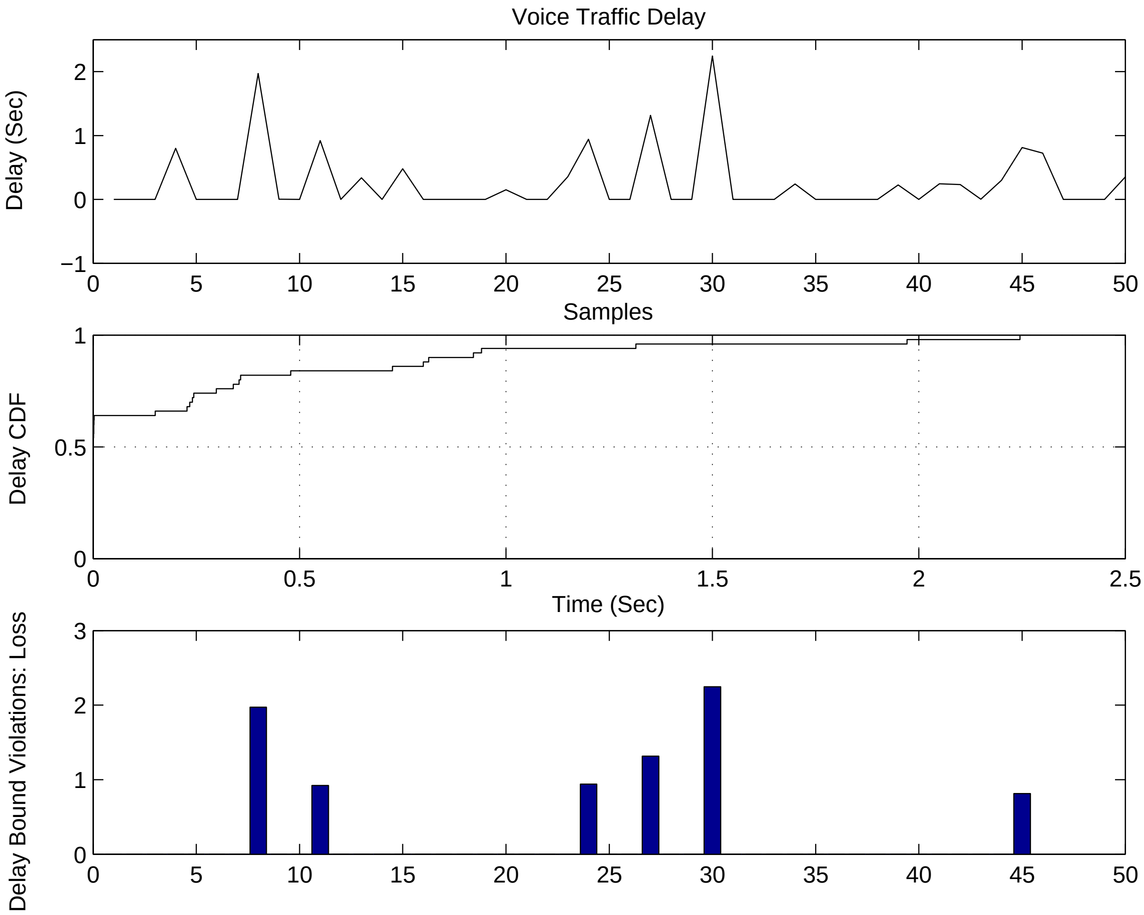

Figure A.6: Voice Traffic Delay

A.2 System Parameters Table

N_p	$N_{w,k}$	μ_k	$N_{s,k}$	M_k	$R^c_{effective,k}$		L_k
2,328	28	1,862.4	1	3	0.5774	48	4,032
3,864	27	1,545.6	2	2	0.7454	96	5,184
3,096	26	2,476.8	1	4	0.6202	48	4,992
3,864	26	3,091.2	1	4	0.7740	48	4,992
1,560	25	1,248.0	1	2	0.6500	48	2,400
2,328	25	1,862.4	1	3	0.6467	48	3,600
3,096	25	1,238.4	2	2	0.6450	96	4,800
3,864	25	1,545.6	2	3	0.5367	96	7,200
2,328	23	931.2	2	2	0.5272	96	4,416
2,328	23	1,862.4	1	4	0.5272	48	4,416
3,096	23	2,476.8	1	4	0.7011	48	4,416
3,864	23	1,545.6	2	3	0.5833	96	6,624
1,560	22	1,248.0	1	2	0.7386	48	2,112
3,096	22	1,238.4	2	2	0.7330	96	4,224
1,560	21	1,248.0	1	3	0.5159	48	3,024
3,096	21	1,238.4	2	3	0.5119	96	6,048
3,096	21	2,476.8	1	4	0.7679	48	4,032
3,864	21	1,545.6	2	3	0.6389	96	6,048
1,560	20	624.0	2	2	0.4063	96	3,840
2,328	20	465.6	4	2	0.3031	192	7,680
2,328	20	931.2	2	2	0.6063	96	3,840
2,328	20	1,862.4	1	4	0.6063	48	3,840
3,096	20	619.2	4	2	0.4031	192	7,680
408	19	326.4	1	2	0.2237	48	1,824
792	19	316.8	2	2	0.2171	96	3,648
792	19	633.6	1	2	0.4342	48	1,824
1,560	19	1,248.0	1	3	0.5702	48	2,736
3,096	19	1,238.4	2	3	0.5658	96	5,472

System Parameters Table-Continued

N_p	$N_{w,k}$	μ_k	$N_{s,k}$	M_k	$R^c_{effective,k}$		L_k
3,864	19	772.8	4	2	0.5296	192	7,296
3,864	19	1,545.6	2	4	0.5296	96	7,296
2,328	18	1,862.4	1	4	0.6736	48	3,456
1,560	17	1,248.0	1	3	0.6373	48	2,448
2,328	17	931.2	2	2	0.7132	96	3,264
3,096	17	1,238.4	2	3	0.6324	96	4,896
3,864	17	1,545.6	2	4	0.5919	96	6,528
2,328	16	1,862.4	1	4	0.7578	48	3,072
3,096	16	619.2	4	2	0.5039	192	6,144
3,864	16	772.8	4	2	0.6289	192	6,144
792	15	633.6	1	2	0.5500	48	1,440
1,560	15	624.0	2	2	0.5417	96	2,880
1,560	15	1,248.0	1	4	0.5417	48	2,880
2,328	15	931.2	2	3	0.5389	96	4,320
3,096	15	1,238.4	2	4	0.5375	96	5,760
3,864	15	1,545.6	2	4	0.6708	96	5,760
1,560	14	312.0	4	2	0.2902	192	5,376
2,328	14	465.6	4	2	0.4330	192	5,376
3,864	14	772.8	4	2	0.7188	192	5,376
3,864	14	1,545.6	2	4	0.7188	96	5,376
792	13	633.6	1	2	0.6346	48	1,248
1,560	13	624.0	2	2	0.6250	96	2,496
1,560	13	1,248.0	1	4	0.6250	48	2,496
2,328	13	931.2	2	3	0.6218	96	3,744
3,096	13	619.2	4	2	0.6202	192	4,992
3,096	13	1,238.4	2	4	0.6202	96	4,992
3,864	13	1,545.6	2	4	0.7740	96	4,992
1,560	12	1,248.0	1	4	0.6771	48	2,304

System Parameters Table-Continued

N_p	$N_{w,k}$	μ_k	$N_{s,k}$	M_k	$R^c_{effective,k}$		L_k
3,096	12	1,238.4	2	4	0.6719	96	4,608
3,864	12	772.8	4	3	0.5590	192	6,912
408	11	326.4	1	2	0.3864	48	1,056
792	11	158.4	4	2	0.1875	192	4,224
792	11	316.8	2	2	0.3750	96	2,112
792	11	633.6	1	2	0.7500	48	1,056
1,560	11	624.0	2	2	0.7386	96	2,112
1,560	11	1,248.0	1	4	0.7386	48	2,112
2,328	11	465.6	4	2	0.5511	192	4,224
2,328	11	931.2	2	4	0.5511	96	4,224
3,096	11	619.2	4	2	0.7330	192	4,224
3,096	11	1,238.4	2	4	0.7330	96	4,224
3,864	11	772.8	4	3	0.6098	192	6,336
792	10	633.6	1	3	0.5500	48	1,440
1,560	10	624.0	2	3	0.5417	96	2,880
2,328	10	931.2	2	4	0.6063	96	3,840
3,096	10	619.2	4	3	0.5375	192	5,760
792	9	633.6	1	3	0.6111	48	1,296
1,560	9	312.0	4	2	0.4514	192	3,456
1,560	9	624.0	2	3	0.6019	96	2,592
2,328	9	465.6	4	2	0.6736	192	3,456
2,328	9	931.2	2	4	0.6736	96	3,456
3,096	9	619.2	4	3	0.5972	192	5,184
3,864	9	772.8	4	4	0.5590	192	6,912
216	8	172.8	1	2	0.2813	48	768
408	8	163.2	2	2	0.2656	96	1,536
408	8	326.4	1	2	0.5313	48	768
792	8	316.8	2	2	0.5156	96	1,536

System Parameters Table-Continued

N_p	$N_{w,k}$	μ_k	$N_{s,k}$	M_k	$R^c_{effective,k}$		L_k
792	8	633.6	1	4	0.5156	48	1,536
1,560	8	624.0	2	4	0.5078	96	3,072
2,328	8	465.6	4	3	0.5052	192	4,608
2,328	8	931.2	2	4	0.7578	96	3,072
3,096	8	619.2	4	4	0.5039	192	6,144
3,864	8	772.8	4	4	0.6289	192	6,144
408	7	326.4	1	2	0.6071	48	672
792	7	316.8	2	2	0.5893	96	1,344
792	7	633.6	1	4	0.5893	48	1,344
1,560	7	312.0	4	2	0.5804	192	2,688
1,560	7	624.0	2	4	0.5804	96	2,688
2,328	7	465.6	4	3	0.5774	192	4,032
3,096	7	619.2	4	4	0.5759	192	5,376
3,864	7	772.8	4	4	0.7188	192	5,376
408	6	326.4	1	2	0.7083	48	576
792	6	158.4	4	2	0.3438	192	2,304
792	6	316.8	2	2	0.6875	96	1,152
792	6	633.6	1	4	0.6875	48	1,152
1,560	6	312.0	4	2	0.6771	192	2,304
1,560	6	624.0	2	4	0.6771	96	2,304
2,328	6	465.6	4	4	0.5052	192	4,608
3,096	6	619.2	4	4	0.6719	192	4,608
216	5	172.8	1	2	0.4500	48	480
408	5	163.2	2	2	0.4250	96	960
408	5	326.4	1	3	0.5667	48	720
792	5	316.8	2	3	0.5500	96	1,440
1,560	5	312.0	4	3	0.5417	192	2,880

System Parameters Table-Continued

N_p	$N_{w,k}$	μ_k	$N_{s,k}$	M_k	$R^c_{effective,k}$		L_k
2,328	5	465.6	4	4	0.6063	192	3,840
216	4	86.4	2	2	0.2813	96	768
216	4	172.8	1	2	0.5625	48	384
408	4	81.6	4	2	0.2656	192	1,536
408	4	163.2	2	2	0.5313	96	768
408	4	326.4	1	4	0.5313	48	768
792	4	158.4	4	2	0.5156	192	1,536
792	4	316.8	2	4	0.5156	96	1,536
1,560	4	312.0	4	4	0.5078	192	3,072
2,328	4	465.6	4	4	0.7578	192	3,072
216	3	86.4	2	2	0.3750	96	576
216	3	172.8	1	2	0.7500	48	288
408	3	81.6	4	2	0.3542	192	1,152
408	3	163.2	2	2	0.7083	96	576
408	3	326.4	1	4	0.7083	48	576
792	3	158.4	4	2	0.6875	192	1,152
792	3	316.8	2	4	0.6875	96	1,152
1,560	3	312.0	4	4	0.6771	192	2,304
216	2	43.2	4	2	0.2813	192	768
216	2	86.4	2	2	0.5625	96	384
216	2	172.8	1	4	0.5625	48	384
408	2	81.6	4	2	0.5313	192	768
408	2	163.2	2	4	0.5313	96	768
792	2	158.4	4	4	0.5156	192	1,536
216	1	43.2	4	2	0.5625	192	384
216	1	86.4	2	4	0.5625	96	384
408	1	81.6	4	4	0.5313	192	768

Table A.1: F-PDCH Parameters with Radio Configuration 10