DYNAMICALLY ADAPTIVE BANDWIDTH ALLOCATION FOR HANDOFF MULTIMEDIA SERVICES FOR QUALITY OF SERVICE PERFORMANCE IN MOBILE CELLULAR NETWORKS

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Abstract
With the tremendous growth of multimedia services in the telecommunication industry, Quality of Service (QoS) provisioning is becoming more important. One of the challenges to achieve QoS requirements is to determine how to allocate system bandwidth to various applications. Moreover, bandwidth being a scarce resource, it should be used efficiently. One way of achieving this is to adopt micro/pico/femto-cellular architectures. A consequence of using small cell sizes is the increased rate of call handoffs as mobile terminals move between cells. In a network supporting multimedia services, the increased rate of call handoffs not only increases the signaling load on the network, but also adversely affects the QoS. To reduce handoff failures and achieve high bandwidth utilization an efficient multi- class adaptive bandwidth-allocation strategy that reclaims bandwidth from on-going calls is designed. Since a mathematical analysis of a real network is difficult, proper simulations are preferred. The focus for this paper is simulation of cellular network functionality on NS-2 integrated with EURANE. 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. To analyze the result of the simulation, the traffic is traced during the process. For every packet, information about it is written to a specified trace file.Primarily two scenarios are created. One is 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. The investigation illustrates that proper adaptation of bandwidth meets the QoS Requirements.

Key words: DiffServ, EURANE, NS-2, UTRAN, adaptive bandwidth allocation and non-adaptive bandwidth allocation

1.0 Introduction

The ever growing multimedia applications have led to great demand of UMTS services. The gaining of popularity of UMTS is not only because of its ability to send voice but also due to its ability to send data at a faster rate than ordinary 3G networks. Users will require to be always best connected and at anywhere in any time, but this is not always the case. With service interruption due to movement of mobile subscribers (Albonda & Yousef, 2015), QoS becomes an issue. QoS in cellular networks must be improved continuously to meet the needs that arise from time to time. This has necessitates for methods to process the data differently. For example, real time with low tolerance to delays has to be given high priority than non-real time traffic.

One of the most important factors in the success of increasing user mobility is seamless handover (Akpan, Kalu, & Inyang, 2014). Minimizing call drops resulting from the handover failures is a key issue for achieving seamless hand-overs across cellular networks. To guarantee QoS, this study attempts to provide continuous access to multimedia services to mobile users through adaptive bandwidth allocation mechanism. The paper is going to focus on UMTS terrestrial radio access network (UTRAN) for handover depiction as handover between cells (base stations) happen there.

1.0 Materials and Methods

2.1 Overview of UMTS

In this section the focus will be on functionality associated with handovers. UMTS network is divided into three parts UE (user equipment) for send and receive messages, UTRAN (UMTS Terrestrial Radio Access Network) for Radio Resource Managements among other tasks and the CN (Core Network) that routes messages and connects UMTS to other networks (Vranješ, Švedek, & Rimac-Drlje, 2010). The UTRAN consists of a RNC (Radio Network Controller) for radio resource control of a cell among other functions and a Node B. A Node B handles the communication to and from all UEs in one or more cells. Node B is also responsible for handover.

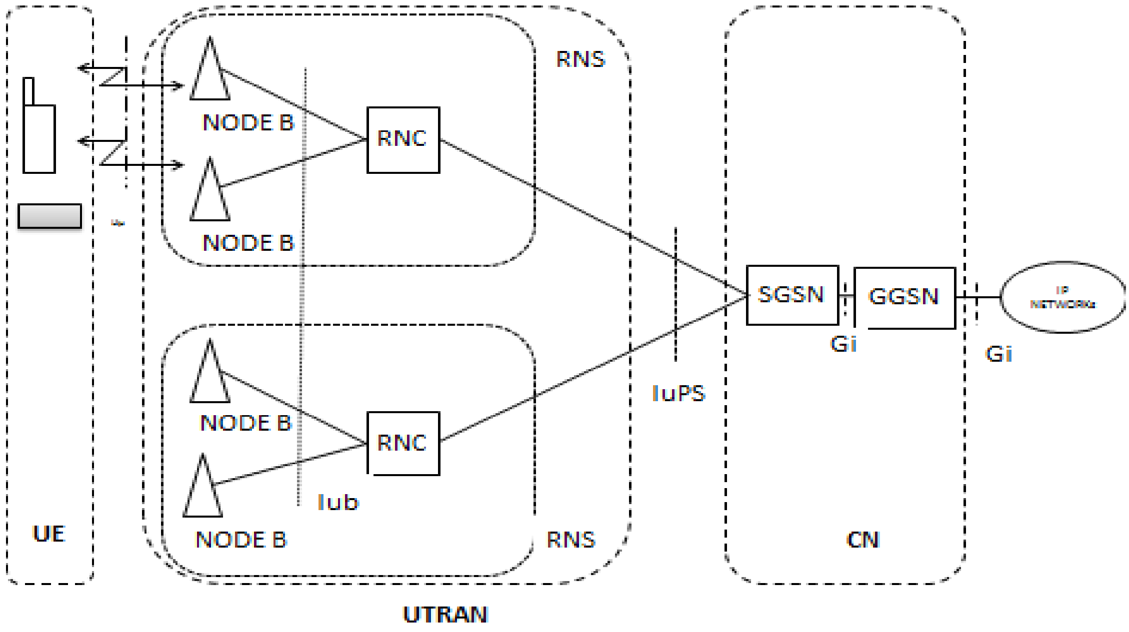

Figure 1: The UMTS architecture

2.2 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). The Mapping of UMTS into DiffServ (Ali, Saleem, & Tareen, 2012) is shown in table 1

Table 1: Mapping of UMTS into DiffServ

UMTS service classes	DiffServ Service Classes
Conversational/ Interactive	EF(voice)
Streaming	AF(video)
Background	BE(ftp)

2.3 NS-2 Simulation Model

2.3.1 Model Description

Understanding the nature of traffic in a system and choosing an appropriate traffic model are important for the simulation study to succeed. A general model with classes of multimedia calls in mobile cellular network will be considered. In this model as discussed below three cells are simulated to evaluate the QoS performance of a mobile cellular network. Figure 2 depicts the three cells arrangement in a UMTS network.

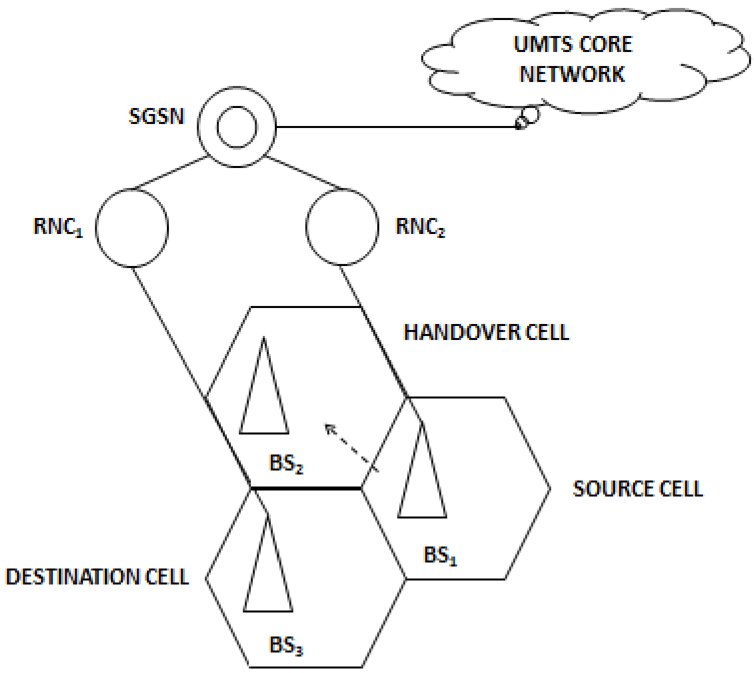

Figure 2: Three cells arrangement in a UMTS network

2.3.2 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 sending traffic and one cell receiving traffic. 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 (Hu & Wang, 2015). That is to say that traffic arrival rate (λ) follows a Poisson process. Thus, packet inter-arrival times are assumed follow an exponential distribution with a mean of $1/\lambda$. The bandwidth required by a call depends on the type of call.

2.3.3 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 N_0 were to initially send traffic to destination N_{12}, after a handoff, N_4 which is located in neighboring 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 handoff, it could be sending traffic to destination N_4 which is located in neighboring cells.

In this model, the rate and direction of handoff is determined 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.

Since a mathematical analysis of a real network is difficult, proper simulations are preferred. The focus for this paper is simulation of cellular network functionality on NS-2 integrated with EURANE. 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 which are explained below .The simulation model consist of 6 routers Edge1 (BS_1), Edge2 (BS_2), Edge3 (SGSN), Edge4 (BS_3), Core1 (RNC_1), Core2 (RNC_2) and six source nodes (N_0-N_5) and three destination nodes (N_{12}-N_{14}) has shown in figure 3. The simulation is conducted using NS-2 version 2.35 patched with EURANE.

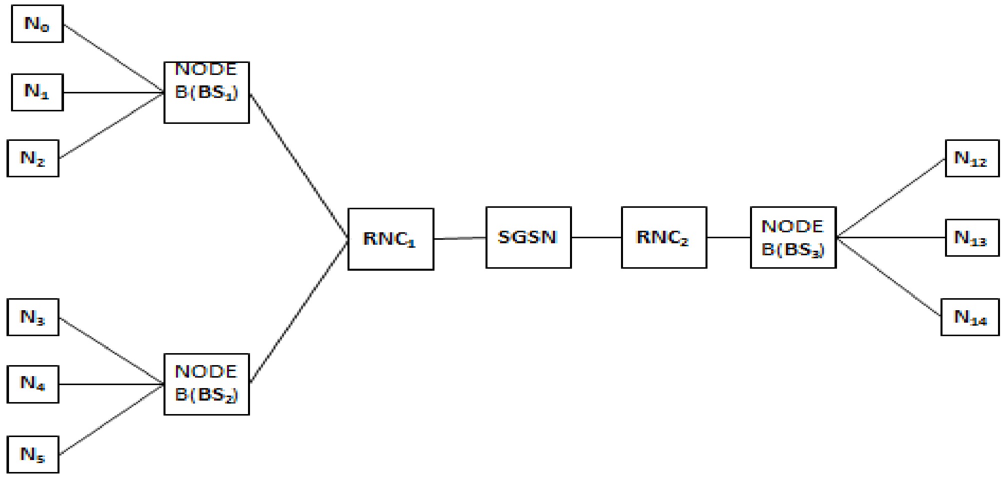

Figure 3: Simulation Model

This are the QoS parameters used (Vijayalakshmi & Kulkarni, 2013):
Throughput is the number of successfully received packets in a unit time and it is represented in bps.

$$\text{Throughput} = \frac{\text{received data}*8}{\text{Data Transmission Period}} \tag{1}$$

Packet loss is the difference between the generated and received packets.
It affects the quality of received video and voice data. Packet loss increases due to increase the traffic congestion

$$\text{Packet Loss} = \text{Generated Packets} - \text{Received Packets} \tag{2}$$

3.0 Results

Primarily we have created two scenarios. One is with adaptive bandwidth allocation and another is non-adaptive bandwidth allocation. Different sets of result from two scenarios give us the opportunity to investigate the performance of these schemes. To analyze quantitatively the result of the simulation, the traffic is traced during the 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. Figure 4, figure 5, figure 6, figure 7, figure 8, figure 9, figure 10 and figure 11 do not mean that packet loss rate increases or throughput increases, but rather shows lost packets and throughput accumulated over time.

Packet Loss

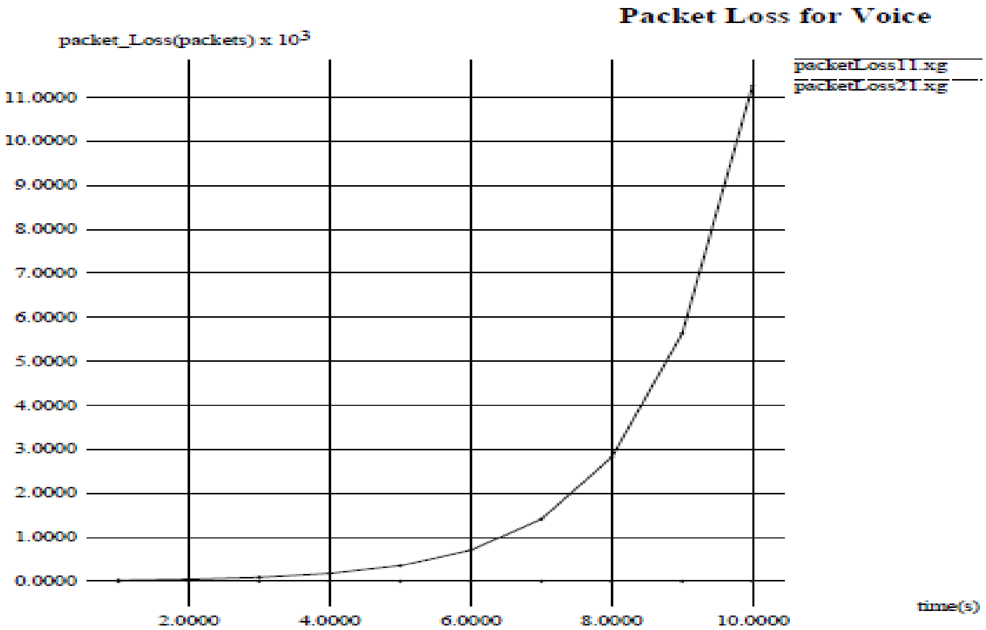

Figure 4: Packet loss for voice

Where packetloss11.xg is for non-adaptive bandwidth allocation and packetloss21.xg is for adaptive bandwidth allocation.

The packet loss for voice for adaptive bandwidth allocation remains at 0 as time increases while for non-adaptive bandwidth allocation, it increases exponentially with time. According to the adaptive bandwidth allocation algorithm real time traffic is getting priority over non-real time traffic. For non-adaptive bandwidth allocation the mechanism for bandwidth provisioning is based on best effort which does not guarantee bandwidth hence the loss experienced.

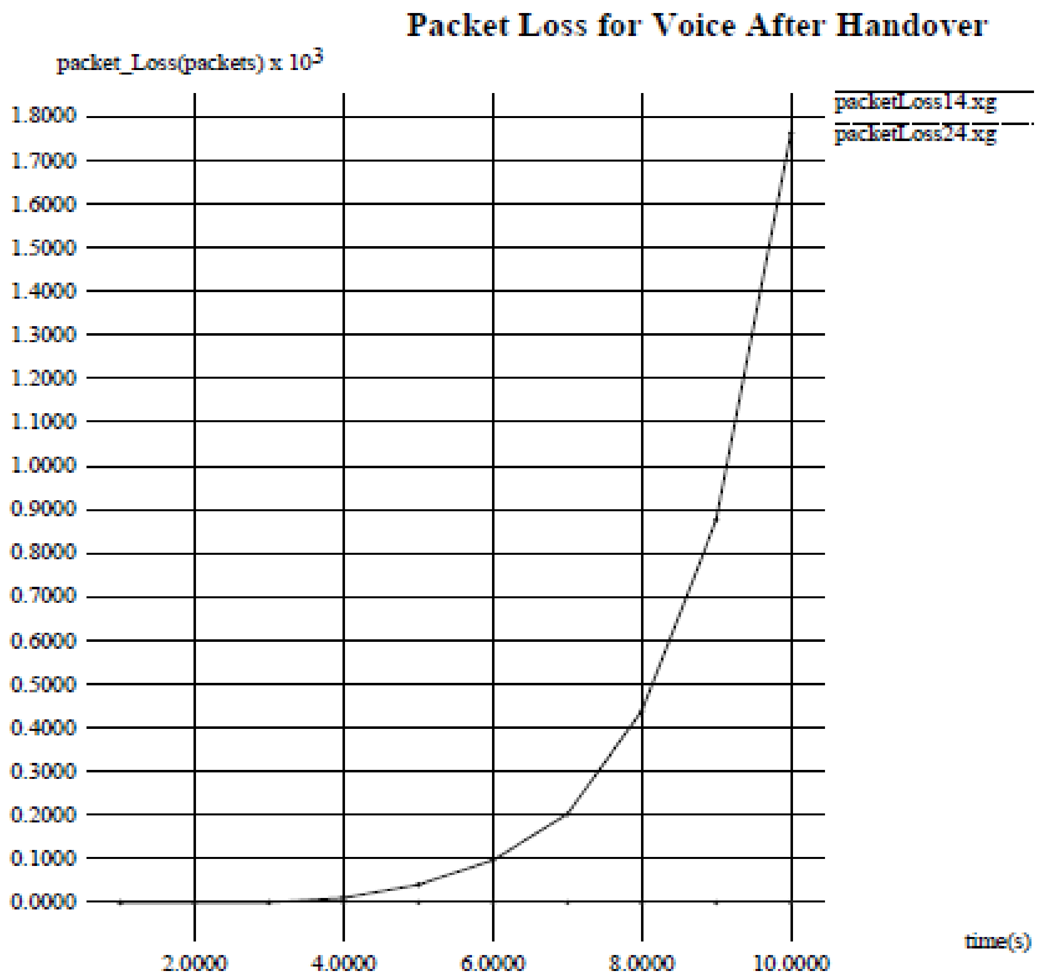

Figure 5: Packet Loss for voice after handover

Where packetloss14.xg is for non-adaptive bandwidth allocation and packetloss24.xg is for adaptive bandwidth allocation.

After handover the packet loss 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 at 0 for adaptive bandwidth allocation as time increases. This is because of its higher priority; voice gets as much bandwidth as needed. As a result, there is no restriction for voice; this is at the expense of other services especially those with the lower priority.

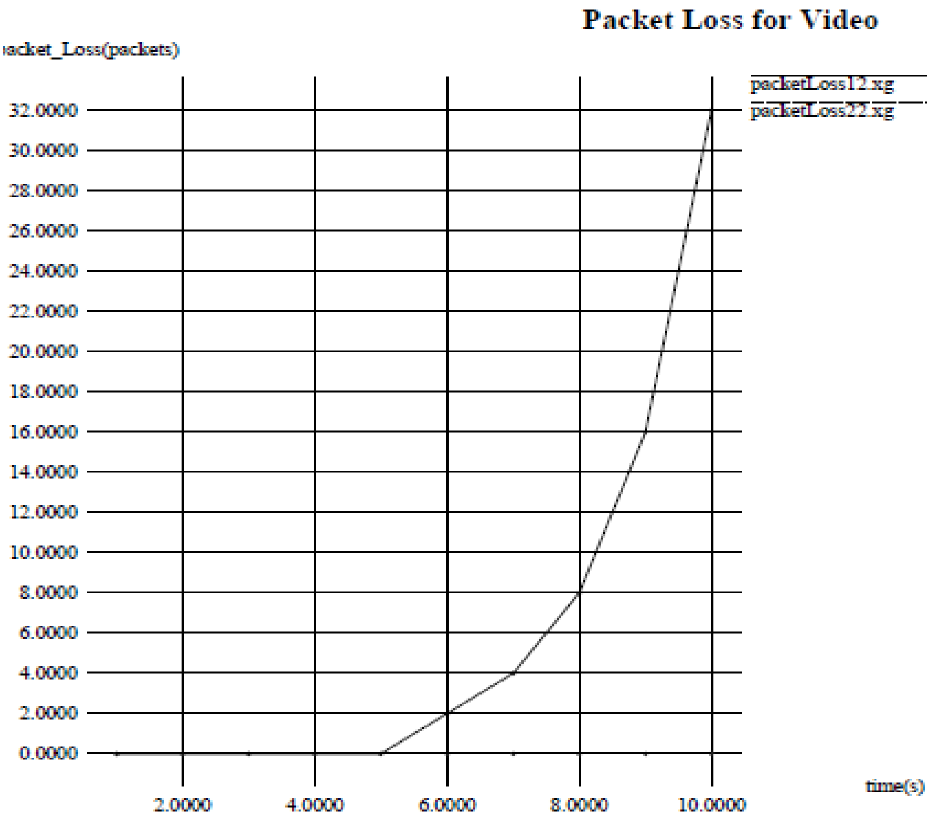

Figure 6: Packet loss for video

Where packetloss12.xg is for non-adaptive bandwidth allocation and packetloss22.xg is for adaptive bandwidth allocation.

The packet loss remains 0 for adaptive bandwidth allocation as time increases; this is due to this algorithm starting degrading services from the lowest priority to the highest. 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. For non-adaptive bandwidth allocation, the packet loss increases rapidly with time.

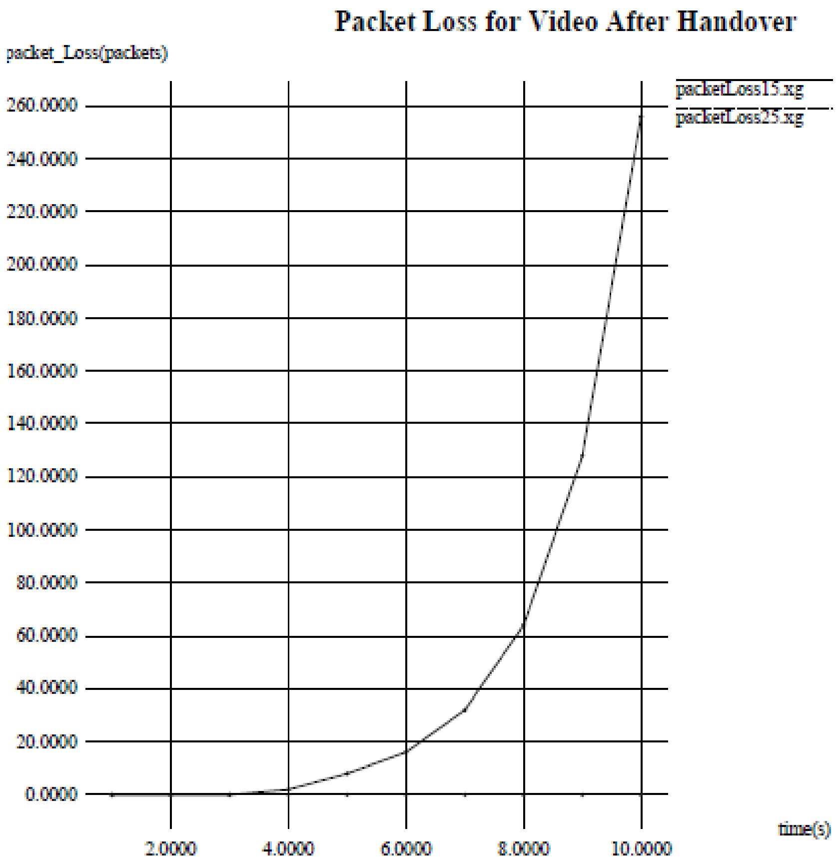

Figure 7: Packet loss for video after handover

Where packetloss15.xg is for non-adaptive bandwidth allocation and packetloss25.xg is for adaptive bandwidth allocation.

After handover 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 at 0 for adaptive bandwidth allocation as time increases. This is because web traffic bandwidth as to be depleted before video traffic can begin to be down-graded.

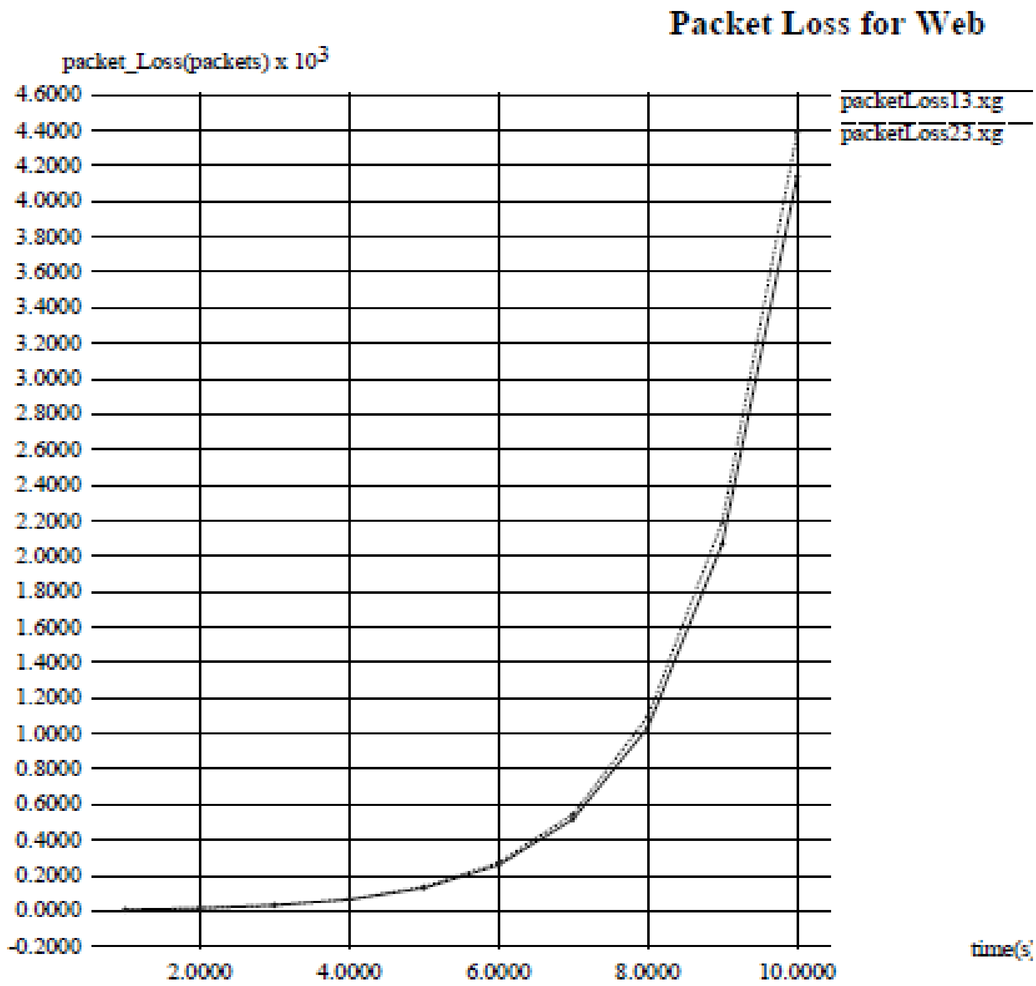

Figure 8: Packet loss for web

Where packetloss13.xg is for non-adaptive bandwidth allocation and packetloss23.xg is for adaptive bandwidth allocation.

The packet loss for web for adaptive bandwidth allocation increases rapidly with time. For non-adaptive bandwidth allocation packet loss also increases rapidly with time. The difference between the two is that packet loss in adaptive bandwidth allocation is more than in non-adaptive bandwidth allocation. 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. Higher-priority packets are not lost unlike other packets. For non-adaptive bandwidth allocation packet loss is dependent on best effort. This means the packet lost are distributed between real time and non-real time traffic.

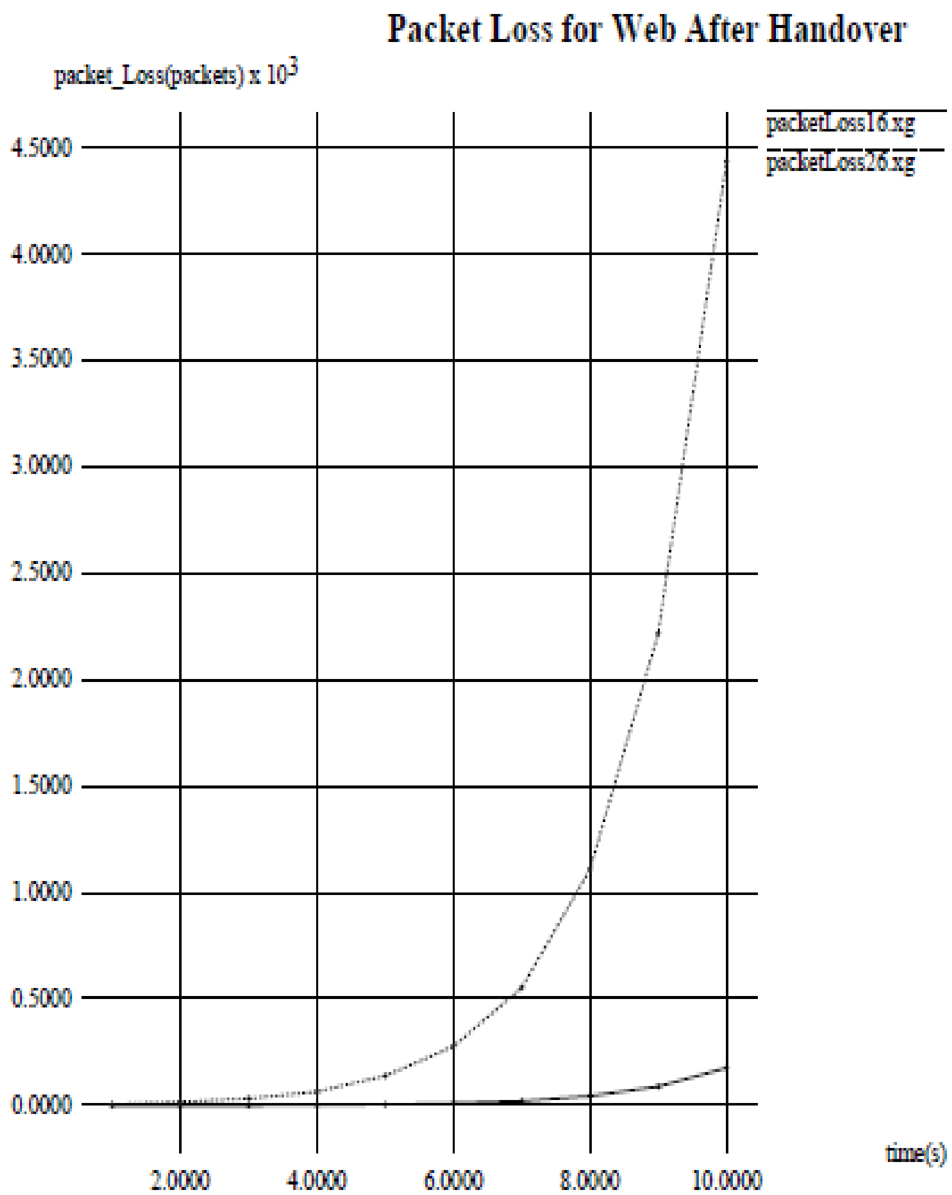

Figure 9: Packet loss for web after handover

Where packetloss16.xg is for non-adaptive bandwidth allocation and packetloss26.xg is for adaptive bandwidth allocation.

After handover the packet loss in web is more for 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 for non-adaptive bandwidth allocation decreases. This is because traffic is distributed between non-real time and real time as per best effort.

Throughput

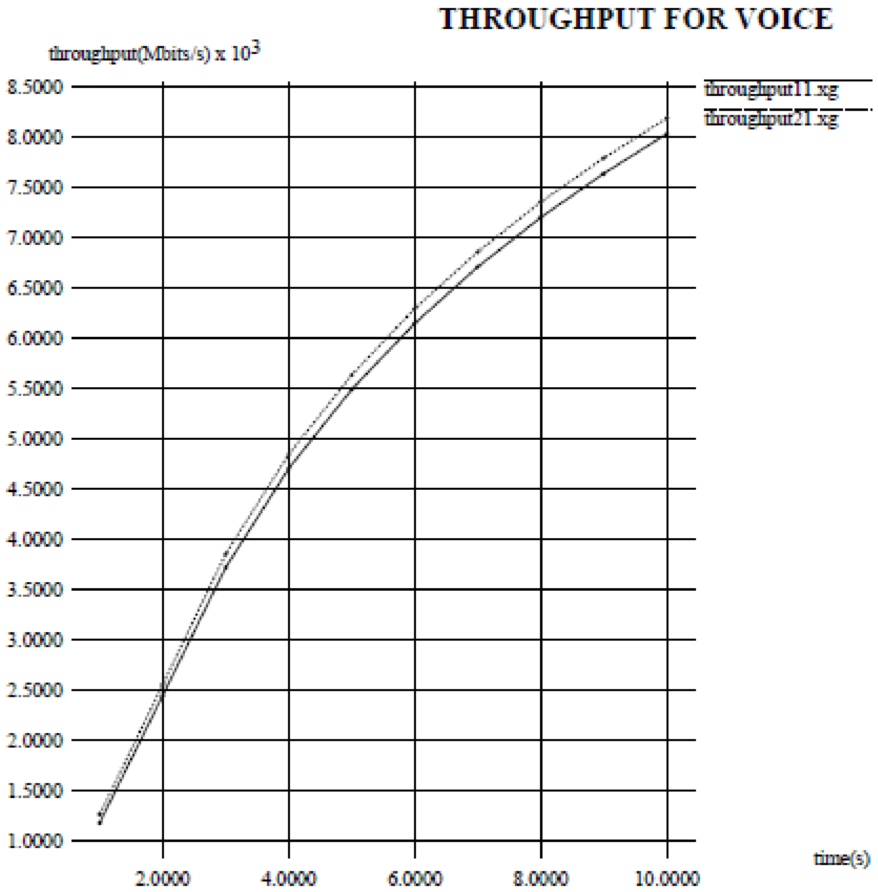

Figure 10: Throughput for voice

Where throughput11.xg is for non-adaptive bandwidth allocation and throughput21.xg is for adaptive bandwidth allocation.

Adaptive bandwidth allocation can guarantee higher throughput for voice than non-adaptive bandwidth allocation as shown in figure above. According to the adaptive bandwidth allocation algorithm real time traffic is getting priority over non-real time traffic. For non-adaptive bandwidth allocation the mechanism for bandwidth provisioning is on best effort which does not guarantee bandwidth.

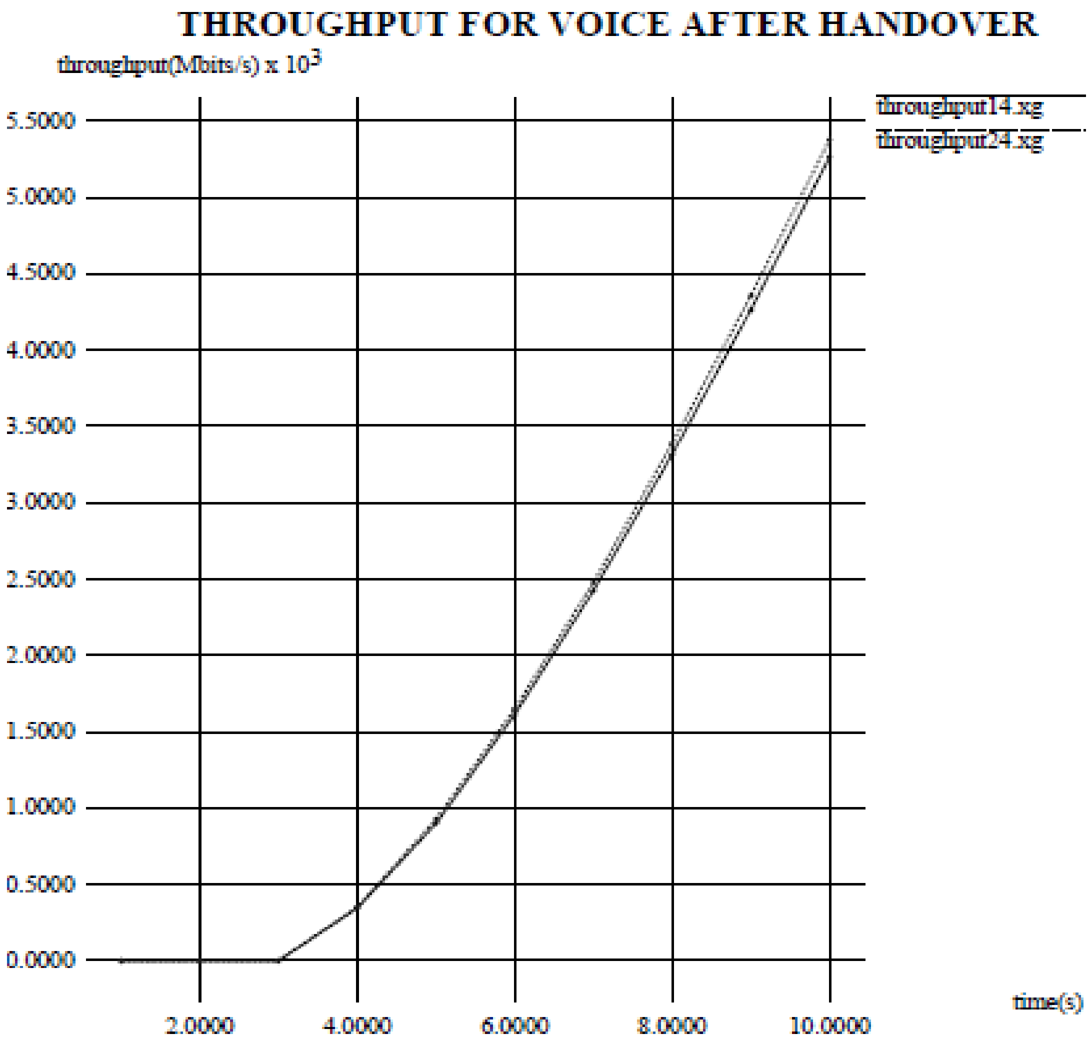

Figure 10: Throughput for voice after handover

Where throughput14.xg is for non-adaptive bandwidth allocation and throughput24.xg is for adaptive bandwidth allocation.

After handover the throughput is less than before handover as traffic increases in the new cell while bandwidth available is the same as the previous cell. But still throughput is more for adaptive bandwidth allocation than non-adaptive bandwidth allocation. This is because of its higher priority; voice gets as much bandwidth as needed.

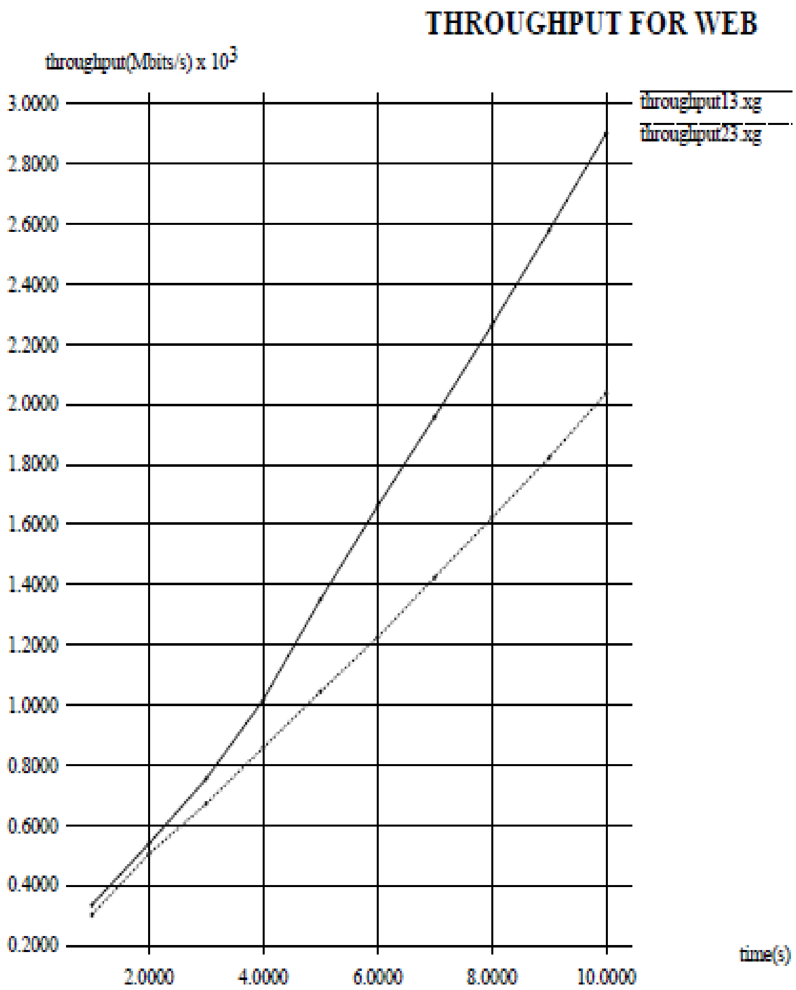

Figure 11: Throughput for web

Where throughput13.xg is for non-adaptive bandwidth allocation and throughput23.xg is for adaptive bandwidth allocation.

The throughput of Adaptive bandwidth allocation algorithm is less compared to the throughput of non-adaptive bandwidth allocation algorithm for web traffic, which is non-real time traffic. This is because real time traffic steals its bandwidth. . Real time voice traffic preempts resources from real time video traffic, which in turn preempts resources from non-real time web traffic.

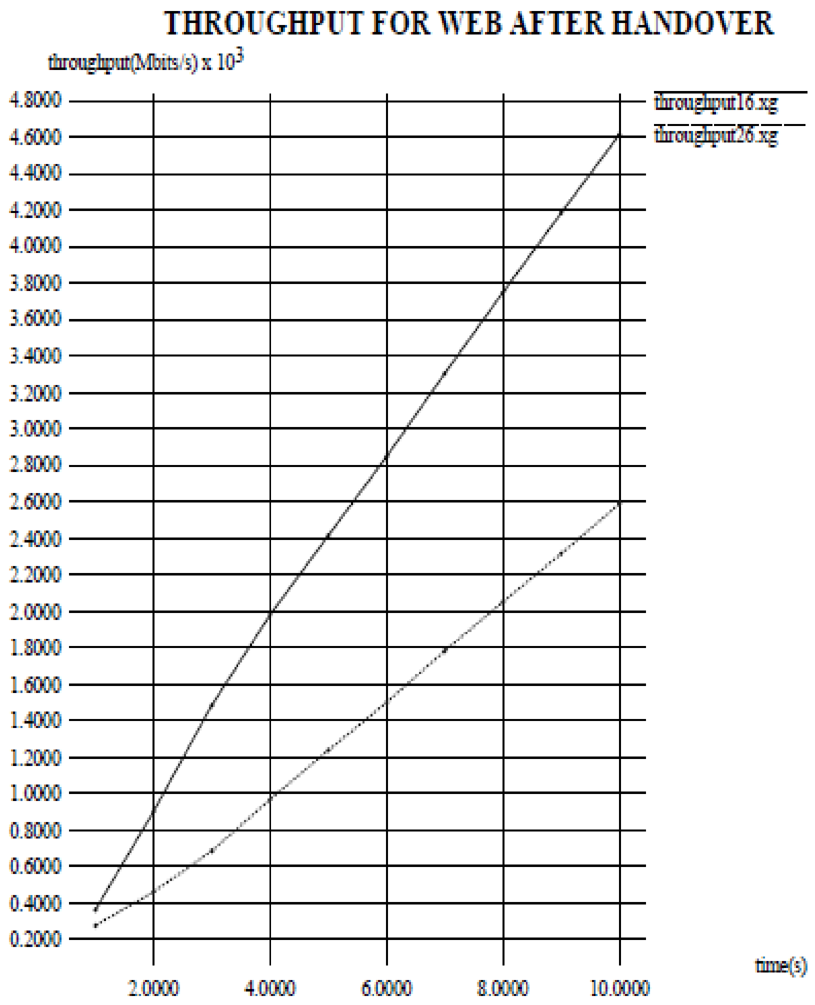

Figure 11: Throughput for web after handover

Where throughput16.xg is for non-adaptive bandwidth allocation and throughput26.xg is for adaptive bandwidth allocation.

After handover the throughput difference between the two algorithms is even greater than before handover as traffic increases with increased rate of arrival in the new cell while bandwidth available is the same as the previous cell. But still throughput is less for adaptive bandwidth allocation than non-adaptive bandwidth allocation. This is because of its higher priority, real time calls gets as much bandwidth as needed. As a result, they steal bandwidth at the expense of other services especially web with the lowest priority.

2.0 Discussion and Conclusions

The Implementation above suggest that, the packet loss for real time traffic remains almost 0 when adaptive bandwidth allocation is employed but increases rapidly with time when non-adaptive bandwidth allocation is employed. For non-real time traffic, the packet loss is more for adaptive bandwidth allocation than non-adaptive bandwidth allocation. This is because non-real time traffic gets down-graded when adaptive bandwidth allocation is employed.

According to the adaptive bandwidth allocation algorithm the real time traffic is getting priority over non-real time traffic. So, the packet loss for real time traffic is negligible or 0 whereas packet loss for non-real time traffic will increase with increase in time. With the non-adaptive bandwidth allocation algorithm, packet loss for real time traffic will increase with increase in in time but the loss is much lower with non-real time traffic because here losses depends on best effort.

It is proven that the adaptive bandwidth allocation algorithm improves bandwidth utilization through prioritization of services. Non real time services are degraded till acceptable service quality is achieved.

5.0 References

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