Performance of Adaptive Bandwidth Allocation for Multimedia Handover Services in UMTS mobile Cellular Networks

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Abstract With the ever growing multimedia services in the telecommunication industry, Quality of Service (QoS) provisioning in cellular networks is becoming more challenging. Studies have suggested that QoS requirements can be met by increasing network capacity through adopting micro/pico/femto-cellular architectures. A consequence of using small cell sizes is the increased rate of call handovers as mobile terminals move between cells. In a network supporting multimedia services, the increased rate of call handovers not only increases the signalling load on the network, but also adversely affects the QoS through sky rocketing handover failures. For seamless connection, one of the solutions will be adequate provision of resources like bandwidth. Moreover, bandwidth being a scarce resource, it should be used efficiently.As the world migrates from Asynchronous Transfer Mode (ATM) transport to Internet Protocol (IP) in UMTS Radio Access Networks (UTRAN), it will introduce more challenges in assigning bandwidth to multimedia services. This is true as IP is of best effort nature. A better solution will be to allocate bandwidth discriminatingly, that is, high priority traffic gets better service while low priority traffic gets reduced service. This paper presents the design, implementation and performance of an Adaptive Bandwidth Allocation scheme for Multimedia Handover Services on an IP-based Radio Access Network of a UMTS Cellular Networks. The methodology adopted is classifying multimedia traffic into different classes using differentiated service (DiffServ) scheme. This is then tested and validated through simulation in Network Simulator 2(NS-2).From the simulation results it has been proven that the adaptive bandwidth allocation minimizes packet losses and give high throughput than non- adaptive bandwidth allocation. It has also been observed that non- adaptive bandwidth allocation does not guarantee service when the network experience congestion due to increased traffic in a cell during handover. After comparing the simulated results, its clear that adaptive bandwidth allocation is found to yield better performance than non- adaptive bandwidth allocation.

Keywords Adaptive Bandwidth Allocation, Network Simulator 2, Internet Protocol, UMTS Radio Access Networks

1. Introduction

With the ever increasing popularity of social media all over the world, the demand for multimedia services can no longer be understated. It is expected that the demand in the near future will be even greater. This will present a great challenge in terms of availability of enough bandwidth that can meet these demands without compromising QoS requirements. Usually, the recommendation is to keep a cellular network uncongested but with the limitations imposed by availability of bandwidth, the quality of services would likely degrade. This has fuelled the need for efficient QoS guarantee through bandwidth provisioning methods. The bandwidth provisioning methods should ensure that bandwidth is shared fairly. This is exciting not only because of an UMTS network ability to provide data services and applications such as voice, video streaming and web at a faster rate than ordinary 3G networks but also being an All-IP network which greatly reduces cost of transmission due to the use of a single network infrastructure [1]. A UMTS network

also provides an environment for delivering both existing and future services [2]. In these cases a UMTS network matches well.

As mobile terminals move between cells a process known as handover, they are engaged to avoid service termination. To provide uninterrupted communication, the destination cell must have enough bandwidth. Moreover, bandwidth being a scarce resource, it should be used efficiently [3]. Bandwidth should be distributed properly among different service requirements. It can be argued that data does not need to be processed in the same way. Therefore there is need to differentiate service offered in a network according to priority. For example, real time multimedia traffic with stringent quality of service (QoS) requirements should be given higher priority over non-real time multimedia traffic. For such QoS provisioning, different methods have been proposed. Among them are the InterServ and DiffServ [4]. InterServ comes with its complexity. This includes: lack of scalability and need to keep large number of flow states. Because of these drawbacks InterServ cannot be used in large networks like the internet. It is of great significance to also note that static procedures do not adapt to traffic dynamics. The aim of this study is to develop a threshold based adaptive bandwidth allocation policy that scalably assigns bandwidth to mobile subscribers with guaranteed QoS through the DiffServ algorithms.

DiffServ answers the needs of adaptation in the cellular networks for the multimedia traffic, when the performance of this traffic is brought into jeopardy by the arrival of handover traffic into a cell. The scheme adapts to a wide range of different demands and the presence of runtime changes in applications requirements. Through service differentiation, it provides a police based management that dynamically supports adaptive QoS provisioning [5]. Policy-based management is the automation and the simplification of the management process using rules that can dynamically change the behaviour of a network. These adaptive policies include both dynamically changing policy parameters and reconfiguring the policy objects at rum-time, where the QoS configuration would be changed as needed. This makes DiffServ a candidate for non-fair sharing of bandwidth in cellular networks.

As diffserv functionality is pushed to the edge nodes instead of every node in the system, it brings the need to configure parameters dynamically in the mobile nodes and base stations. The following is a brief explanation on how DiffServ do dynamic bandwidth policing adaptation [5]: The core routers monitors the network conditions, if service degradation is detected, the relevant edge routers will reject or degrade lower priority traffic, in order to meet the QoS requirements. This solution results in an increased performance for high priority traffic.

The rest of this paper is structured as follows: The description of the related works is done in section 2. Section 3 deals with Differentiated Services. Section 4 gives an overview of mapping UMTS network classes into Differentiated Services classes. In Section 5 the at-

tention is paid to the simulation model. Detailed discussion of results are presented in Section 6 by comparing the adaptive scheme to non-adaptive. Finally, conclusions and recommendations are made in Section 7.

2. 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 like interfering frequency. In the scheme formulated in [6], an ideal concept is developed to investigate the performance of an Adaptive channel reservation Scheme for Multi-Class Traffic by using adaptive radio parameters. The scheme uses an analytical model to verify performance. By reserving channels the concept introduces the storage of state information of allocated channels in routers such as the reservation status and number reserved channels. Maintaining state information is difficult with large number of traffic flows. A call admission control algorithm that utilizes an adaptive multi-level bandwidth-allocation scheme for non-real-time calls is presented in [2]. To verify performance in this scheme, they used a numerical analysis for comparison of results. A mathematical analysis of a real network is difficult; proper simulations are preferred which will be the approach in this paper.

A new scheme, Dynamic Channel Allocation Scheme (DCAS) for call admission control is devised in [7]. The scheme dynamically allocates guard channel for handover calls and introduces the concept of channel borrowing strategy. Although concepts based on guard channels are simple and capable of minimizing call drops, they affect bandwidth utilization negatively [2]. Arafat Abu Mallouh and Christian Bach have proposed DCAS that make use of artificial intelligence to assign channels optimally [8]. This strategy is implemented in an environment with uniform and non-uniform load distribution. It depends on the following factors: size, coordination, frequency reuse, and hand over to make the allocation process efficient and reliable. As the base stations are obligated to communicate intelligently with other base stations in order to allocate channel intelligently it results in increased signaling overhead due to communication with the neighboring cells.

In [9], dynamic bandwidth allocation algorithm for interactive multimedia applications with/without background load in the cellular networks is presented. In this strategy, maintaining the QoS is based on policies which are derived from history of traffic using traffic analyzer and application history monitor. By taking into account history of traffic and analysis of previous traffic involves complex calculations; this increases the signaling load on the network which in turn reduces network efficiency. Bandwidth reservation is crucial for improvement of the performance of cellular networks. In [10], bandwidth reservation strategy is considered which first reserves

some amount of bandwidth for handoff calls then the bandwidth can be increased for handoff calls by the base station based on the user mobility. This is to say that the base station dynamically increase the reserved bandwidth for handoffs when the initially reserved bandwidth is not enough, minimizing delay and at the same time increasing the system throughput. The main drawback of this scheme is the scalability problems. There is a lot of processing overhead. This paper attempts to fill in these gaps.

3. Differentiated Services for Packet Forwarding

DiffServ is scalable IP based technology developed by Internet Engineering Taskforce (IETF). which can efficiently provide QoS in networks by providing bandwidth discriminately to different categories of traffic [5]. Instead of allocating bandwidth to every traffic flow, it categorizes traffic which can be identified as classes then forward it according to the class specification. The forwarding treatment of the packets is according to pre-specified forwarding Per Hop Behaviours (PHBs). All packets in each traffic class, receive the same forwarding behaviour in routers. This makes it able to guarantee QoS without over-provisioning of bandwidth to a particular traffic flow. Therefore, DiffServ could provide differentiated QoS guarantee for voice, video and web traffic. Each traffic stream is assigned a distinct dropping probability determined by its priority where high priority streams are favoured at the expense of low priority streams.

In Diffserv, packets are forwarded in three steps [4] as illustrated in Figure 1.

3.1. Packet Classification
This is important because it helps in marking, metering, shaping and dropping packet. Therefore it is crucial in controlling traffic. Marked traffics are classified into classes depending on the Differentiated Service code point (DSCP) value. Voice traffic is marked with Expedited Forwarding (EF) DSCP value of Expedited Forwarding class. Video traffic is marked with Assured Forwarding (AF) DSCP value of Assured Forwarding and Web traffic is marked with Best Effort (BE) DSCP value of Best Effort class. It is done at the edge routers at the sender side.

3.2. Packet Marking
It is the allocation of a DSCP to a packet. Packet marker stamps packets with desired DSCP code-point value and adds it to a particular Differentiated Service(DS) behavior aggregate. The incoming traffic is marked by different DSCP value for different types of traffic. Packets with the same DSCP value are given the same treatment. Marking is done at the edge routers of the network at the sender

side.

3.3. Packet Scheduling
Schedulers are responsible for sending packets using physical queue. The routers assign separate physical queues for each traffic class and the available bandwidth is distributed among the queues. The traffic scheduler chosen corresponds to the desired level of service differentiation.
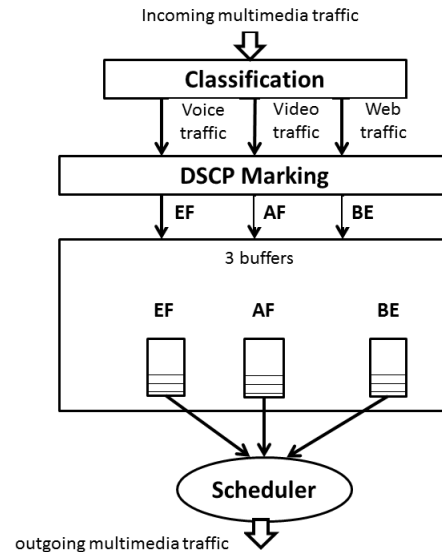

Figure 1.   Diffserv packet forwarding steps

4. Mapping UMTS into DiffServ

It is crucial to note that UMTS commonly uses IP transport as it is cheap. This makes DiffServ appropriate for QoS implementation. UMTS is grouped into four QoS classes while DiffServ is grouped into three classes. The UMTS service classes include Conversational, Streaming, Interactive and Background while the DiffServ service classes include Best Effort (BE), Expedited Forwarding (EF) and Assured Forwarding (AF) [11]. The Mapping of UMTS into DiffServ is shown in Table 1.

Table 1. Mapping UMTS into DiffServ

UMTS service classes	DiffServ service classes
Conversational	EF (voice)
Streaming	AF (streaming video)
Background/ Interactive	BE (web)

5. Simulation model

5.1. Model Description

Understanding the nature of traffic in a system and choosing an appropriate traffic model are important for the simulation study to succeed. General models with classes of multimedia calls in mobile cellular network are con-
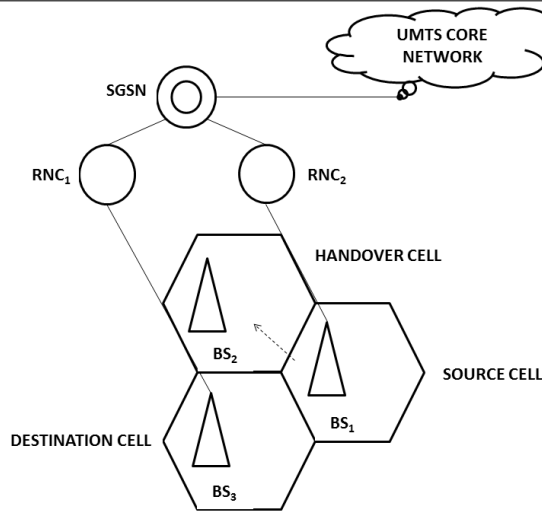
Figure 2.   Three cells arrangement in a UMTS network

sidered. In this model as discussed below three cells are simulated to evaluate the QoS performance of a mobile cellular network. The three cells arrangement in a UMTS network are shown in Figure 2.

5.2. Traffic Model

Before the analyzing of the performance of a mobile cellular network, it is crucial to come up with a traffic model. The study will consist of three cells which will consist of two cells (one being traffic originating cell and the other being the handed-over cell) sending traffic and one cell receiving traffic (the destination cell). The calls made carry constant bit rate for voice and variable bit rate video and web content. The model makes use of common assumptions that handoff calls follow a Poisson process. That is to say that traffic arrival rate (λ) follows a Poisson process. Thus, packet inter-arrival times are assumed to follow an exponential distribution with a mean of $1/\lambda$. This is illustrated by the following example. The bandwidth required by a call depends on the type of call.

- Let packet arrival rate (λ) = 500 packs/s
- Then mean inter-arrival time $1/\lambda$ =0.002s
- If packet size is 500 bytes
- Then transmission rate=500 bytes x 500 x8 packs/s =2mbps

The following discussion gives a brief definition of the Poisson process in three different but equivalent ways [12].

1) It is a pure birth process:
   In an infinitesimal time interval (dt) there may occur only one arrival. This happens with the probability (dt) independent of arrivals outside the interval.

2) The number of arrivals (n) in a period from 0 to t obeys the Poisson distribution ($P_n(t)$).

$$P_n(t) = \frac{(\lambda t)^n}{n!} e^{(-\lambda t)} \qquad (1)$$

Where: t is used to define the interval 0 to t ,n is the total number of arrivals in the interval 0 to t and λ is the total average arrival rate in arrivals/sec.

3) The inter-arrival times are independent and obey the Exponential distribution ($P_0(t)$): Let us consider a special case of Poisson distribution, the probability of no arrivals taking place over a given interval. It is easy to see that by substituting n with 0, we get the following equation:

$$P_0(t) = e^{(-\lambda t)} \qquad (2)$$

5.3. Mobility Model

To simulate handover, 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 neighbouring cell's BS with varying probability. With reference to Figure 3, this mean that if a mobile source from N_0 were to initially send traffic to destination N_{12}, after a handover, N_3 which is located in neighbouring cells could be sending traffic to destination N_{12}. This is similar to if N_{12} were to send traffic to destination N_0, after a handover, it could be sending traffic to destination N_3 which is located in neighbouring cells.

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

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 network functionality on NS-2 version 2.35. 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 throughput and packet loss. 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 ($N_0 - N_5$) and three destination nodes ($N_{12} - N_{14}$) which illustrates an IP-based Radio Access Network of a UMTS network.

This is the brief description QoS performance parameters used:
In this case **throughput** which is represented in bps is the number of successfully received packets in a unit
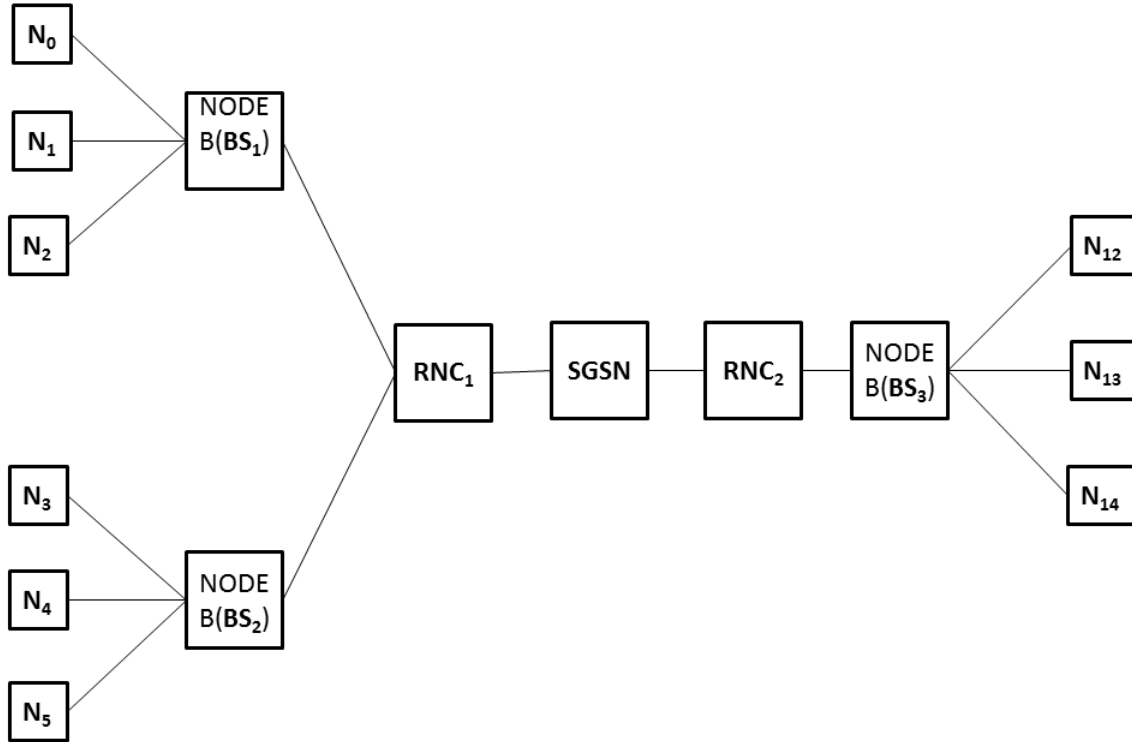
Figure 3.   Simulation Model for an IP-based Radio Access Network of UMTS network

time

$$Throughput = \frac{received\,data * 8}{Data\,Transmission\,Period} \quad (3)$$

while **packet loss** is the difference between the generated and received packets. It affects the quality of received video and voice data [2]. Packet loss increases due to increase the traffic congestion [3]

$$PacketLoss = GeneratedPackets - ReceivedPackets \quad (4)$$

Table 2. 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)

5.4. Simulation Parameters

The simulation settings and parameters are summarized in table 2.

6. Results and Discussion

Primarily two scenarios have been created. One is with adaptive bandwidth allocation and another is non-adaptive bandwidth allocation. Different sets of results from the two scenarios give us the opportunity to investigate the performance of these schemes. To analyse 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.

Packets are categorized depending upon whether they are very urgent, real-time (voice and video), non-real-time (web). Once the categorization is done the packets are sent through the separate queues according to their priority. Also it is of great importance to note that this analysis concentrates on voice handover and how it affects other multimedia traffic.

6.1. Packet Loss

It is of great importance to note that Figure 4 and Figure 5 do not mean that packet loss rate increases with
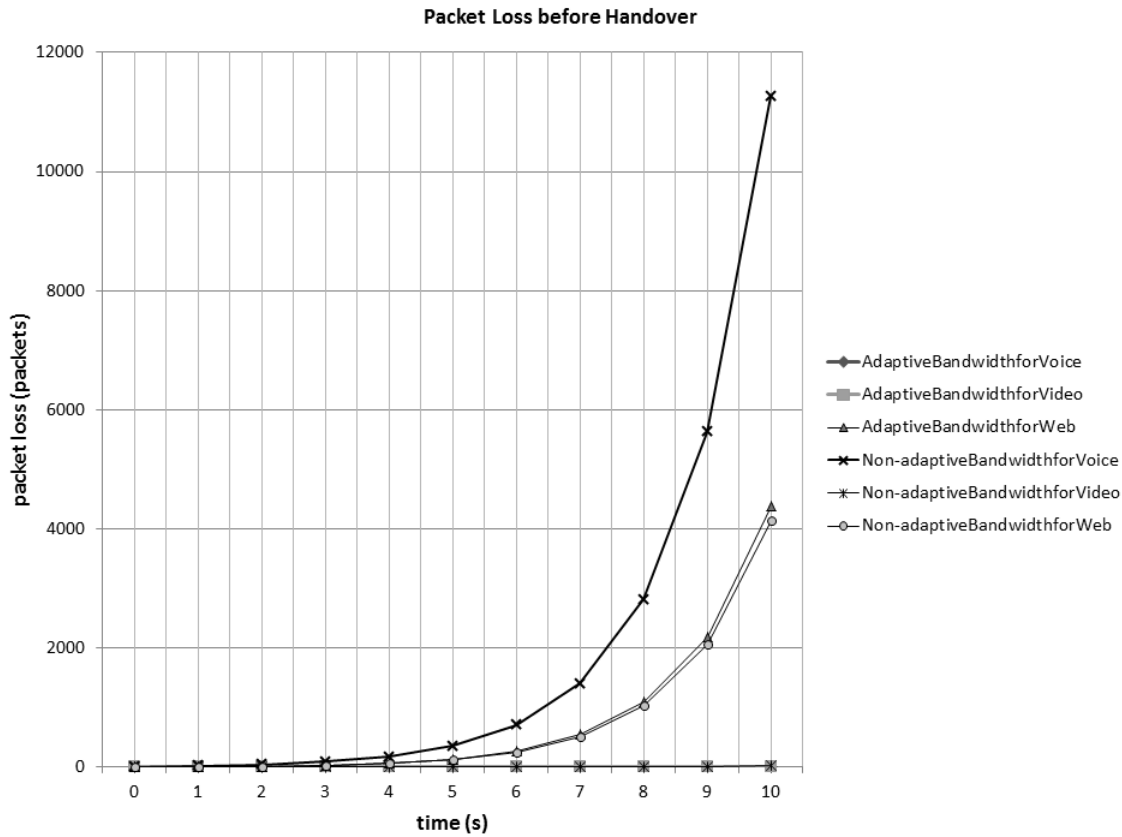
Figure 4.   Packet Loss in a voice originating cell

time, but rather shows lost packets that are accumulated over time. As illustrated by Figure 4, the packet loss for voice traffic in adaptive bandwidth allocation remains zero as time increases for the duration of the simulation while it grows exponentially with time until it reaches 11.3k at 10 seconds for non-adaptive bandwidth allocation. Also, the packet loss for video traffic remains zero in adaptive bandwidth allocation as time increases while for non-adaptive bandwidth allocation, the losses are small or nearly negligible. The packet loss for web in both adaptive bandwidth allocation and non-adaptive bandwidth allocation rises rapidly with time. The difference between the two is that packet loss in adaptive bandwidth allocation is more, around 4.3k at 10 seconds than in non-adaptive bandwidth allocation which is 4.2k as depicted by Figure 4.

Both voice traffic and video traffic are real-time traffic while web traffic is non-real-time traffic. According to the adaptive bandwidth allocation algorithm real time traffic gets priority over non-real time traffic hence no loss for both voice traffic and video traffic while web traffic experience severe losses. For non-adaptive bandwidth allocation mechanism, bandwidth provisioning is based on best effort which does not guarantee bandwidth hence the loss experienced for the three types of traffic. In best effort, packets for all types of traffic are given equal treatment. It will also be noted that from 0 to 2

seconds the losses are steady as the network system is initializing.

It should be noted that the bandwidth provisioned in the traffic originating cell and handed-over cell is the same. After handover there is increased traffic in the new cell, this makes it extremely congested. The packet loss for voice traffic remain zero as before handover in adaptive bandwidth allocation as time increases while for non-adaptive bandwidth allocation, packet loss increases rapidly with time as can be seen from Figure 5, until it reaches 1.7k at the end of simulation. After handover the packet loss for video traffic remains steady at zero in adaptive bandwidth allocation as time increases while in non-adaptive bandwidth allocation it increases even more than before handover up to 0.3k from 0 as simulation completes. The packet loss for web traffic is more in adaptive bandwidth allocation at around 4.5k at 10 seconds than before handover while packet loss in non-adaptive bandwidth allocation decreases to 0.2k than before handover which was 4.2k.

Voice traffic being an inelastic traffic uses a fixed amount of bandwidth. So it gets a higher priority and as much bandwidth as needed to avoid packet drops as it does not adapt to network conditions. As a result, there is no bandwidth restriction for voice traffic; this is at the expense of other services especially those with the lower priority hence zero loss in adaptive bandwidth allocation.
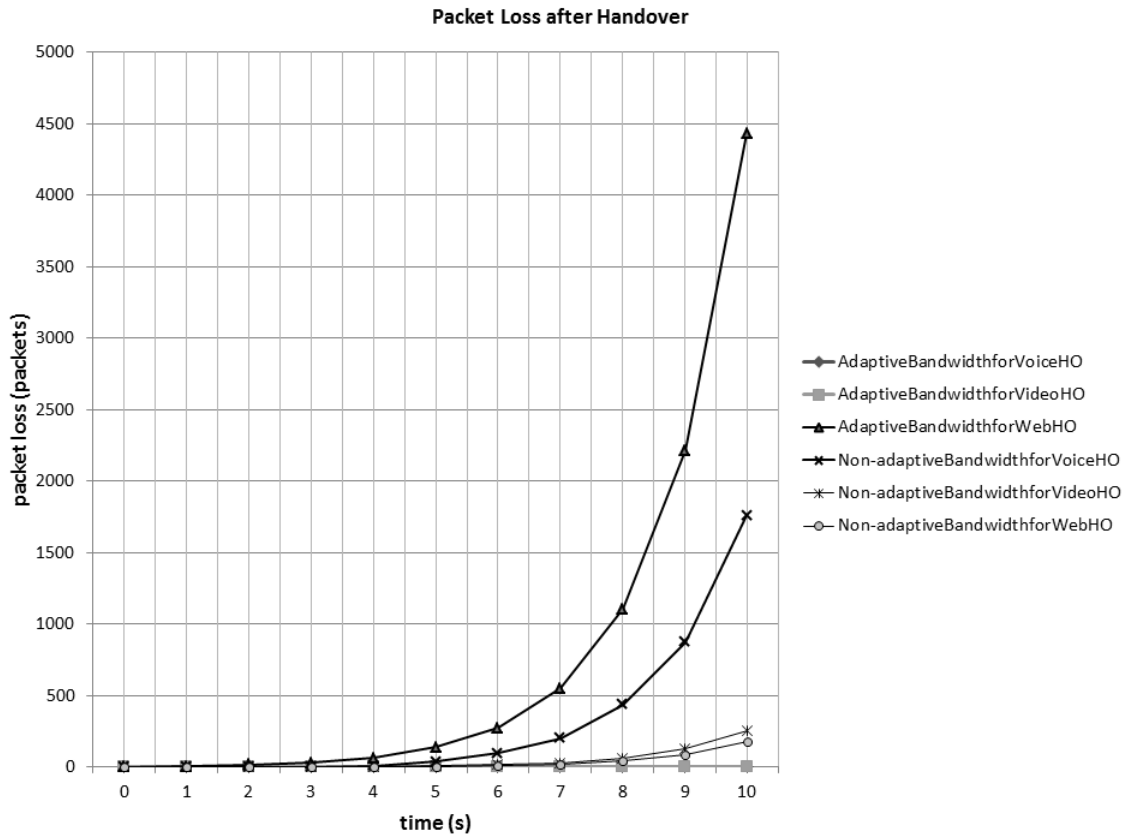
Figure 5.   Packet Loss in a voice handed-over cell

The packet loss in video is more for non-adaptive bandwidth allocation than before handover as traffic increases in the new cell while bandwidth available is the same as the previous cell. Packet loss remains fixed at zero for adaptive bandwidth allocation as time increases as it is a real-time traffic; this is due to adaptive bandwidth allocation algorithm starting degrading bandwidth from the lowest priority to the highest services. In this case, the bandwidth assigned to web traffic will have to be depleted before the video traffic with medium priority will start to be down-graded.

After handover the packet loss in web is more for adaptive bandwidth allocation than before handover as traffic increases in the new cell. This can be attributed to the fact that lower-priority packets are discarded at higher rates than packets with medium and high priorities in adaptive bandwidth allocation. In this case web traffic is a low priority traffic. Packet loss for web in non-adaptive bandwidth allocation decreases after handover than before handover. For non-adaptive bandwidth allocation packet loss is dependent on best effort. This means the packet loss probability between real time and non-real time traffic is equal.

Web traffic is of TCP type which is elastic. It tries to occupy as much of the available bandwidth as it can, but depending on network conditions it adapts accordingly. That is, with high packet losses or less bandwidth it adapt to a slower rate of transmission and vice versa, this explain why explain why it is treated as a low priority traffic.

6.2. Throughput

It should be taken into account that Figure 6 and Figure 7 do not mean that throughput increases with time, but rather shows throughput that is accumulated over time. From Figure 6, it is clear that adaptive bandwidth allocation guarantees higher throughput for voice than non-adaptive bandwidth allocation which are 8.3Mb/s and 8 Mb/s respectively at the end of 10 seconds simulation. Throughput for voice traffic increases linearly up to 3 seconds then starts to fall as time progresses. It is also shown in the graph that for the video traffic in adaptive bandwidth allocation, throughput is much higher (5 Mb/s) than in non-adaptive bandwidth allocation which is 0.3 Mb/s at 10 seconds. Comparing throughput for web traffic in adaptive bandwidth allocation and non-adaptive bandwidth allocation, it is higher for non-adaptive bandwidth allocation (2.9 Mb/s) against 0.3k at the end of simulation.

It is observed that throughput is much higher in voice traffic followed by video traffic and web traffic in that order respectively in adaptive bandwidth allocation, but that is not the case in non-adaptive bandwidth allocation. This is because in adaptive bandwidth allocation the order of priority from high to low is voice traffic followed by video traffic and finally web traffic while for non-
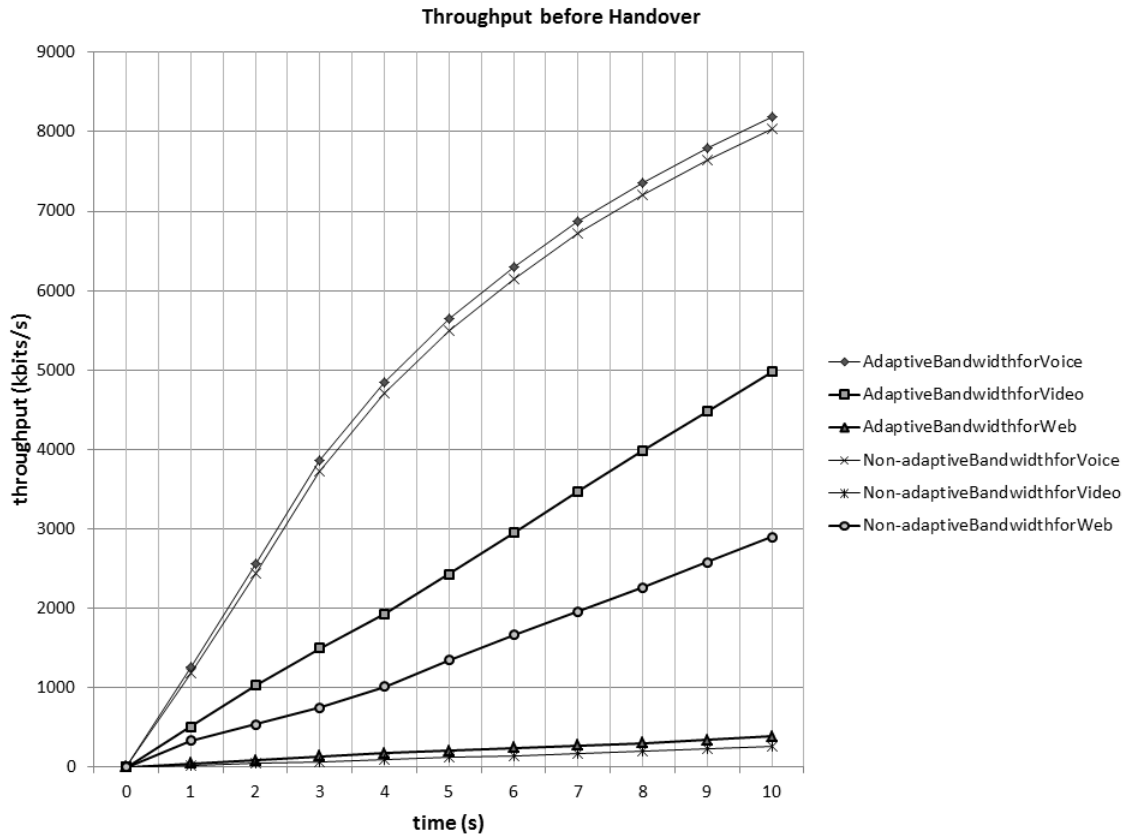
Figure 6.   Throughput before Handover

adaptive bandwidth allocation the order follows the best effort concept. The increase linearly of throughput for voice traffic up to the third second and then falling is attributed to voice being handed-over to new cell from the present cell which occurs at 3 seconds.

It can be seen from Figure 7 that the throughput for voice traffic remains zero up to the third second then rises rapidly. At around 3.5 seconds it surpasses both the non-adaptive video traffic and adaptive web traffic graphs and continue to rise up at a faster rate up to 9 seconds where again it surpasses the non-adaptive web traffic . It is also observed that the throughput for voice traffic in adaptive bandwidth allocation is higher than in non-adaptive bandwidth allocation, 5.4 Mb/s and 5.3 Mb/s in that order at 10 seconds. As can be seen in Figure 7, for video traffic throughput is high in adaptive bandwidth allocation than non-adaptive bandwidth allocation which is 5.5 Mb/s and 0.3 Mb/s respectively at 10 seconds. For web traffic throughput is higher in non-adaptive bandwidth allocation at 4.6 Mb/s than in adaptive bandwidth allocation at 0.4k when simulation ends. Web traffic throughput is also affected by handover as it starts to drop after the third second in non-adaptive bandwidth allocation.

The throughput for voice traffic remains zero up to the third second then rises rapidly because before handover there in no voice traffic in the new cell. Handover occurs in the third second. Both voice traffic and video traffic have high throughput in adaptive bandwidth allocation than in non-adaptive bandwidth allocation as they are prioritized but for web traffic the opposite is the case since web traffic is given low priority. Web traffic throughput is higher in non-adaptive bandwidth allocation than in adaptive bandwidth allocation due to equal treatment of traffic in in non-adaptive bandwidth allocation. The sudden drop in throughput for web traffic immediately after handover is as a result of its bandwidth being stolen by high priority voice traffic.

7. Contribution of the Work

This paper presents a dynamically adaptive bandwidth allocation scheme during handover in IP-based DiffServ UMTS networks. Since DiffServ algorithms do not have a mobility module, to simulate mobility in this paper, 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 neighbouring cell's BS with varying probability. The rate and direction of handover is determined by probability. A high handover rate represents a high probability of handover and vice versa. A congestion of traffic in a cell is a representation of traffic moving in the direction of the cell as opposed to other directions.

Also the scheme formulated in this paper adapts to
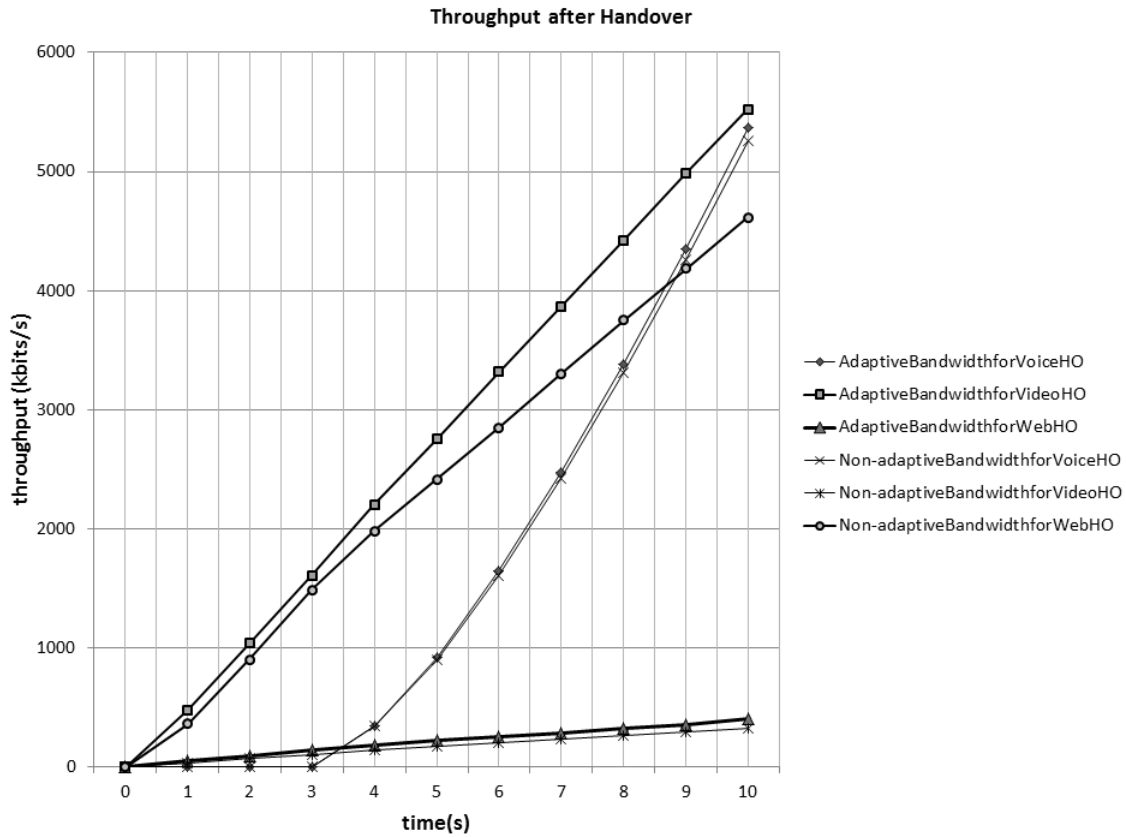
Throughput after Handover

Figure 7.   Throughput after Handover

a wide range of different demands and the presence of runtime changes in applications requirements. Through service differentiation, it provides a police based management that dynamically supports adaptive QoS provisioning where by each class of service offered would be governed by service level agreements contracted between service providers and mobile subscribers that spell out exact QoS assurance in terms of throughput and packet loss bounds.Finally, the adaptive bandwidth allocation developed is efficient, scalable, and able to provide bandwidth on demand.

8. Conclusions and Recommendations

The functioning of this scheme is that, if there is enough bandwidth in a cell, the handed-over voice traffic is admitted without any problem. But if the available bandwidth in that cell is scarce then some bandwidth from web traffic in that cell is borrowed. Still if the web traffic is insufficient, then extra bandwidth is borrowed from video traffic in that cell. Hence, this strategy gives high priority to handed-over voice traffic then video and finally web with the lowest priority. It has been observed that the adaptive bandwidth allocation minimizes packet losses and give high throughput than non- adaptive bandwidth allocation. It has also been proven that non- adaptive bandwidth allocation does not guarantee service when the network experience congestion due to

increased traffic in a cell during handover. After comparing the simulated results, its clear that adaptive bandwidth allocation is found to yield better performance than non- adaptive bandwidth allocation. That is the reason the proposed adaptive bandwidth allocation scheme will be of great interest in multimedia services as they continue to grow at an alarming rate.

It should be noted that this paper has only handled voice traffic handover. Further analysis should be done on the effect of other multimedia services handover on QoS parameters such as end to end delay, jitters and packet delivery ratio.

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