Congestion Avoidance using DSRM for WCDMA Networks

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Abstract: This paper proposes a congestion avoidance mechanism for multimedia traffic that addresses unicast as well as multicast best effort flows specially for Wide band CDMA network. Present schemes uses a single queue per output port to buffer packets destined for that port. This often causes congestion leading to packet loss &delay. Real time applications are very much sensitive with loss &delay occurs during transmission & reception. Therefore a necessity to develop a buffer management system that will an effectively accommodate both real time & non real time applications. In this paper, we propose a advance threshold buffer management scheme. This scheme uses dynamic source routing mechanism (DSRM) to minimize congestion during transmission & reception for wireless networks.

To add fairness to the scheme, high priority packets are dropping during preactive period &low priority packets guaranteed a minimum buffer space. We simulated the scheme and found that DSR mechanism performs well in terms of packet loss, delay, congestion time and buffer utilization.

Keywords: Congestion time, WCDMA, buffer management, DSR, QoS

I. INTRODUCTION

Cellular wireless networks have become an indispensable part of the communication infrastructure. Wideband CDMA is an important air-interface technology for cellular wireless networks. The present trend is towards the deployment of real-time multimedia applications, in addition to the non-real-time applications that the Internet has been transmitting since its inception. Multimedia applications such as voice, video, audio-video on-demand, and video conferencing, demand not only high bandwidth but also real-time delay constraint. In order to properly transmit these real-time multimedia applications, a variety of resources have to be made available. Bandwidth and buffers are the main network resources that need to be considered in the transmission of these applications (flows). Many real-time applications require large amounts of buffers to store packets effectively. Efforts have been made to introduce quality of service (QoS) for WCDMA network and researchers have proposed new architectures for wireless applications. As CDMA-based cellular networks mature, the current point-to-point links will evolve to an IP-based Radio Access Network (RAN). Replacing the point-to-point links between the base stations and the radio network controllers These WCDMA based QoS architectures are designed to make network components QoS aware. The wireless network Task force has been the driving force behind finding a suitable QoS architecture. They have proposed the Integrated Services (IntServ) [1], and the Differentiated Services (DiffServ) [2]. In addition, the Rainbow Services [3] has also been proposed. A common feature in all these proposed QoS models is that, there is the need to allocate resources, either implicitly or explicitly. Appropriate resource allocation is one way of providingQoS in a network. In a network, congestion is created if bandwidth and buffers are not properly managed within the hosts or at the routers of the network. A communication link can become congested if the amount of traffic introduced on the link exceeds its capacity. This excess traffic causes queuing delays to increase rapidly as buffers fill up, and in extreme cases can cause the buffers to overflow, losing packets. When this occurs, the network will not be able to provide the guarantee made to some applications.

Buffer management techniques involve scheduling of resources to applications, packet dropping and buffer Allocation. Scheduling is used to move packets out of the buffers in a specified order and to manage delay within the buffers. Packet dropping is used to maintain the buffer size,

in order to efficiently handle both real-time and non-real time packets. Many buffer management schemes that have been proposed have focused either on packet dropping [4, 5, 6], packet scheduling [7] or both [8]. Buffer allocation,

as an independent mechanism, has not been properly investigated. Buffer allocation is an aspect of buffer management that focuses on managing buffers by allocating

buffer volume (logical unit of space) to each application class, so as to reduce packet loss due to lack of buffer space.

Previously proposed buffer management schemes [4, 5, 6, 8],are not capable of efficiently managing buffers when real-time multimedia applications are involved, as many of these schemes were developed with non-real-time applications in mind. Hence, we propose DSRM, an advance threshold buffer allocation scheme, to efficiently manage buffers in the future wireless network. Though real-time and non-real-time applications share the same buffer space, in the DSRM, they are logically separated. This sharing of common buffer space is done by

dynamically varying buffer thresholds so as to reduce packet loss rate of high priority packets, as well as increasing the utilization of the buffer space. Additionally, the scheme uses the concept of marked packets to prevent the dropping of useful packets when congestion is severe. This paper therefore proposes a class-based QoS control scheme that allows the systematic allocation of buffers for various types of flows. It also manages packet loss rates Within the buffers to efficiently transmit real-time and nonreal-time applications for the wireless Network. The remainder of this paper is organized as follows:
Section II looks at some previously conducted work in this Field Section III describes the proposed scheme with architecture of DSRM scheme Section IV for Simulation model & Flow chart and presents our results in Section V. Finally the paper concludes in Section VI.

II. RELATED WORK

Considerable research effort is currently underway to Develop a suitable buffer management scheme that will accommodate high bandwidth applications on the Internet, As well as reduce large amounts of packet loss during Congestion [9, 10, 11, 12, 13]. Work has also been done on managing both bandwidth and buffer to provide QoS [14, 15]. There is also ongoing research in the wireless field on the allocation of buffers [16, 17, 18]. In this section we review some of the existing buffer management schemes. We focus mainly on buffer allocation strategies in the output buffers of routers on the Internet. The class-based QoS control scheme by Minami *et al.*[10], uses a single buffer that is shared among all the output ports. This scheme uses a hierarchical structure in the order of ports, classes and flows. In this scheme, the total buffer size is shared by output ports according to the ratio of each output link speed. The basic volume is allocated per flow. This increases the complexity of the scheme. Also delay is introduced due to per flow calculations.Yaprak *et al*. [12] proposed a dynamic buffer allocation scheme that uses a shared architecture with virtual partition among output ports. Instead of using pre-emption, they created a shared buffer pool where excess of buffer space
from any logical queue goes into. The problem with this technique is that the shared buffer pool may be monopolized by non-real-time applications thereby reducing the QoS of real-time applications. They neither considered using preemption nor the marking of packets. Complete Sharing using Virtual Partition (CSVP) scheme by Wu and Mark [11] was introduced to manage multi-traffic flows in ATM networks. In this scheme, if there are M number of users and the available buffer space is B, the B buffer spaces are virtually partitioned into m segments corresponding to the M number of users. A newly arriving cell is admitted by pushing out a different type of cell that is borrowing that cells buffer volume. The authors of the scheme did not consider the possibility of marking packets as either IN-profile or OUT-profile. The scheme proposed by Garcia-Marcias *et al*. [16],
shares buffer in a linear manner. In this scheme class packets enter into the output buffer and each of these class packets are allocated buffer volume. This scheme completely partitions its buffer space among the various classes. This reduces the efficiency of the scheme. Furthermore, pre-emption of low priority packets was not considered.The goal of the authors of [18] was to develop a QoS model to support multimedia applications over Cellular IP by using class-based service without reservations. They used per-class buffers with finite sizes. Using static buffers, it is not possible to provide excess buffer to other applications especially real-time applications. Pre-emption was not used.
This paper focuses on the use of mainly buffer allocation to minimize both delay and loss rate. Unfortunately buffer allocation alone cannot totally reduce delay in transmission. It is the proper combination of buffer allocation, packet dropping and scheduling mechanisms that would produce an optimal strategy. This study uses simple packet dropping and scheduling mechanisms so that the effect of the buffer allocation strategy will be largely felt.

III. PROPOSED DSRM SCHEME

An Advance Threshold Buffer allocation is the proposed Scheme with the help of DSR mechanism, which is structured into a hierarchy of ports and classes. The scheme uses a buffer often adjusts timings by implementing a queue or FIFO algorithm in memory, simultaneously writing data in to the queue at one rate & reading it at another rate. A threshold based packet discarding mechanism is used. Applications are grouped into two classes. Each class can access the other class Buffer volume when a class assigned buffer volume is depleted. A routers output buffer consists of ports with multiple queues per port. Each queue is assigned to a class of traffic. Hence ports consist of classes and classes consist of packets. We do not group class packets into flows. Packets in a class are treated as a single group, just as in DiffServ [2]. In this study only two classes are used (Class 1 and Class 2). Class 1 has a higher priority than Class 2; Class 1is for real-time applications while class 2 is for non-realtime applications. Packets of Class 1 flows are marked as either IN-profile or OUT-profile. During congestion period, OUT-profile packets are dropped first before IN-profile packets so as to minimize the loss rate of useful IN-profile packets. High-priority Class 1 packets can pre-empt low priority Class 2 packets.

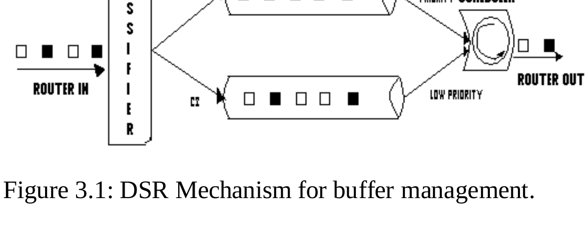


Figure 3.1: DSR Mechanism for buffer management.

Figure Shows DSRM Scheme with the help of router. It consists of a classifier, two logical buffers (C1 and C2) and a scheduler. Arriving packets are classified as either class 1 and sent to C1 logical buffer or as class 2 and sent to C2. Packets of each logical buffer are then scheduled. This scheme uses virtual partitioning to separate the buffer space for packets of each class. We used Drop tail with dynamic threshold todrop the packets of Class 1 and Class 2 flows. When a Class 1 high-priority packet enters its designated buffer space and finds it full, it looks for any available space in Class 2.s logical buffer, to borrow. If there is an available space within Class 2.s logical buffer,packets of Class 1 will then use this space and the upper threshold of Class 1 is increased accordingly. The space can be borrowed until the minimum threshold of class 2 is reached. During this period, Class 1 starts dropping its OUT-profile packets. When the minimum threshold of Class2 is reached, all arriving class 1 packets will be dropped. If Class 1 buffer is free, Class 2 packets could also use it.

V SIMULATION MODEL AND FLOWCHART

To evaluate the performance of our model. The simulation design consisted of mainly source, a router and buffer as shown in Figure 2.

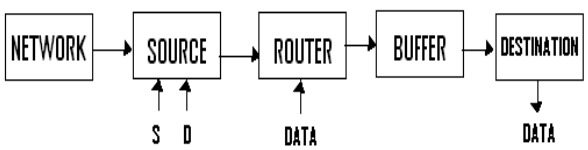
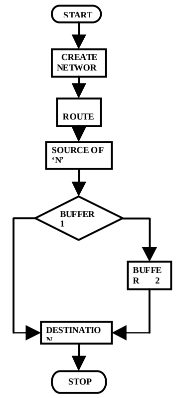


Fig 4.1: Simulation model for DSRM Scheme

For above fig wireless network refers to any type of computer network that is wideband WCDMA network whose interconnections between nodes are implemented without reuse of wires each node participates in routing by forwarding data for other nodes and so the destination of which nodes forward data is made dynamically based on the network connectivity. And each data packets contains address information that a router can use to determine if a source and destination are on the same network , the routers exchange information about regret system address . Here we use buffering. There may be overlapping if the limits exceed which causes loss of data. To overcome this problem we use buffer management. Here source is using for information processing which encode message data and transmit the information via channel to more receivers at the last the destination must separate the desired data from others, amplify it and demodulate it & to recover the original data.

FLOWCHART FOR DSRM



ALGORITHM

Step 1: start.

Step 2: create a network.

Step 3: specify source and destination.

Step 4: selects final path.

Step 5: source contains ‘n’ packets.

Step 6: select buffer 1 (intermediate node 1). If n=1 then go to step 8 else go to step 7.

Step 7: select buffer 2 (intermediate node 2).

Step 8: destination.

Step 9: end.

V. RESULT

A. Real time streaming-

Streaming media are multimedia that are constantly received by, and normally presented to, an end-user while being delivered by a streaming provider (the term "presented" is used in this article in a general sense that includes audio or video playback). The name refers to the delivery method of the medium rather than to the medium itself. The distinction is usually applied to media that are Distributed over telecommunications networks, as most Other delivery systems are either inherently streaming (e.g., radio, television) or inherently non-streaming (e.g., books, video cassettes, audio CDs). The verb 'to stream' is also derived from this term, meaning to deliver media in this manner. Internet television is a commonly streamed medium Security remains one of the main challenges with this new methodology however DRM systems are the best way to keep the content secure.

B .Streaming Band-Width

Streaming media storage size (in the common file system measurements mega bites, megabytes, gigabytes, terabytes, and so on) is calculated from the streaming bandwidth and length of the media using the following formula (for a single user and file):

- storage size (in megabytes) = length (in seconds) × bit rate (in bit/s) / (8 × 1024 × 1024)
- Since 1 mebibyte = 8 × 1024 × 1024 bits.

If the file is stored on a server for on-demand streaming and this stream is viewed by 1,000 people at the same time using a Unicast protocol, the requirement is:

300 kbit/s × 1,000 = 300,000 kbit/s = 300 Mbit/s of

- bandwidth
- The calculation for Live streaming is similar.
- Assumptions: speed at the encoder, is 500kbps.
- If the show last for 3 hours, with 3000 viewers then the calculation is:
- Number of MB transferred = encoder speed (in kbps) * number of seconds * number of viewer/(8*1024)
- Number of MB transferred = 500 (kbps) * 3*3600 (= 3 hours)*3000 (nbr of viewers)/(8*1024) = 1977539 MB

C. Buffer Utilization with the help of DSRM

In this scheme the main purpose of communication is the adjustment of routing and buffer management.

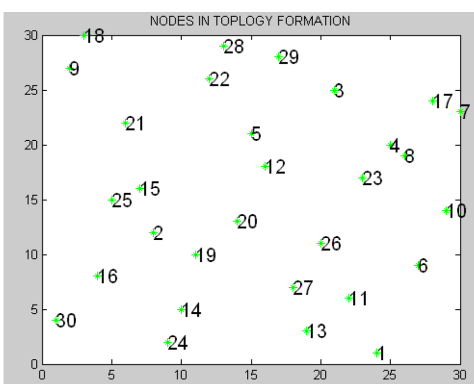


Figure 5.1-Network of the system.

The figure 5.1 shows the plot of the network of the system. In this network is created by selecting the nodes randomly. This is the main step in the DSRM scheme. So only after creating the network. We can proceed for routing and buffer management.

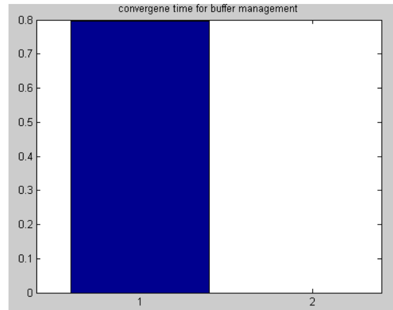


Figure 5.2 -Convergence time for buffer management.

The figure 5.2 shows the plot of convergence time for buffer management. In this we can know the required time for the buffer management. After appearing this graph we have to analyze the result by clicking on the option named as analyze result.

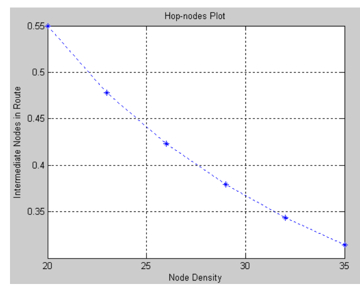


Figure 5.3 -Hop-nodes plot of the system.
The figure 5.3 shows the hop-nodes of the system.

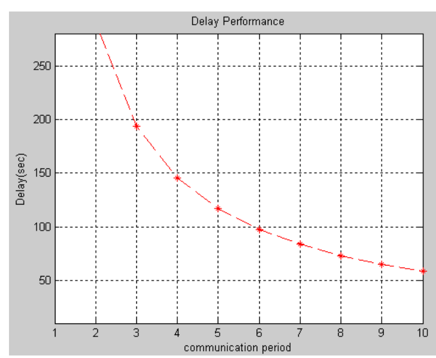


Figure 5.4 -Delay performance of the system.

The figure 5.4 shows the plot of delay performance of the system. In this plot we can know the time delay (in seconds) of communication of the systems. By this the results and conclusions of the system gets completed.

VI. CONCLUSION

Congestion is the major drawback in case of multimedia communication. It affects the quality of real-time& non real time multimedia applications. Many mechanisms have been proposed but very few have use of buffer allocation strategies to reduce the effect of congestion. The Advance buffer management Scheme presented and evaluated in this paper allows effective QoS management with the help of DSRM scheme to evaluate packet loss rate (PLR) in terms of delay & congestion time & buffer utilization. Simulation results show that the DSR scheme is the best scheme for Wideband WCDMA Network in future wireless network applications. Future work on Advance buffer allocation scheme could use delay as a performance metric. The DSRM scheme performance could further be investigated using a combined packet dropping and scheduling mechanism.

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