Channel Allocation Method in CDMA Networks using Transmission Control Protocol

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Abstract: As the Internet grows both in terms of users and available applications, one of the challenging issues is providing a fair and efficient allocation of the available network resources. The Transport Control Protocol (TCP), is the default transport layer protocol used on the Internet to provide reliable end-to-end communications. Modern digital communication technologies now allow wireless channel schedulers to rapidly change the allocated channel resources in response to channel conditions as well as user demands. This is achieved by changing parameters such as the coding rate, spreading factor, modulation scheme and link-layer retransmission rate. In current code division multiple access (CDMA) networks the high rate channels are achieved by reducing the spreading factor of the orthogonal Walsh codes. This essentially implies that only a few users can simultaneously share the high-rate channels at any time. Specifically, we propose a wireless channel scheduler that allocates different channel rates from a set of optimized rates to TCP sessions in response to their sending rates. In the context of a single TCP session, we develop a model to compute its long-term throughput under such a scheduler and use it for joint optimization of control variables to compute the set of optimal channel rates that are used by the scheduler. Therefore, the goal of this paper is to increase the TCP sending rate allocated in channel resources by viewing their performance and speed rate.

Keywords: channel allocation scheme, wireless communication networks, transmission control protocol

1. Introduction

Lack of an adaptive or optimized way of channel allocation method, may cause high usage of bandwidth as well as power consumption for transmitting packets. Most of the existing system used to find a channel using buffer backlog system (identify the channel rate by using the number of packets stored in the buffer within the unit of time), it may cause the packet loss due to congestion, if not get a channel with proper bandwidth.

First generation wireless communication networks were configured to operate at one specified channel rate. Schedulers were not required in the first generation wireless communication networks because only one channel rate was available. Advances in wireless communication technology were introduced by third generation wireless communication networks that provide multiple rates and scheduling policies for dynamically transitioning between the multiple rates. This is accomplished by substituting specifications such as the coding rate, spreading factor, modulation scheme, and link-layer retransmission rate[1]. Typically, the scheduling policies for the third generation wireless communication networks select a high channel rate to quickly clear buffer backlogs. Essentially, this policy attempts to optimize output to a user by always utilizing the fastest available channel rate. A scheduling policy that always aims to clear buffer backlog can be sub-optimal for TCP because the TCP sender may not effectively utilize the high channel rate. Additionally, the high channel rate may introduce more data errors than a low channel rate. The challenge of improving service quality in wireless communication networks is a never-ending research project that necessitates the development of effective channel allocation schemes.[2]

The issue is how to make the most of limited channel capital in order to ensure smooth contact while maintaining high service quality.

Transmission Control Protocol is the dominant protocol over both wired and wireless links. However, due to the loss of packet transmission and noise breakage in TCP we propose several solutions to rectify this shortcoming. A simple two state TCP-aware channel scheduler is proposed which enables wireless channel schedulers in cellular networks to rapidly change the allocated channel resources. This is achieved by changing parameters such as the coding rate, spreading factor, modulation scheme and link-layer transmission rate.

Section 2 gives the Literature Review, Section 3 gives the proposed algorithm, Section 4 elaborates the experimental results and the conclusion and future enhancement of this paper are made in Section 5.

2. Literature Survey

Mobile devices (laptops, smartphones, and tablets) have become more popular, and applications (web, e-mail, video on demand, and social networks) have become more popular over the cellular network, using more frequency spectrum resources[3]. Transmission Control Protocol is the dominant protocol over both wired and wireless links. However, due to loss of packet transmission and noise breakage in TCP we propose several solutions to rectify this shortcoming. A simple two state TCP-aware channel scheduler is proposed which enables wireless channel schedulers in cellular networks to rapidly change the allocated channel resources. This is achieved by changing parameters such as the coding rate, spreading factor, modulation scheme and link-layer transmission rate.

The major advantage of this project is to maximize the throughput of TCP connections through careful selection of the channel rates and a mechanism to adaptively allocate different channel rates in response to TCP state. TCP uses an additive increase multiplicative decrease (AIMD) algorithm that gradually increases its transmission rate based on receiver feedback and rapidly throttles back when it perceives losses.

Specifically, a wireless channel scheduler is proposed that allocates different channel rates from a set of optimized rates to TCP sessions in response to their sending rates. In the case of a single TCP session, a model is developed to compute its long-term throughput and use it for joint optimization of control variables. The rate allocation framework is extended to incorporate the presence of multiple TCP sessions.

These present problems were solved by providing a system and a method for dynamically allocating channel resources based on a current sending rate. It calculates the TCP throughput in an adapt wireless channel, improving throughput associated with communication session.

Optimum Bandwidth for CDMA was adapted from a presentation by Klein S .Gilhousen at the International Conference on Personal, Mobile Radio, and Spread Spectrum Communications. The effects of multipath propagation on the CDMA signal are quite different from those of narrowband signals. While increasing spreading bandwidth leads to an asymptotic improvement in Erlang capacity of CDMA per megahertz, multipath propagation phenomena have increasingly detrimental effects. The direct sequence modulation and Rake receiver do mitigate the multipath that leads to narrowband Rayleigh fading. However the complexity of the Rake increases with bandwidth.

[4] K. Mattar, A. Sridharan, H. Zang, I. Matta and A. Bestavros, “TCP over CDMA2000 Networks: A Cross-Layer Measurement Study”, Proc. Eight Int’l Conf. Passive and Active Network Measurement (PAM ’07), Apr. 2007.

[5] M. Ghaderi, A. Sridharan, H. Zang, D. Towsley, and R. Cruz,“Modeling TCP in a Multi-Rate Multi-User CDMA System,” Proc. Networking, May 2007.

From the above mentioned papers we derive the method to calculate the coding rates and gain percentage. The formulas and the analysis part are studied and derived from these papers.

[6] Modeling TCP in a Multi-rate Multi-user CDMA System by Majid Ghaderi, Ashwin Sridharan, Hui Zang, Don Towsley, and Rene Cruz, University of Massachusetts Amherst, Sprint Advanced Technology Labs University of California San Diego

The above mentioned paper is referred to identify the channels through which the channel rates are being passed. Depending on the speed of sending rates it decides through which channel it has to pass through. This paper also derives the equation to calculate the coding rate values according to the packets that are passed through the system channels and also the gain percentage.

[7]Telecommunications Industry Assoc, TIA EIA IS-2000, www.tiaonline.org/standards/sfg/imt2k/cdma2000/, ar.2000.

From the above mentioned paper we can refer the impact of CDMA on TCP by taking the present CDMA2000 as an example. The paper also specifies about the single user scheduler and how the window size and packet size of a coding rates are calculated.

3. Proposed System

3.1 System Overview

This section describes the overall view of a proposed system and its subsystem. The different subsystem gives their unique functionality which when put together provides an effective, user-friendly and reliable system which fulfills the objectives. Fig. 2.1 is a network environment diagram of a proposed system. The below figure, shows an overview of the system

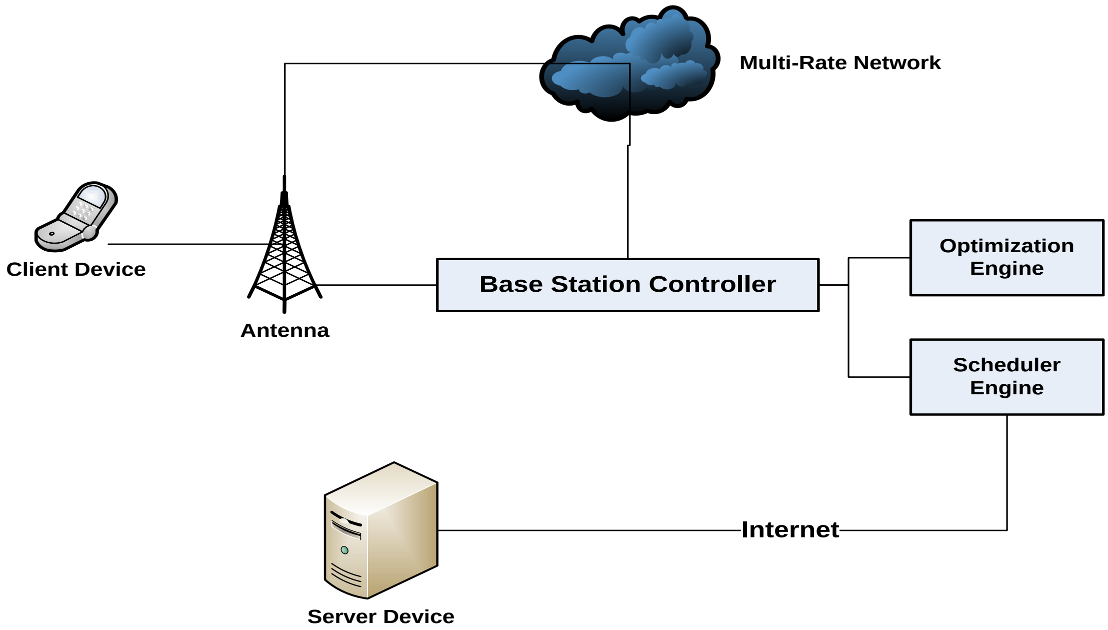

Fig 2.1: A network environment for the proposed system

In Fig 2.1, the wireless operating environment includes a base station controller, an antenna, a client device, a multi-rate network, a server device and the Internet. The multi-rate network interconnects the antenna, the base station controller and the client device. The Internet enables the base station controller to communicate with the server device. The multi-rate network can switch between a several channel rates. In certain situations, selections of control parameters alter the channel rate of the multi-rate network.

The base station controller (BSC) facilitates communication between mobile communication sessions between the client device and the server device. Additionally, the BSC may include an optimization engine and scheduler engine. The optimization engine generates a subset of the available channel rates provided by the multi-rate network that maximizes TCP throughput for mobile communication sessions. The scheduler engine selects one or more channel rates to form the subset of the available channel rates based on a TCP sending rate associated with the sending device or receiving device.

The antenna may be communicatively connected to the BSC to enable communication between the client device multi-rate networks. The antenna receives and sends wireless signals associated with communication sessions managed by the BSC.

The server device may include a database, web server or any other server device that provides file transfer or query services in response to a request. The server device is communicatively connected to the Internet, which provides access to the BSC. The server device may establish a communication session with the client device via the BSC.

The client device may include a mobile phone, personal digital assistant or any other mobile communication device. The client device is communicatively connected to the multi-rate network and may establish a communication session with the server device. The client device may include a wireless communication interface that connects the client device to the multi-rate network.

The TCP sending rate is collected for a specified period to calculate a long-term TCP sending rate. The optimization engine utilizes the long-term TCP sending rate information to generate a subset of the channel rates from available channel rates associated with the multi-rate network.

3.2 Performance

The developed system is designed to be full-featured, efficient and simple, supporting the current generation for transmitting packets. The performance of the developed software depends on system memory and system performance. The system may also perform efficiently based on the varying speed and type of internet access to which the client computer has access.

The below mentioned are approximate values:

- The simulation should clearly describe how packets are being transmitted through different channels.
- The client and server configured system should be safe.

Efficient use of memory can be done for the storage of packets.

- The response time, throughput and capacity depends on the speed and efficiency of the network.
- Packet size depends upon the system configuration and other parallel applications running on the system.

4. System Architecture

The following section describes the architecture of the system with the help of data flow diagrams of each level.

4.1. Data Flow Diagram

The data flow diagram is a graphical representation of the flow of data through an information system. It can also be used for the visualization of data processing**.** With a data flow diagram users can visualize how the system will operate, what the system will accomplish and how the system will be implemented.

A data flow diagram is concerned with designing a sequence of final transformation that converts system input into the required output. These diagrams illustrate how data flows through a system and how the output is derived from the input through a sequence of functional transformation. A data flow diagram can also be used for the visualization of data processing (structured design). It is common practice for a designer to draw a context-level DFD first which shows the interaction between the system and outside entities.

DFD Level - 0

Figure.4.1 shows the context diagram of the whole project. This figure shows the working of the project at the outermost level. Thus providing a basic description of the project. Here sender sends the packets through controller; controller sends the packets to the receiver based on any one of the methods at a time either by direct method or by an algorithm. We are using two types of channels namely fundamental, supplemental. If packet sending speed is low packet passes through fundamental. If packet sending speed is high, the packet passes through supplemental. If we choose the direct method, packet directly passes through a channel. If we choose an algorithm, there are three scenarios:

- Initially channel will be fundamental even though sending speed is low.
- If fundamental kept busy the packet passes through supplemental. Even if the rate is low.
- If the channel is denied a request to send the packet is blocked.

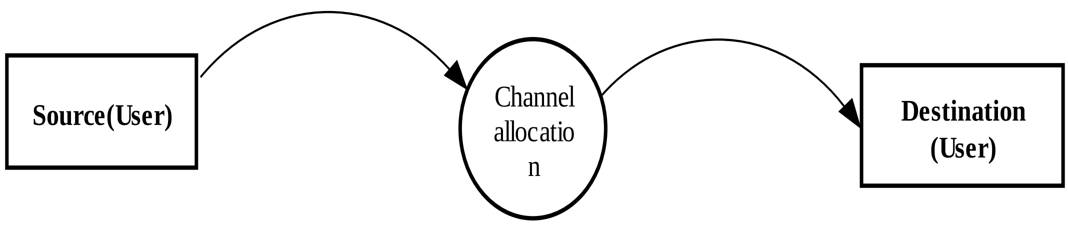

Fig 4.1: Level 0 DFD

DFD Level – 1

The level 1 DFD describe what happens in the level second in the outermost. The client sends a packet to the controller which in turn sends to the server either directly or by an algorithm. The receiver then stops the packet transmission.

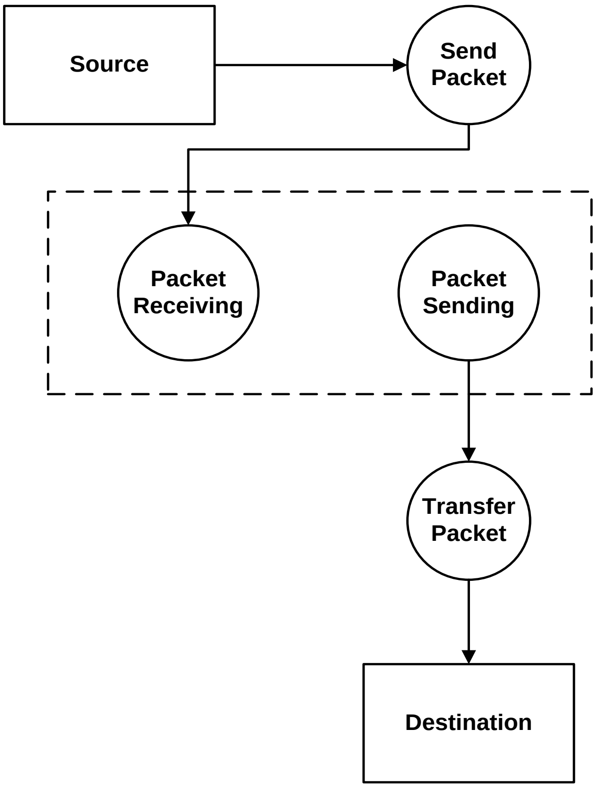

Fig 4.2: DFD Level 1

DFD Level-2

Level 2 DFD shows the details of the steps involved in packet transmission. Level 2 description is as follows:

- Firstly the sender sends a packet in which the sending speed rate will be specified.
- The packet sent by the sender is captured by the controller.
- The controller works in a parallel situation.
- The sending rate of the captured packet is calculated together with the controller temporarily stores the captured packet in the buffer if there is an excess of packets being transmitted.
- These packets are then opted to send either by sending directly through the system channel or by a scheduling algorithm.
- The packet to be transmitted through an algorithm specifies which channel it has to be passed depending on the spending speed rate of the system.
- The packet is then transferred from controller to the receiver which receives the packet and stores it in the destination.

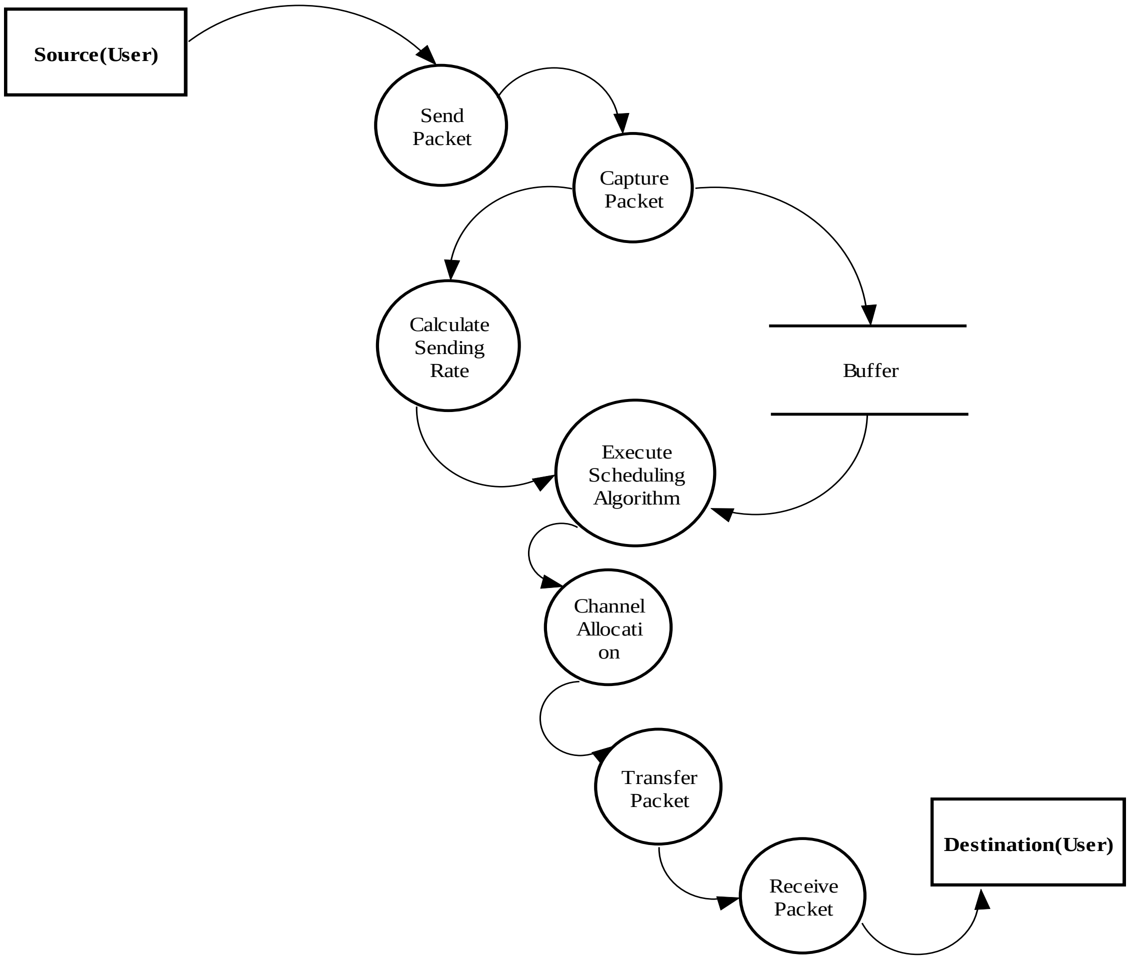

Fig 4.3: DFD Level 2

4.2 Structured Chart Diagram

Structured Chart Diagram shows the control flow in the system. The control flow of this system is shown in Figure 4.1. The calling sequence is from top to down and left to right. The execution of the mechanism follows these steps:

- On execution of the system, a packet sending page is first displayed which is necessary for the service provider to send a packet through this much milliseconds.
- After the packet sending step, the service provider has to select the option whether it is sent by direct or by an algorithm.
- Depending upon the choice, the packet is received and the gain of the coding rate is calculated.

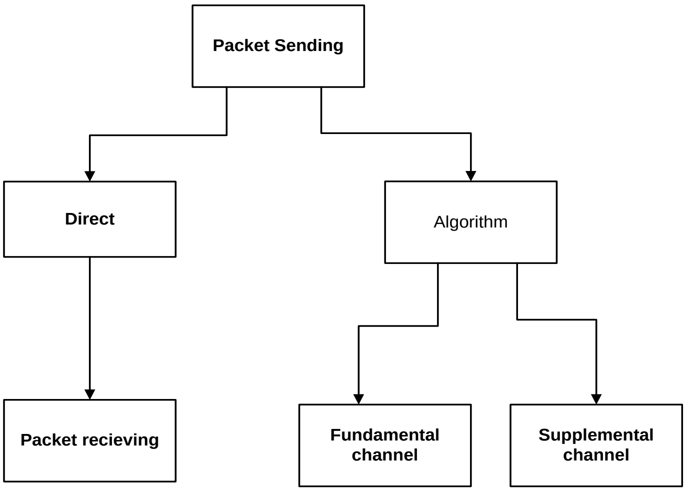

Fig 4.2.1 Structured Chart Diagram

4.3 Flowcharts of the Project

The following section describes the flow various steps taking place in the implementation of the system in the form of a flow chart.

The sequence of steps carried out in this system is depicted by the following flow chart:

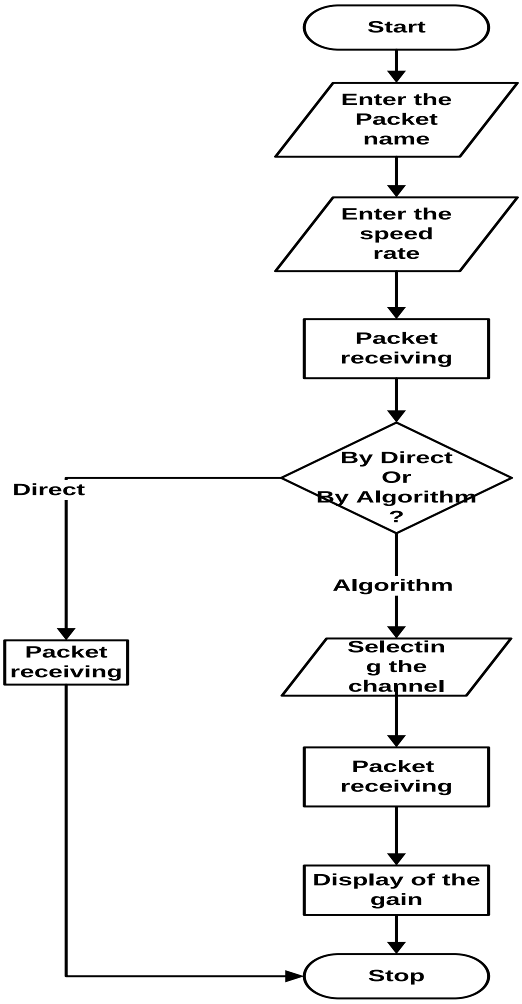

Fig 4.3.1 Flow chart of the proposed System

The implementation begins with a service provider sending a packet. The service provider has to select an option of speeding rate to select the channel through which the packet is being sent. If the service provider selects the direct method, the packet is received directly in the receiving packet and the system calculates its coding rate. If the service provider selects the algorithm method, it gives an option which channels can be kept busy and according to the packet size the two coding rate and gain are calculated.

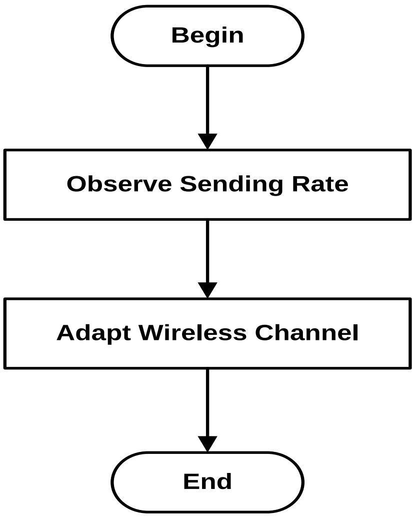

Fig 4.3.2 illustrates a method to adapt wireless channels

The above fig 4.3.2, a method is initiated, when a communication session is triggered. The base station controller observes sending rate associated with a sending device. At adapt wireless channel, the base station controller adapts the allocated wireless channel rates based on a variance observed in the sending rates. Later the method terminates.

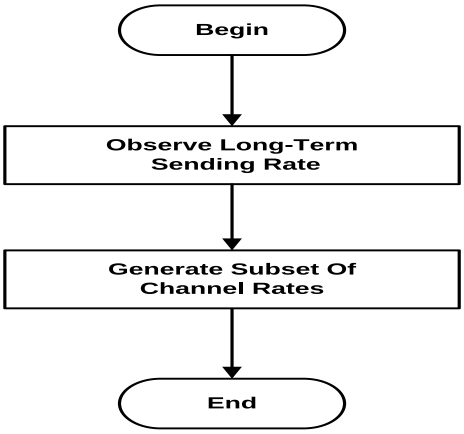

Fig 4.3.3 illustrates a method to generate subset of wireless channels that maximizes throughput

In Fig 4.3.3, a method is imitated, at the begin step, when the base station controller is initialized and a communication session was previously initiated. The base station controller observes the long-term sending rate associated with the sending device. In certain embodiments, the long term sending rate is calculated from propagation delays and window sizes collected over a period of time. Inturn, a subset of available channel rates that maximize throughput is generated based on the long-term sending rate. Later the method is terminated.

5. Results

This section shows the result obtained on running the system. It briefly discusses the user interfaces with the help of screenshots of software.

Since its client/server architecture we can run the program in two systems. The figure shown below is the packet send by the server

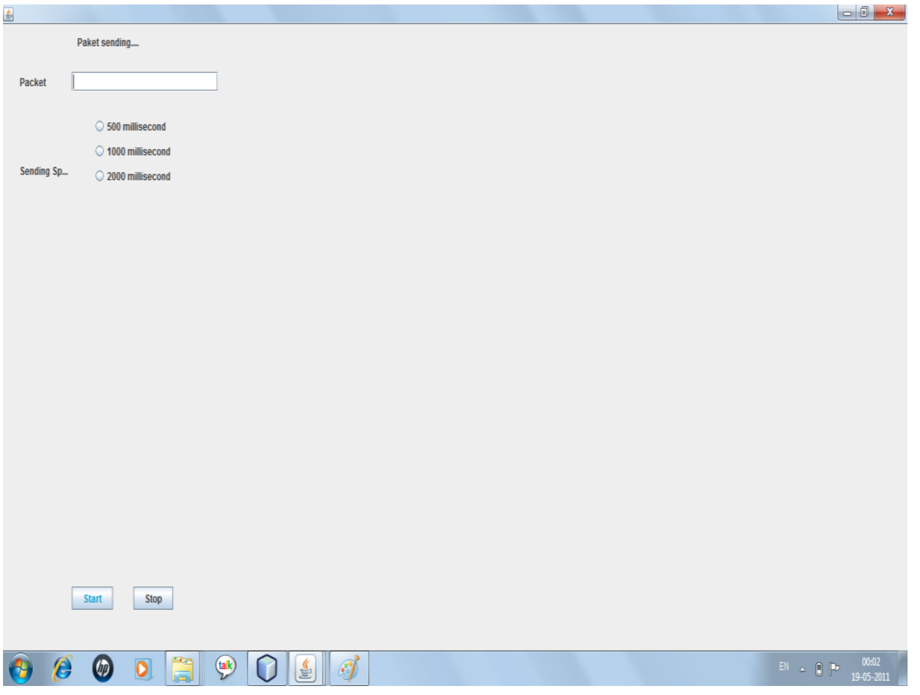

Fig 5.1 Screenshot of packet sending

The figure 5.2 shows that a packet hello is being send at a speed rate of 500 millisecond. The system itself identifies whether it should be passed through a supplemental or fundamental channel according to their speeding rate..

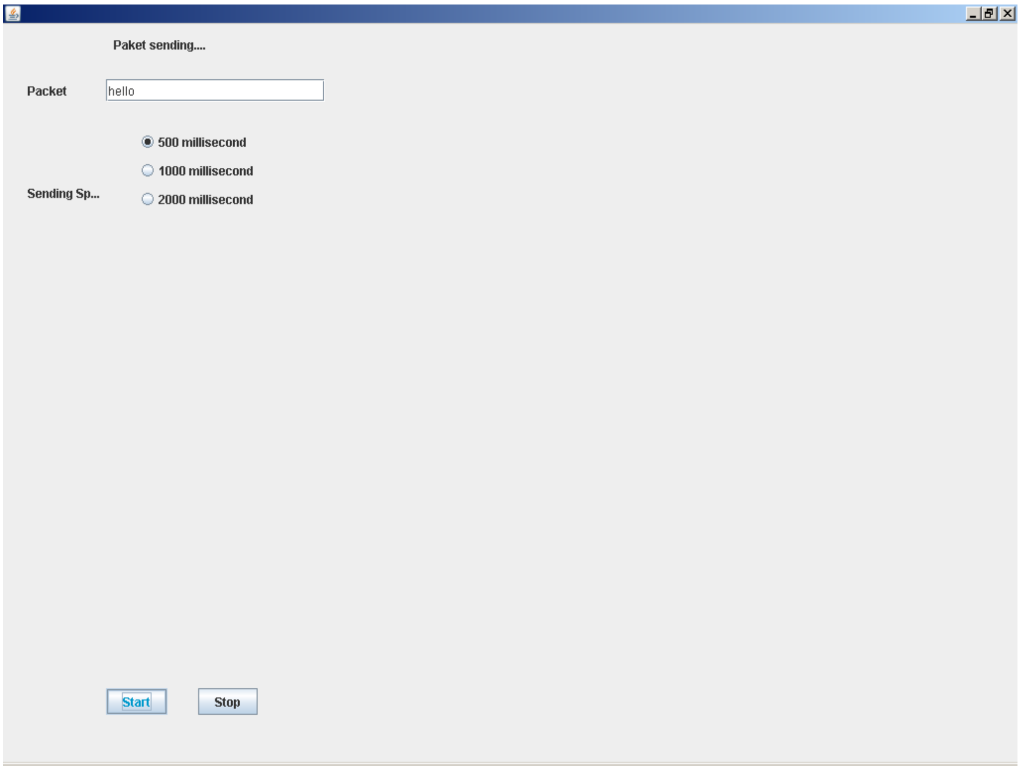

Fig 5.2 Screenshot of packet "hello" being send.

It can be seen from the figure 5.3 that the packet hello which was send by the sender has been received in the controller without any delay and an option sending direct is selected to send the packet through the system channel. This enables the system to calculate the single coding rate and the value will be stored in the text file of the source package. The receiver receives the file which has been send by the controller through the socket connection. Finally the stop button is pressed to stop the sending of the packet from the sender as shown below.

The receiver file is installed in the second system. By pinging the two systems the values are reached from server to the client. The IP address in the receiver file has to be changed and should be given that of the sender's IP address.

After the stop button is clicked on the receiver the packet sending comes to a halt. Once again we enter another packet with a different name along with a sending speed rate of 500 millisecond similar to that of figure 5.2. as done.

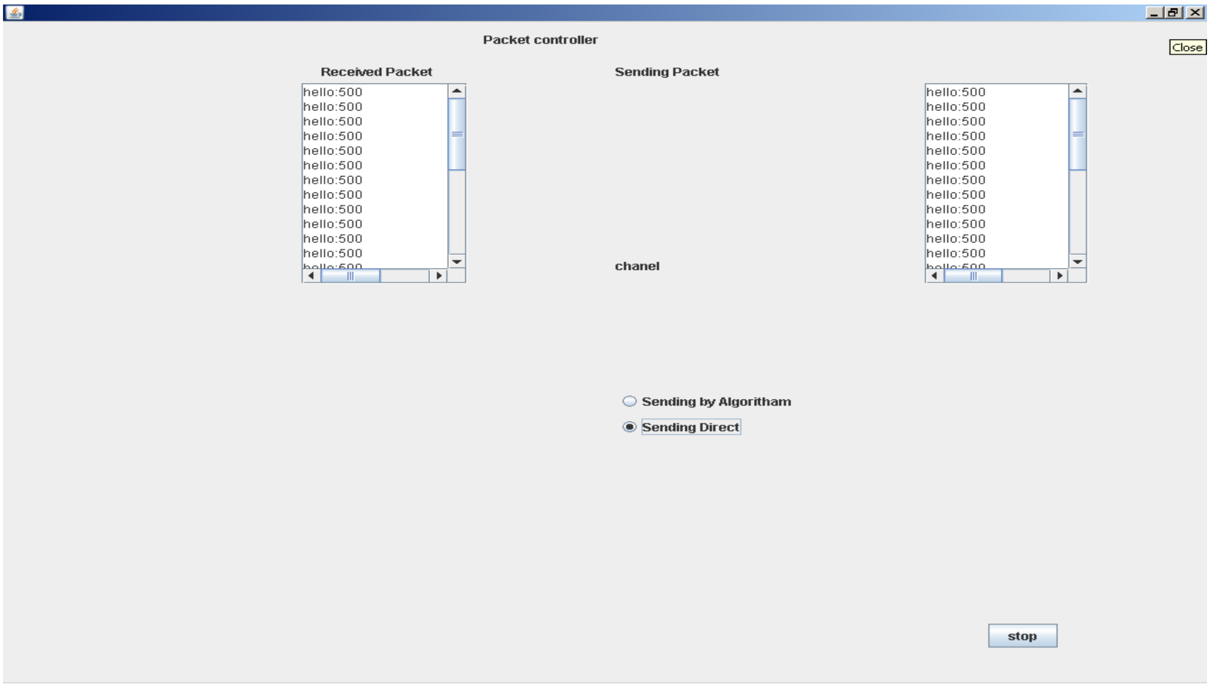

Fig 5.3 Screenshot of packet receiving and sending in the controller module.

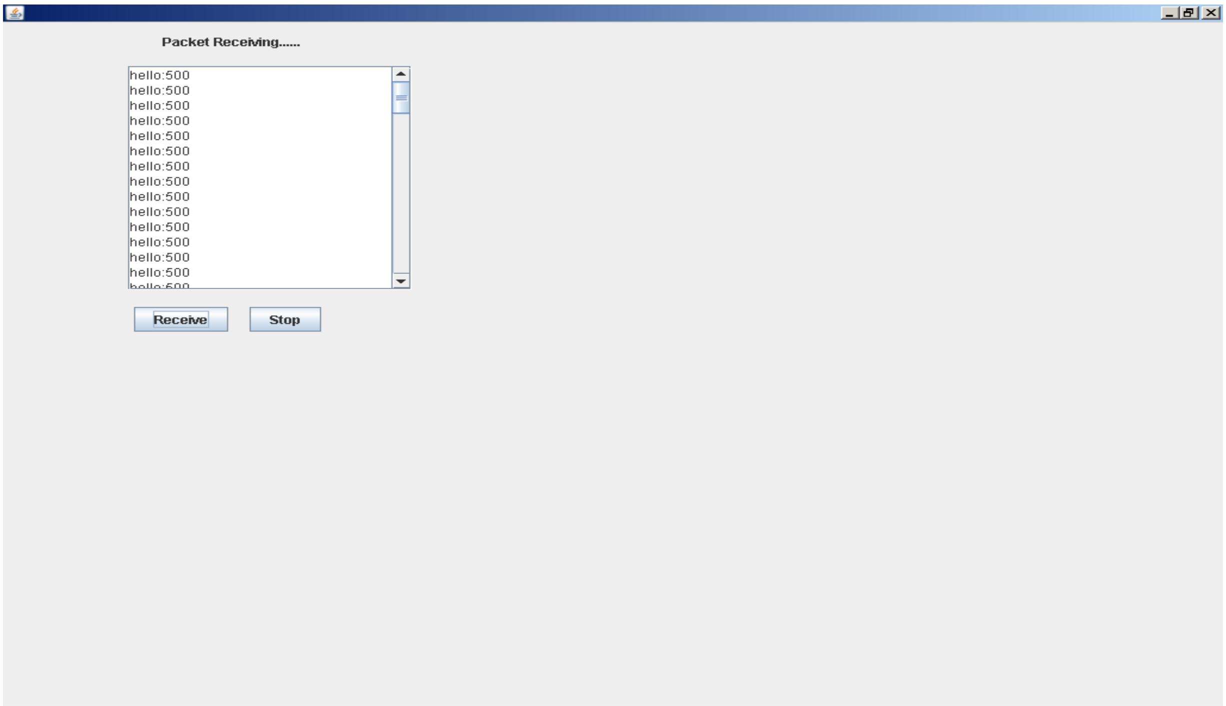

Fig 5.4 Screenshot for packet receiving

The packets are being received once we press the start button. We click the option sending by algorithm as shown in the figure below. This enables the system to decide through which channel it should pass either supplemental or fundamental channel. If the channel rate is high the packets passes through the supplemental channel and if the channel rate is low it passes through the fundamental channel.

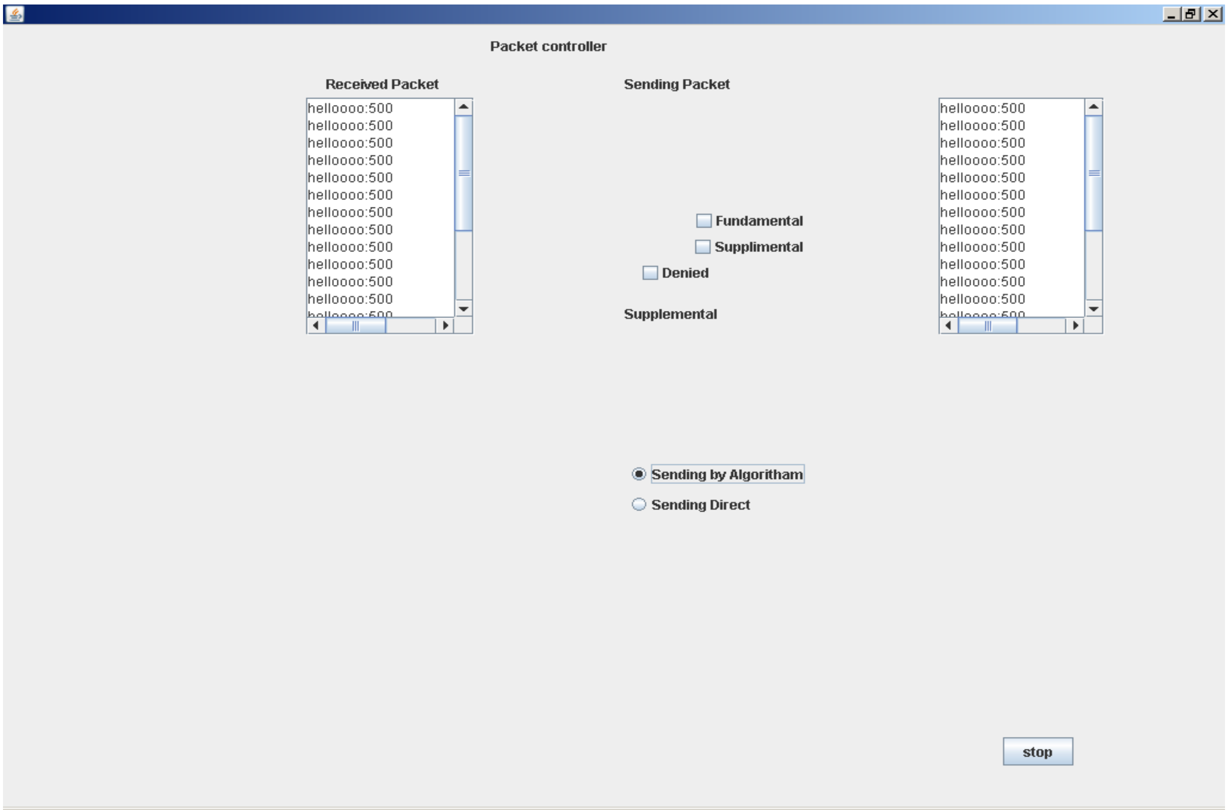

Fig 5.5 Screenshot for sending by algorithm

As it can be seen since the speeding rate is high the channel passes through the supplemental channel. These packets are received by the receiver and a dialog box appears displaying the single coding rate, two coding rate and gain.

It is seen the gain value increases by seven percentage and two coding rate value changes depending upon the packet size. The window size is initialized to 8. The TCP packet size is set to 1024 bits. Hence we calculate the

Two coding rate = (window_Size(packectSize/1024))/rtt_avg*

The RTT(Round Trip Time) is calculated to obtain a minimized packet transmission from client to the server. The average of RTT is calculated as the total time for a packet to be sent by total time it takes for the acknowledgment of packet to be received.

The gain is calculated by comparing the single coding rate and the two coding rate. The gain is calculated is as follows:

*Gain =100 * ((Two coding rate – Single coding rate) / Single coding rate)*

The figure below represents, if the gain appears to be zero it means that the packet did not pass through any disturbing channels. If it appears to be negative then it indicates that the speed rate of the system is high which can cause data errors. Normally when a packet is transmitted through a channel its gain value increases after every count=10.

Fig 5.6 : Screenshot of coding rate value 1

As the system's ping value changes the single coding rate value is again analyzed. The system again transmits the packet through an algorithm method.

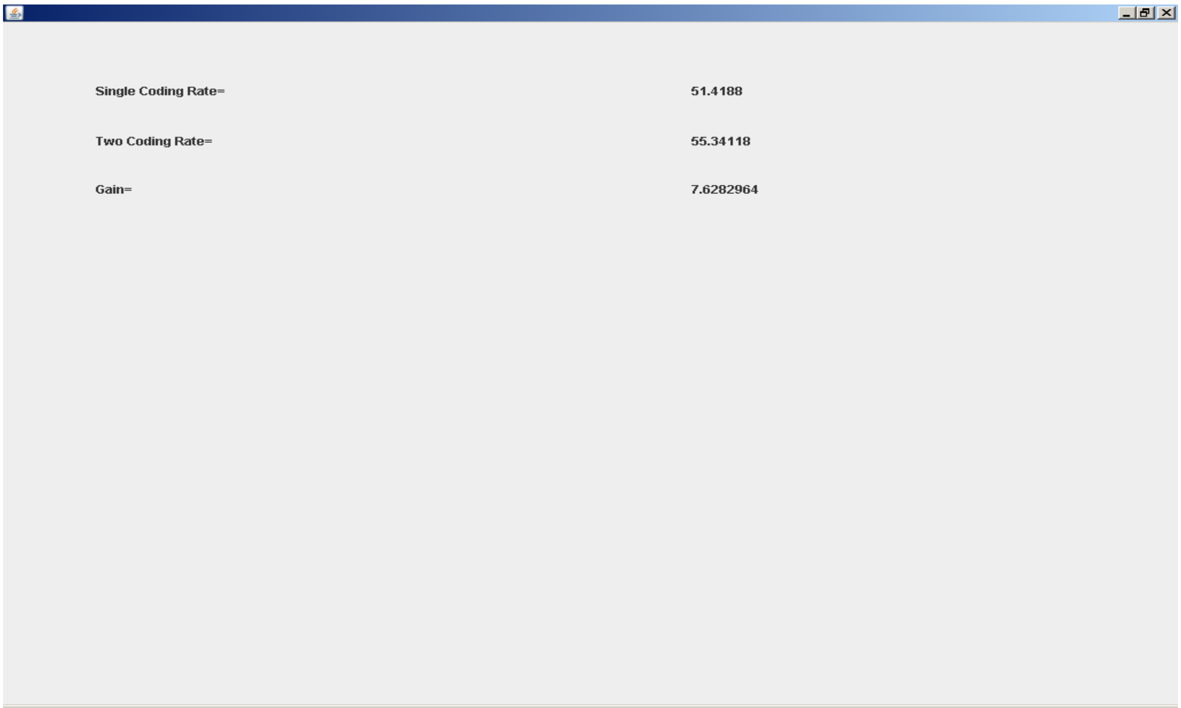

Fig 5.7: Screenshot of coding rate value 2

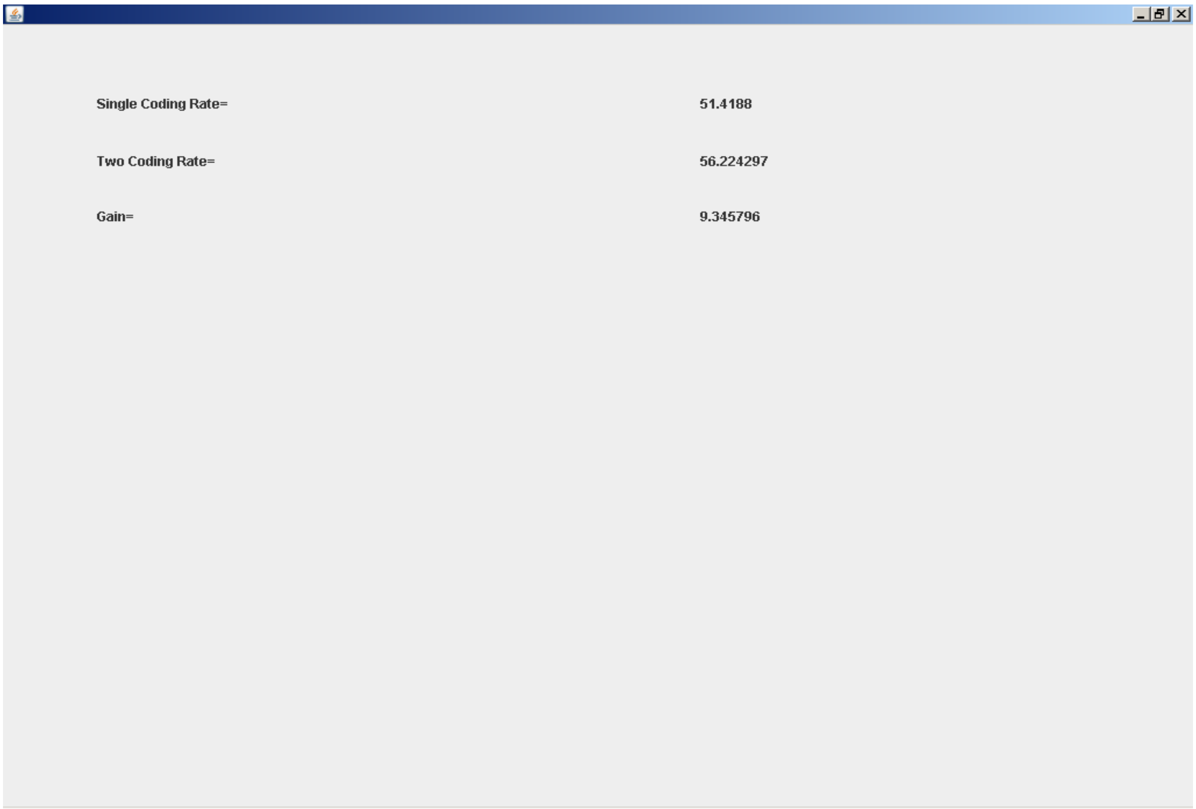

Fig 5.8: Screenshot of coding rate value 3

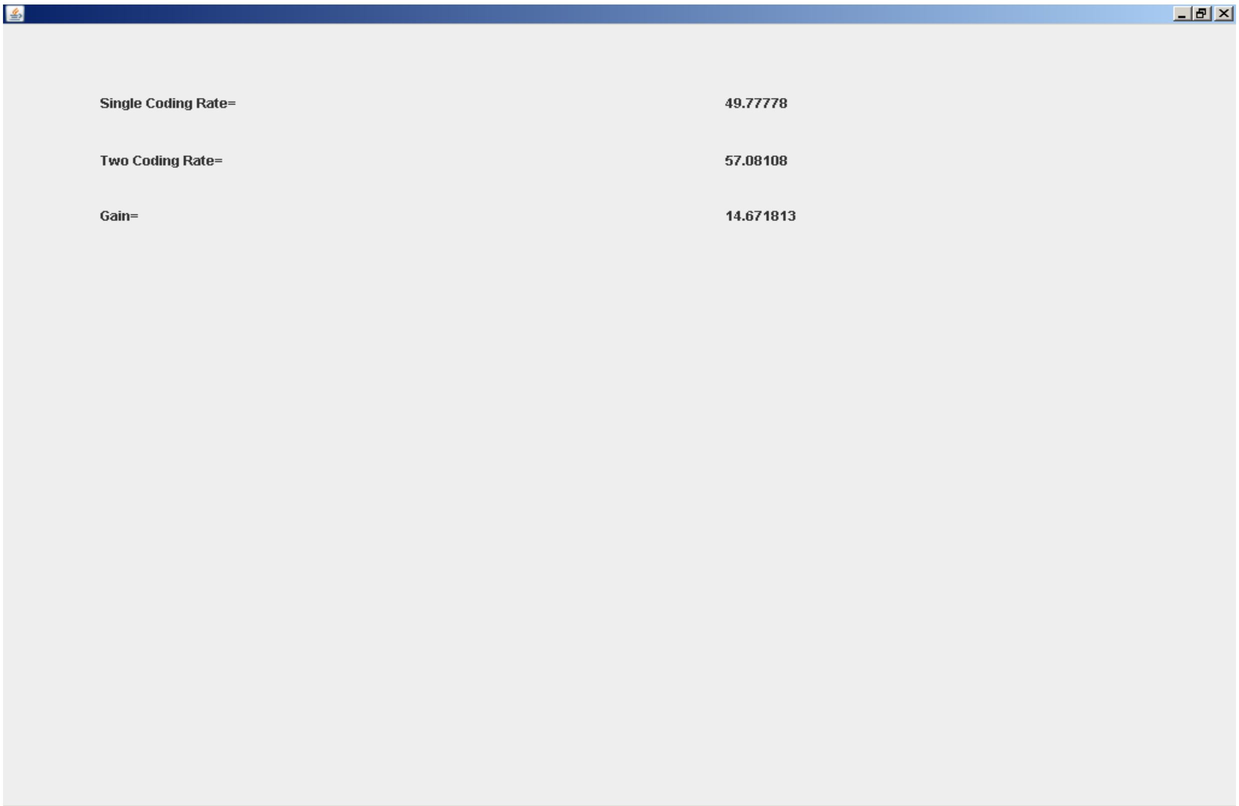

Fig 5.9: Screenshot of coding rate value 4

As the figures 5.7, 5.8, 5.9 shown above indicates that the packet when it is dynamically allocated through the channel the gain seems to be increasing which in turn increases the TCP throughput and packet transmission have high accuracy.

6. Conclusion and Future work

This proposed techniques for optimizing the rate adaptation feature of modern CDMA systems in a TCP-aware fashion. The proposed system adapts it channel rate in response to the TCP sending rate, allowing it to trade-off channel rate in a TCP-friendly manner. For the single TCP session case, an analytical model was developed for a two state system that explicitly accounts for the interaction between our proposed scheduler and TCP dynamics as well as the presence of two distinct channel rate, packet error probability and RTT. Improvements of 15 to 20 percent were observed compared to a system that does not exploit rate adaptation.

However, simulation is needed to prove strongly the result of the dynamic channel allocation. In addition, ports in our system should work in cooperation to detect that packets are being transmitted faster and with higher accuracy. These drawbacks are what mainly focused on to improve the TCP throughput. In this paper, we have only dealt the packet transmission with single-user scheduler. However, this can also be implemented with multi-user scheduler. To do this, we have to propose two flow-level schedulers, which are extensions of the single-user scheduler, for channel allocation among multiple TCP sessions. The proposed channel allocation schemes yield bandwidth savings and higher throughput compared to a network with a single static channel rate.

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