Adaptations at MAC layer
for
Third generation wireless networks

A Report Submitted

in Partial Fulfillment of the Requirements
for the Degree of
Bachelor of Technology

by
Amit Jain (97040)

under the guidance of
Dr. Dheeraj Sanghi

to the

Department of Computer Science and Engineering

Indian Institute of Technology, Kanpur

April, 2001.

Certificate

Certified that the work contained in the report entitled "*Adaptations at MAC layer for Third Generation wireless networks*" by Amit Jain, has been carried out under my supervision and that this work has not been submitted elsewhere for a degree.

Dr . Dheeraj Sanghi

April, 2001

Acknowledgements

I would like to express my gratitude to my supervisor Dr. Dheeraj Sanghi for his guidance, motivation and invaluable suggestions at all stages of this project. I would like to thank him for his enthusiasm throughout the discussions I had with him during the course of the project.

I would also like to thank Dr. Thierry Turletti (INRIA, France), without whose help I would not have been able to get started on the project. He provided constant support and invaluable suggestions throughout the different stages of the project.

Abstract

The third generation(3G) wireless network is envisaged as the next generation wireless solution for meeting the demand of its users. It promises to provide wireless access to multimedia applications at rates upto 2Mbps. But the problems of intermittent disconnection, high error rate in wireless networks can cause degradation of protocols and compromise the QoS (Quality-of-Service) guaranteed to the users. So, there is a need for adaptive protocols, which, on determining the current adverse wireless network conditions, adapt themselves in order to overcome these changes and reduce their effect. Our focus in this work has been to design adaptation techniques for media access control (MAC) protocol since their capability to adapt to the adverse effects of wireless conditions can make the high-level network protocols less susceptible to the changes in the wireless conditions. We have chosen SMPT (Simultaneous MAC Packet Transfer) technique for our study. Our results show that there is increase in throughput using the adaptive techniques mentioned in this report.

Contents

Page No.

Chapter 1

Introduction

One of the driving forces of the next generation of wireless communication and computing networks is the promise of high speed multimedia services. Third generation(3G) systems, such as the International Mobile Telecommunication System 2000 (IMT-2000) network (formerly known as the FPLMTS -- Future Public Land Mobile Telecommunication System) and the Universal Mobile Telecommunication System (UMTS), promise to provide multimedia services to mobile and fixed users via wireless access to the global telecommunications infrastructure. The IMT-2000 is a universal, multifunction, globally compatible digital mobile radio system that plans to integrate all traffic types and all wireless systems under a common set of formats. The UMTS is a similar global wireless solution, and is being standardized by the Europe by the European Telecommunications Standards Institute (ETSI). The 3G systems are expected to be deployed by 2003. Among the requirements for the third generation systems is the ability to support multimedia traffic.

Ultimately, 3G is expected to include capabilities and features such as:

- Enhanced multimedia (voice, data, video, and remote control)
- Usability on all popular modes (cellular telephone, e-mail, paging, fax, videoconferencing, and Web browsing)
- Broad bandwidth and high speed (upwards of 2 Mbps)
- Routing flexibility (repeater, satellite, LAN)
- Operation at approximately 2 GHz transmit and receive frequencies
- Roaming capability throughout Europe, Japan, and North America

The 3G networks are expected to support a variety of voice and date services (at high data rates) and yet maintain high quality. However, as against the wired channel, the wireless channel is much more error prone. The performance degradation of wireless communication can be caused both by poor wireless environment and large number of wireless users that share the communication medium. Thus, providing the QoS to the application becomes an important issue. So, adaptive protocols are needed which, depending on the adverse wireless conditions, would adapt themselves to mitigate the effects of these changes.

The adaptation can be provided at any layer of the OSI reference model (see Appendix). In this work, we propose adaptive techniques for improving the performance of MAC protocols through awareness of mobile communication environment. We chose MAC layer because being closer to the physical layer, it has a better idea of channel conditions and thus, it can adapt easily and quickly. Also, its adaptation can make high-level network protocols less susceptible to changes in the wireless conditions.

The MAC protocols proposed for CDMA systems include Multidimensional PRMA with prioritized Bayesian Broadcast[19] , Wireless Multimedia Access Control with Bit Error Rate Scheduling (WISPER), WCDMA MAC protocol[10], Simultaneous MAC Packet Transfer(SMPT) etc [11]. Most of these protocols use appropriate scheduling algorithms and reservation of traffic types to improve performance under different types of multimedia services. These protocols differ in the way they allocate resources(codes) to the mobiles. But none of them is adaptive to the channel conditions. Our effort in this work has been to make the existing scheme, SMPT, adaptive.

Chapter 2

Medium Access Control Protocol (MAC)

2.1 Definition

In a wireless system that consists of a number of mobile terminals that transmit traffic of any type on a shared medium to a centralized base station, a procedure must be invoked to distribute packet transmission among all users. This procedure is known as a medium access control (MAC) protocol.

2.2 Classification

MAC protocols are often classified according to their method of resource sharing, as well as their multiple access technology.

The resource sharing methods include *dedicated assignment, random access, and demand-based assignment*. Dedicated channels assign each user a pre-determined and fixed allocation of resources, regardless of the user's need to transmit. Dedicated assignment schemes are appropriate for continuous traffic, but can be wasteful for bursty traffic. On the other hand, random access channels, allow all users to contend for the channel by transmitting as soon as packets are available to send. Random access is suitable for bursty data traffic, but is not desirable for delay-sensitive traffic. Demand-based assignment schemes assign resources according to requests, or reservations, submitted by users. Once the requests are transmitted (using either dedicated or random access channels) and processed, users can be assigned resources according to the results. Demand-based channels are useful for variable rate traffic and the hybrid conditions of multimedia traffic. However, the additional overhead and delay caused by the reservation process can degrade performance.

In addition to the resource sharing method, the multiple (or multi) access scheme of a MAC protocol establishes a method of dividing the resources into accessible sections. Three accepted methods for resource division are *Frequency Division Multiple Access* ***(FDMA)****, Time Division Multiple Access (****TDMA****), and Code Division Multiple Access* ***(CDMA)***. FDMA schemes divide the resource into portions of spectrum, referred to as channels. TDMA schemes divide the resourceinto time slots. Finally, CDMA divides the resource into a collection of codes through which assigned users can co-exist on the same channel. First generation mobile systems (late 1970s) and early second generation systems, such as the Advanced Mobile Phone System (AMPS), used FDMA schemes to support analog communication. For second generation systems (began in 1990s) , the most used multi-access schemes have been TDMA and CDMA. Wide-band CDMA (W-CDMA) has been chosen as the basic radio-access technology for 3G networks.

Chapter 3

Overview Of CDMA

In a CDMA system, the symbols which carry information bits occupy a bandwidth more than the minimum needed for the transmission. There are two techniques to achieve this, namely, Direct-Sequence (DS) and Frequency-Hopping (FH) spread spectrum. In this report, we are only concerned with the first technique. In a receiver-oriented DS-CDMA system, Base Station (BS) may have several receivers, each of which listens to a specific code. A mobile host (MH) transmits information bits by applying a code to them to generate symbols covering a large bandwidth. Only the receiver, which listens to this code, can recover the bits from the symbols; other receivers perceive them as noise. It is important that two mobiles should use different codes for transmission at the same time; otherwise, the receiver cannot distinguish them.

Even if a bit transmitted by a MH is spread with a code different from any other code used at the same time, it may still be corrupted because the bits simultaneously transmitted by other MHs generate some level of noise on the intended receiver. This is called multiple-access interference (MAI).

Chapter 4

Simultaneous MAC Packet Transfer (SMPT)

The idea behind SMPT [11] approach is that CDMA systems allow multiple channels in parallel as long as the total number of used channels does not exceed a specific threshold. In opposite to common wireless CDMA systems, where each wireless mobile terminal uses a single channel for all supported flows of a multimedia application, this approach allows the mobile to allocate multiple channels, assuming a CDMA system which has more channels than users.

In this approach, the sender transmits a higher protocol data unit within a given delay bound. The segment is fragmented into data link packets. It is assumed that allowed transmission time for a higher protocol entity is a multiple of the data link packet transmission time. As long as the wireless channel is error-free, packets can be sent sequentially on one channel. In case of an error prone link, ARQ transmissions of data link packets are done within the code domain using multiple channels. Furthermore, if it is assumed that fading durations on the wireless link are smaller than the transmission time of an entire transport segment, then MAC level retransmissions on multiple channels are performed only if the wireless link is less error prone to achieve a high spectral efficiency. The advantage of SMPT is that it can recover from gaps caused by errors on the wireless link within a given TCP or UDP segment.

Multiple channels are used by the mobile at its own discretion. Using multiple channels in parallel, however, can degrade the overall system capacity, because each additional active channel will lead to a degradation of the signal-to-noise ratio (SNR), which results in high bit error probabilities. Thus, multiple channel usage must be avoided when the channel is bad.

There are many possible ways to use multiple channels. The two different types of approaches are 'Self Healing' and 'Start Up'.
(As an example we demonstrate the transmission of 16 link packet data units (LPDUs))

Self Healing is only used if sequential transmission (using one channel) falls behind due to channel errors. The Self healing mechanism reduces the accumulated delay jitter using the capacity to send on additional channels. There are two main approaches to Self healing process:

- **Slow Healing(SH):** It uses the available resources incrementally after detecting a good channel state.

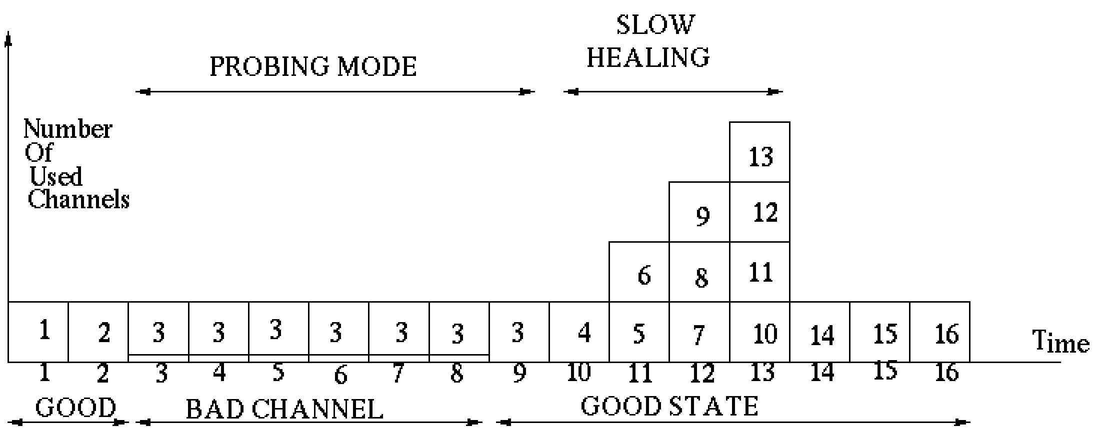


Figure : Slow Healing

- **Fast Healing(FH):** It uses all available resources immediately after detecting a good channel state.

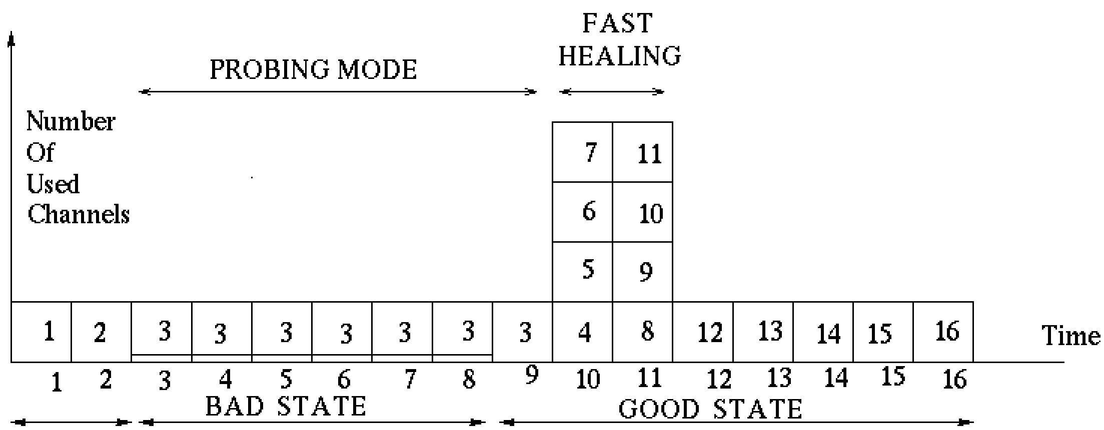


Figure : Fast Healing

The ‘Start Up’ method sends on additional channels *whenever* the channel is in good state. There are two main approaches to ‘Start Up’ process :

- **Slow Start (SS) :** It uses the available resources incrementally whenever the channel is in good state.

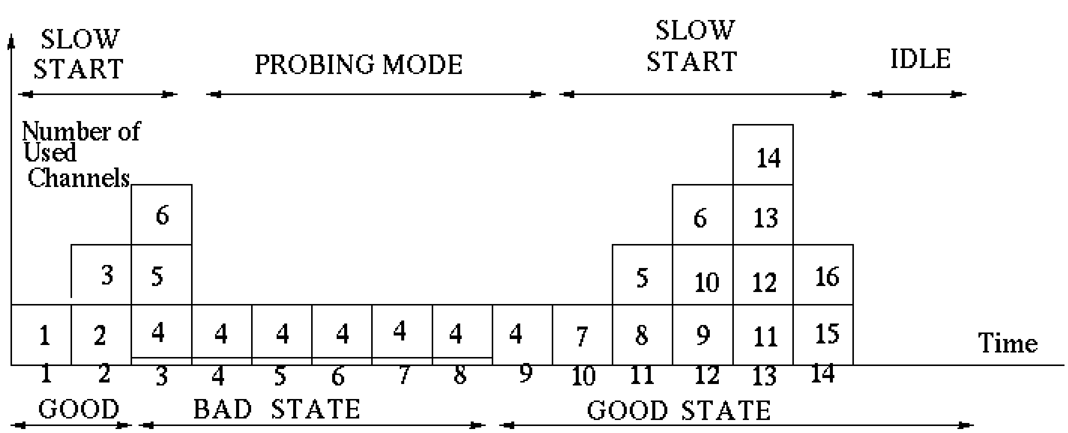


Figure : Slow Start

- **Fast Start (FS):** It uses all the available resources whenever the channel is in good state.

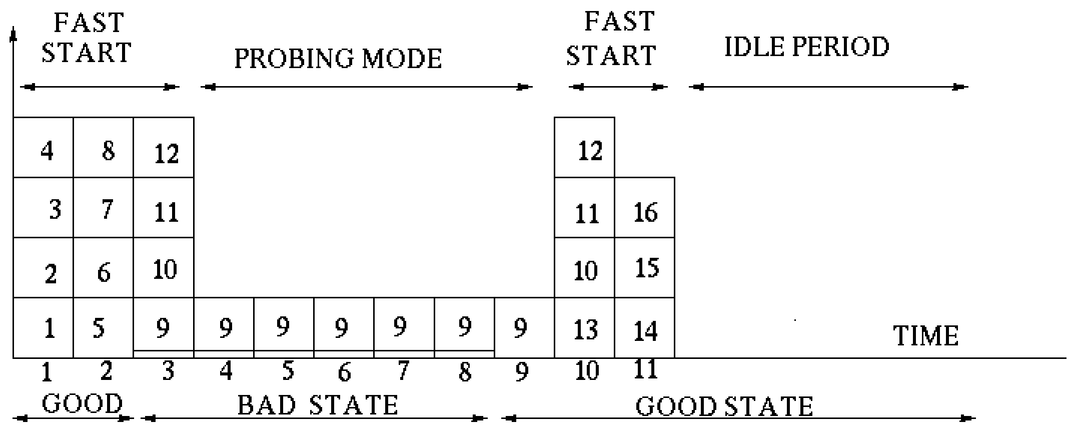


Figure : Fast Start

The SMPT technique switches between *normal* and *probing* mode depending on the channel state. As soon as the sender side data link entity realizes that the channel state has changed from good to bad, it sends probing packets (Probe Mode). A single channel is used when the channel is bad. The probing mode continues until one probing packet is transmitted successfully. It is assumed that a successfully transmitted packet is indicator of good state, while a lost packet is equivalent to an indicator of a bad state. To achieve a high SNR, multiple channels are used only in the good channel states.

Chapter 5

Details of Simulation

5.1 Tool used

We have used 'ns' (network simulator) for simulations.
(refer Chapter 8 for 'ns' description)

5.2 Simulation Scenario

One mobile connected by bi-directional link with a base Station. The effect of more number of mobiles is taken into consideration in the calculation of SNR (Signal-to-Noise ratio)

5.3 Simulation Parameters

- *Source Traffic*

  The traffic source used is CBR (Constant Bit Rate).

  Rate : 3360 bits/sec

  Packet Size : 140 bytes (1120 bits)

- *Bandwidth*

  The bandwidth of one channel (corresponding to one code) is 1120 bit/sec.

- *Maximum number of parallel channels per mobile* = 6
- *Simulation Time :* 1000 sec
- *Signal-to-Noise (SNR) ratio expression*

  SNR = 7.0 – (0.02 * no_of_channels_per_mobile * no_of_mobiles)

  This relation is not exact. It is a 'crude' relation devised by us after running the error module for several times and noticing how the channel goes bad with different values of SNR. We then related the SNR value with the number of mobile in the system.

- *Error Model :* Rayleigh fading Model and Additive White Gaussian Noise (AWN) is used to detect whether the packet is in error or not. When FEC is used, convoltutional coding and Viterbi dcoding is used to detect whether the packet is in error or not.

5.4 Assumptions

- It is assumed that the information about the correctness or corruptness pf the packet is immediately available. This can be assumed if the cell size is small.
- It is assumed that all the mobiles have the same load pattern and thus have the same channel usage pattern. (This is the reason why the number_of_channels (per mobile) is multiplied with number_of_mobiles to give the effect of total number of codes used in the system.)
- The expression, relating SNR to number of mobiles, is not exact.

5.5 Graphs

The graphs are plotted between ***Goodput*** *(Y-Axis)* and the ***number of mobiles*** *(X-Axis)* in the system.

(Goodput is defined as the number of correctly received packets per unit time.)

Chapter 6

Simulation of SMPT Techniques

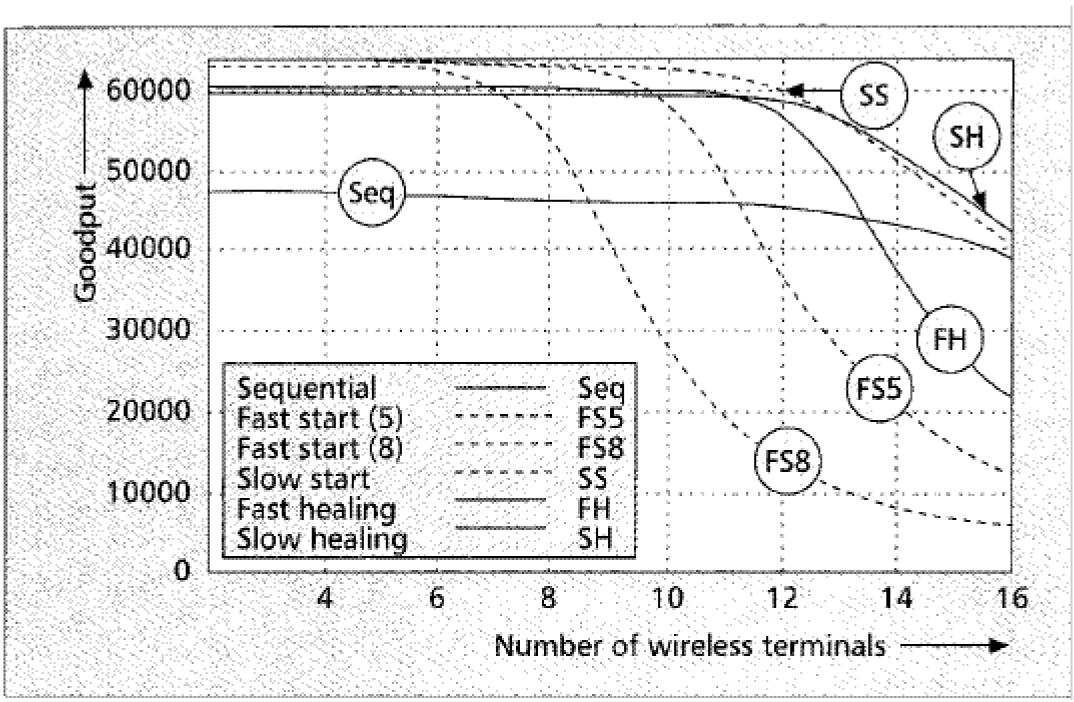


*Fig : Goodput vs number of wireless terminals (**Actual graph** as given in the SMPT paper)*

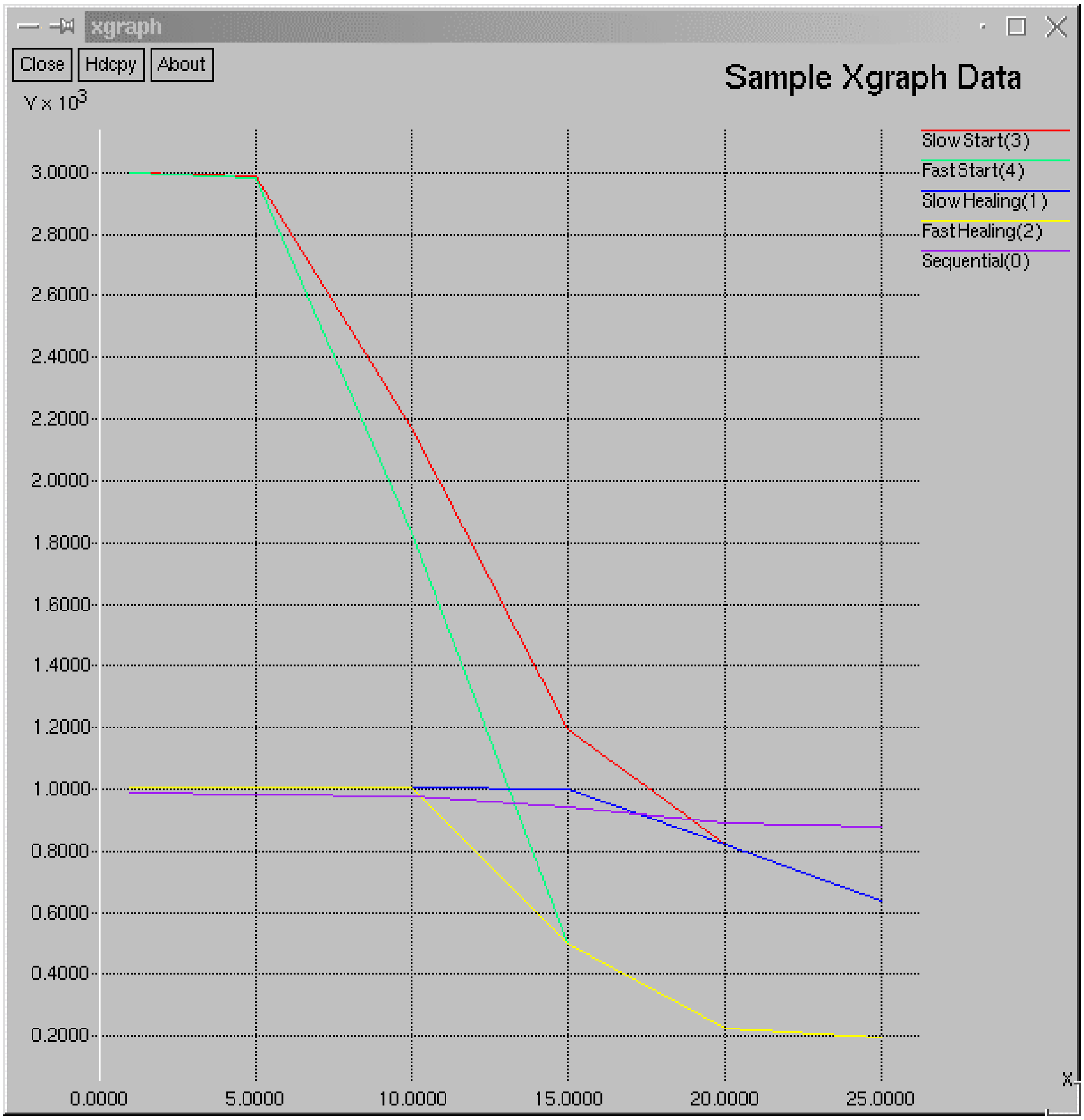


Figure : Goodput vs number of wireless terminals ***(obtained by simulations in ns)***

For small number of mobiles, the goodput for all SMPT strategies is much higher than the sequential case. With an increasing number of wireless devices some strategies lead to results even worse than the sequential case. Slow start and Slow Healing perform better than sequential for more number of mobiles. Fast Start and Fast healing perform better only for small number of mobiles. However, after certain number of mobiles , all the curves fall below the sequential case. This is quite obvious since more channel usage creates more interference in the already congested cell causing more packet errors.

6.1 Comments on comparison between the actual and simulated results

The values on both graphs should be ignored because our model and simulation parameters are different from that used in the actual implementation. We observe that the pattern of the simulated results agree quite closely with the pattern of actual result.

However, there appears some disparity in the results at the initial stage since the curve for Slow Start(SS) and Fast Start(FS) is quite above the Sequential Transmission. This is because of the load pattern that we have chosen. The packets are produced at rate, which is higher than the bandwidth allotted (corresponding to a single code). Thus Slow Start and Fast Start have very high throughput because they always have many packets to transmit simultaneously on many codes. If the load was chosen comparable to that of bandwidth of one code, then SS and FS curves would lie close to Sequential curve.

Chapter 7

Adaptations Proposed

7.1 Two-Channel Probe

The SMPT technique uses single channel during probing mode i.e when the channel is bad. The packets sent during the probing mode are sent without using any channel coding scheme. Depending on how long the probing mode continues, the packets can suffer increased delays. This is undesirable especially for real-time data traffic.

Our idea is to use two channels during probing mode. The goal is to increase the goodput even during the bad channel mode. One first channel, send an uncoded packet. On the second channel, send the FEC coded packet. (We used rate=1/2 FEC scheme in our simulations).

Suppose N th packet is corrupted. Then during probing, transmit two packets within a given time slot, using two channels. Applying FEC coding to N th packet, we get two packets N(a)and N(b). Then on the first channel transmit (N+1) th packet and on the other, transmit N(a) packet. Now four possibilities arise

- *If only FEC coded packet is received correctly*

Since the uncoded packet is received corrupted, thus this means that the channel is bad and only the FEC coded packets can be transmitted correctly. Since FEC coded packet was transmitted successfully, thus we are able to increase the throughput during the bad channel state. So, we continue using the two channel probing mode. (In the next slot we send, (N+1) th packet and FEC coded N(b) packet.. In the next slot we will send, (N+2) th packet and FEC coded (N+1)(a) packet. Then in next slot we will send (N+2) th packet and FEC coded (N+1)(b) packet). This continues until both are received correctly or both received incorrectly.

- *If both the packets are received correctly*

Since the uncoded packet is received correctly, thus this means that the channel has become good. Thus, there is no need for coding and the protocol should revert back to the normal mode and start with the transmission of N th packet.

- *If both the packets are received incorrectly*

Since even the FEC coded packet gets transmitted corrupted, so this means that channel condition is *very* bad. So, either continue sending the same set of packets or start using higher FEC scheme.

- *If only the uncoded packet is received correctly*

This case is not probable because if FEC coded packet is corrupted then uncoded packet will be corrupted too.

We do not use more than two channels in the probing mode because using more codes can increase the level of interference in the bad state of the channel. And we have to use at least two codes for this idea to work.

7.1.1 Simulation Results

Simulation Scenario **:** As given earlier.(refer Chapter 5)
Simulation Parameters **:** As given earlier (refer Chapter 5)
Graphs: The following graphs are plotted between goodput and the number of mobiles in the system.

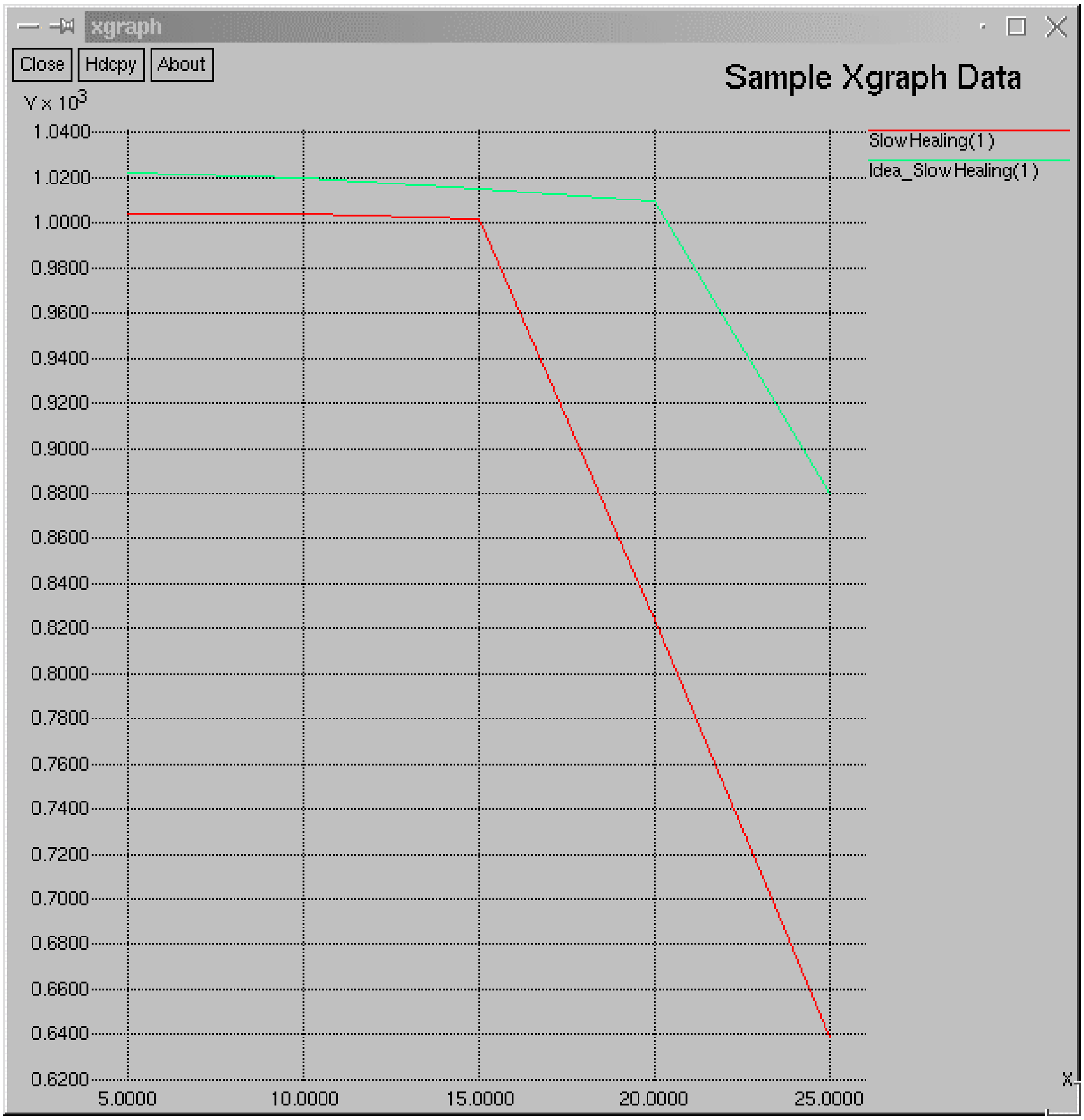


Figure : Slow Healing Graph (Goodput Vs number of wireless terminals)

Figure : Fast Healing Graph (Goodput vs number of wireless terminals)

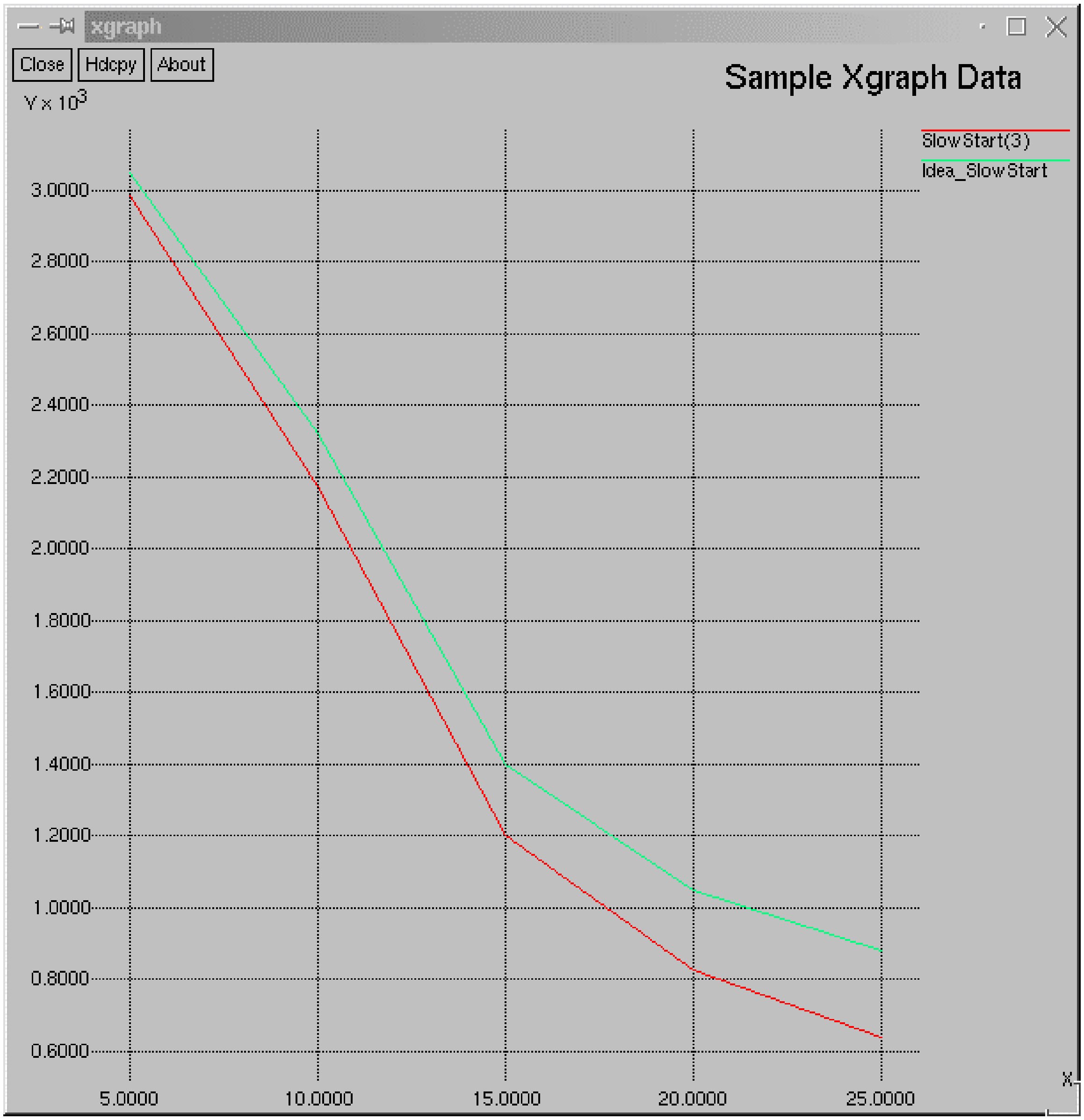


Figure : Slow Start Graph (Goodput vs number of wireless terminals)

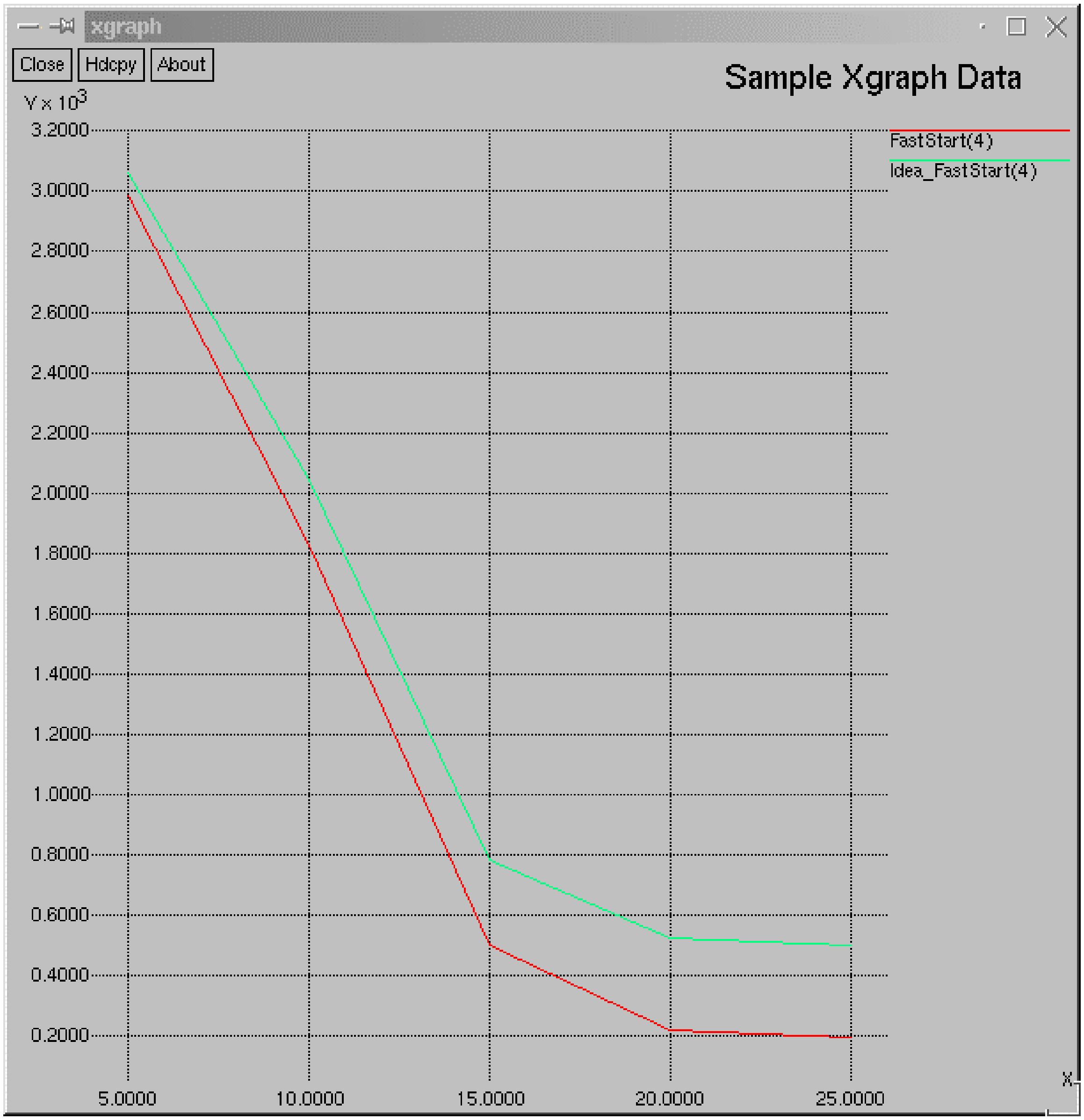


Figure : Fast Start Graph (Goodput vs number of wireless terminals)

7.1.2 Comments about results

As is evident from the graphs, all the techniques show improvement in goodput compared to simple SMPT technique. We had expected that the difference would diminish as the number of mobiles increases as more mobiles would have added more interference and made the channel more error prone. And thus the packets sent on the second channel would have added more interference and would have been thus corrupted. But the simulation results showed that the difference actually increases with the number

of mobiles. This lead us to further thinking and provided further insight into the understanding of our idea. As the number of mobiles increase, the channel becomes more error prone and thus the erroneous phases are long and thus we are able to transmit *more packets* during the erroneous phase. The reason why these packets escape corruption is because they are FEC coded. Our conjecture is that the difference would sharply decrease at the point when the number of mobiles (interference level) would become so much that even the FEC coded packets get corrupted. However, we could not simulate that and hence these results.
(There is a slight increase in throughput with number of mobile in the graph for Fast Healing. This could be attributed to effects of randomizations etc.)

7.2 Using multiple codes when "available"

When there is a general congestion in the system, all the mobiles will back off and will use only one code for transmission (Probing Mode). This reduces the interference level in the system.

But suppose, there is no congestion in the system and only a particular mobile is the one who is facing the bad channel condition may be because it is behind a wall or something like that. The Base Station has the information that whether a particular mobiles' channel is bad either due to congestion in the system or due to some problem with that mobile itself. So, utilizing this information, this particular mobile need to operate in Probing Mode, in which only one channel is used for transmission. So, if the load in the system is low (this information is also available from Base Station) then this particular can use the all the channels for transmission. It has to use FEC scheme so that probability of packet being transmitted correctly increases. It can use higher, costlier FEC schemes since it can now more bandwidth (more channels).

There would definitely be an increase in the interference level of the system when this mobile will start using more channels but since the load in the system is low, so it can be assumed that transmission by other mobiles is not affected.

7.2.1 Simulation Results

Simulation Scenario : There were two types of mobiles that were considered in the simulaion. The ***first type* of *mobiles(I)*** were the ones which suffered error-prone channel due to some erroneous conditions which they only are facing. Our idea was applied on these mobiles..

The ***second type of mobiles(II)*** were the ones which did not face individual erroneous conditions. The only erroneous conditions they faced were due to high interference level in the system.

Number of I type of mobiles were kept fixed (2) while the number of II type of mobiles were kept variable (5,10,15,20,25). For I type of mobiles, Rayleigh error model (burst errors) is used while for II type of mobiles Additive White Gaussain Noise is used.(which depends on the number of mobiles in the system). Both types of mobiles were connected by bi-directional link with the Base Station.

(*The number of mobiles, which are plotted on X-axis in the graph, refer to the second type of mobiles).*

Simulation Parameters : As given earlier (refer Chapter 5)

Graphs : The graphs are plotted between the goodput (number of correctly received packets in 1000 sec) and number of mobiles (II nd type).

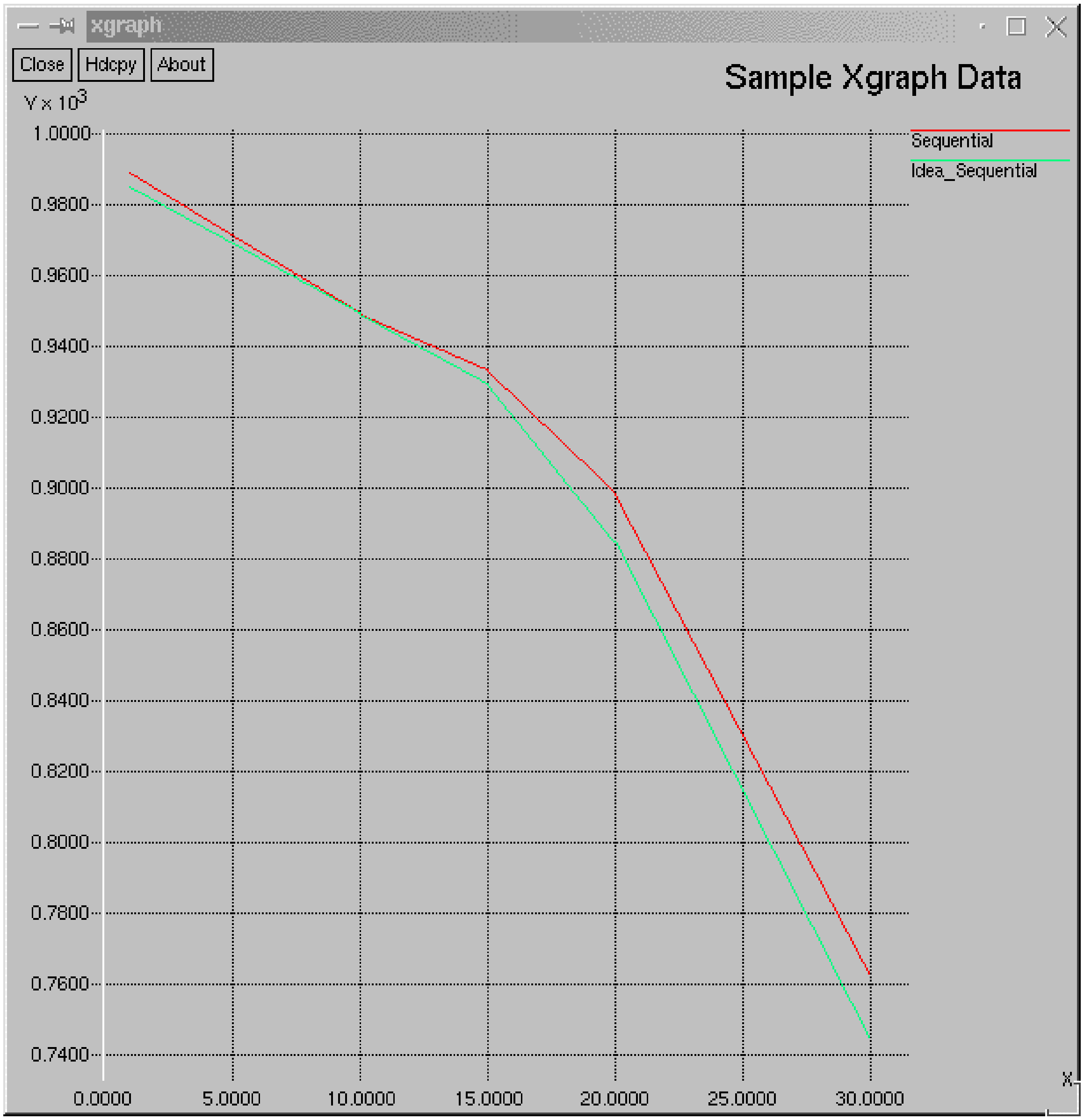


Figure : Sequential Graph (Goodput vs number of wireless terminals)

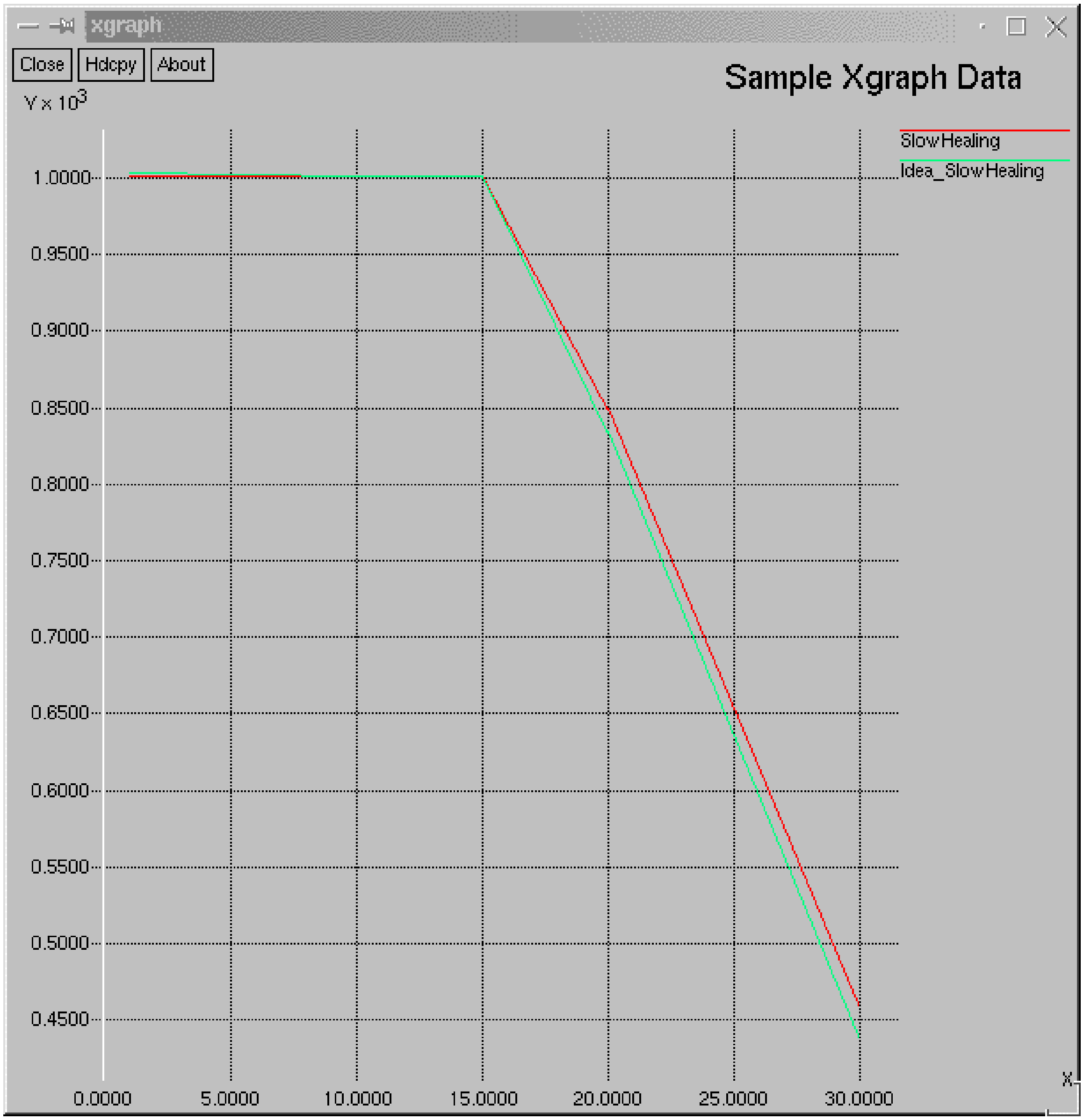


Figure : Slow Healing Graph (Goodput vs number of wireless terminals)

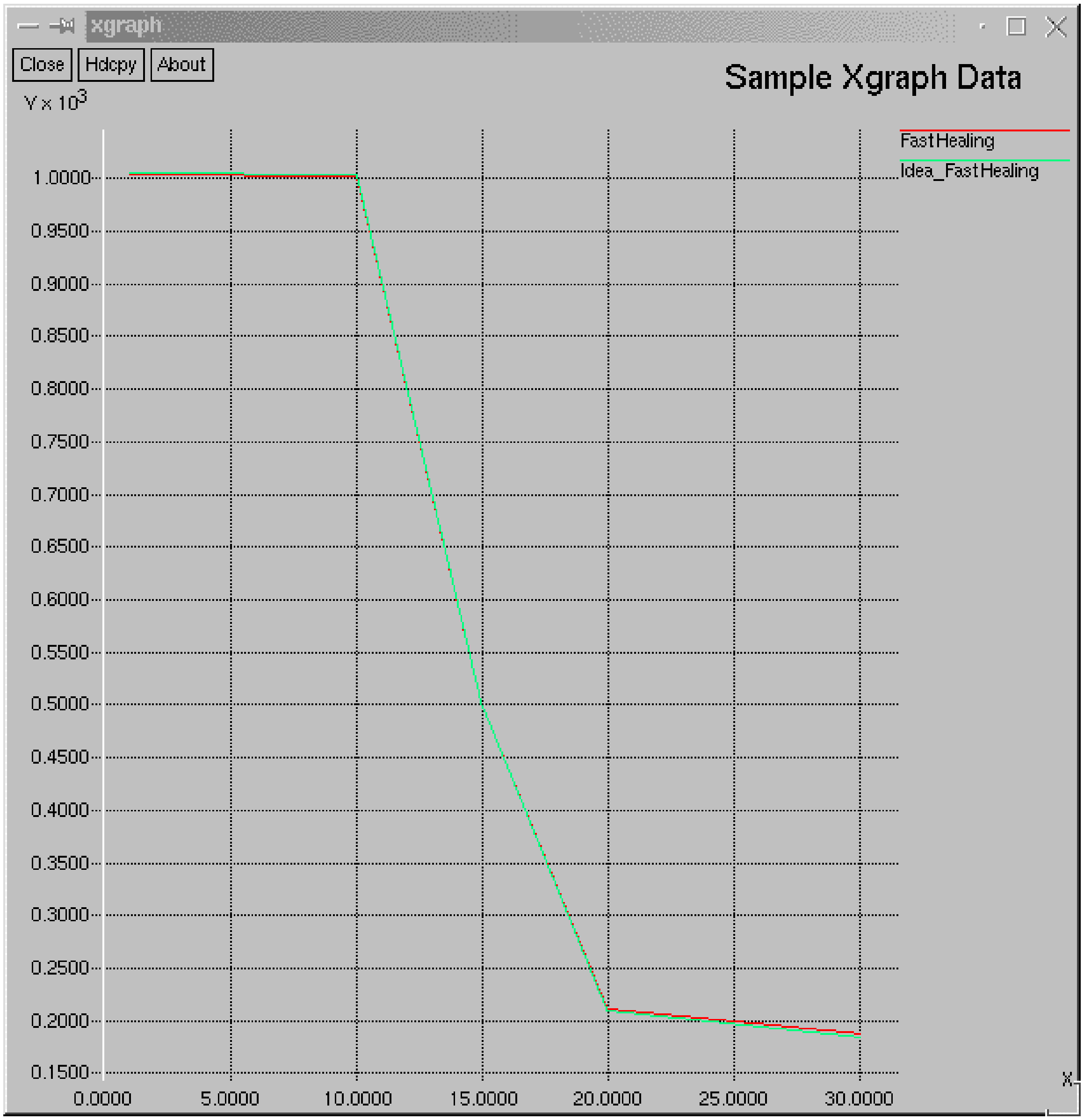


Figure : Slow Start Graph (Goodput vs number of wireless terminals)

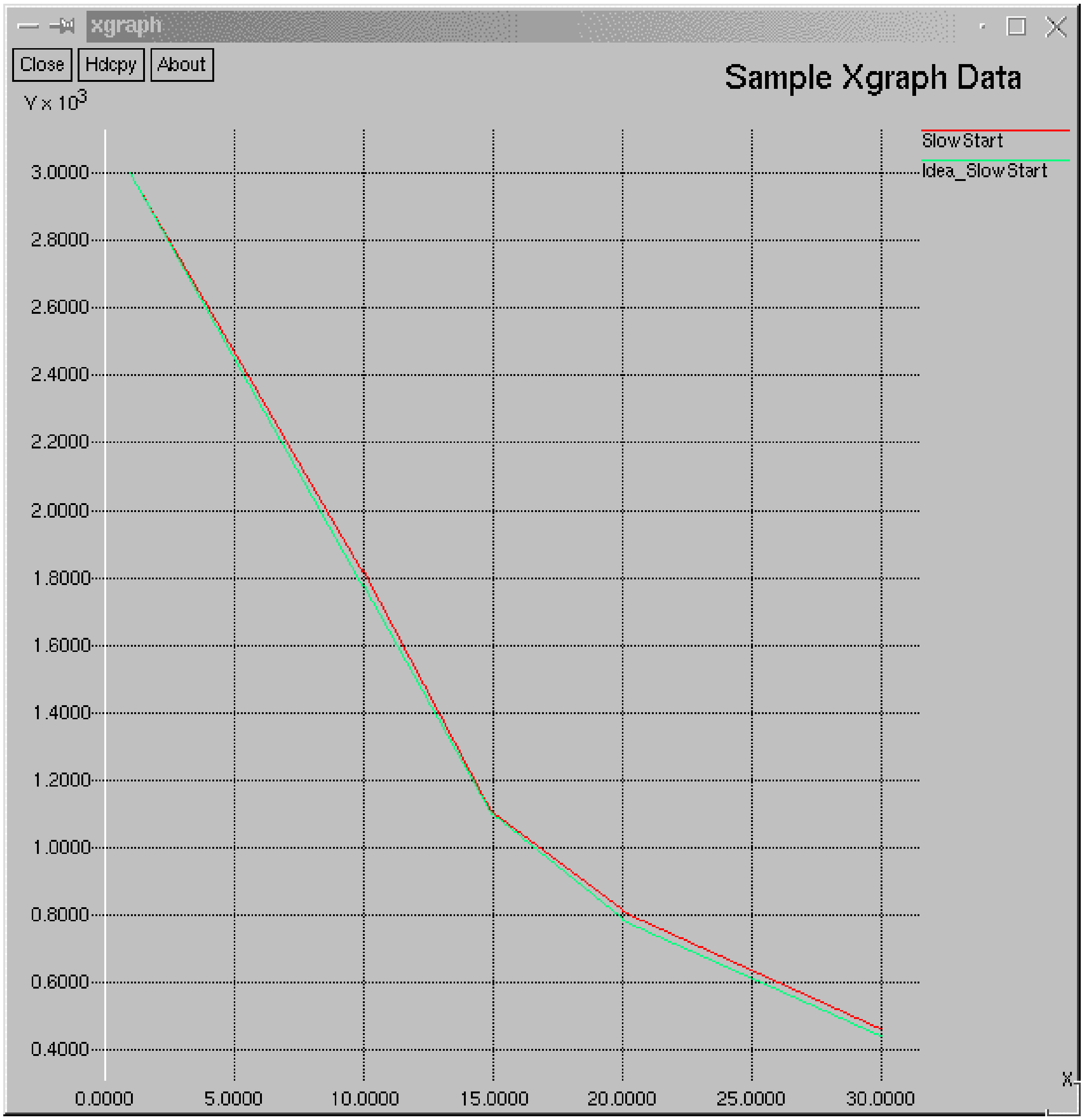


Figure : Slow Start Graph (Goodput vs number of wireless terminal)

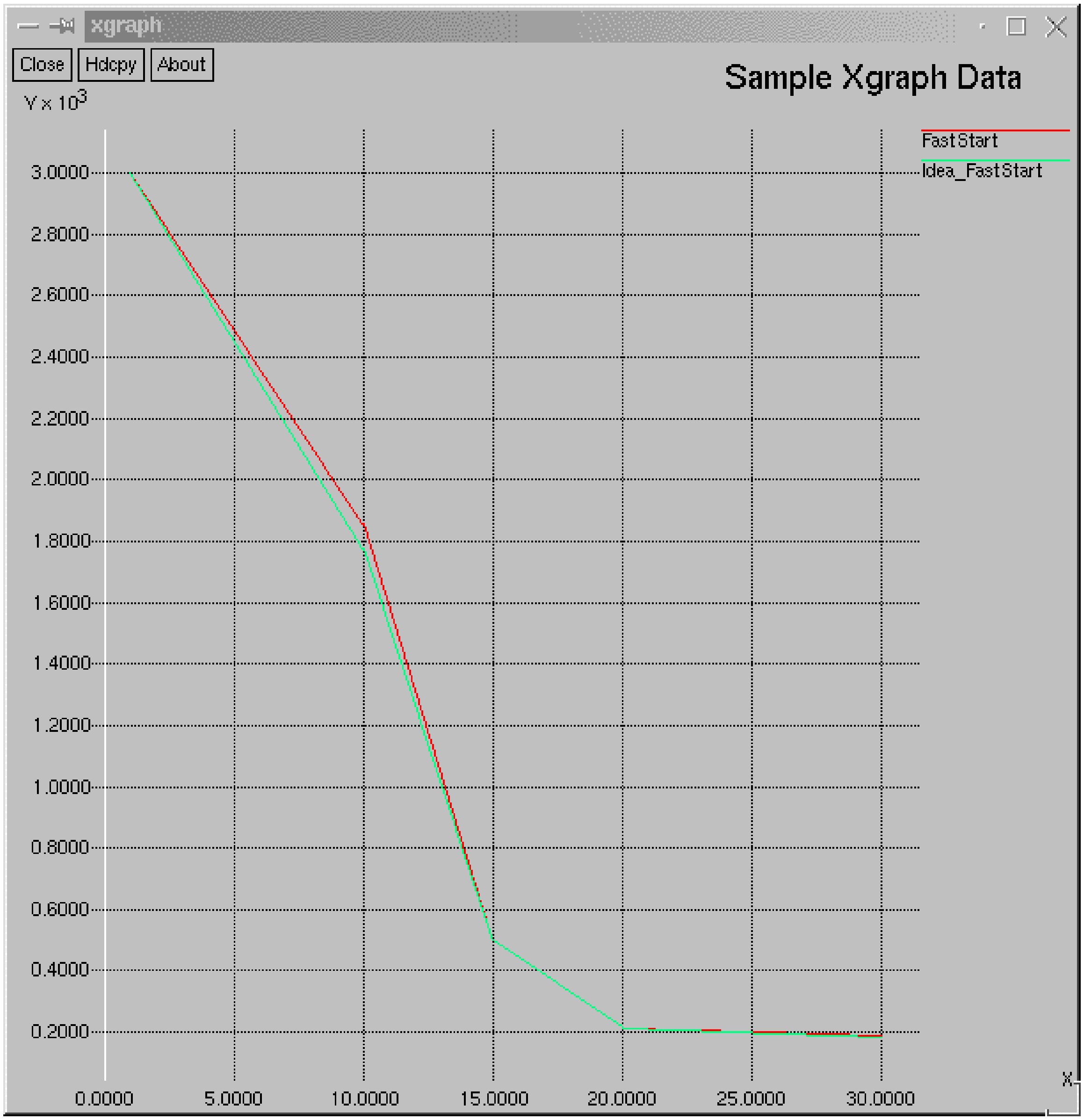


Figure : Fast StartGraph (Goodput vs number of wireless terminal)

7.2.2 Comments about results :

As seen in the graphs, there is no difference between the two curves when the number of mobiles(II type) in the system is less. Thus, I st type of mobiles can safely use the "free" bandwidth when the load is low and thus increase their throughput(it is not shown in the following graphs). But as the number of mobiles (II nd type) increase, the throughputof

II nd type of mobiles is affected by the usage of extra codes by Ist type of mobiles. Thus, it shows the suitability of our idea that mobiles can use extra codes when load is low, without affecting the throughput of II nd type of mobiles.

Also note that the difference is quite easily shown in Slow Start, Slow Healing and Sequential schemes. As these techniques use resources moderately, thus the effect of more code usage(more interference) is easily shown than in Fast Start and Fast Healing.

7.3 Multi-Copy Transmission

The SMPT approach assumes that ACK / NACK is immediately available. But practically, it is not possible, and acknowledgements would be coming late.
The erroneous channel condition would require retransmission. As we know, that timer controlled retransmissions can incur long delays, which is undesirable for real time traffic. So what can be done is that data can be retransmitted fast before the timeout occurs, even if an acknowledgement has not been received. It is our conjecture that by retransmitting data quickly through lossy wireless media, the throughput can be increased.

We have not tested this idea but still we can see intuitively that this idea would lead to some gains in the goodput and would reduce packet delays.

7.4 Adaptively changing the FEC scheme

It would be a good idea to use the FEC schemes adaptively depending on the traffic requirements and the channel conditions

- Rate 1/3 convolutional coding can be applied for low-delay services having moderate error-requirements

- A concatenation of rate 1/3 convolutional coding and outer Reed-Solomon coding and interleaving can be applied for high-quality services.
- Turbo codes can be used for high-rate and high quality services.

We have not tested this idea but still we can see intuitively that this idea would lead to some gains in the goodput.

Chapter 8

Implementation Details

8.1 Description of 'ns'

'ns' is widely used simulation tool, developed in collaboration by researchers at UC Berkeley, US California, and Xerox Parc.

'ns' is an object oriented simulator written in C++, with an OTcl interpreter as a front end. The simulator supports a class hierarchy (compiled hierarchy) in C++ and a similar hierarchy(interpreted hierarchy) within the OTcl interpreter.
The two hierarchies are closely related to each other. There is one-to-one correspondence between a class in the interpreted hierarchy and the one in the compiled hierarchy.

'ns' uses two languages because simulator has two different kinds of things it needs to do. On one hand, detailed simulations of protocols requires a systems programming language which can efficiently manipulate bytes, packet headers, and implement algorithms that run over large data sets. For these tasks, the run-time speed is important and turn-around time (run simulation, find bug, recompile, re-run) is less important. C++ is fast to run but slower to change, making it suitable for detailed protocol implementation.

On the other hand, a large part of network research involves slightly varying parameters or configurations, or quickly exploring a number of scenarios. In these cases, iteration time (change the model and re-run) is more important. Since configuration runs once (at the beginning of simulation), run-time of this part of the task is less important. OTcl runs much slower but can be changed very quickly (and interactively), making it ideal for simulation configuration. 'ns' (via *tclcl*) provides glue to make objects and variables appear on both languages.

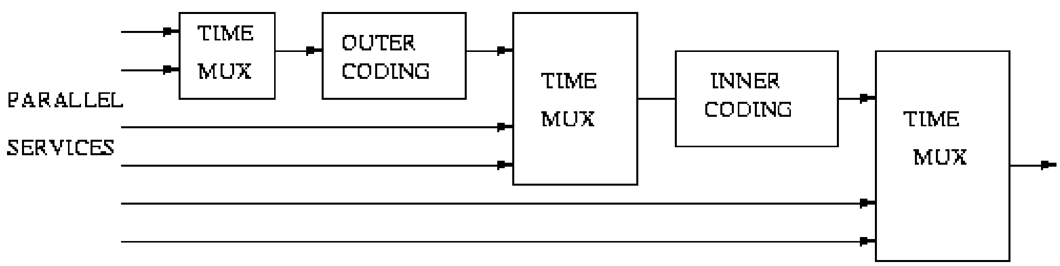
Though a widely used tool among the network community, it is not a finished and polished project yet. It lacks some support for wireless networks. *So, the effort in our project has also been to enhance 'ns'.*

8.2 Modules Implemented in 'ns'

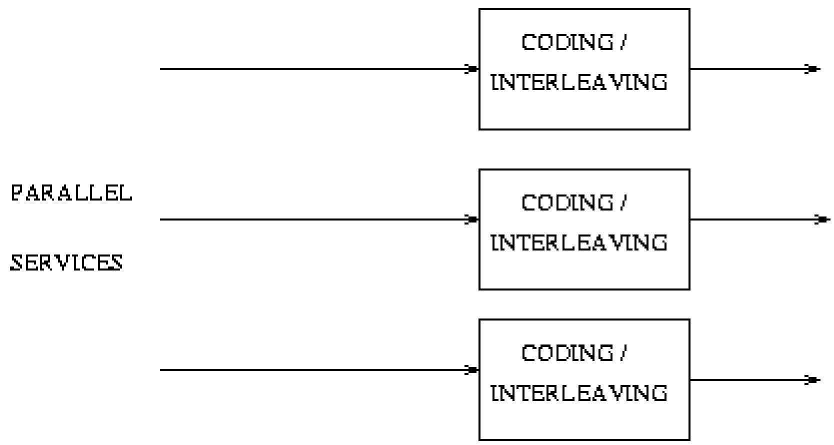
8.2.1. Multirate Services

Multirate design means multiplexing different connections (within the same host) with different QoS requirements in a flexible and spectrum efficient way. Multiple services belonging to the same session can be time-multiplexed or code multiplexed.

Time multiplexing avoids multicode transmission thus reducing peak to average power of the transmission. Since in this case, the cumulative bit rate (i.e after multiplexing) is variable so variable bandwidth is allocated by assigning a set of codes with variable spreading gain. Only one code can be used at a time. The switching is done to a code with appropriate spreading gain to support the instantaneous data rate of multimedia traffic. This is referred to as variable spreading gain (VSG) implementation.



A second alternative for service multiplexing is to treat parallel services completely separately, with separate channel coding/interleaving and map them to separate physical data in multicode fashion. In this case, a number of codes are allotted with fixed (same) spreading gain. One or more of the codes can be used simultaneously to support higher data rate services.



We have provided modules to support multirate services both in a time multiplex and code multiplex.

For time multiplexing, we have added a module 'Multirate.{cc,h}. It is basically a subclass of 'Queue' which consists of many packet queues. We have also added a new header 'QoS' which consists of five fields- *connection_id, ber_ , delay_bound, bw_allotted, scheme_.*

- The *connection id_* field specifies the unique id of the application for a particular node.
- The *ber_* field specifies the bit error rate tolerable by the application.
- The *delay_bound_* field is a boolean variable which specifies whether the data is delay bounded or not.
- The *bw_alloted_* field specifies how much bandwidth is allotted for the a particular application. It should be non zero for real time services for which a connection is needed. It can be zero for example packet data services for whom no reservation are made. Some applications are VBR(Variable bit rate) services which transfer data intermittently, so their unused slots could be used to service the packet data. The packet data can be serviced also on a common shared (contention based) channel.

- The scheme_ field specifies which scheme Sequential(0), Slow Healing(1), Fast Healing(2), Slow Start(3), Fast Start(4) to use.

While creating an application and an agent for it, one has to specify above mentioned qos_parameters (*the interface for this is provided in OTcl*). With each connection_id_ (unique application), for whom bandwidth is allotted, a separate packet queue is created and all the packets corresponding to that connection_id are enqueued in that queue. A separate queue is created in which all those packets are enqueued for whom no reservation is made but are waiting for some empty slots. With all such queues being set up, different application data would be served in a round robin manner according to bandwidth that they have been allotted. If there is no data to transfer during a particular application's turn, the chance is given to other data services (for whom no reservation has been made), who are waiting for these empty slots.

The information such as ber_ and delay_bound_ can be used to ensure guaranteed QoS to each service. For example if bit error rate is specified to be zero and data is delay insensitive (as for packet data services) then Automatic Repeat Request(ARQ) mechanism is used. For other services, different channel coding schemes can be used depending on their delay and bit error rate characteristics. Due to uncertain delivery delays of ARQ schemes, a Forward error correction (FEC) scheme is used for real time services. The purpose of forward error correction (FEC) is to improve the capacity of a channel by adding some carefully designed redundant information to the data being transmitted through the channel (The process of adding this redundant information is known as *channel coding).*

8.2.2 Segmentation and Reassembly

A new type of packet header, 'sar', is added to each packet as it is fragmented. Each packet is fragmented into a fixed fragment of size 160 bytes. If the packet is not a exact multiple of 160, then the last fragment is padded and made of 160 bytes too.

8.2.3 Automatic Repeat Request (ARQ) scheme

In *ARQ scheme*, each MAC packet (fragmented upper layer packet) is assigned a unique id and transmitted. The retransmission of the packet occurs after a timeout or if NACK for it has been received. After some unsuccessful retransmissions, MAC will abort the retransmission and pass the control to higher layers. The receiver side maintains the two sequence numbers, one (s1) for the next sequence number expected to be received and the second one (s2) is the next frame needed for sequential delivery. s2 also represents the oldest missing packet. At the receiving side, if packets received are enough to form a higher layer packet (from which they have been fragmented), then MAC sends the packet to upper layer. Duplicate packets are detected and discarded. Besides this various logs and statistics are recorded to infer some possible results.

8.2.4 Convolutional coding and Viterbi decoding

Convolutional coding of code rate=1/2 is implemented. The code rate, k/n, is expressed as a ratio of the number of bits into the convolutional encoder (k) to the number of channel symbols output by the convolutional encoder (n) in a given encoder cycle. The other parameter of convolutional code i.e. constraint length can be changed at will. Basically, the constraint length parameter, K, denotes the "length" of the convolutional encoder, i.e. how many k-bit stages are available to feed the combinatorial logic that produces the output symbols.

8.2.5 Rayleigh Fading Model

In order to actually simulate the error prone and fading wireless channel, we have implemented some error and fading models. We have modeled the effect of multipath fading on a mobile radio channel by a *Rayleigh distribution*. It is approximated by a first-order Markov process with continuous amplitude. We used a simple two-state Markov chain model for this.

8.2.6 Additive White gaussian noise (AWGN)

Convolutional encoding with Viterbi decoding is a FEC technique that is particularly suited to a channel in which the transmitted signal is corrupted mainly by *AWGN*. One can think of AWGN as noise whose voltage distribution over time has characteristics that can be described using a Gaussian, or normal, statistical distribution, i.e. a bell curve. This voltage distribution has zero mean and a standard deviation that is a function of the signal-to-noise ratio (SNR) of the received signal. We have added the support for AWGN in 'ns'.

Chapter 9

Conclusion and Future Directions

We have *demonstrated* that MC-CDMA (Multi-Code CDMA) based 3G (third generation) wireless communication systems can benefit from *the introduced* adaptations in SMPT mechanisms. Mechanisms that use the wireless resource in a moderate way, such as Slow Start and Slow healing, achieve the best results.

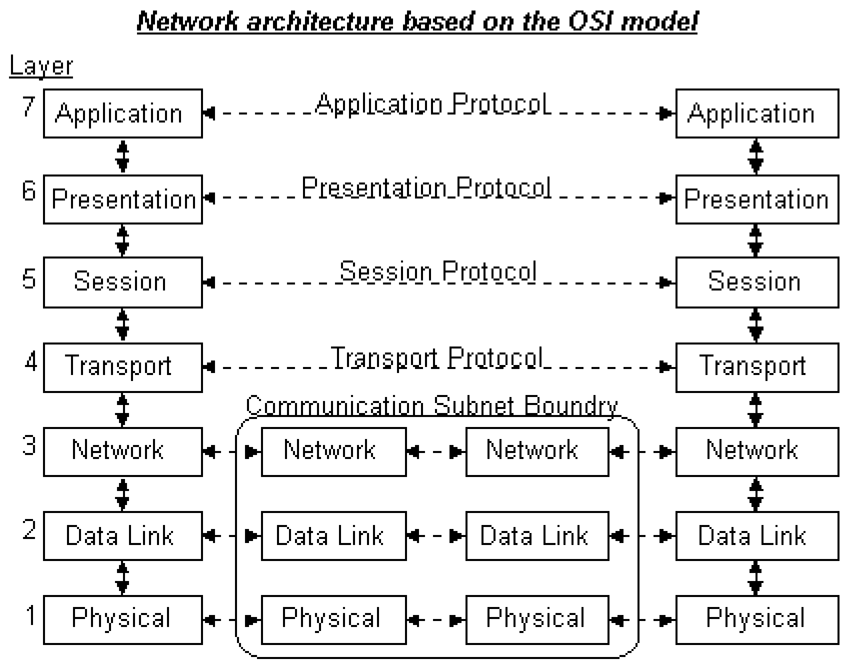
Our work can be extended in several ways. Here we describe few ideas.

- Some of the proposed ideas could not be simulated due to lack of time. One obvious extension is implementing them in 'ns' and comparing their performance with simple scheme.
- In the current work, code allocation schemes are not discussed so work can be done on this too.
- A connection admission control (CAC) policy is needed for a MAC protocol to determine if a new connection can be given channel access without violating the QoS requirements of the existing conditions. For CDMA systems, it becomes all the more complicated and involves rigorous mathematical background.
- The mobile nodes were considered stationary during the current simulations. The simulations could be performed with more complicated topology taking into account the mobility of nodes and the hand-offs. The results, thus obtained, would be more realistic.

We hope that with using the above-mentioned ideas, we can have a new adaptive MAC protocol, which can handle different types of traffic flow and can adapt itself according to changes in the wireless conditions and provide QoS to them.

Appendix

A. 1 OSI reference model



MAC layer is part of Data Link layer.

Bibliography

1. 3rd Generation Partnership Project : Technical Specification Group Radio Access Network ; MAC Protocol Specification , 3G TS 25.321 v3.3.0, www.3gpp.org
2. 3rd Generation Partnership Project : Technical Specification Group Radio Access Network ; Physical Layer – General Desrciption, 3G TS 25.201 v3.0.2, www.3gpp.org
3. 3rd Generation Partnership Project : Technical Specification Group Radio Access Network ; Services Provided by the Physical Layer, 3G TS 25.302 v3.4.0., www.3gpp.org
4. 3rd Generation Partnership Project : Technical Specification Group Radio Access Network ; Radip Interface Protocol Architecture, 3G TS 25.321 v3.3.0, www.3gpp.org
5. Tero Ojanpera, Johan Skold, Jonathan Castro, Laurent Girard, Anja Klein, "Comparison Of Multiple Access schemes for UMTS", IEEE 47th Vehicular Technology Conference Vol 1-3.
6. Ramjee Prasad, Tero Ojanpero, "An overview of CDMA Evolution Toward Wideband CDMA", IEEE Communication Surveys.
7. Erik Dahlman, Per Knutsson, Fredrik Ovesjo, Magnus Persson, Christian Roobol, "WCDMA – The Radio Interface for Future Mobile Multimedia Communications", IEEE Transcations on Vehicular Technology, Vol 47 No 4, November 1998
8. Seizo Onoe, Koji Ohno, Katsuhiko Yamagata, Takehiro Nakamura, "Wideband-CDMA Radio Control Techniques for Third Generation Mobile Communication Systems", IEEE Transcations on Vehicular Technology, Vol 47.
9. Tero Ojanpera, Ramjee Prasad, "An Overview of Air Interface Multiple Access for IMT-2000/UMTS"

10. Christian Roobol, Per Baming Johan Lundsjo, Mathias Jaohansson, "A proposal for an RLC/MAC Protocol for Wideband CDMA Capable of Handling Real time and Non Real Time Services"
11. Frank H.P. Fitzek, Rolf Morich "Comparison of Multi-Code Link Layer Transmission strategies in 3G wirelessCDMA", IEEE Communications Magazine, Oct 2000
12. A. Samukic, "UMTS universal mobile telecommunications system: Development of standard for the third generation," IEEE Transactions on Vehicular Technology, vol. 47, pp. 1099-1104, November 1998.
13. E. Dahlman, B. Gudmundson, M. Nilsson, and J. Sk"old, "UMTS/IMT-2000 based on wideband CDMA," IEEE Communications Magazine, vol. 36, pp. 70-80, September 1998.
14. P. Jung, P. W. Baier, and A. Sreil, "Advantages of cdma and spread spectrum techniques over fdma and tdma incellular mobile applications," IEEE Transactions on Vehicular Technology, vol. 42, pp. 357-363, August 1993.
15. J. Sanchez, R. Martinez, and M. Marcellin, "A survey of MAC protocols proposed for wireless ATM," IEEE Network Magazine, pp. 52-62, November-December 1997.
16. J. De Vile, "A reservation based multiple access scheme for a future universal mobile telecommunications system," in Proceedings of the 7th IEE Conference on Mobile and Personal Communications, (Brighton, U.K.), pp. 210-215, December 1993.
17. N. D. Wilson, R. Ganesh, K. Joseph, and D. Raychaudhuri, "Packet CDMA versus dynamic TDMA for multiple access in an integrated voice/data PCN," IEEE Journal on Selected Areas in Communications, vol. 11, pp. 870-884, August 1993.
18. Z. Zhang and Y.-J. Liu, "Performance analysis of multiple access protocols for CDMA cellular and personal communication services," in IEEE INFOCOM'93, vol. 3, pp. 1214-1221, 1993.

19. A. Brand and A. Aghvami, "Multidimensional PRMA with prioritized bayesian broadcast--a MAC strategy for multiservice traffic over UMTS," IEEE Transactions on Vehicular Technology, vol. 47, pp. 1148- 1161, November 1998.
20. R. Pichna and Q. Wang, "A medium-access control protocol for a cellular packet CDMA carrying multirate traffic," IEEE Journal on Selected Areasin Communications, vol. 14, pp. 1728-1736, December 1996.
21. A. E. Brand and A. H. Aghvami, "Performance of a joint CDMA/PRMA protocol for mixed voice/data transmission for third generation mobile communication," IEEE Journal on Selected Areas in Communications, vol. 14, no. 9, pp. 1698-1707, 1996.
22. J. M. Capone and L. F. Merakos, "Integrating data traffic into a CDMA cellular voice system," Wireless Networks, vol. 1, no. 4, pp. 389-401, 1995.
23. Z. Liu and M. E. Zarki, "Performance analysis of DS-CDMA with slotted ALOHA random access for packet PCNs," Wireless Networks, vol. 1, no. 1, pp. 1-16, 1995.
24. M. Soroushnejad and E. Geraniotis, "Multi-access strategies for an integrated voice/data CDMA packet radio network," IEEE Transactions on Communications, vol. 43, no. 2/3/4, pp. 934-945, February/March/April 1995.
25. N. D. Wilson, R. Ganesh, K. Joseph, and D. Raychaudhuri, "Packet CDMA versus dynamic TDMA for multiple access in an integrated voice/data PCN," IEEE Journal on Selected Areas in Communications, vol. 11, no.6, pp. 870-884, August 1993.
26. W.-B. Yang and E. Geraniotis, "Admission policies for integrated voice and data traffic in CDMA packet radio networks," IEEE Journal on Selected Areas in Communications, vol. 12, no. 4, pp. 654-664, May 1994.