Performance of an Admission Control Scheme for Bandwidth in Multimedia Handover Services in UMTS mobile Cellular Network

Edwin Omosa, Kibet Langat and Stephen Musyoki

Abstract—The ever growing multimedia services in the telecommunication industry have congested the mobile cellular networks in proportions never witnessed before. This worrisome trend has resulted in degraded services especially during handover in networks as bandwidth is limited. To avoid starving critical services after handover, bandwidth has to be shared fairly. Specifically, users requiring more bandwidth for transmission should not be limited by bandwidth, conversely, those requiring less bandwidth should not be over allocated. A worthwhile strategy to these challenges would be a properly designed admission control that meets Quality of Service (QoS) requirements. An admission control ensures that the acceptance of a new traffic into a network cell does not deny services to already established traffic and that the network cell can satisfy its different QoS requirements. The paper presents the design, implementation and performance of an admission control algorithm based on bandwidth broker that provides QoS of multimedia traffic in a cellular network during handover. The algorithm manages multimedia traffic through Differentiated Service (DiffServ) policies. Instead of having admission mechanisms in all core routers processing traffic requests as in Integrated Service (IntServ), a centralized Bandwidth Broker (BB) is preferred. The bandwidth is partitioned between different multimedia traffic.
Keywords—Admission control, Bandwidth Broker, DiffServ, QoS
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I. Introduction

Establishment and management of connections is important if quality of service is to be maintained in cellular network as mobile equipment are in constant motion during communication sessions experiencing handovers from one cell to another. However, if available bandwidth is insufficient to accommodate the handovers, forced termination of services occurs. For better bandwidth provisioning, the adaptive bandwidth allocation scheme should work in conjunction with an admission control scheme. In an all-IP UMTS mobile cellular network the admission control should be simple and scalable. Admission control is a mechanism for determining of whether a new traffic flow requesting service should be admitted or rejected. In other words it allows the acceptance or refusal of new traffic into a network. For mobile users' satisfaction, traffic flows should be granted the requested QoS without affecting earlier guarantees [1]. The main aim of admission control algorithm is to meet mobile uses satisfaction while maintaining efficient bandwidth use. The factors considered before providing requested QoS includes but not limited to current traffic load, current QoS and requested QoS.

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In the past, voice and data services had distinct network infrastructures for their support, of late the trend is to use the same infrastructure to provide services in packet switched network that is IP-based. For a long time, IP networks have only supported best-effort services, that is, different services with different QoS requirements get equal treatment by the network. Since multimedia traffic demands QoS guarantee from a network, for it to be supported successfully, it is necessary to provide QoS guarantees between the end-systems. One of the important factors to improve the quality of service of an IP-based UMTS mobile cellular network is to vary the allocated bandwidth. This makes DiffServ the best QoS provisioning strategy in Internet Protocol (IP) networks for support of multimedia services.

DiffServ has been the preferred model for implementing IP-based UMTS mobile cellular networks as UMTS service classes can be mapped easily into DiffServ service classes [2]. To provide better scalability than other mechanisms like IntServ, DiffServ concentrates with individual traffic flows at the edge router while core routers do the forwarding. Since a BB manager is in use in DiffServ, it automatically becomes the admission control of choice. BB may admit and maintain the traffic flow at a reduced level of its service until its stolen bandwidth is returned. When the bandwidth of a link becomes exhausted, low priority traffic can be downgraded to lend some of its bandwidth to high priority traffic. For example, when there is not enough bandwidth for traffic flow, instead of discarding the traffic flow, BB may admit traffic and degrade its service until enough bandwidth become available. This is important for real time traffic flows. This can significantly reduce network congestion. The flexibility of multimedia traffic allows it to tolerate the limitation of bandwidth by upgrading or downgrading system bandwidth for QoS guarantee. The System performance is improved through favoring high priority traffic at the expense of low priority traffic [3]. A traffic flow is only prohibited into the network only if bandwidth is scarce even after degrading its services

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This paper demonstrates that the admission control algorithm developed determines whether the available bandwidth in each link is enough to admit new traffic flow while providing the requested QoS.

II. RELATED WORKS

The problem of intelligent usage and allocation of the available bandwidth in a wireless environment is still a challenge due to mobile subscriber mobility and radio born errors. An ideal concept is developed in [1] to optimize bandwidth use by implementing a fuzzy logic controller of the crucial factors affecting the performance. It uses MATLABs fuzzy logic toolbox for the investigation the system. But this concept comes with its shortcomings as there is no systematic approach to fuzzy logic design, instead empirical approaches are used. A strategy for call admission control using Diffserv in wireless networks is presented in [2]. Here traffic is classified into Transmission Priority flow (TP) and Bandwidth Priority flow (BP). TP flows denote real time flows and BP denote non- real time flows. The Control mechanisms are RIOC queuing and Time Sliding Window (TSW) algorithm for TP and BP respectively. In the proposed scheme, same admission control mechanism is applied for both non-real time and real time traffic.

Broker architecture for quality of service has been proposed in [4] with suggestions and improvement in the existing architectures like Common Open Policy Services and DiffServ technologies. The emphasis is on resource allocation and resource admission control involving admission control servers located at different levels of hierarchy. Having many servers located at different levels of hierarchy for admission control instead of one central server increases complexity of the scheme. This in turn results to additional costs. In the scheme formulated by Okumus and Dizdar, they attempts solve the preemption and QoS problem through intra-domain resource manager (IDRM) [5]. IDRM monitors the available capacity and the reserved resources. It then uses this information for admission control. The main drawback of this scheme is the scalability problems. There is a lot of signaling overhead.

In [6], the paper weighs parameter based admission control (PBAC) with measurement based admission control
(MBAC), as well to situation when no admission control is used. PBAC does not guarantee optimal bandwidth utilization due to unpredictability of new traffic and for MBAC; there are high chances of making errors when taking measurements. An admission control techniques for UMTS is presented in [7]. The algorithms are verified through fuzzy logic and genetic algorithms. The simulations are through MATLAB. Just as is the case for any artificial intelligent techniques, both fuzzy logic and genetic algorithms cannot promise constant optimization response time. This limits them in real time applications.

III. DIFFERENTIATED SERVICES AND QOS

From the definition according to [8] DiffServ, is an IP QoS architecture based on packet marking that allows packets to be prioritized according to user requirements. The architecture provides QoS by classifying traffic in some order, then marking it with code points according to its class. The order of classification then determines the level and QoS that the traffic receives in the network. When the network becomes overloaded, more low priority traffic is dropped than high priority traffic [8].

The DiffServ module in ns consists of three major parts:

Policy:

Policy is specified by network administrator to specify which traffic receives a particular level of service.

Edge router:

Classify packets by marking them with a code point to reflect the desired level of service.

Core router:

Differentiate incoming packets based on code point and forwarding them accordingly.

IV. BANDWIDTH BROKER

In the DiffServ framework, resource management module is absent [5]. That is to say that there are no admission control strategies to regulate traffic in the network. For admission control, BB can do [5]. The BB is defined by the Internet2 Qbone BB Advisory Council [9]. BB collects data about QoS state. It uses this information to allow or prohibit new traffics into a network. From a routers perspective, QoS support consists of three basic components: categorize traffic, define resources for each category, and placement of traffic into its corresponding category.

Research has been done on bandwidth broker and a lot of schemes on its working have been suggested. BB manages the bandwidth in a particular DiffServ environment by regulating traffic through prohibiting or allowing a bandwidth request. The BB should control multimedia traffic through DiffServ policies defined based on priority of the traffic. It evaluates the available bandwidth and depending on the classification of the traffic, it prioritizes it accordingly. A BB is composed of policies for particular per hop bahaviors and a broker manager to assist in communication with other BBs. The most important advantage of a BB is that it eliminates bandwidth reservation in core routers, through managing data in a centralized system.

The Main modules of a BB are the admission control and routing [9]. The former maintains the QoS state of the network domain and is responsible for the admission control and resources reservation. The latter decides the path that the admitted flow will traverse towards the receiver.

V. BANDWIDTH BROKER BASED ADMISSION CONTROL

The purpose of a BB is admission and controlling of flows and also routing them. This is crucial because it ensures fairness in bandwidth usage between call requests and the overall bandwidth utilized by the whole network. The BB also stores data about the network including traffic flows and QoS. Most Bandwidth Brokers use simple admission control modules that accounts for the network condition and the pre-defined Service Level Agreement (SLA). [8].

VI. DESCRIPTION OF THE WORKING OF AN ADMISSION CONTROL MODULE

When a new flow requests admission, a QoS request message is sent to the BB. The request message contains details of source/destination IP addresses, source/destination ports, requested rate and burst size, and finally the time duration for the session. The BB authenticates the request message and recalculates the available bandwidth in each link. It then checks if there is a path where the new flow can be admitted or not and if there exists unallocated bandwidth sufficient to meet the request. If a request passes these tests, allocation is done; otherwise the available bandwidth is reduced by the requested amount. If the reduced bandwidth is not enough, the request is rejected. Once a request has been accepted, the Bandwidth Broker has to make sure that it will be met by the network. Admission controls significance to the Bandwidth Broker operation is to ensure fairness between the requests and the degree of network utilization. Fairness in this case is downgrading of the admission control service for the users. Finally, the BB sends a message to the sender and updates its database.

VII. METHODOLOGY

A. *Traffic Model*

Since a mathematical analysis of a real network is difficult, proper simulations are preferred. The focus for this paper is simulation of IP-based Radio Access Network of UMTS functionality on NS-2. The base stations are integrated with IP routers. Each base station router maps the UMTS packets into IP packets for transportation in the IP network technology. In this simulation, different scenarios have been designed for different classes of traffic and the performance is measured using parameters such as end to end delay and jitters which are explained below .The simulation model consist of 6 routers$Edge_1$ (BS_1); $Edge_2$ (BS_2); $Edge_3$ (SGSN), $Edge_4$ (BS_3); $Core_1$ (RNC_1); $Core_2$ (RNC_2) and six source nodes (S_0 - S_5) and three destination nodes (D_0 - D_2) has shown in figure 1. The research makes use of ns-allinone-2.35 release that contained some of these tools:

- Nam. Used for representations of the network topology and operations.
- X-graph. Used for graphical representations of simulated data

Also the topology is composed of one centralized BB configured at edge routers SGSN and BS_3 .Three multimedia traffic are simulated. They include

1) S_0 - D_0 source destination pair which represents voice traffic with EF PHB,
2) S_1 - D_1 source destination pair which represents video traffic with AF PHB, and
3) S_1 - D_2 source destination pair which represents web traffic with BE PHB.

After handover the traffic sources S_1, S_2 and S_3 are not moved but are shifted with a certain probability. As shown in the figure 1 S_0 is shifted to S_3, S_1 is shifted to S_4 and S_2 is shifted to S_5. To form the source destination pair S_3 - D_0, S_4 - D_1 and S_5 - D_2 respectively.

The voice traffic is marked with DSCP 10, video traffic is marked with DSCP 20 and web traffic is marked with DSCP 30. If a traffic flow refuses to conform to its profile as defined, it is assigned a reduced bandwidth. For example voice with DSCP 10 is downgraded to DSCP 11; video with DSCP 20 is downgraded to DSCP 21 voice and in the same way web with DSCP 30 is downgraded to DSCP 31. If a traffic flow does not still conform to the downgraded traffic profile then it is discarded.

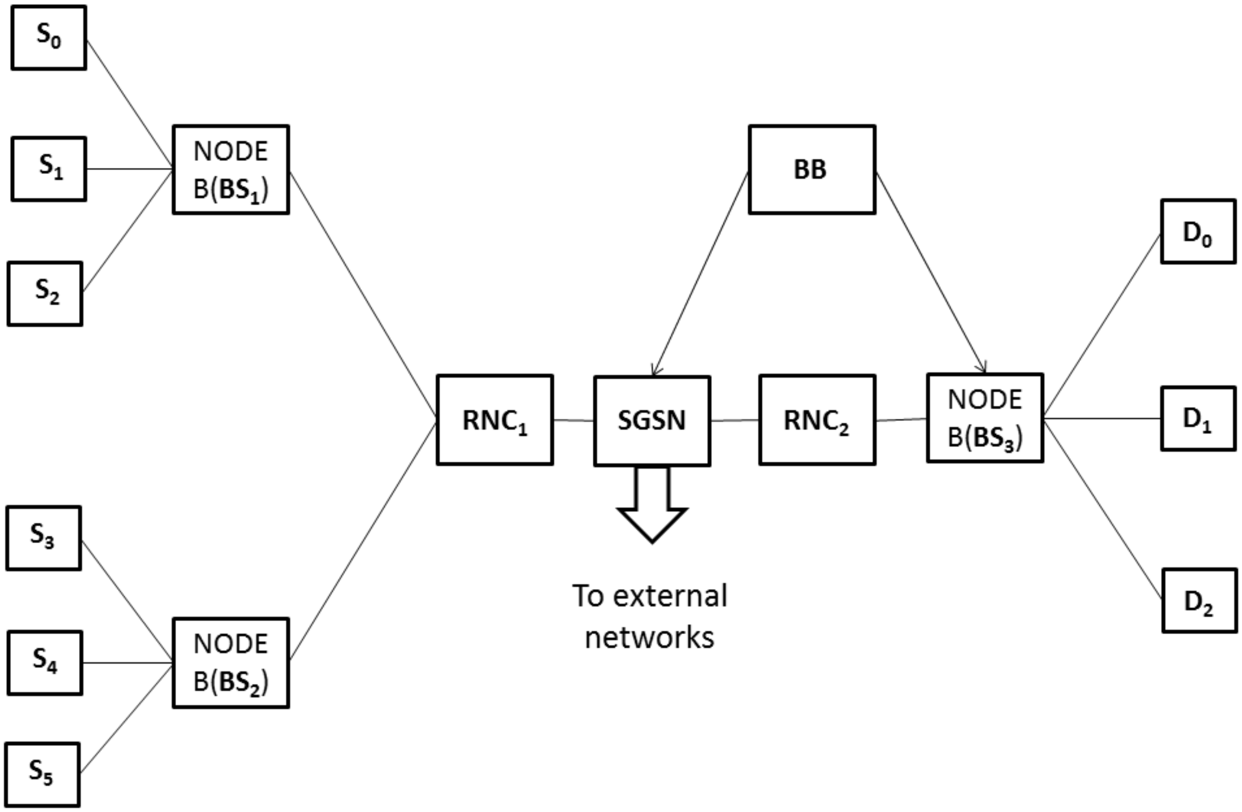

Figure 1: Simulation Model for an IP-based Radio Access Network of UMTS network with Bandwidth Blocker

B. Mobility Model

To simulate handoff, mobile nodes that represent mobile users are not moved instead part of traffic (source node) is shifted from its current cell's BS to another mobile node attached to a neighboring cell's BS with varying probability. With reference to figure 3, this mean that if a mobile source from S0 were to initially send traffic to destination D0, after a handoff, S3 which is located in neighboring cells could be sending traffic to destination D0. This is similar to if D0 were to send traffic to destination S0, after a handoff, it could be sending traffic to destination S3 which is located in neighboring cells.

In this model, the rate and direction of handoff is controlled by probability. A high handoff rate represents a high probability of handoff and vice versa. A congestion of traffic in cell is a representation of traffic moving in the direction of the cell as opposed to other directions.

C. Admission Control model

Admission control module decides when to prohibit or allow a flow [6]. On receiving a traffic flow request along with it service level requirements, BB checks for current assigned bandwidth of traffic specified with it PHB. It then uses configured policies to determine the available bandwidth for new traffic arrivals. These policies then allocate bandwidth according to criticality of the traffic to ensure fairness. This this saves bandwidth to be assigned to later flows.

The only difference between the figure 2 and the standard DiffServ module is the added admission control module which works with the meter. When traffic arrives at an edge router, the meter measures traffic to determine whether the traffic is in-profile or out-of-profile and passes the measured traffic information to the BB to conduct admission control. The measured information includes the aggregate class of the traffic. If an incoming traffic is out-of-profile and access is denied to its appropriate class, the traffic QoS can be downgraded to accommodate the incoming traffic. If the traffic cannot be downgraded below a certain threshold the traffic is dropped. For the case of downgrading, remarking will be done or, traffic is dropped at the dropper. The marking depends on the classs specified initial code point or a downgraded code point [8].

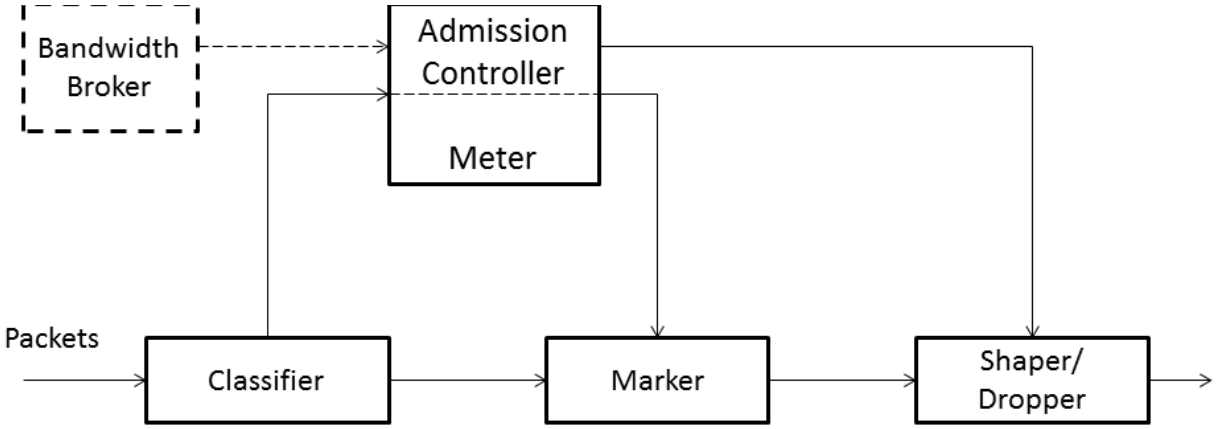

Figure 2: Admission Control in DiffServ

D. Simulation Parameters

The simulation settings and parameters are summarized in table 1.

Table 1: Simulation Parameters

PARAMETER	Specification
Users Per Cell	3
Number of Cells	3
Simulation Time(seconds)	10
Packet Size	512 bytes
Traffic Source	FTP, CBR and Exponential
Rate in cell before handover	1.8Mb
Rate in handed-over cell	2Mb
Queue Length voice	20 (min) - 40 (max)
Queue Length video	11 (min) 20 (max)
Queue Length web	1 (min) 5 (max)

VIII. RESULTS AND DISCUSSION

The performance parameters for the three types of traffic during 10 seconds simulation time are: end-to-end delay and jitter.

End to end delay is the time taken by a data to arrive at its destination. A lower value of end to end delay implies better performance.

Packet end-to-end delay is calculated as:

$$Delay = \text{packet transmission time} \quad \text{packet arrival time}$$

Jitter is the absolute value of the difference between the arrival time difference of two adjacent packets in a traffic flow and their departure time difference. A lower value of jitter implies better performance. It is calculated as:

$$Jitter = (a_k - a_j) - (d_k - d_j)$$

Where j and k are consecutive traffic packets arriving at a node, a_k and a_j are the arrival times of the j and k packets at the node respectively and d_k and d_j are the departure times of the j and k packets at the node respectively.

In these results analysis two scenarios have been created. One is with Admission control and another is without admission control. Different sets of results from the two scenarios give us the opportunity to investigate the performance of these schemes. To analyze quantitatively the simulation results, the traffic is traced during the transmission process. For every packet that passes a trace object, information about the packet is written to the specified trace file. Final output results from trace files are visualized in plotted graphs. Also it is of great importance to note that this analysis concentrates on voice handover and how it affects other multimedia traffic.

A. Jitter

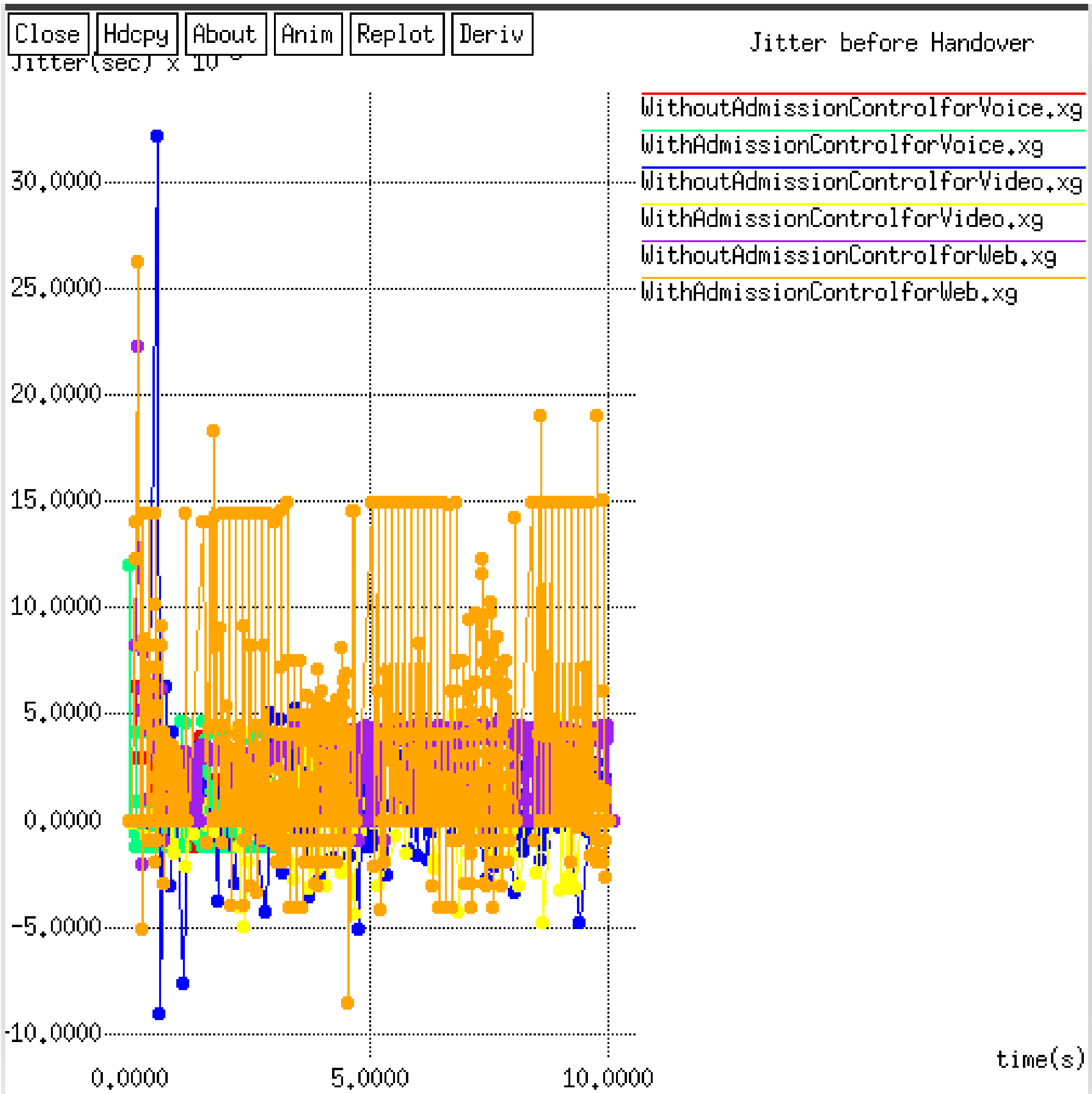

Figure 3: Jitter before Handover

From Figure 3, web traffic has the worst performance in terms of jitter with admission control than without admission control. The values range from 15 to -5 milliseconds approximately with admission control while without admission control the range is from 4 to-1 milliseconds approximately. For real time traffic (voice and video) it ranges from 5 to -5 milliseconds approximately. It should be noted there are jitter spikes in between this values due to traffic bursts. With the prioritization of real-time traffic during admission with admission control, web is given less preference hence it waits longer in queues. Without admission control jitter is less for web compared to with admission control because both real and non-real time traffic are treated equally.

From Figure 4, web traffic has the worst performance in terms of jitter with admission control than without admission control after handover. The values range from 15 to -5 milliseconds approximately with admission control while without admission control the range is from 5 to -1 milliseconds approximately. For real time traffic (voice and video) it ranges from 5 to -7 milliseconds approximately. The major difference between jitter before handover and jitter after handover is that as there is more traffic in the cell after handover than before, web traffic experience higher traffic burst (spikes). Also due to preference of real-time traffic with admission control, non-real time traffic (web) waits longer in its queue. It can be seen that jitter is variable, which is to say that the delay between different packets admitted is different.

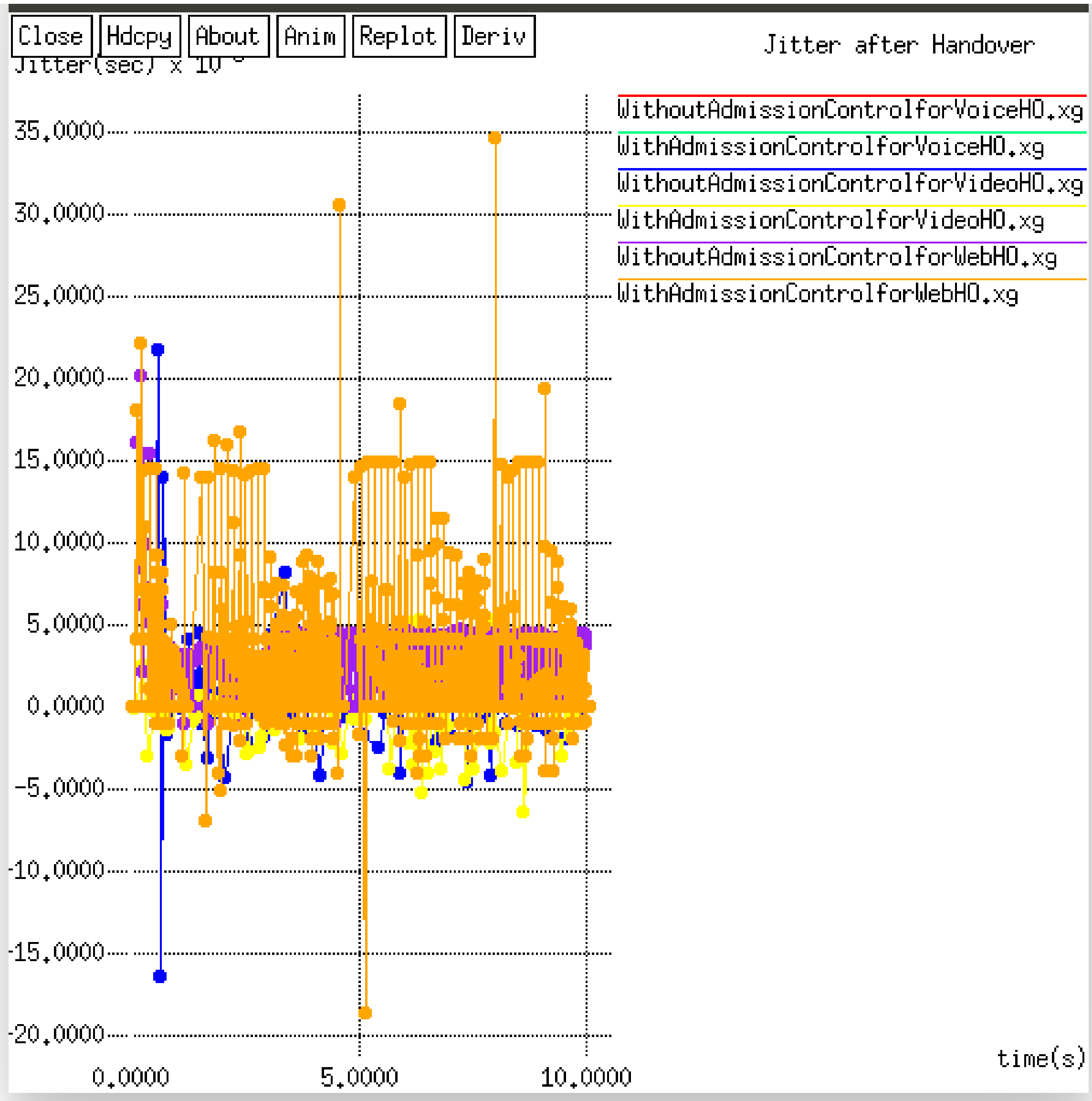

Figure 4: Jitter after Handover

B. End to End Delay

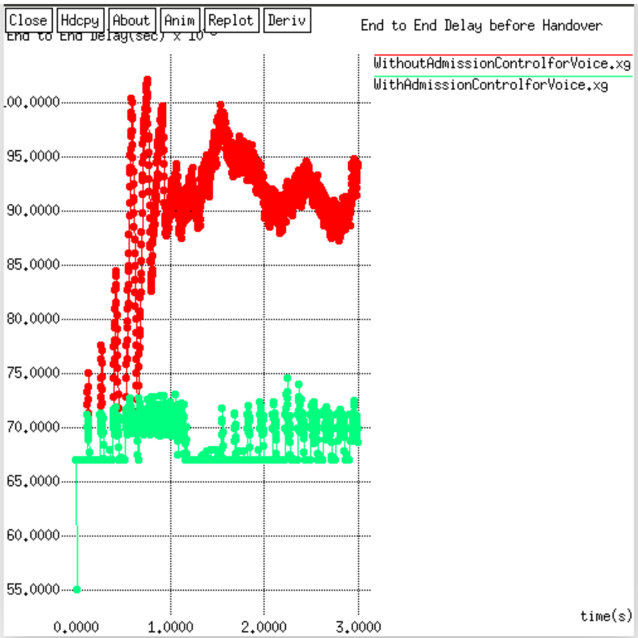

Figure 5: Delay for Voice before Handover

Figure 5 shows the end-to-end delay of voice traffic with admission control in a cell before handover being less than without admission control. The values range between 67 and 73 milliseconds when admission control is employed. Without admission control the end to delay varies greatly from 0 to 0.8 seconds (between 70 to 103 milliseconds), afterwards it stabilizes between 87 and 95 milliseconds with high spikes at about 2.5 seconds. The lesser end-to-end delay of voice traffic with admission control is due to voice being prioritized while without admission control; it is on best effort. The graph ends at the 3 seconds because voice is handed-over to another cell at that point.

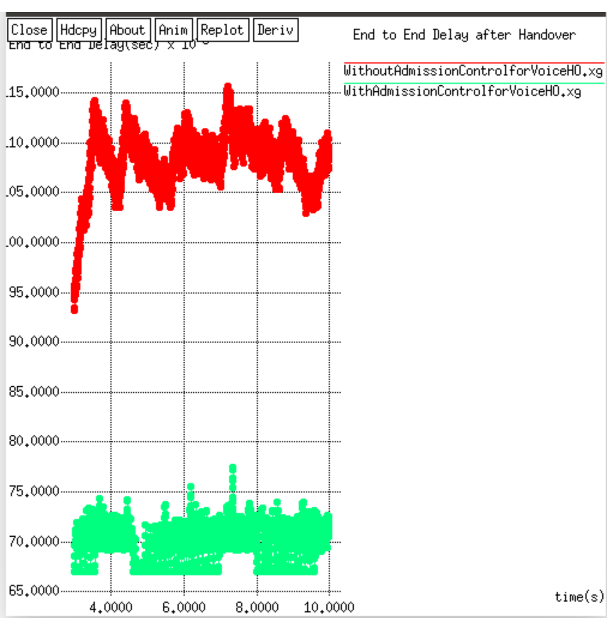

Figure 6: Delay for Voice after Handover

From Figure 6 the plotting starts at about 3 seconds, this is the time that the destination cell receives handed-over packets. After handover, the end-to-end delay with admission control is less than without admission control.

The end-to-end delay with admission control fluctuates between 67 and 75 milliseconds with spikes in between while without admission control it fluctuates between 103 and 114 milliseconds. Over all end-to-end delay of voice traffic is high after handover than before as the handed-over cell becomes congested.

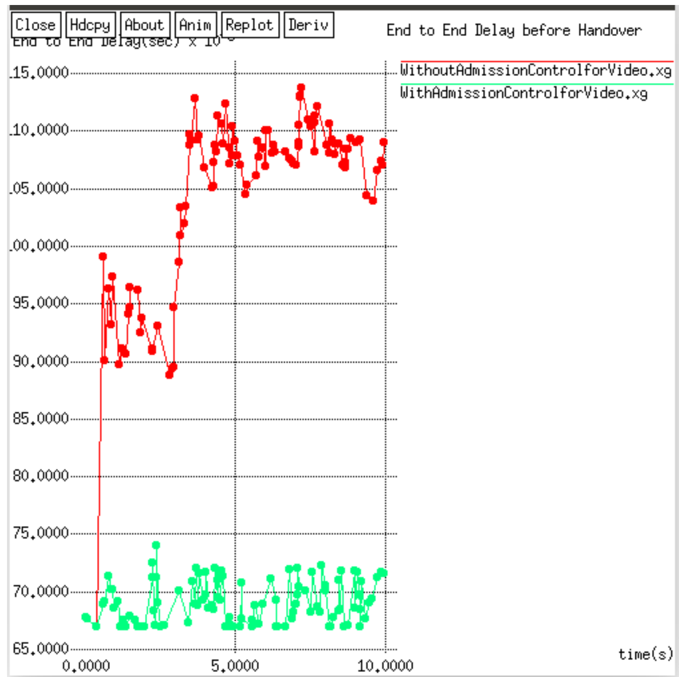

Figure 7: Delay for Video before Handover

It is observed from figure 7 that end to end delay for video traffic with admission control is less than without admission control. The end to end delay for video traffic with admission control is nearly uniform ranging from 67 to 73 milliseconds while for without admission control the end to end delay for video traffic varies non-uniformly, this is due to regulation of video traffic with admission control.

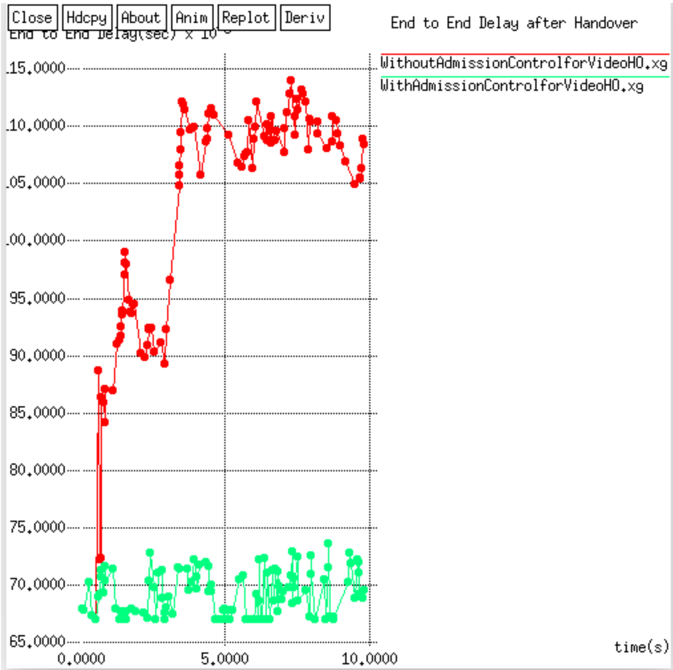

Figure 8: Delay for Video after Handover

It is observed from figure 8 that end to end delay for video traffic with admission control ranges from 67 to 74 milliseconds while for without admission control, there are high end to end delay with spikes for the first three seconds, then stabilizes after three seconds between 105 to 114 milliseconds. It can be concluded that admission control give video high preferential service with lesser end to end delay, realizing a higher QoS as congestion is reduced.

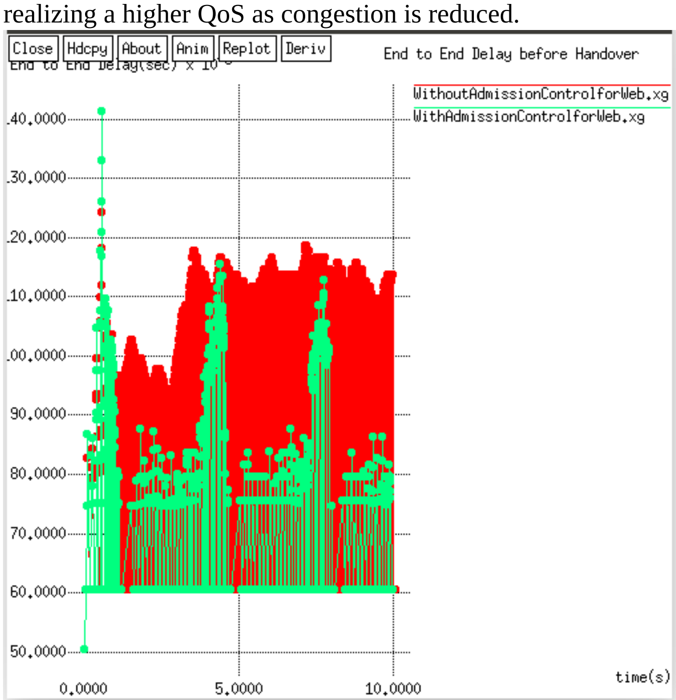

Figure 9: Delay for Web before Handover

It is observed from figure 9 that end to end delay for web traffic with admission control is less compared to without admission control. The major difference between is that with admission control there are times when delay varies uniformly at about 60 and 87 milliseconds and experience high delay spikes at about 1, 4 and 7 seconds when the networks experiences burst data. Also after handover, the end to end delay for web traffic with admission control is less compared to without admission control. The major difference between end to end delay before and after handover is that end to end delay for web traffic with admission control is more after handover than before, ranging between 60 to 90 milliseconds with end to end delay spikes in between the duration of simulation compared to 60 and 87 milliseconds with end to end delay spikes in between the duration of simulation before handover. The with end to end delay spikes is due to the variable nature of web traffic.

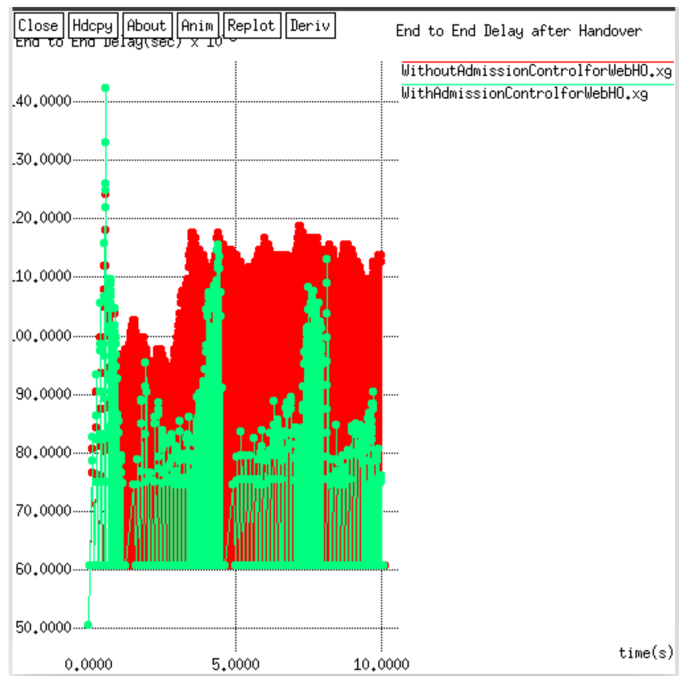

Figure 10: Delay for Web after Handover

IX. Conclusions and Recommendations

Multimedia traffic demands QoS guarantee from a network. For multimedia traffic to be supported successfully, it is necessary to provide QoS guarantees between the end-systems. By using admission control, multimedia traffic can be assigned bandwidth according to demand and controlled before being admitted into a network. Admission control directs traffic to confirm to defined profile or downgrade it but still maintaining an acceptable quality of service in the network. If traffic goes out of its defined profile then it can be downgraded and finally discarded thus bringing the traffic under control. Thus with admission control traffic congestion can be minimized, this improves the QoS especially for high priority traffics.

The admission control mechanism used in this paper is the bandwidth broker. It is responsible admission of user request and management of bandwidth and assignment. Whenever traffic flow request admission, it first ask the bandwidth broker which determine if the traffic flow can be allowed. To evaluate the performance of the admission control scheme, jitter and delay are used through simulation in ns2, the results show that using the proposed admission control scheme reduces both the delay and jitter guaranteeing users QoS requirements. As the Admission control mechanism investigated is centralized, future work should focus on a distributed Admission control mechanism.

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