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

EDWIN NYANG'WARA OMOSA

MASTER OF SCIENCE
(Telecommunication Engineering)

JOMO KENYATTA UNIVERSITY OF AGRICULTURE AND TECHNOLOGY

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Adaptive bandwidth allocation for handover multimedia services for quality of service performance in mobile cellular networks

Edwin Nyang'wara Omosa

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DECLARATION

This thesis is my original work and has not been presented for a degree in any other University.

Signature .... Date...................................

Edwin Nyang'wara Omosa

This thesis has been submitted for examination with our approval as the University supervisors:

Signature .... Date ....

Dr. Philip Kibet Lang'at

JKUAT, Kenya

Signature .... Date ....

Prof. Stephen Musyoki

TUK, Kenya

DEDICATION

I would like to dedicate this thesis to my family especially my parents Mr. Benford Omosa and Mrs. Navency Omosa whose love, prayers, sacrifice and moral support kept me going.

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First of all, all praises to God for giving me strength and whom I owe my existence.

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LIST OF ABBREVIATIONS

3*G*	Third Generation
3*GPP*	Third Generation Partnership Project
A	Acceptable
ACAS	Adaptive Channel Allocation Scheme
AF	Assured Forwarding
AWGN	Additive white Gaussian Noise
BA	Bandwidth Adaptation
BAF	Bandwidth Allocation Factor
BB	Bandwidth Broker
BE	Best Effort
BP	Bandwidth Priority
BS	Base Station
BW	Bandwidth
CAC	Call Admission Control
CBR	Constant Bit Rates
CDMA	Code Division Multiple Access
CN	Core Network
CSPP	Complete Sharing with Preemptive Priority

CSSR	Call Setup Success Rate
DBW	Dedicated Bandwidth
DCA	Dynamic Channel Allocation
DCAS	Dynamic Channel Allocation Scheme
DiffServ	Differentiated Services
DSCP	Differentiated Service Code Point
EF	Expedited Forwarding
ETSI	European Telecommunication Standards Institute
FBA	Fixed bandwidth allocation
FCA	Fixed Channel Allocation
FDD	Frequency Division Duplex
FIFO	First In First Out
GSM	Global System for Mobile Communication
HCDP	Handoff Call Dropping Probability
IDRM	Intra-domain Resource Manager
IETF	Internet Engineering Task Force
IntServ	Integrated Services
IP	Internet Protocol
ITU	International telecommunication Union
KPIs	Key Performance Indicators

$MATLAB$	Matrix Laboratory
$MBAC$	Measurement Based Admission Control
$MPLS$	Multi-protocol Label Switching
MS	Mobile Station
MSC	Mobile Switching Centre
MT	Mobile Terminal
$PBAC$	Parameter Based Admission Control
PHP	Per Hop Behavior
$PSTN$	Public switch Telephone Network
QoS	Quality of Service
RC	Reserved Channels
$RIO-C$	Red with In/Out and Coupled
RNS	Radio Network Subsystems
RRM	Radio Resource Management
$SGSN$	Serving GPRS Support Node
SLA	Service Level Agreement
SMS	Short Message Service
SNR	Signal to Noise Ratio
SSB	Scanning with Self-back off
TDD	Time Division Duplex

TE	Terminal Equipment
TP	Transmission Priority
TSW	Time Sliding Window
U	Unspecified
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VBW	Variable Bandwidth
VoIP	Voice over Internet Protocol
WCDMA	Wideband Code Division Multiple Access
WiMAX	Worldwide Interoperability for Microwave Access

ABSTRACT

With the explosive growth of multimedia services in the telecommunication industry due to the increase in the role of communication in the recent past, Quality of Service (QoS) provisioning and ensuring fair bandwidth allocation has become more and more challenging. Moreover, bandwidth being a valuable and a scarce resource, it should be used efficiently. It has emerged that use of small cells is a better solution for achieving a higher capacity but it has its limitations. The consequence of using small cell sizes is the increased rate of call handovers as mobile users move between cells. In a network supporting multimedia services, frequent handovers not only result in network overload, but, and more importantly, may adversely affect the QoS of the calls due to handover failures. Therefore it cannot be argued that in a congested environment efficiency can be increased by increasing capacity. The best solution is to administer good resource management schemes. This necessitates for methods to manage handovers so that the mentioned problems do not occur.

To reduce handover failures and achieve high bandwidth utilization, a design of an efficient multi-class integrated framework for Call Admission Control (CAC) and adaptive bandwidth allocation for multimedia services in mobile cellular networks is presented. The scheme exploits a new prioritized adaptive bandwidth-allocation strategy that allows the reclamation of more bandwidth from on-going non-real time calls up to an optimally determined threshold.

From the simulations, the result obtained in this thesis report shows that under the adaptive bandwidth allocation scheme developed, the quality of service parameters such as throughput, packet loss, end to end delay and jitter can be controlled to satisfy the QoS requirement according to users' requirements compared to Conventional

Internet Protocol(IP) only scheme.

Chapter 1

INTRODUCTION

1.1 Background of the Problem

The tremendous growth of the telecommunication industry due to technological revolution in the area of cellular network has come with its challenges which have to be successfully solved. Among the challenges are the limitation of the bandwidth available and the provision of quality services [1]. It can be argued that as mobile users continue to grow and the demand for multimedia traffic (voice, video, and data) increases, the demand for the bandwidth required to support them also increases, adding more strain to the already scarce bandwidth. Hence there is a need to efficiently use the bandwidth available. The bandwidth should be utilized in an optimal way to ensure that more number of users may be serviced. In a network with constant traffic flows, bandwidth can be assigned to the cells depending on traffic they carry during the busy hours [2]. During times of low traffic, their bandwidth is underutilized and cannot be used to accommodate excess traffic of other channels.

According to current trends, it can be predicted that the next generation of traffic in future wireless networks will be mostly generated by personal multimedia applications [3]. For multimedia traffic 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 cellular service is to increase the system capacity and vary the allocated bandwidth [4].

The system capacity can be increased by reducing the cell size to accommodate more mobile users [5]. But by using small cell sizes there is a higher probability of increased number of call handovers when mobile users move from one cell to the next cell. This in turn increases the signaling load on the network, making it very difficult for the network to guarantee the QoS promised to a call at setup or admission time. Since handovers are usually given higher priority at the expense of new calls another problem arises, the increase in number of new calls being blocked [5] [6] [7]. These challenges can be dealt with using an efficient and effective bandwidth (or channel) allocation strategy.

In this thesis, an adaptive bandwidth allocation scheme for multimedia applications is formulated. The overall aim of the scheme is to achieve maximum efficiency in bandwidth usage in cellular networks while maintaining a certain assured quality of service.

1.2 Motivation for the Work

The motivating factor for this work was the Communications Authority of Kenya Quality of Service Report of 2013/2014. The report showed that not even one of the four telecommunication operators in the country satisfied the set minimum quality of service standards [8]. The quality of service report was based on eight QoS performance parameters and a clearly stated criteria that was approved in 2008/09 [8]. Speech quality, completed calls, call drop rate, call block rate, handover success rate, received signal strength, call setup time and Call Setup Success Rate (CSSR) are some of the most important performance parameters used by mobile operators. The report demonstrated that cellular networks in Kenya suffer from a higher call drop rate, low number of completed calls and low call setup rate. The clarity of speech delivered during a call was also found to be poor.

To comply with the ever increasing consumer need, mobile service providers have to continuously improve mobile cellular network QoS to required levels. They need to come up with QoS policies to ensure that real-time services receive the highest priority when they compete for network bandwidth.

1.3 Statement of the Problem

As users for multimedia services increase, it results in a drastic demand for network bandwidth. Bandwidth provisioning becomes a major issue as it is a limited resource. This has fueled the need for more efficient bandwidth allocation strategies. To achieve efficiency in a cellular network, it will be required that bandwidth be shared fairly among individual users according to necessity. Specifically, users requiring more bandwidth for transmission should not be limited by bandwidth resources, conversely, those requiring less bandwidth should not be over allocated.

Establishment and management of connections is important if quality of service is to be maintained in a cellular network as mobile equipment are in constant motion during communication sessions experiencing handovers from one cell to another. The cells to which the mobile traffic would be handed-over to must have sufficient bandwidth available if QoS is to be maintained. However, if available bandwidth is insufficient to accommodate the handover, forced termination occurs. Bandwidth reservation has been a method of choice in the recent past to mitigate this problem. Reserving bandwidth for handover calls has its short comings too. First, it is highly inefficient as it would require large amounts of bandwidth to be reserved. Secondly, reservation of bandwidth in the network helps in seamless interactive multimedia services provisioning, but if the bandwidth set aside is too wide, the number of new call blocked will be high due to a large amount of bandwidth reserved for handover calls though the traffic in the network might be low. In this case, the bandwidth is underutilized by not giving service to either handover call or new call. If the amount of bandwidth reserved is too small, the handover call success cannot be assured during high traffic situations in the network.

Due to random access of cellular networks by mobile users, traffic fluctuations are inevitable. This will lead to instances of burstiness in the network that causes congestion. Under this undesirable condition bandwidth cannot be over-provisioned as this would be inefficient. Therefore when traffic surge occurs, traffic with stricter QoS requirements must be given preferential treatment.

Different multimedia services have different QoS requirements. For example voice traffic may require low latency, while video download may require high assurance. Other multimedia services like web traffic may tolerate up to a reasonable amount of delay. To provide these different levels of multimedia traffic services to satisfy all QoS requirements for each level is a hard task.

CAC and bandwidth adaptation (BA) for handover calls are some of the provisioning strategies to limit the number of call connections into the networks and to degrade low priority traffic respectively. A worthwhile CAC strategy has to balance between the admitted calls and the rejected calls for it to provide the desired QoS requirements. So, coordination amongst all parties such as admission control and bandwidth adaptation should be observed.

1.4 Objectives

1.4.1 Main Objective

The main objective is to develop a new adaptive bandwidth assignment strategy for multimedia services that guarantee high capacity and high efficiency in cellular network.

1.4.2 Specific Objectives

1. To develop a threshold based adaptive bandwidth allocation policy that scalably assigns bandwidth to mobile subscribers with guaranteed QoS.

2. To develop an Admission Control algorithm and integrate it with adaptive bandwidth allocation in order to provide the desired QoS requirements.

3. To evaluate the performance, through simulation, of the superiority of this solution against conventional IP network.

1.5 Research Questions

To achieve the desired results for this research work it depends on successfully answering the well formulated research questions below:

- How do the existing QoS parameters of the Universal Mobile Telecommunications System (UMTS) cellular networks provide support for multimedia transmission?

- How can the QoS of multimedia services be improved on UMTS cellular networks?

- How does a Cellular network handle increased traffic load with limited bandwidth?

- How does a Cellular network achieve high system bandwidth utilization?

- How is QoS guaranteed through admission control and bandwidth adaptation?

1.6 Justification

Each multimedia application expects a desired level of QoS. The service provider is interested in providing the desired QoS, as efficiently as possible. This requires efficient methods for resource allocation and at the same time guaranteeing provision of QoS.

Nowadays the need for adaptive allocation of resources for multimedia applications is an imperative need because of the variable nature of cellular networks. Multimedia applications exhibit large variations in their data rate, something which makes their resource management extremely difficult. Thus multimedia applications in cellular networks need to implement highly scalable and adaptive techniques in terms of transmission rates. Taking all these into account it is clear that coming up with an adaptive mechanism for multimedia transmission poses a great challenge.

1.7 Scope of the Work

This thesis research concentrates on allocating bandwidth adaptively in a UMTS network. The UMTS network is modelled using three cells to evaluate the QoS performance. Differentiating traffic classes is achieved through Diffserv. As Diffserv is not designed for mobile nodes, to simulate handovers, nodes that represent mobile users are not moved, instead part of traffic (source node) is shifted from its current cell's Base Station (BS) to another node attached to a neighboring cell's BS with varying probability.

Only IP-based voice, video and data services will be applied and evaluated in an integrated voice, video and data in the UMTS network. In this circumstance, voice traffic refers to voice over IP (VoIP) traffic, video traffic refers to video streaming and data traffic refers to data based services such as e-mail, web-access, and file transfer.

1.8 Organization of the Study

The thesis is organized into the following chapters. Chapter one introduces the thesis. Chapter two presents the literature review. Chapter three outlines the methodology used. Chapter four provides and analyses results to evaluate the performance of the algorithm under different multimedia applications. Chapter five is the summary of the main ideas, conclusions made and recommendations for further study to be done in this area.

Chapter 2

LITERATURE REVIEW

2.1 Introduction

Some cellular wireless networks are designed to support adaptive multimedia by influencing ongoing calls to constantly change their bandwidth in response to traffic variation e.g. UMTS [9] [10]. There is a need to improve QoS under this adaptive multimedia framework. This can be achieved through a bandwidth adaptation.Furthermore the scarcity and large fluctuation of link bandwidth in wireless networks have led to the development of adaptive multimedia services where the bandwidth in a link is dynamically allocated to adapt to traffic variations.

In order to be able to add to this endeavor, this chapter starts by briefly discussing theories for adaptive bandwidth allocation algorithms which support Multimedia applications in cellular network. After that, the focus is given to investigating; and understanding the inefficiencies associated with the current adaptation techniques.

2.2 The Theoretical Review

2.2.1 Overview of UMTS

UMTS is a third-generation (3G) mobile communications system. The design of 3G wireless networks supports multimedia traffic [9] [10]. In this section the focus will be on UMTS functionality associated with handovers. As shown in figure 2.1, UMTS network is divided into three parts [10]: user equipment (UE) to send and receive messages, UMTS Terrestrial Radio Access Network (UTRAN) for bandwidth management among other tasks and the Core Network (CN) that routes messages and connects UMTS to other networks. Radio Access Network (UTRAN) consists of a Radio Network Controller (RNC) for radio resource control of a cell and mobility 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 responsible for handover. Figure 2.1 demonstrates the UMTS architecture where

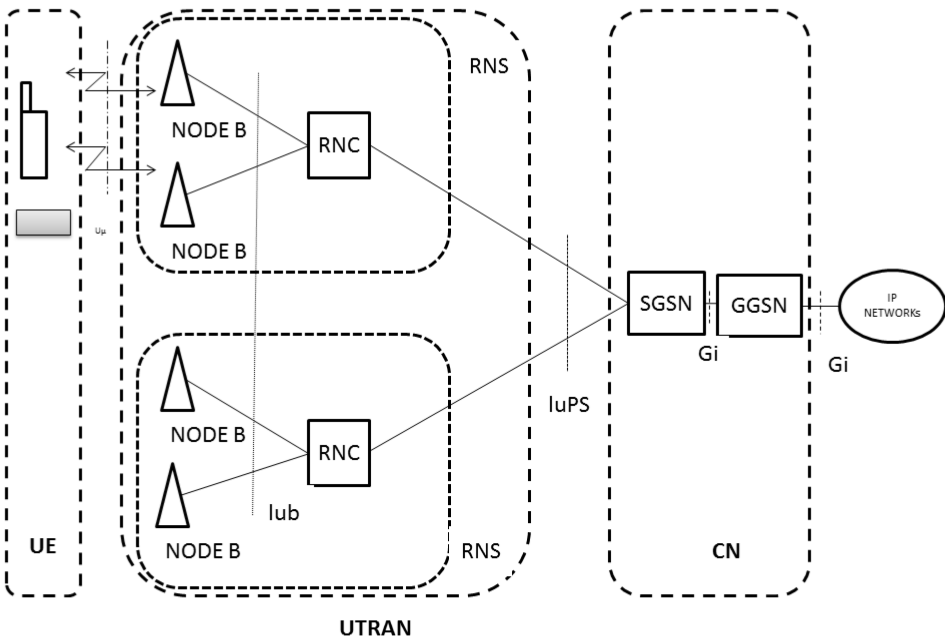

Figure 2.1: The Basic UMTS Network architecture

- IP - Internet Protocol
- Iub - interface connecting RNC with node B
- IuPS - interface connecting RNC with SGSN
- GGSN - Gateway GPRS Support Node
- Gi - interface used to exchange data with external networks
- RNS - Radio Network Subsystems

- SGSN - Serving GPRS Support Node

- Uu - interface connecting node B with User Equipment

2.2.2 UMTS QoS Requirement

The UMTS QoS is provided by the UMTS bearer service as defined in the Third Generation Partnership Project (3GPP) specification [10] [11] [12] [13]. A bearer service involves all ways that enable the provision of an agreed QoS. The concepts here include but not limited to the signal control and QoS schemes. UMTS permits a user/application to negotiate bearer properties during connection establishment. It is also possible to renegotiate bearer properties if the network is unable to meet the QoS expected depending on availability. A situation where renegotiation may apply is during handover. For example during handover the network checks the available bandwidth in a cell against the required bandwidth, then a user can either be accepted or rejected using admission control.

The architecture of a UMTS bearer service with its layers that serve unique purpose is depicted in Figure 2.2 [13];
Where

- BS - Bearer Service

- FDD - Frequency division duplexing

- Iu - interface that connects RNC to core network (CN)

- MT - Mobile Termination

- TDD - Time division duplexing

- TE - Terminal Equipment

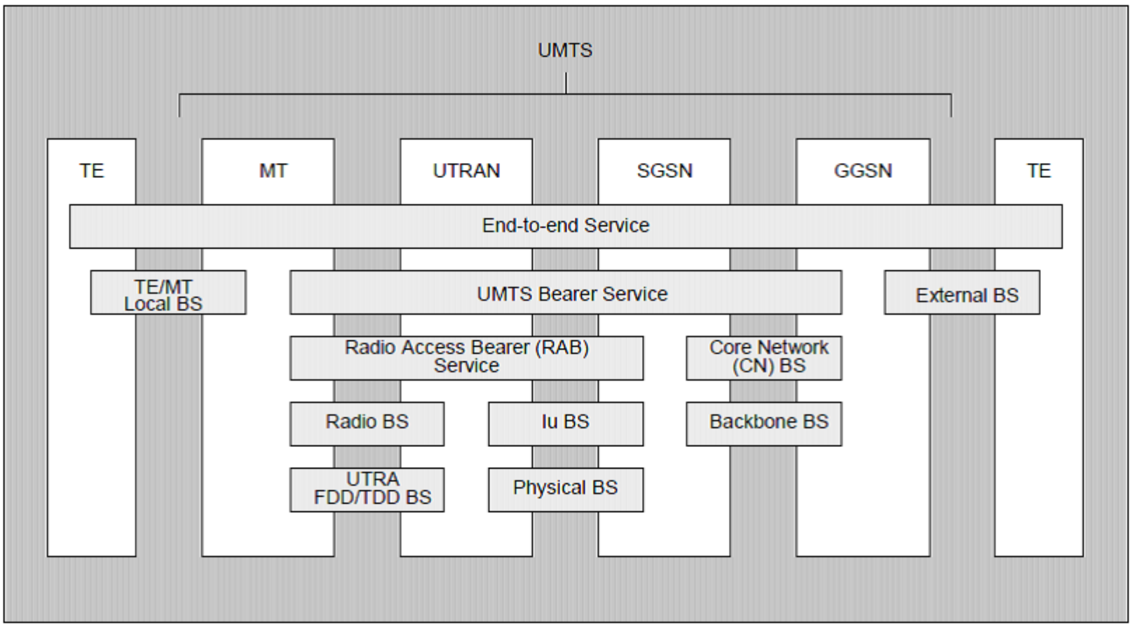

Figure 2.2: The Basic UMTS Network QoS Architecture

In UMTS there are four traffic classes that have been defined by 3GPP that attempts to satisfy different traffic QoS requirements. The classes are; conversational, streaming, and interactive and background classes[10] [11]. Applications and services in these classes can be categorized differently according to their QoS requirements. The main feature that differentiates these classes is how responsive their traffic is to delay[10]. Conversational class is the most responsive, followed by streaming class, then interactive class. The background class is the most unresponsive to delay.

- Conversational Class: As its name suggests, it provide conversational services. Real-time conversation takes place when human beings are talking. It is only type of the four classes where the required characteristics are strictly imposed by human perception. They include real-time services such as voice calls over IP or video conferencing. It is characterized by a low end-to-end delay.

- Streaming Class: It is a one-way real-time services transport. Example of services includes multimedia streaming services such as video streaming which is processed in a steady and continuous stream. Due to buffering, a certain amount of delay is tolerable. This also means that they tolerate more jitter. Jitter can be easily smoothed out by buffering. With streaming, data can be displayed before the entire file can be transmitted.

- Interactive Class: It applies when an client, which could be either a machine or a person, asks for information from a server. At the destination of the requested information the client is always anticipating the information requested within a certain time frame. It is characterized by round trip delay time. Applications that require a reasonable response time come under this class. Low bit error rate is essential for this class. Examples of applications are serving through the web , getting information from the database and querying a server.

- Background Class: It applies when an end-user receives data that does not demand to be acted upon immediately. Example of services includes sending E-mails and Short Message Service(SMS). This class requires to that the packets should be transmitted with a low bit error rate. For this class, delay is not so much a concern.

2.3 Bandwidth Adaptation Concepts

2.3.1 Bandwidth Adaptation Algorithm

This algorithm increases or decreases the assigned bandwidth of an ongoing call in a cell depending on the network traffic [14] [15]. That means that its function is to allocate and reallocate the desired bandwidth to multimedia call when the cell is or not overloaded respectively. The algorithm performs two main roles. The first one is the reduction procedure that is activated when the incoming handover call reaches at a loaded cell. The second one is expansion procedure which is activated when there is an outgoing call handing-over to another cell or there is a call completion in a given cell.

2.3.2 Adaptive Multimedia Applications

A multimedia application is one that can change its bandwidth requirement in a network during its lifetime [16]. To model a multimedia application it is crucial to know what it entails. A multimedia application content entails a collection of two or more media sources that make up a multimedia presentation. It includes but not limited to audio traffic, video traffic and data traffic which can be in the form of text or data files. By observation it is true that parts of multimedia application will vary from one application to another to an extent that even two application of the same media type will generate different amount of data and with a different method. For example the same video compressed with two different codec such as mp4 and avi. This necessitates for development of a single traffic model to cater for the different media types

2.3.3 Dynamic Bandwidth Allocation

This is a technique of bandwidth allocation on the amount of bandwidth need by the user depending on network conditions at a particular instance in time.That is to imply that bandwidth is allocated based on the number and type of activities taking place at that instance [17] [18].

Basically there are two types of bandwidth allocation methods: static and dynamic. In Static bandwidth allocation, a channel is dedicated some amount of predefined bandwidth, regardless of whether it uses it or not, and thus the bandwidth resources are never released for use by other services [17].

Dynamic bandwidth allocation is continuous process part of bandwidth management. As calls connections are still on, the network can assign a portion of the available free bandwidth to each of the calls to ensure that each call has sufficient bandwidth to function efficiently. Once a particular call completes, the freed bandwidth can be used by other calls as my be required. The most important advantage of dynamic bandwidth allocation is that applications that need considerable bandwidth at a certain point but can work with much less at a later time can be adapted. Bandwidth that is not being utilized can easily be allocated to where it is in high demand [17] [18] [19].

2.3.4 Bandwidth Allocation Mechanisms in Cellular Wireless Networks

As the telecommunication industry is fast evolving from voice centered communication to applications of multimedia, there is increased bandwidth requirements per user, limiting the system capacity. System capacity can be improved through shrinking cell size [20]. This increases the handovers, compromising the QoS. In order to adapt to changes in traffic pattern, the status information of traffic in a cell should be considered in QoS provisioning. For example it should be determined whether the traffic is real time or non-real time.

It is unrealistic to eliminate handover call dropping completely. The best step we can take is to keep them below a certain level, which this research has addressed. Since handovers are given high priority at the expense of high new call blocking because users are more sensitive to an ongoing call being dropped than blocking a new call, it leads to poor channel utilization [5] [6] [7]. Thus, the problem formulation is to maximize the bandwidth utilization and at the same time to guarantee QoS requirements for wireless cellular network. This can be solved through an effective and efficient bandwidth allocation strategy.

Future wireless networks are projected to support a variety of multimedia applications and more mobile users. Advanced and future wireless network are designed to provide adaptable radio resource allocation capabilities that can efficiently support multimedia traffic. This can be justified by the UMTS network [21]. Here an algorithm is developed that allows flexible reallocation of resources to guarantee QoS in a network. QoS provisioning in an adaptive multimedia in cellular wireless network is to develop CAC that work in conjunction with Bandwidth Adaptation (BA). In UMTS systems, set of services are available to users whichever part of the world they may be. The main advantage of UMTS is its fast processing speed for calls. Current rates of transfer for broadband information are 2 Mbps [21]. This speeds are capable of supporting movie downloads and video conferencing. Four traffic classes are identified in UMTS, which are conversational, streaming, interactive, and background classes.

2.3.5 Adaptive Bandwidth Provisioning for Radio Access Network

The design of advanced cellular networks like the UMTS provides flexibility in radio resource allocation. These capabilities allow bandwidth of on-going calls to be reconfigured thus ensuring efficiently support for adaptive multimedia traffic [21]. The radio access network of the UMTS is usually referred to us as the UTRAN [21]. The UTRAN is responsible for handovers. The architecture of the UTRAN is of pyramid or hierarchical nature [22]. Bandwidth provisioning in UTRAN starts at RNC at the top to the base station at the bottom. As mobile subscribers (UE) are constantly roaming, handovers from one Node B (base station) to another may occur. It is expected that handovers will occur more frequently in a densely populated area or when the mobile subscriber is moving at a high speed. During handover, the destination Node B should have enough bandwidth to accommodate the handover call. But due to the burst nature of multimedia traffic, traffic pattern in a cell changes from time to time. Researchers have proposed schemes where bandwidth is reserved in advance before handover to mitigate this situation. However, this has been found to result in low bandwidth utilization when traffic in a cell is high and also when the mobile subscriber is moving at a high speed. To overcome this problem and maintain QoS, an adaptive bandwidth allocation will be a better solution. For example in a UMTS network, bandwidth for a call can be dynamically upgraded and downgraded during the call session as long as it falls within a tolerable range in response to changes in traffic load. When handovers occur, the new traffic entering the new cell, share the bandwidth with the other traffic already present in the cell.

2.3.6 Handover Prioritization

There are two techniques to this;

1. **Guard Channel Concept:** Here a number of channels in a cell are reserved exclusively for handover request [3].

   The total carried traffic is reduced if fixed channel assignment is used. However, if channels are assigned dynamically,the bandwidth utilization in guard channel mechanism can be improved. If the number of channels to be reserved is low, the number of handover call rejected is high. If the number of channels reserved is high it may result in bandwidth wastage and rejection of large number of new calls. There is also a general scheme known as fractional guard channel [23]. Here an originating call is admitted with certain probability that is determined by the number of channels that are engaged.

2. **Queuing:** Queuing of handovers is possible because of a constricted window of time between which the received signal level drops below handover threshold and the time the call is brought to an end due to a low signal level [23]. The delay size is determined from the traffic pattern of a particular service area. Calls are accepted whenever there is a free channel. Its objective is to decrease the probability of a call being ended prematurely due to an inadequate bandwidth.

2.4 Mathematical Meaning of Bandwidth

Let us begin by understanding the main parameter of network performance, known as channel capacity. Channel capacity is the upper boundary on the rate of information that can be transmitted in a communication channel. The channel capacity that can be comfortably supported by a channel is limited by Shannon-Hartley Capacity Theorem [24]. The Theorem is given by the equation below for a noiseless channel [24] [25]:

$$C = B \log_2 (1 + \frac{S}{N}) \tag{2.1}$$

where C- is the maximum bit rate referred to as channel capacity measured in bits/s, B represent the bandwidth of the channel which measured in Hz, S represent the average received signal power over the bandwidth expressed in Watts, N is the average noise power over the bandwidth while $\frac{S}{N}$ represent the mean-square signal to noise ratio which is not in dB. Shannon's work showed that the values of S, N, and B set a limit upon the transmission rate. For a channel that is susceptible to Additive white Gaussian Noise (AWGN) the Shannon-Hartley theorem is re-written as [24] [25]:

$$C = W \log_2 (1 + \frac{P}{N_0 W}) \tag{2.2}$$

where P denotes the average received power measured in Watts, N_0 the noise power spectral density expressed in Watts/Hz, W a certain bandwidth and $\frac{P}{N_0 W}$ is the received signal-to-noise ratio (SNR). In wireless communication bandwidth is the numerical difference that range between limiting frequencies (upper and lower) within certain waveband. The Shannon formula represents the theoretical maximum capacity that can be achieved [24].

2.5 Diffserv

DiffServ is a scalable Internet Protocol (IP) based technology developed by Internet Engineering Task Force (IETF) which can efficiently provide QoS in networks by providing bandwidth discriminately to different categories of traffic [26] [27]. 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 follow a pre-determined forwarding Per Hop Behaviours (PHBs) [26] [27]. 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, 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 [26].

2.5.1 Differentiated Services and QoS

From the definition according to [28] [26] [29] 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 [28].

The DiffServ module in Network simulator (NS2) consists of three major parts [26] [27] [27]:

- **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 forward them accordingly.

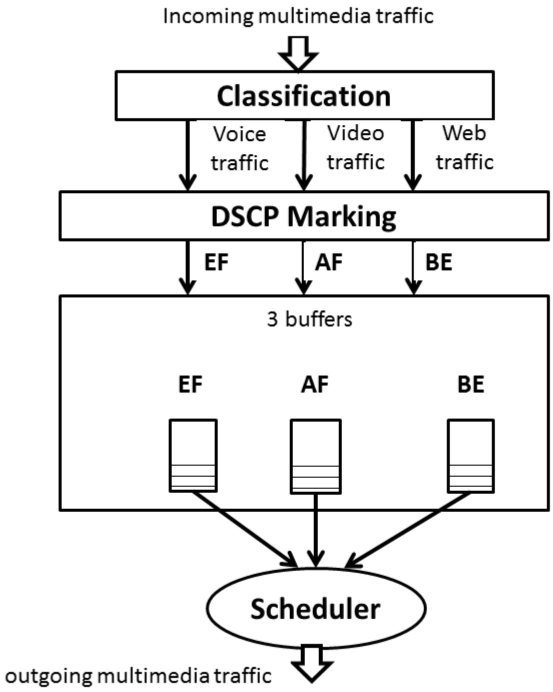

Figure 2.3: Diffserv packet forwarding steps

In Diffserv packets are forwarded in three steps [27] [29] which are illustrated in Figure 2.3.

2.5.2 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 EF DSCP value of Expedited Forwarding (EF) class. Video traffic is marked with AF DSCP value of Assured Forwarding (AF) class and Web traffic eg.

files download is marked with BE DSCP value of Best Effort(BE) class. It is done at the edge routers at the sender side.

2.5.3 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.

2.5.4 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.

2.6 Mapping UMTS into DiffServ

UMTS network support various services including data, voice and video applications. To enhance support for these services, 3GPP has adopted the IETF Differentiated Services (DiffServ) [9] service Model.

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 are 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 is shown in Table 2.1 [9] [30].

Table 2.1: Mapping UMTS into DiffServ

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

2.7 Admission Control

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 or handover traffic flow requesting service from a cell should be admitted or rejected [31]. In other words it allows the acceptance or refusal of new or handover traffic into a network. For mobile user's satisfaction, traffic flows should be granted the requested QoS without affecting earlier guarantees [32] [33] [34]. 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.

In the past, voice and data services had distinct network infrastructures for the 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. In best-effort 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 Differentiated Service (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 [35]. 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 bandwidth broker (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 [36]. A traffic flow is only prohibited into the network only if bandwidth is scarce even after degrading its services.

2.8 Bandwidth Broker

In the DiffServ framework, resource management module is absent [37]. That is to say that there are no admission control strategies to regulate traffic in the network. For traffic control, a bandwidth broker (BB) is used which has been defined by the Internet2 Qbone Bandwidth Broker Advisory Council [37] [38]. 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 [38]. The former ensures QoS requirements are satisfied in a network and takes the obligation of admission control and resources reservation. The latter decides the path that the admitted traffic flow will take to the receiver.

2.9 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) [28]. The SLA is a contract between the service provider and the mobile subscribers with emphasis on how to meet the agreed QoS.

2.10 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, burst size and 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. On a condition that a request is granted, the Bandwidth Broker ensures that it will be met by the network. Admission controls significance to the Bandwidth Broker operation is to eliminate bias between the requests and the degree of network utilization. This is achieved through degradation of the admission control service for the users. Finally, the BB sends a message to the sender and updates its database.

2.11 Mobility

The evolution of UMTS envisions providing anytime and anywhere mobile multimedia services. To provide uninterrupted communication, the destination cell must have enough bandwidth [39] [40]. As mobile terminals move between cells a process known as handover, they may be moving from a high bandwidth cell to a cell that is crowded, which may be experiencing bandwidth congestion. For example, mobile users from a cell with the satisfactory level of bandwidth may find it hard to continue with the required QoS soon after moving into a cell having little or no bandwidth to offer. Despite this mobile terminals, expect to have the same QoS wherever they roam to. In this situation, a fair bandwidth allocation algorithm will be desirable that attempts to optimize the overall satisfaction for the users in the cell when it is impossible to provide the desired bandwidth to all. Bandwidth reservation can be a solution in order to maintain QoS. But, reserving bandwidth for a large number of mobile users is highly inefficient, as this would demand a large amount of bandwidth to be reserved [39]. This answers the need for better bandwidth allocation schemes. Bandwidth is a limited resource; therefore it should be used efficiently. In a crowded cell with insufficient bandwidth, high priority traffic that demand high QoS should not be denied bandwidth to low priority traffic that may only require minimum services. So, bandwidth should be shared fairly among different service requirements according to necessity. Specifically, users requiring more bandwidth for transmission should not be limited by bandwidth resources, conversely, those requiring less bandwidth should not be over allocated.

2.12 Handover

This is the process of transferring an ongoing call from one cell to another during mobile user's movement. It is initiated when a mobile user's signal weakens in a certain base station and that mobile user can be provided a stronger signal by another neighbouring base station [39] [41] [42]. This means that the main reason for a handover is the deterioration of signal quality in the current cell [43] [44] [45].

2.12.1 Types of Handovers in UMTS Network

The different types of handovers in UMTS network have been covered in [41] [42].They include; horizontal handover, vertical handover, inter system handover, soft handover, softer handover and hard handover.

2.13 Existing Related Works

2.13.1 Methods of Bandwidth Allocation

Here literature is reviewed. It seeks to explain the reasons why some methods for bandwidth allocation are insufficient.

2.13.1.1 Integrated Service (IntServ)

The existing Internet architecture provides a best effort service [46]. All traffic is treated equally on a First In First Out (FIFO) queuing method. This implies that there is no mechanism for distinguishing between delay sensitive and best effort traffic. Also best effort service does not guarantee for end-to-end QoS [47] [48] [49]. To provide QoS for IP based networks efforts were directed towards IntServ model [47] [50] [51] [48] [49] [52]. IntServ provides QoS using resource reservation and call admission control by focusing on individual packet flows [46] [50]. Good research has been done on this model, however it has been found to have some drawbacks. The main issue is the scalability of bandwidth [47] [50] [51] [48] [49] [50] [52]. For IntServ to function, all routers in a network must store state information. This means that IntServ would be preferred on a small-scale, but cannot be scaled up to a system the size of the internet as it would be daunting task to keep track of all of the reservations such as Flow identification (using IP address, port etc), previous hop identification, reservation Status and reserved Resources [46] [53].

2.13.1.2 Traffic Prediction Methods

Traffic predication methods can alleviate congestion as well guarantee QoS and ensure optimal bandwidth utilization. Traffic prediction for long period is associated with forecasting of traffic models to evaluate future capacity requirements while short period prediction is associated with dynamic resource allocation. In [54] a Short-term traffic speed forecasting hybrid model using chaos-wavelet methodology that supports supervised learning is presented. Wei and Chen researched on prediction over short time where large traffic flows in subway using empirical methods and neural networks [55]. In [56], a prediction method that is short term is used in a rural highway and it takes the artificial neural network methodology. Other prediction models include regression analysis models [57] and time series models [58]. Though great research has been done on short period prediction models, still some challenges exist. First, to forecast future trends accurately, it would require long prediction duration but majority of research in short period prediction models is done in one-step prediction. Second, multimedia traffic fluctuates with some randomness and since short period prediction models depends on the data of the prior time instant as its input, there is a high likelihood of large prediction errors.

2.13.1.3 Priority Queuing

In Queuing tasks follow the first-in first-out basis [59] [60]. However, some tasks may be more important or timely than others. Priority queuing provides a mechanism for ordering tasks based on priority that ensures high priority tasks are served first at the expense of low priority tasks [59] [60] [61] [62]. For example Customers who pay more pay more are get served first. But it has its concerns too, if services cannot be differentiated, customers may get bad services. Also a specific level of service cannot be assured if the more customers join the high priority service class. Lastly, priority queuing satisfies the needs of a higher priority customer first without regards of the needs of low priority customer.

2.13.1.4 Fuzzy Logic Bandwidth Allocation Methods

In order to effectively control and manage multimedia traffics in a cellular network, there is a need for intelligent bandwidth allocation methods that guarantees QoS. In the previous researches done, proposals on fuzzy logic bandwidth allocation methods have been studied. D. S. Shuaibu, et al have proposed a fuzzy logic partition-based call admission control [63]. The scheme partitions bandwidth in a network into three separate classes that corresponds to constant bit rate (CBR), variable bit rate (VBR) and handover services. An admission control scheme based on fuzzy logic was enacted in the handover portion to intelligently keep dropping probability as low as possible based on the available bandwidth. An ideal concept is developed in [34] to optimize bandwidth use by implementing a fuzzy logic controller of the crucial factors affecting the performance. It uses MATLAB to study the system.

It is generally known that the performance of a fuzzy controlled system depends on the rules defined for it [64] [65] [66] [67] . The rules defined may not always be the optimal. Also, fuzzy controlled system make use of different sets of functions for various networks and for different QoS requirements, this makes it hard to choose a function that is satisfactory.

2.13.1.5 Fixed Channel Allocation (FCA)

Here a number of predetermined channels are assigned permanently to each cell in the network during the planning phase[68]. The advantage with FCA is that it is simple. The allocation is static and cannot be changed. The drawbacks with FCA is that in an IP network where multimedia communication occurs, the traffic is bursty and the traffic load fluctuates from time to time. This results in great inefficiencies.

2.13.1.6 Dynamic Channel Allocation

Dynamic Channel Allocation (DCA)tries to overcome the deficits of FCA. Here channels are centrally placed and the allocation to calls that comes/arrives at a cell is dynamic [68] [69] . After each call is completed, the channel is returned to the central pool.

For user mobility management an analysis of a QoS Handover scheme is done that exploits Service Degradation and Compensation to reduce the handover dropping probability [6]. A novel scheme that dynamically assign channels for call admission control is devised in [7]. Arafat Abu Mallouh, et al has proposed DCAS that make use of artificial intelligence to assign channels optimally [68]. This strategy is implemented in an environment with uniform and non-uniform load distribution. It is based on the following factors: size, coordination, frequency reuse, and handover to bring the allocation process into compliance with expectations. As the base stations are obligated to communicate intelligently with other base stations in order to allocate channel intelligently it leads to the rise in cost of signaling as information has to be exchanged with the neighbouring cells. In [70], dynamic bandwidth allocation algorithm in the cellular networks for multimedia applications with/without traffic in the background is presented . The system is simulated for many application that take place concurrently and it is deduced that if services are to be provided with reasonable delay time, a frequent adjustment of policy at call admission has to be done, accounting history and criticality of applications. As can be viewed this strategy does not classify multimedia traffic.

Among the most crucial issues in offering real-time communication services in a mobile network is support for uninterrupted handover between BSs to preserve communication. To manage user mobility in WiMAX networks a Dynamic Bandwidth Allocation Scheme for Efficient Handoff in IEEE 802.16e Networks is deduced [69].

Here they give high priority to traffic with high Bandwidth Allocation Factor (BAF). The number of users is increased by assigning bandwidth dynamically banking on the Arrival Rate (λ). To make it more efficient, a Scanning with Self-back off (SSB) scheme is used.

The Dynamic channel allocation scheme is under the constraint that it does not violate frequency reuse conditions [68] [70]. However it also has its shortcomings. First, it has a degree of randomness in the reuse distance. This leads to the fact that frequency reuse is often not maximized compared to FCA where cells using the same channel are separated by the minimum reuse distance. Secondly, DCA involve algorithms that are more complicated for making decisions on the availability of a channel that is most suitable for producing desired results. These algorithms may involve a lot of calculations and may require large computing resources in order to be real-time.

2.13.1.7 Guard Channel

Here part of total the channels/bandwidth in a cell is reserved for exclusive use of handover calls [[70]. The remaining channels/bandwidth is shared equally between handover calls and new calls.

Bandwidth reservation is a crucial for improvement of the performance of cellular networks. In [71], bandwidth reservation strategy is considered which first reserves some amount of bandwidth for handover calls then the bandwidth can be increased for handover calls by the base station based on the user mobility. This is to say that the base station dynamically increase the reserved bandwidth for handovers when the initially reserved bandwidth is not enough, minimizing delay and at the same time increasing the system throughput. The importance of [71] is to set aside some bandwidth for mission-critical application and best effort traffic. The problem in the number of guard channels to be chosen.

The problem of managing of the available bandwidth in a wireless environment is still a challenge due to mobile subscriber mobility. Various researches have been done in this area to optimize bandwidth in cellular mobile network. For instance 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. It gives priority to handover through admitting handover calls of a class with low priority to the guard channels of its next higher class with a certain probability that depends on how channel capacity is occupied and the mobility of calls.
The guard channel scheme like the many schemes discussed previously has its limits too; if the number of channels reserved is low, large numbers of handover calls are dropped. If the number of channels reserved is high, it may result in waste of bandwidth and blocking of large number of new calls [7] [71].

To Further explore on studies that have been done, a call admission control algorithm that utilizes an adaptive multi-level bandwidth-allocation scheme for non-real-time calls is presented in [5]. This scheme is able to reduce handover call dropping probability (HCDP) to a negligible level but it fail to account for call blocking probability. A strategy for call admission control using Diffserv in wireless networks is presented in [35]. 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 Red with In/Out and Coupled (RIO-C) 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.

A Bandwidth Broker architecture for quality of service has been proposed in [72] with suggestions and advancement of architectures that have already existed like

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 as a substitute for a central server makes the scheme more complicated. This in turn results to additional costs. In the scheme formulated by Okumus and Dizdar, they attempted to solve the preemption and QoS problem through intra-domain resource manager (IDRM) [37]. 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 also a lot of signaling overhead.

In [73], the paper weighs measurement based admission control (MBAC) against parameter based admission control (PBAC) and also it includes circumstances where admission control is not involved. 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. Admission control techniques for UMTS are presented in [74]. 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.

2.14 Summary

From the literature review it is evident that a lot of techniques have been developed for bandwidth allocation and their shortcomings highlighted. To add to those challenges, multimedia traffic being of variable nature, its bandwidth allocation complicates the process further. As mobility of users is an ever changing process, there should be a mechanism that is sensible to this process. A good bandwidth allocation scheme has to assign bandwidth adaptively depending on the network conditions.

The provision of sufficient quality for mission-critical services and achievement of efficient bandwidth utilization are the driving forces behind this research work undertaken in this thesis. This research work takes the Differentiated service approach to offer flexible multimedia traffic differentiation in a UMTS network by developing an adaptive bandwidth allocation algorithm for handover multimedia services. Multimedia traffic is grouped into only three service classes. This is a very small number and simplifies the management of traffic especially when configuring desired policy for a particular class of traffic according to agreements between service providers and their customers. As flows are processed in an aggregated manner, signalling overhead at nodes is reduced significantly. Also in this adaptive bandwidth allocation, customers are able to know the quality of service they will expect during congestion and the service provider can make informed decisions about the bandwidth to be provided. The simulations are designed in network simulator (NS-2.35). Researchers have extensively used this simulator which means its functionality has been thoroughly tested. This shows that it is a proven simulator.

Chapter 3

METHODOLOGY

3.1 The Research Designs

3.1.1 Model Description

Understanding the nature of traffic in a system and choosing an appropriate traffic model is important for the simulation study to succeed. A general model with classes of multimedia traffic in mobile cellular network was considered. In this model as discussed below three cells were simulated in NS2 to evaluate the QoS performance of a mobile cellular network. NS2 was the preferred simulator in this research because of the following reasons; First, traffic flows can be visualized. Secondly, it is an open source software. Lastly, researchers have used it extensively which means its functionality has been thoroughly tested. This shows that it is a proven simulator. Figure 3.1 depicts the three cells arrangement in a UMTS network where

$SGSN$-Serving GPRS Support Node

RNC_1- Radio network controller of the BS_3

RNC_2-Radio network controller of theBS_1 and BS_2

BS_1- Originating base station of traffic

BS_2- Base station that is handed over the traffic

BS_3- The destination base station of the traffic

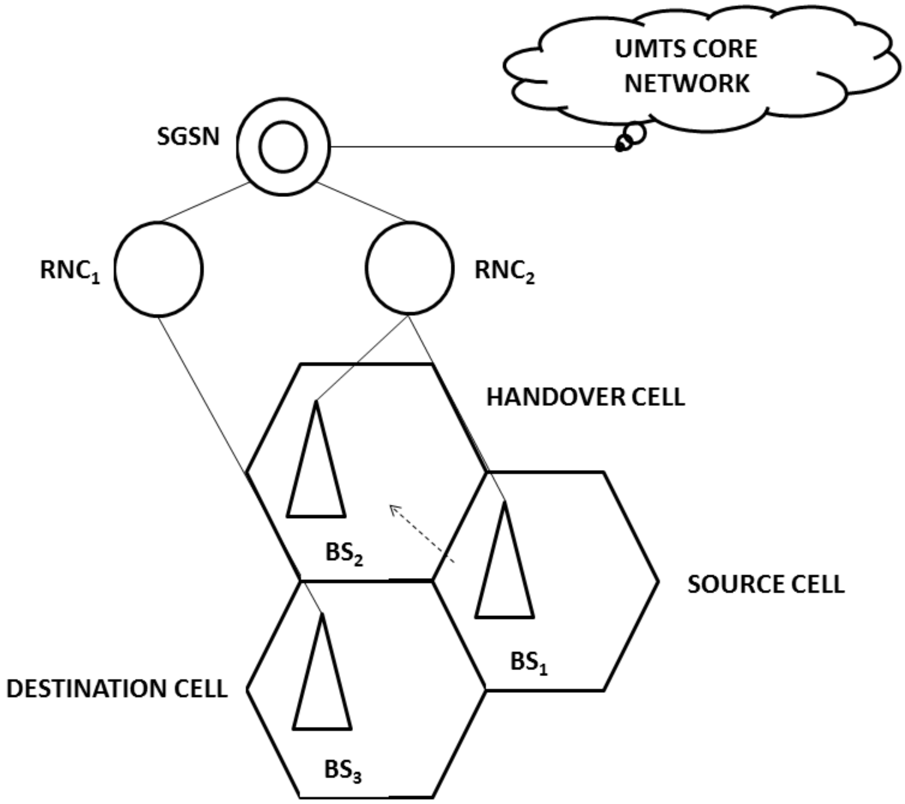

Figure 3.1: Three cells arrangement in a UMTS network

3.1.2 Traffic Model

Before analyzing the performance of a mobile cellular network, it was crucial to come up with a traffic model. The study consisted of three cells; the traffic originating cell, the traffic hand-over cell and the traffic destination cell.

The following assumptions were made on handover behaviors in the traffic modelling

The calls made carried constant bit rate for voice and variable bit rate video and web content (files downloading). The model made use of common assumptions that handover calls follow a Poisson process [33] [75]. Thus, packet inter-arrival times were 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 packets/s
- Then mean inter-arrival time $1/\lambda$ = 0.002 s/packet
- If packet size is 500 bytes
- Then transmission rate = 500 bytes x 500 packets/s x 8 bits = 2 Mbps

The following discussion gives a brief definition of the Poisson process in three different but equivalent ways [75].

1. It is a simple birth process with zero deaths:
   In an infinitely small window of time (δt) only one arrival may come up . This occurs with the probability (δt) that is does not depend on arrivals outside the small window of time.

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

   $$P_n(t) = \frac{(\lambda t)^n}{n!} e^{(-\lambda t)} \tag{3.1}$$

   Where: t is used to define the window of time between 0 and t ,n is the sum of the number of arrivals between 0 and t and λ is the sum of average arrival rate.

3. The inter-arrival times are independent and obey the Exponential distribution $(P_0(t))$: Let us consider a special case of Poisson distribution where we assume that no arrivals occurs in a given time period. Without any doubt, it is a straightforward answer that by substituting n with 0 in equation 3.1, the following equation is deduced:

   $$P_0(t) = e^{(-\lambda t)} \tag{3.2}$$

Let T_n be the call holding time with an indiscriminate variable exponentially distributed with parameter μ(in 1/seconds)

$1/\mu$ is the average duration of a call in seconds

Let T_h be the cell residence time with an indiscriminate variable exponentially distributed with parameter η (in 1/seconds)

$1/\eta$ is the average time a terminal stays in a cell in seconds

Let T_c be the channel holding time and is defined as $T_c = min(T_n, T_h)$. It can be proved that it is exponential distributed.

$$E[T_c] = \frac{1}{\mu_c} = \frac{1}{\eta + \mu} \tag{3.3}$$

A call handover probability is determined by these features, (a) the average cell residence or sojourn time $(1/\eta)$ and (b) the average call duration $(1/\mu)$.

As an exponential distribution assumption is made to both the call duration and the cell residence time,then the probability of a call handover(P_h) at a given time interval is:

$$P_h = P(T_n > T_h) \tag{3.4}$$

$$= \int_0^\infty f_h(T_h)[\int_{T_h}^\infty f_h(t|T_h)dt]dT_h \tag{3.5}$$

$$= \int_0^\infty \eta e^{-\eta T_h}[\int_{T_h}^\infty \mu e^{-\mu t}dt]dT_h \tag{3.6}$$

$$= \frac{\eta}{\eta + \mu} \tag{3.7}$$

Traffic Flow in and out of cell

New traffic

Let λ_n be the average intensity of new traffic (in calls per sec)

Let P_b be the call block probability

The new traffic intensity into a cell is $\lambda_n(1 - P_b)$

Handover traffic

Let λ_h be the average rate of handover towards the cell

It can be assumed that on average the rate of handover towards the cell is equal to the rate of handover due to mobile users leaving the cell.

Let P_{hf} be the handover failure probability (it should be noticed that P_{hf} is it not the same as P_h). The rate of handover entering the cell is $\lambda_h(1-P_{hf})$.

The handover rate λ_h is equal to the total new call rate times the probability that a call will make handover, P_h.

$$\lambda_h = P_h[\lambda_n(1-P_b)+\lambda_h(1-P_{hf})] \Longrightarrow \lambda_h = \frac{P_h(1-P_b)}{[1-P_h(1-P_{hf})]}\lambda_n \qquad (3.8)$$

The handover-call arrival rate into a cell is calculated as:

$$\lambda_h = \frac{P_h(1-P_b)}{[1-P_h(1-P_{hf})]}\lambda_n \qquad (3.9)$$

3.1.3 Approximations

If the call drop and block probabilities are negligible $(small, P_b, P_{hf} << 1)$, then;

$$\lambda_h \approx \frac{P_h}{1-P_h}\lambda_n \quad P_b; P_{hf} << 1 \qquad (3.10)$$

The forced termination probability P_d is the probability that a call is dropped due to some handover before it is terminated. It can be calculated as:

$$P_d = \sum_{i=0}^{\infty} P_h^i(1-P_{hf})^{i-1}P_{hf} = \frac{P_h P_{hf}}{1-P_h(1-P_{hf})} \qquad (3.11)$$

If P_{hf} is small:

$$P_d \approx \frac{P_h}{(1-P_h)}P_{hf} \qquad (3.12)$$

3.1.4 Mobility Model

To simulate handover, mobile nodes that represented mobile users were not moved instead part of traffic (source node) was 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 3.2, this means that if a mobile source from S_0 were to initially send traffic to destination D_0, after a handover, S_3 which is located in neighbouring cells could be sending traffic to destination D_0. This is similar to if D_0 were to send traffic to destination S_0, after a handover, it could be sending traffic to destination S_3 which is located in neighbouring cells.

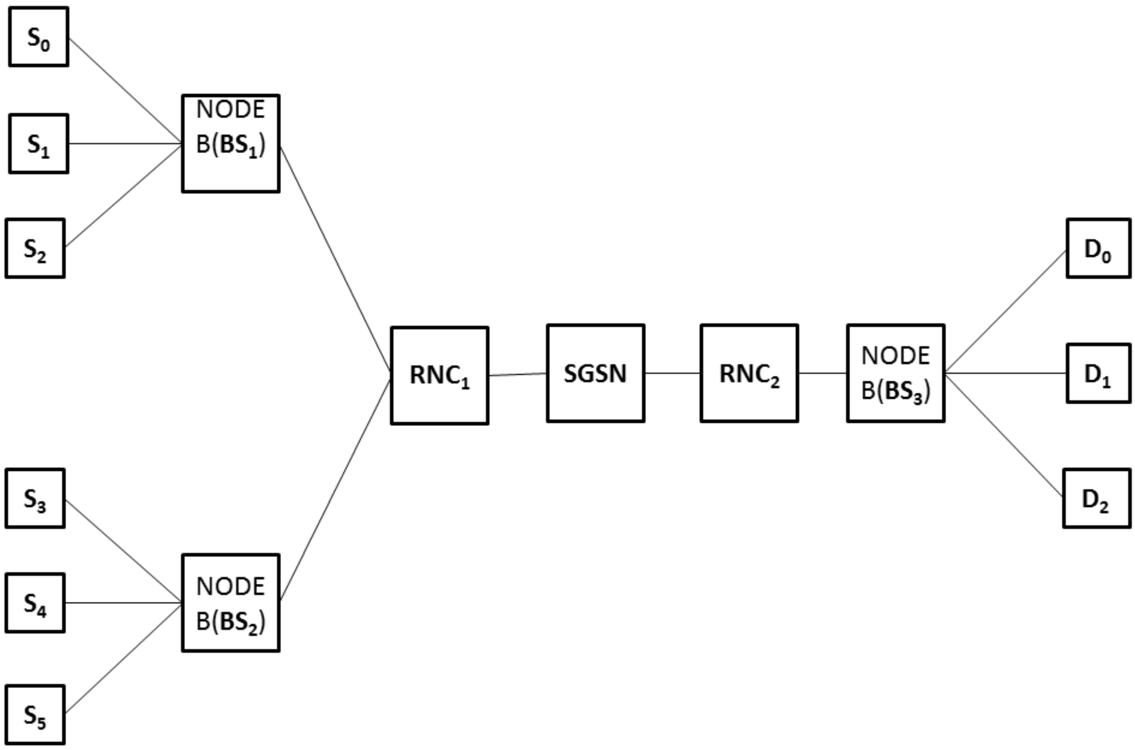

Figure 3.2: Simulation Model of an IP-based Radio Access Network of a UMTS

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

Because investigating a network through numerical analysis is a daunting task, this study took the simulation approach. The center of attention in this thesis was the simulation of an IP-based Radio Access Network of UMTS functionality on NS-2. The base stations were integrated with IP routers. Each base station router mapped the UMTS packets into IP packets for transportation in the IP network technology. In this simulation, different scenarios were designed for different classes of traffic and the performance was measured using parameters such as throughput, packet loss, end to end delay and jitter which are explained below.

3.1.5 Simulation Parameters

The simulation settings and parameters are summarized in Table 3.1.

Table 3.1: Simulation Parameters

Parameter	**Specification**
Users Per Cell	3
Number of Cells	3
Simulation Time(seconds)	10
Packet Size	512 bytes
Voice Traffic Source	CBR
Video Traffic Source	Exponential on/off
Web Traffic Source	FTP
Transport Protocol for Voice and Video	UDP
Transport Protocol for Web	TCP
Rate in cell before handover	1.8Mbps
Rate in handed-over cell	2Mbps
Queue Length voice	20 (min) - 40 (max)
Queue Length video	11 (min) 20 (max)
Queue Length web	1 (min) 5 (max)

3.1.6 Simulation of a Scalable Adaptive Bandwidth Allocation Policy

The simulation model consisted 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 $(D_0 - D_2)$ as shown in figure 3.2. The simulation was conducted using NS-2 version 2.35 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. To obtain clear graphs, Matlab was also used.

Three multimedia traffic were simulated. They included

1. $S_0 - D_0$ source destination pair which represented voice traffic with EF PHB,
2. $S_1 - D_1$ source destination pair which represented video traffic with AF PHB, and
3. $S_2 - D_2$ source destination pair which represented web(files download) traffic with BE PHB.

where PHB is the Per Hop Behaviours,EF is Expedited Forwarding, AF is the Assured Forwarding and BE is the Best Effort

After handover the traffic sources S_0, S_1 and S_2 were not moved but were shifted with a certain probability. As shown in Figure 3.2 S_0 was shifted to S_3 , S_1 was shifted to S_4 and S_2 was 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 was marked with Differentiated Service code point(DSCP) 46, video traffic was marked with DSCP 20 and web traffic was marked with DSCP 0. If a traffic flow refused to conform to its profile as defined, it was assigned a reduced bandwidth. For example video with DSCP 20 was downgraded to DSCP 21 and in the same way web with DSCP 0 was downgraded to DSCP 1. If a traffic flow did not still conform to the downgraded traffic profile then it was discarded.

3.1.7 Admission Control model

Admission control module decides when to prohibit or allow a flow [73] [74]. On receiving a traffic flow request along with its service level requirements, a BB checks for current assigned bandwidth of traffic specified with its PHB. It then uses known policies to determine the available bandwidth for new traffic arrivals. These policies then allocate bandwidth according to criticality of the traffic to ensure fairness.

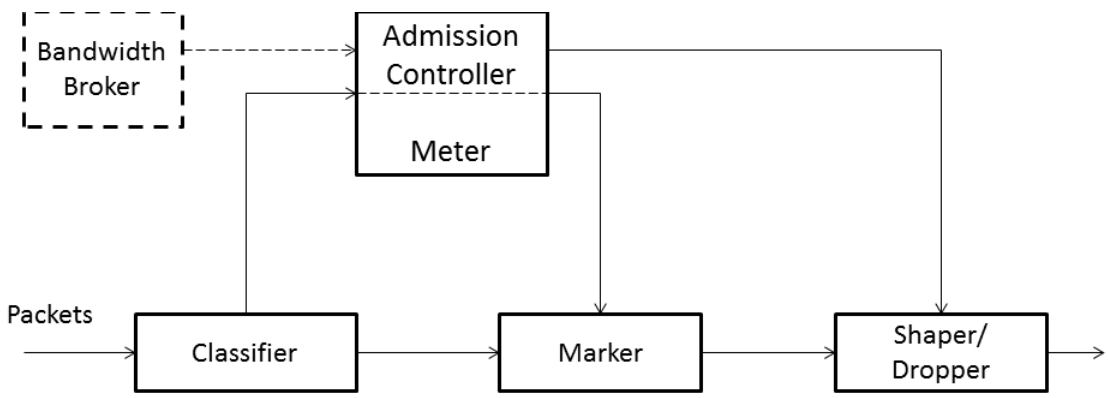

Figure 3.3: Admission Control in DiffServ

The difference between Figure 3.3 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 the 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 is downgraded to accommodate the incoming traffic. If the traffic can not be downgraded below a certain threshold the traffic is dropped. For the case of downgrading, remarking is done. Or,the traffic is dropped at the dropper. The marking depends on the class's specified initial code point or a downgraded code point.

The simulation of a Call Admission Control algorithm consisted of the model shown in Figure 3.4. The research made use of NS-allinone-2.35 release. The difference between Figure 3.2 and Figure 3.4 is that the topology in Figure 3.4 is composed of one centralized BB configured at edge routers *SGSN* and BS_3 .

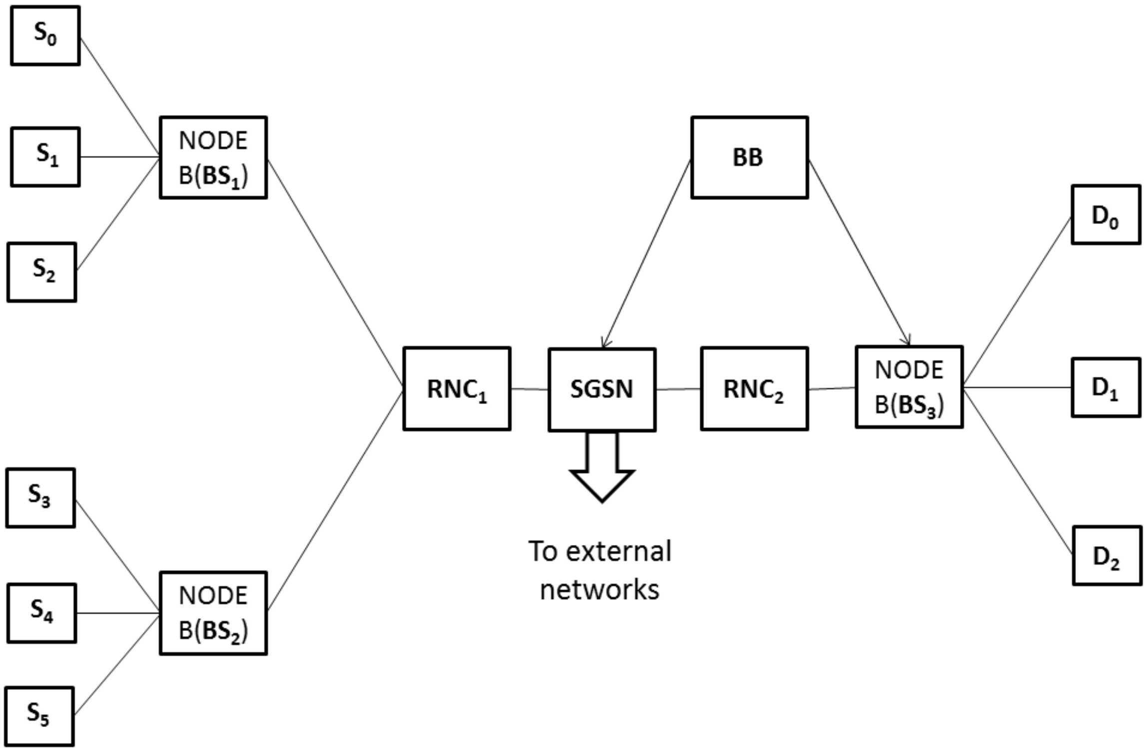

Figure 3.4: Simulation Model of an IP-based Radio Access Network of a UMTS with a bandwidth broker

3.1.8 Adaptive bandwidth-based Call admission control (CAC) modeling

Here the handovers were serviced in accordance to the class of traffic.

3.1.8.1 Voice Traffic

First attempts are made to assign bandwidth to the voice traffic using the available bandwidth dedicated for voice traffic. If the available bandwidth dedicated for voice traffic is not enough to service the handover request, the web (files download) traffic is degraded to the minimum bandwidth that is allowed in the cell to make available enough bandwidth to service the handover. If the available bandwidth dedicated for voice traffic plus the amount of bandwidth accumulated by degrading web traffic with dedicated bandwidth into web traffic with minimum allowed bandwidth is not enough to service the handover request, the video traffic is degraded to the minimum bandwidth that is allowed in the cell to make available enough bandwidth to service the handover. If the available bandwidth dedicated for voice traffic plus the amount of bandwidth accumulated by degrading web traffic with dedicated bandwidth into web traffic with minimum allowed bandwidth plus the amount of bandwidth accumulated by degrading video traffic with dedicated bandwidth into video traffic with minimum allowed bandwidth is not enough to service the handover request, the handover request is dropped. The components of this scheme includes;

CAC-call admission control
$BW_{A.V}$- available bandwidth dedicated for voice traffic
$BW_{MAX.ASN.V}$ maximum bandwidth assigned to voice traffic
BW_{DW} bandwidth accumulated by degrading web traffic
$BW_{A.W}$ - available bandwidth dedicated for web traffic
$BW_{MIN.W}$- minimum bandwidth that web traffic can be degraded to
BW_{DVD} - bandwidth accumulated by degrading video traffic
$BW_{A.VD}$ - available bandwidth dedicated for video traffic
$BW_{MIN.VD}$- minimum bandwidth that video traffic can be degraded to

Figure 3.5 illustrates the adaptive bandwidth-based Call admission control model for voice traffic.

3.1.8.2 Video Traffic

First attempts are made to assign bandwidth to the video traffic using the available bandwidth dedicated for video traffic. If the available bandwidth dedicated for video traffic is not enough to service the handover request, the web traffic is degraded to the

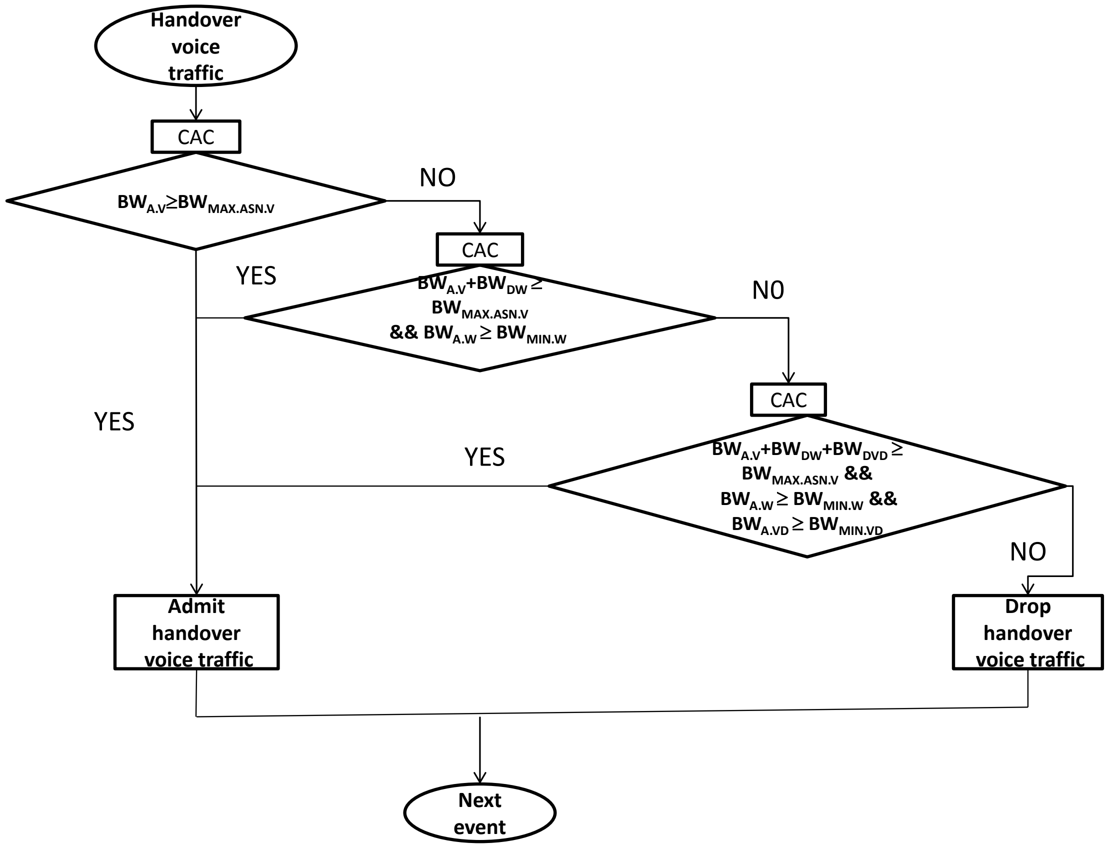

Figure 3.5: Flow chart for adaptive bandwidth-based Call admission control for voice traffic

minimum bandwidth that is allowed in the cell to make available enough bandwidth to service the handover. If the bandwidth utilized by web traffic calls goes below this threshold value, then no degradation is done to service a video traffic handover. The handover request is rejected and the video traffic is forced to terminate. This process is shown in Figure below. The components of this scheme includes;

$BW_{ASN.VD}$ - bandwidth assigned to video traffic

Figure 3.6 illustrates the adaptive bandwidth-based Call admission control model for video traffic.

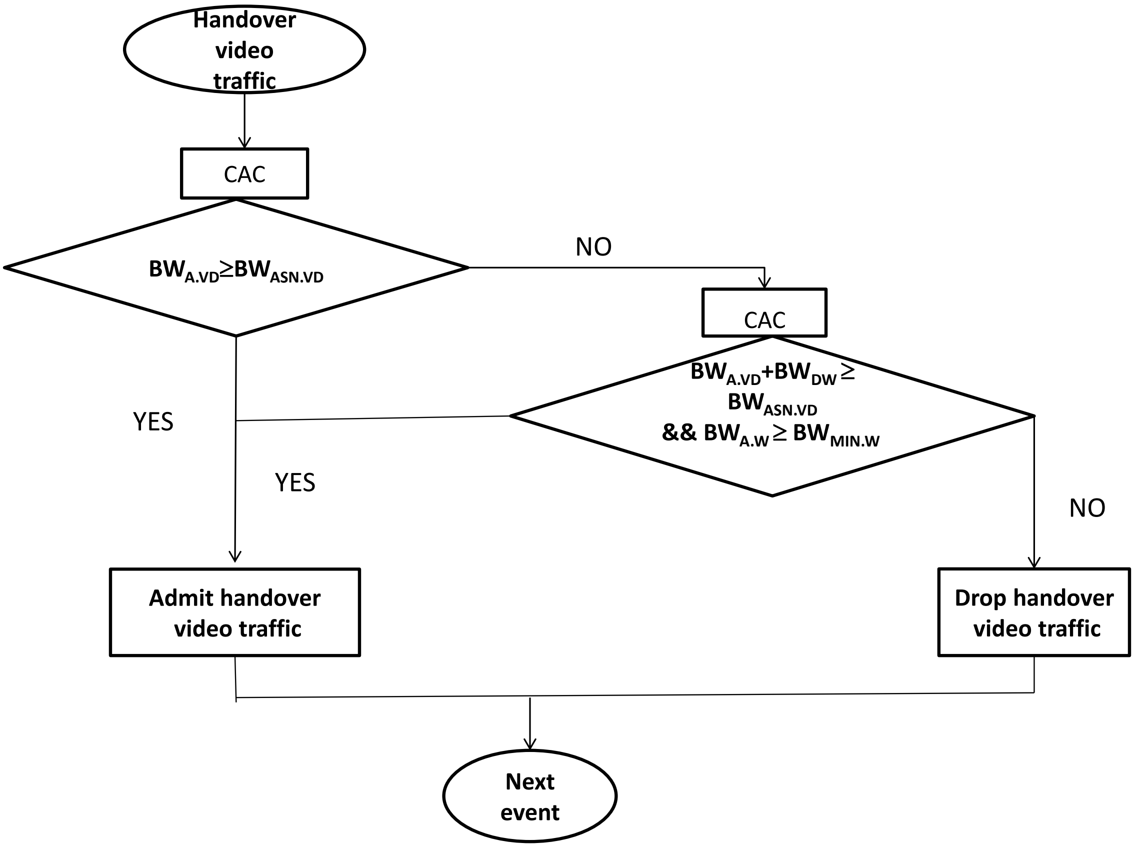

Figure 3.6: Flow chart for adaptive bandwidth-based Call admission control for video traffic

3.1.8.3 Web Traffic

First attempts are made to assign bandwidth to the web traffic using the available bandwidth dedicated for web traffic. If the available bandwidth dedicated for web traffic is not enough to service the handover request, the web traffic is degraded to the minimum bandwidth that is allowed in the cell to make available enough bandwidth to service the handover. If the bandwidth utilized by web traffic calls goes below this threshold value, web traffic handover is dropped. This process is shown in Figure 3.7. Where, $BW_{ASN.W}$- is the bandwidth assigned to web traffic

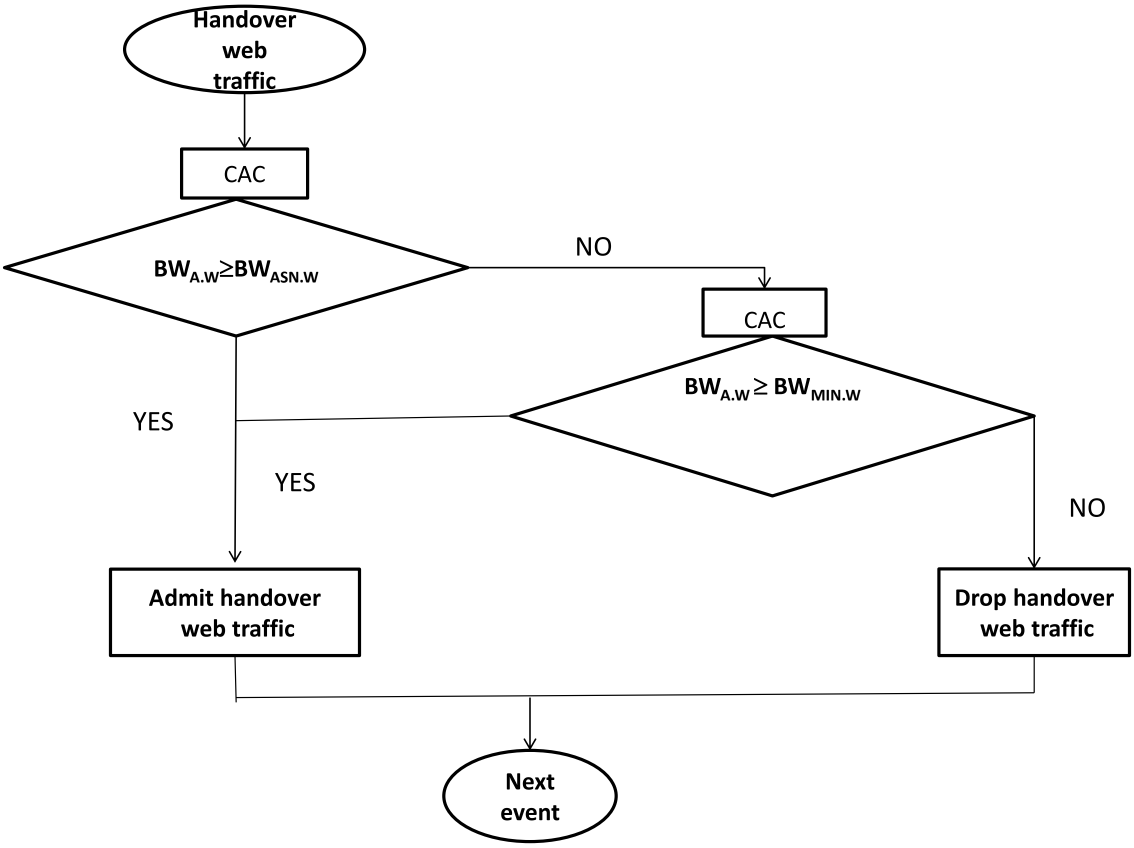

Figure 3.7: Flow chart for adaptive bandwidth-based Call admission control for web traffic

3.1.9 QoS measuring instruments

Throughput is the number of successfully received packets in a unit time and it is represented in bps.

$$Throughput = \frac{received\ \ data * 8}{Data\ \ Transmission\ \ Period} \tag{3.13}$$

where the 8 represent the number of bits in a byte

Packet loss is the difference between the packets transmitted and packets received. Packet loss is caused by traffic congestion in a network.

$$PacketLoss = Packets\ \ Transmitted - Packets\ \ Received \tag{3.14}$$

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:

$$Delay = Packet\ \ Transmission\ \ Time - Packet\ \ Arrival\ \ Time \tag{3.15}$$

Jitter is an absolute value which is defined as the variation between the arrival time of two packets that are next to each other in a traffic flow and their departure time. A lower value of jitter implies better performance. It is calculated as:

$$Jitter = (a_k - a_j) - (d_k - d_j) \tag{3.16}$$

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.

Chapter 4

RESULTS AND DISCUSSION

Primarily two scenarios were simulated. One was adaptive bandwidth allocation and another was Conventional IP (just plain IP network). Different sets of results from the two scenarios gave us the opportunity to investigate the performance of these schemes. To analyse quantitatively the simulation results, the traffic was traced during the transmission process. For every packet that passes a trace object, information about the packet was written to a specified trace file. Final output results from trace files were visualized in plotted graphs.

Packets were categorized depending upon whether they were very urgent, real-time (voice and video)and non-real-time (files download from web). Once the categorization was done the packets were sent through the separate queues according to their priority.

4.1 Voice Handover and its Effects on other Multimedia Traffic

It should be noted that voice handover occurred 3 seconds after the start of simulation. Also video traffic being real time traffic as voice traffic, it's handover will not be discussed in this chapter as voice traffic will demonstrate the treatment of premium services but graphs drawn from simulations data are attached at the appendices

4.1.1 Packet Loss

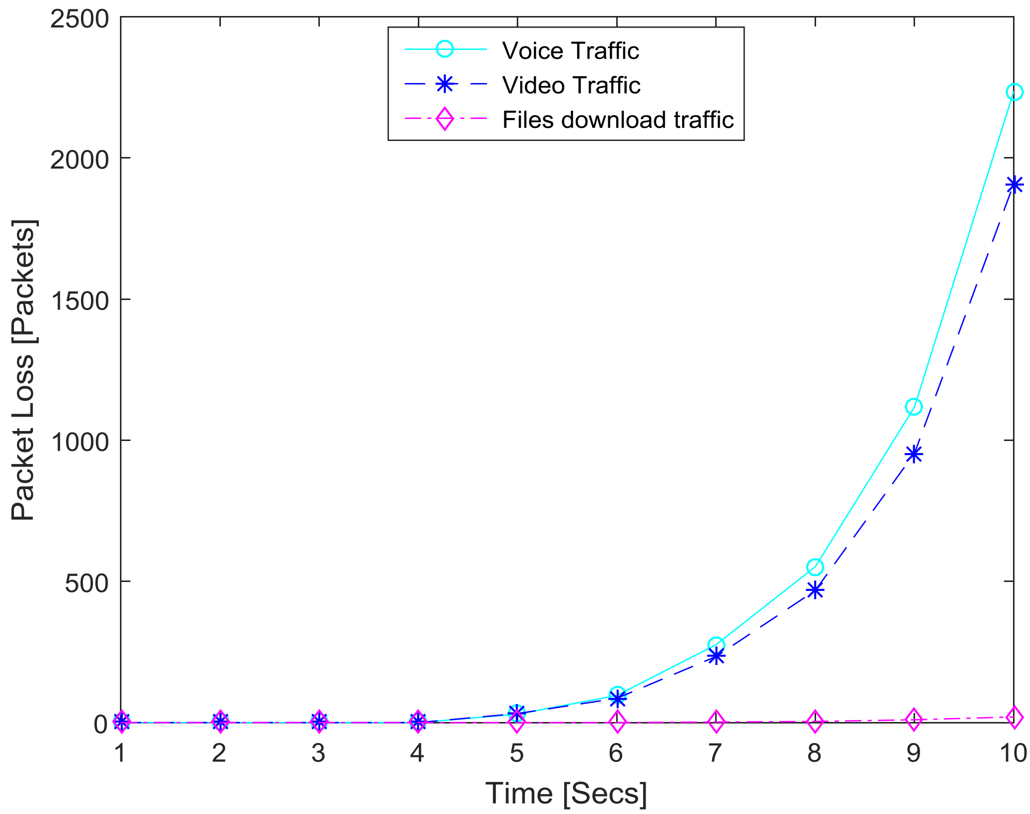

Figure 4.1: Packet loss in conventional IP for voice traffic handover

Figure 4.1 showed the packet loss in conventional IP for voice traffic handover. It was observed that voice traffic had the highest dropped packets followed by video traffic while files download traffic had the lowest packet drop. The packet drop for voice traffic and video traffic grew exponentially while for files download traffic, it increased slightly with time until it accumulated to 2200, 1900 and 20 respectively after 10 seconds.

Figure 4.2 presented packet loss in adaptive bandwidth allocation for voice traffic handover. From 4.2, it was apparent that the line for voice traffic overlapped with the line for video traffic. It was seen that only files download traffic incurred packet losses while voice traffic and video traffic had zero packet drop. Files download traffic packet

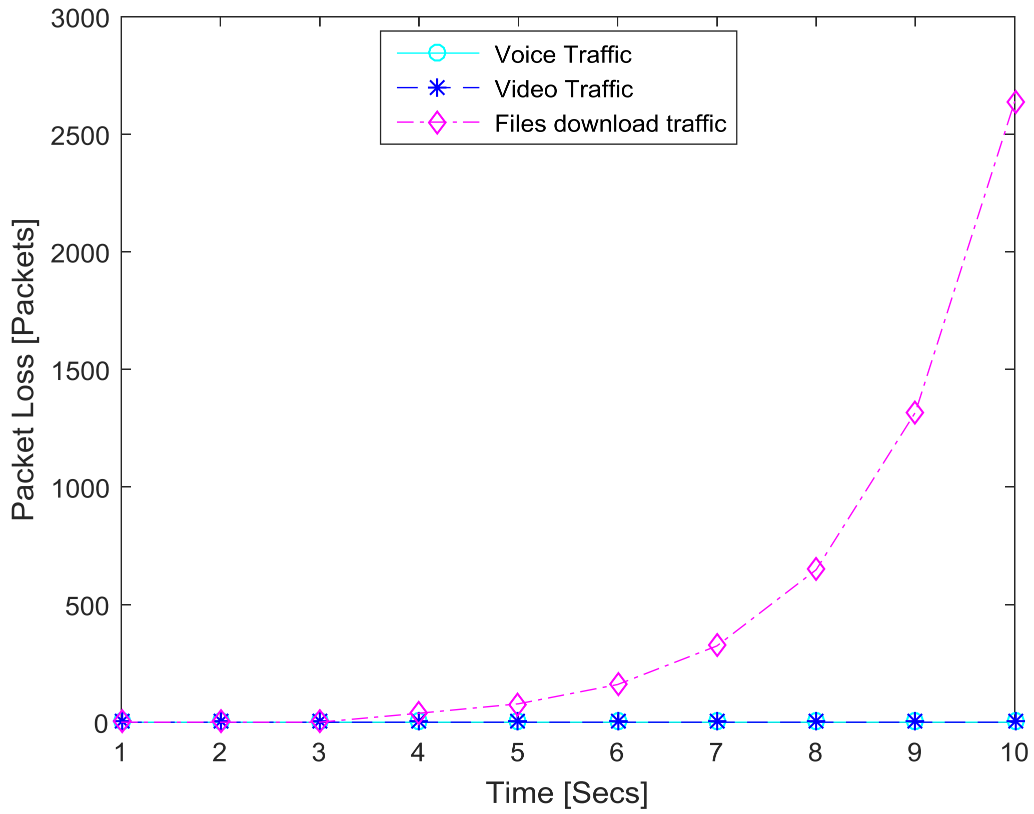

Figure 4.2: Packet loss in adaptive bandwidth allocation for voice traffic handover

drop, increased exponentially with time until it reached 2640 at the 10 seconds mark. It was noted that before handover, there were no packets dropped in both adaptive bandwidth allocation and conventional IP as the bandwidth available was enough to accommodate the traffic.

Both voice traffic and video traffic are real-time traffic while files download traffic is non-real-time traffic. According to the adaptive bandwidth allocation algorithm real time traffic gets priority over non-real time traffic hence there was no loss for both voice traffic and video traffic while files download traffic experienced severe packet losses in the adaptive bandwidth allocation as it packets were rejected by admission control. In conventional IP the mechanism for bandwidth provisioning is based on best effort which does not guarantee bandwidth hence the loss experienced for the three types of traffic. Packet loss in conventional IP is highest for voice traffic because UDP transport protocol does not attempt error recovery for erroneous packets, instead it discards them while files download traffic has the lowest packet loss due to TCP transport protocol's error checking and recovery mechanism through retransmission of erroneous packets. Video traffic had medium packet loss in conventional IP as its packet transmission was not constant. The packet loss ratio in adaptive bandwidth allocation had the reverse order to the priority.

4.1.2 Throughput

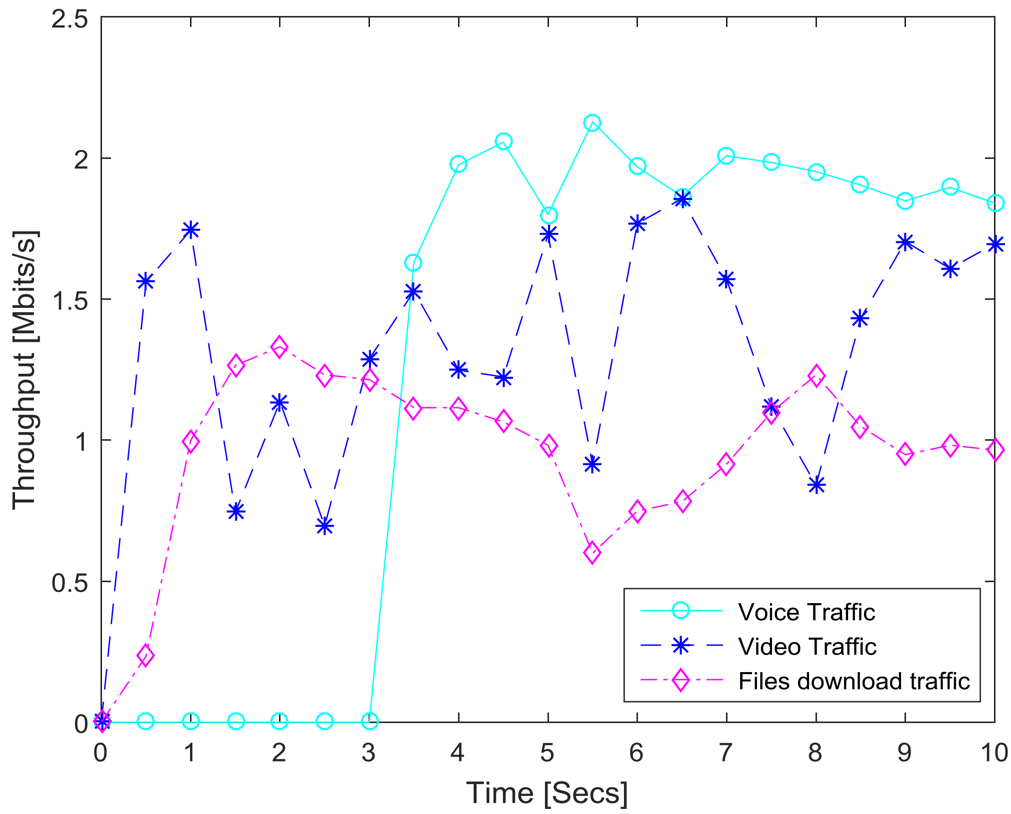

Figure 4.3: Throughput in conventional IP for voice traffic handover

From Figure 4.3, in conventional IP, the throughput for voice traffic fluctuated between 1.75Mb/s and 2Mb/s. The throughput for video traffic zigzagged between 0.7Mb/s and 1.75Mb/s while files download traffic throughput ranged between 0.5Mb/s and 1.2Mb/s in the course of 10 seconds simulation though there was a slight decrease at around 3 seconds when handover took place as a result of TCP control mechanisms. Throughput measures how fast we can transmit data. Voice traffic which used UDP protocol that is fast, had the highest throughput while files download traffic had the lowest throughput due to its use of TCP protocol that has error correction and acknowledgement mechanism making it slow. Video traffic uses UDP but its bit rate are variable during the on and off periods. During off period there is no transmission hence low throughput at that time.

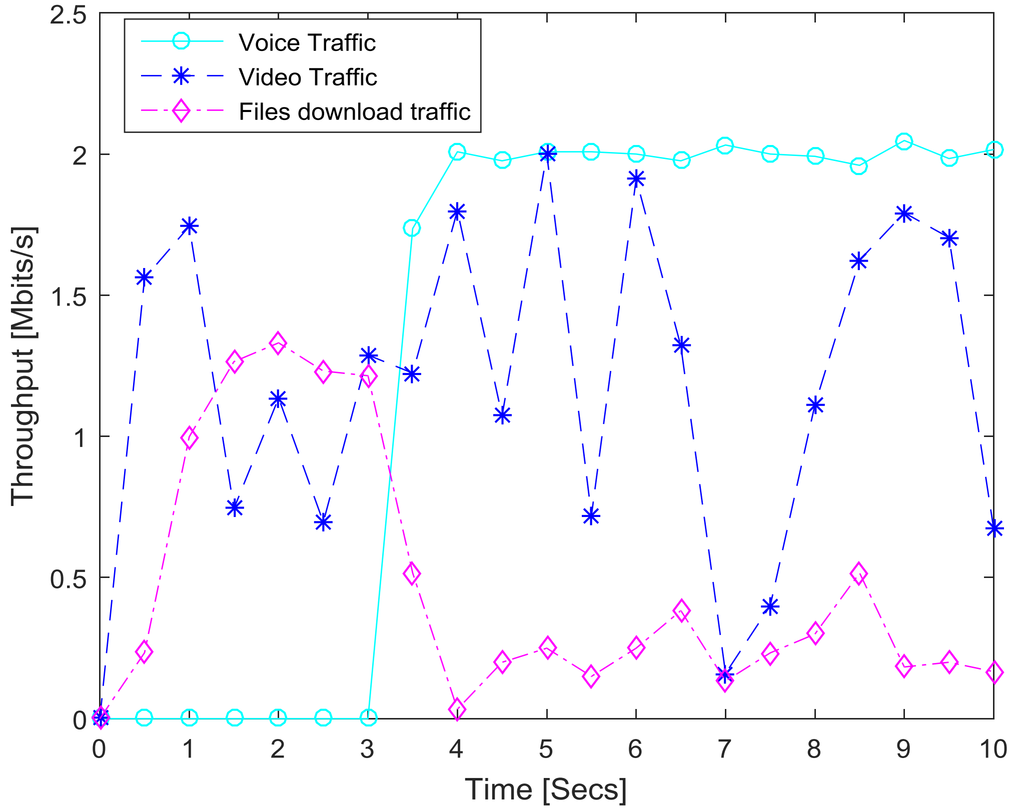

Figure 4.4: Throughput in adaptive bandwidth allocation for voice traffic handover

As depicted by Figure 4.4, in adaptive bandwidth allocation, throughput for voice traffic remained steady at 2Mb/s after its handover. For video traffic, throughput reached a peak of 2Mb/s and sank to a trough as low as 0.25Mb/s during the simulation . Files download traffic throughput rose to 1.2Mb/s for the first 3 seconds then fell to range between 0.1Mb/s and 0.5Mb/s after handover for the rest of the simulation.

Generally the throughput for voice traffic remained constant at 2Mb/s while files download traffic fell a lot after handover as the control mechanisms implemented regulated it by dropping some of its packets.

The adaptive bandwidth allocation algorithm gave priority to voice, video and files download traffics in that order while in conventional IP , traffic was treated equally on first come first serve basis. That explained the high throughput for voice traffic and video traffic while files download traffic throughput sank a lot as offered load in the bottleneck link surged beyond the available bandwidth in adaptive bandwidth allocation after handover. This was due to high priority traffic starving low priority files download traffic by allowing them to have small amount of link capacity. It was observed that in adaptive bandwidth allocation high priority voice traffic had the highest throughput (stable at 2Mb/s) while low priority files download traffic had the lowest throughput compared to conventional IP. It was also true that before handover throughput was the same for both adaptive bandwidth allocation and conventional IP as bandwidth was sufficient. Of great importance is to take note that quality of service in a limited environment is a trade-off.

4.1.3 End to End Delay

Figure 4.5 compared the end to end delay of voice traffic, video traffic and files download traffic in conventional IP for voice traffic handover. The end to end delay of voice traffic ranged between 65 milliseconds and 113 milliseconds after handover .The end to end delay of video traffic ranged between 65 milliseconds and 70 milliseconds for the first three seconds then shot up to 115 while for web traffic, the values zigzagged between 60 milliseconds to around 72 milliseconds for the first 3 seconds and 60 milliseconds and 118 milliseconds thereafter.

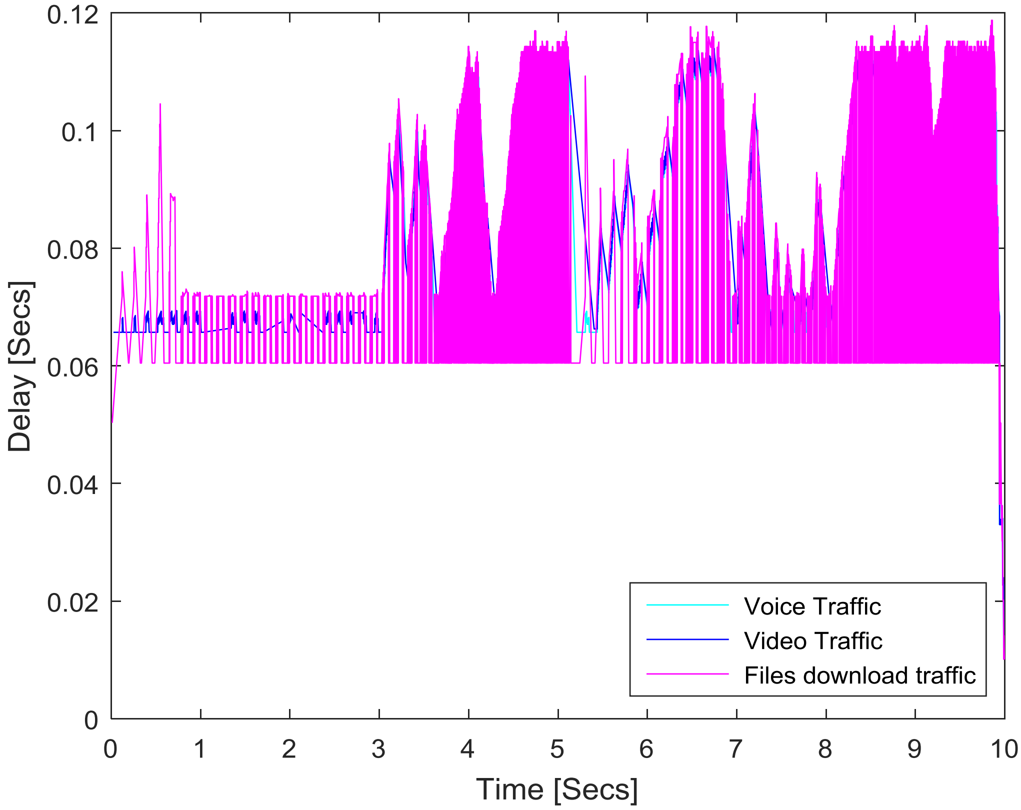

Figure 4.5: Delay in conventional IP for voice traffic handover

Figure 4.6 highlighted the end to end delay among voice traffic, video traffic and files download traffic in adaptive bandwidth allocation. The end to end delay of voice traffic ranged between 65 milliseconds and 69 milliseconds after handover .The end to end delay of video traffic ranged between 65 milliseconds and 70 milliseconds while for files download traffic the values zigzagged between 60 milliseconds up to 72 milliseconds before handover. After handover end to end delay of video traffic ranged between 65 milliseconds to 82 milliseconds while for files download traffic the values zigzagged between 60 milliseconds up to 84 milliseconds. It was observed that end to end delay before handover in both adaptive bandwidth allocation and conventional IP were the same as available bandwidth was sufficient. After handover end to end delay in adaptive bandwidth allocation was less than in conventional IP. Also files download traffic had more delay between consecutive traffic as evidenced by clustering together

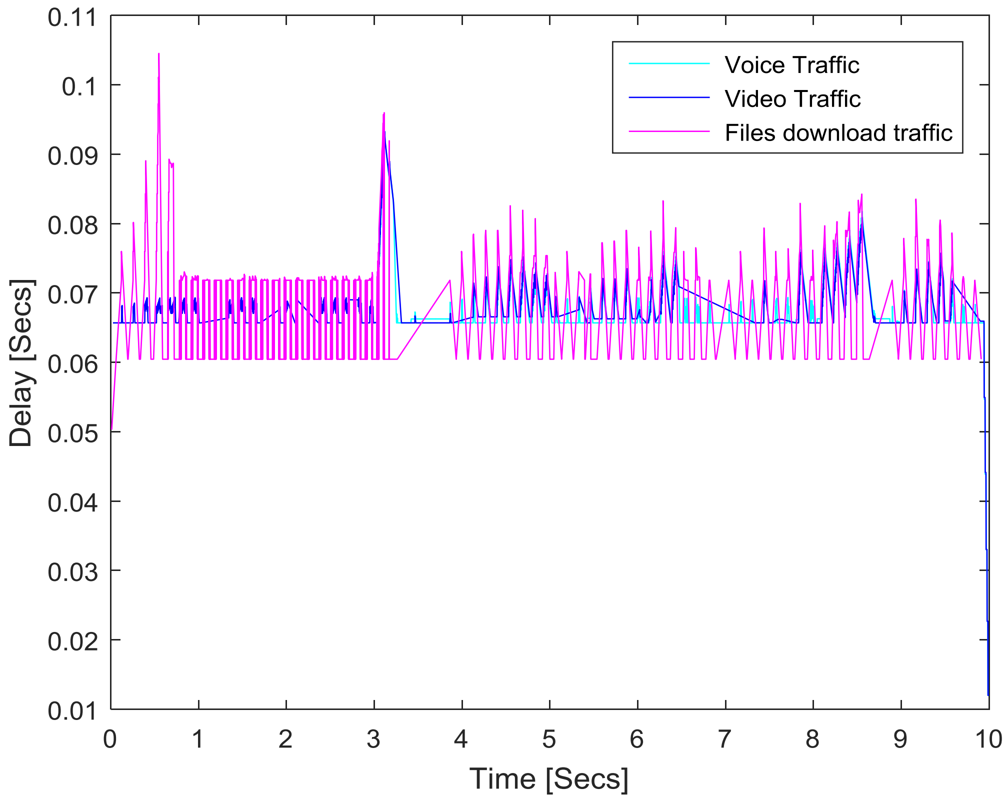

Figure 4.6: Delay in adaptive bandwidth allocation after voice traffic handover

of traffic in conventional IP than adaptive bandwidth allocation. This was due to an admission control in adaptive bandwidth allocation that admitted traffic that could be comfortably be accommodated in the network.

From figure 4.6, files download traffic also experienced traffic spikes in between because of burst data causing high end to end delay during that interval. The clustering together of lines was due to severe end to end delay between consecutive packets. In the first seconds, there was a complex conditioning of the traffic that took place, introducing a significant delay.

The reason why the adaptive bandwidth allocation network showed a substantial impact on reducing premium packets delay was due to its routers using the priority queue mechanisms. It treated the entire premium class packet with the first priority queue according to the SLA. The first priority queue transferred its packets first; therefore the packets reached their destination quickly and spent less time waiting in the queue. The voice packets still suffered some queuing delays when there were other voice packets ahead of them in the same queue.

4.1.4 Jitter

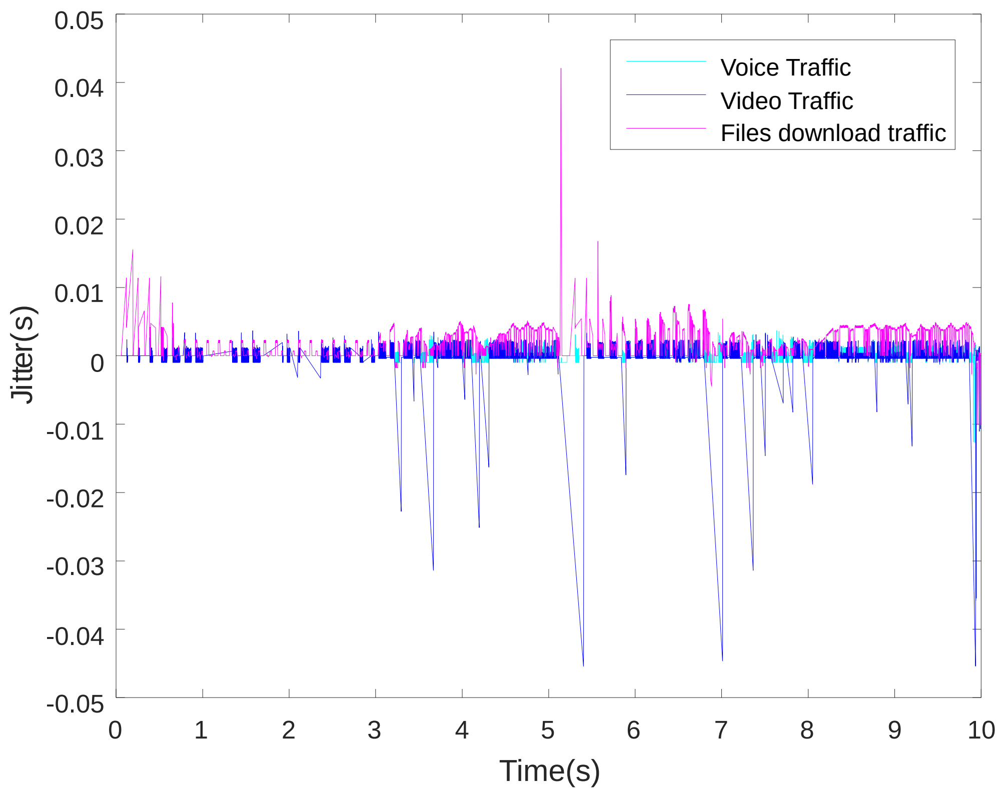

Figure 4.7: Jitter in conventional IP for voice traffic handover

Figure 4.7 illustrated the jitter for voice traffic, video traffic and files download traffic in conventional IP. The video traffic jitter ranged between 4 milliseconds to -3 milliseconds for the first three seconds while files download traffic jitter ranged between 2.4 milliseconds to -2.4 milliseconds. After handover voice traffic jitter ranged between 4 milliseconds to -1 milliseconds for the rest of the simulation period while video traffic jitter ranged between 4 milliseconds to -45 milliseconds. Files download traffic jitter ranged between 40 milliseconds to -5 milliseconds. The files download traffic had the highest jitter because of reduced speed of transmission caused by acknowledgement and correction of errors while voice traffic had the lowest jitter as its transport protocol is unidirectional hence fast.

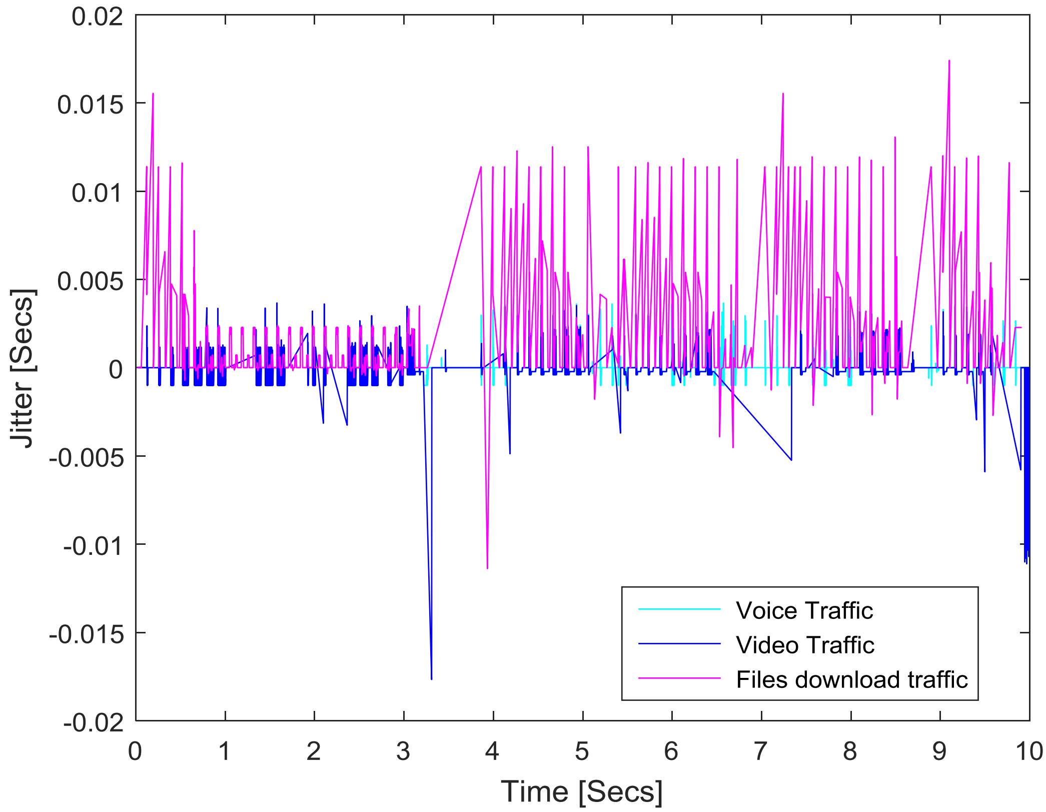

Figure 4.8: Jitter in adaptive bandwidth allocation after voice traffic handover

Figure 4.8 showed the jitter for voice traffic, video traffic and files download traffic in adaptive bandwidth allocation. Jitter for video traffic zigzagged from 4 milliseconds to -3 milliseconds while the jitter for files download traffic zigzagged from 2.4 milliseconds to -2.4 milliseconds. After handover voice traffic jitter ranged between 4 milliseconds to -1 milliseconds for the rest of the simulation period while video traffic jitter ranged between 4 milliseconds to -18 milliseconds. Files download traffic jitter ranged between 18 milliseconds to -12 milliseconds. It was deduced that the jitter range for the three traffics in adaptive bandwidth allocation is less than in conventional

IP due to admission control that limit the traffic into adaptive bandwidth allocation hence traffic in the network was served faster. Jitter was less before handover as bandwidth was sufficient to cater for the traffics.

In adaptive bandwidth allocation files download traffic was given less preference hence it waited longer in queues hence the high jitter range observed. Adaptive bandwidth allocation allowed the premium services to take enough bottleneck bandwidth. It starved the files download traffic by allowing them to have only a small amount of link capacity as its QoS needs were not as crucial as those of real-time traffic. As files download was given less preference, it waited longer in queues.

4.2 Files download traffic Handover and its Effects on other Multimedia Traffic

It should be noted that files download traffic handover occurred 3 seconds after the start of simulation.

4.2.1 Packet Loss

Figure 4.9 showed the packet loss in conventional IP for files download traffic handover. As the offered load at the bottleneck link increased with time after handover, the packet loss for voice traffic increased exponentially while video traffic packet loss was slightly in conventional IP network with values of 2000 and 860 respectively. The files download traffic packet loss was negligible(40). The packet loss for voice traffic increased more rapidly with time than both video traffic and files download traffic because it was transported by CBR/UDP which does not have any congestion control mechanism unlike FTP/TCP. Though video traffic too used UDP, it was of exponential on/off. During off period there was reduction in packet loss.

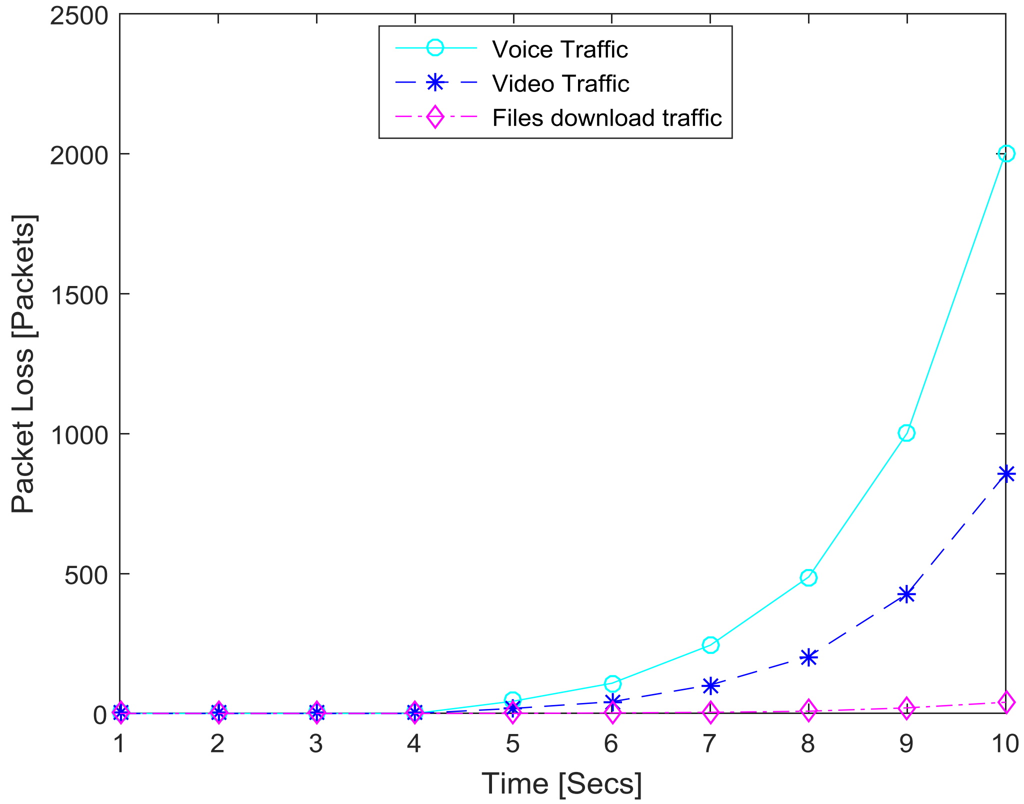

Figure 4.9: Packet loss in conventional IP for files download traffic handover

Figure 4.10 presented packet loss in adaptive bandwidth allocation for files download traffic handover. In adaptive bandwidth allocation, the packet loss for voice and video traffic remained at zero, though the offered load at the bottleneck link increased over time beyond the available bandwidth after handover had taken place. On the other hand, the packet loss for files download traffic increased rapidly with time when the offered load was beyond the available bandwidth at the bottleneck link to 2300 after 10 seconds after handover.

Adaptive bandwidth allocation limited packet loss for each of the services until total offered load was up to the bottleneck limit as time passed when traffic rose in the cell due to handover. Thereafter as the offered load rose, the voice and video packets continued to get guarantee of low packet loss at the expense of file download traffic. When offered load exceeded the bottleneck bandwidth with time when handover

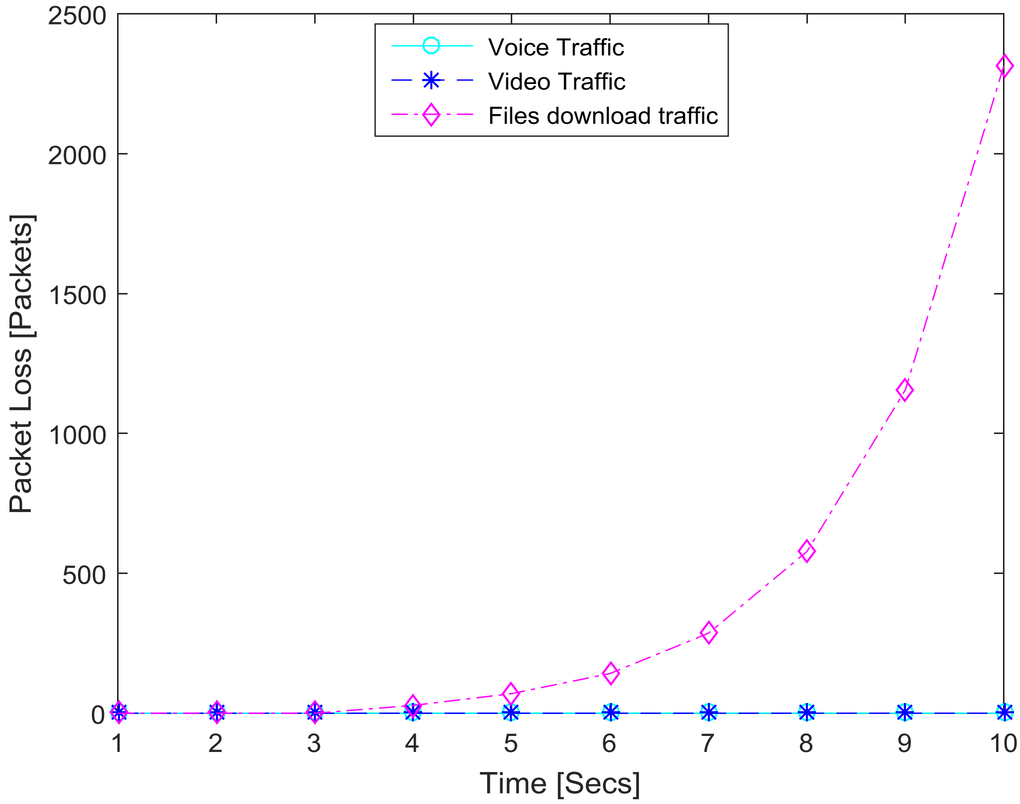

Figure 4.10: Packet loss in adaptive bandwidth allocation for files download traffic handover

occurred, the priorities for voice and video traffic assured higher QoS for their classes while the files download traffic class incurred higher packet loss as the bandwidth broker rejected some of its traffic.

From figure 4.9 and figure 4.10, it was concluded that the packet loss increased due to the increase of traffic congestion. It was also deduced that packet loss can be reduced for premium class of traffic and traffic congestion avoided by implementing proper QoS mechanism in the network. In this case, adaptive bandwidth allocation was implemented that had the administrative control of packet dropping preference and forwarded the packet based on SLAs.

4.2.2 Throughput

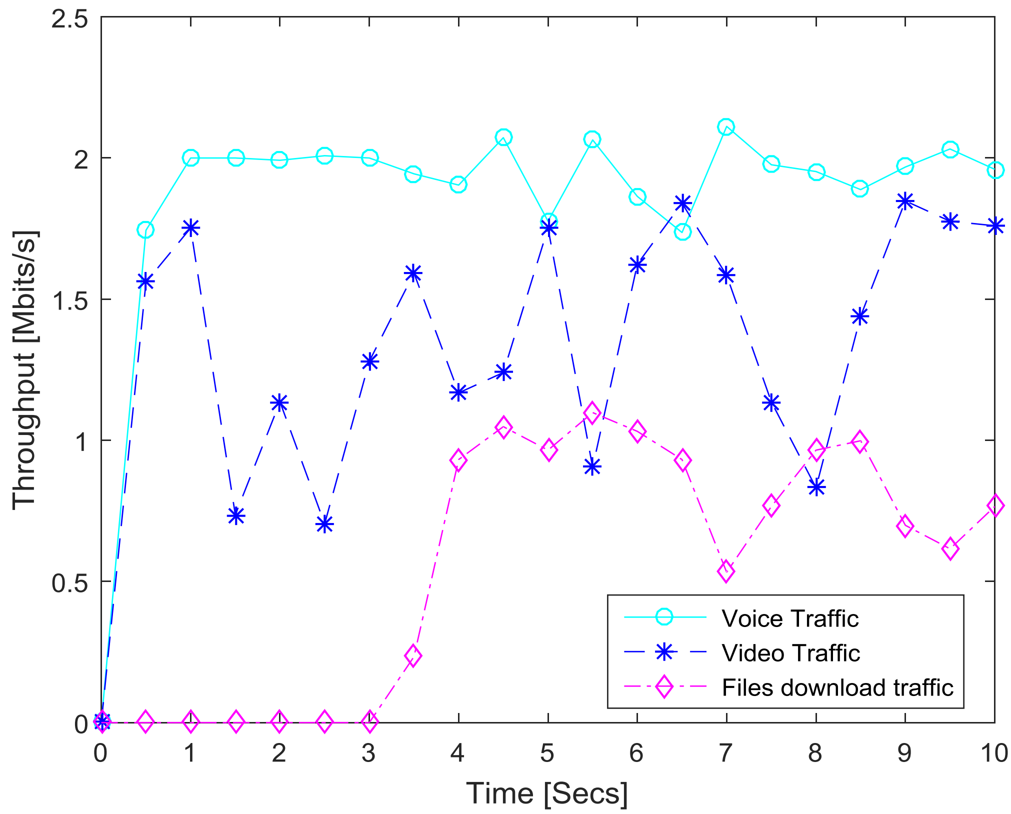

Figure 4.11: Throughput in conventional IP for files download traffic handover

Figure 4.11 showed the throughput in conventional IP where it fluctuated between 1.7 Mb/s and 2 Mb/s for voice traffic. Video traffic throughput zigzagged between 0.75 Mb/s and 1.75Mb/s. The files download traffic throughput shot to 1 Mb/s then slipped to as low as 0.5 Mb/s.

Figure 4.12 showed the throughput in the adaptive bandwidth allocation where it grew to 2 Mb/s for voice traffic during the simulation period and stabilised there. Video traffic throughput ranged between 0.3 Mb/s to 2 Mb/s while files download traffic throughput varied between 1 Mb/s and 0 Mb/s.

As depicted by Figure 4.12, in adaptive bandwidth allocation, the highest priority

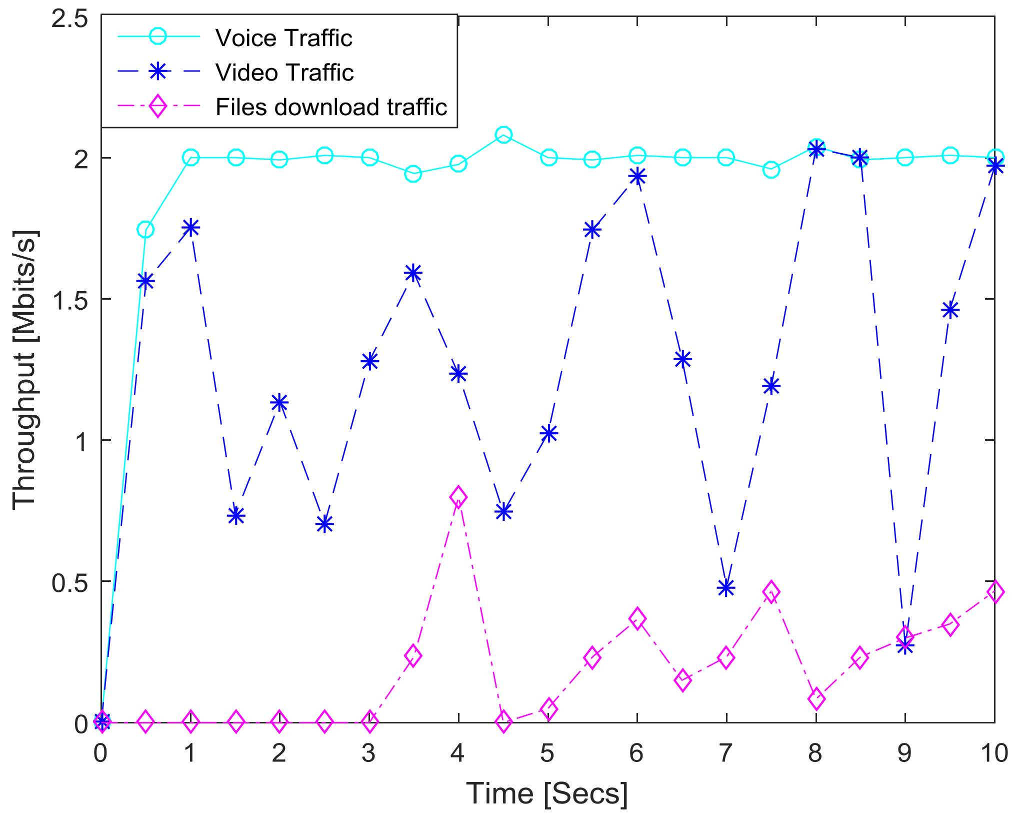

Figure 4.12: Throughput in adaptive bandwidth allocation for files download traffic handover

voice traffic had the highest throughput which plateaued at 2Mb/s followed by medium priority video traffic that varied; the files download traffic with the lowest priority had the least throughput that plunged most after handover as traffic increased in the cell when compared to conventional IP. When traffic at the bottle neck link was in excess of the available bandwidth, the throughput of voice traffic which was the highest priority flow was unaffected because of its best preference QoS. The video traffic which was the medium priority had a higher throughput than the files download traffic as the admission control mechanism employed limited files download traffic into the network hence the low throughput. The increase of offered load with time caused files download traffic to suffer high packet drops as they were rejected by the admission control. So the throughput was in accordance with priority order.

It was then generally concluded that the throughput in conventional IP is not granted for mission critical applications compared to the throughput in adaptive bandwidth allocation. The reason is that, throughput is affected by the packet loss; the packet loss in conventional IP scenario was much higher than the packet loss in adaptive bandwidth allocation scenario for mission critical applications due to prioritization during the high congestion period. For files download traffic the throughput after handover was lower in adaptive bandwidth allocation than in conventional IP as a result of control mechanism in adaptive bandwidth allocation that limited the files download traffic that could be admitted into the network.

4.2.3 End to End Delay

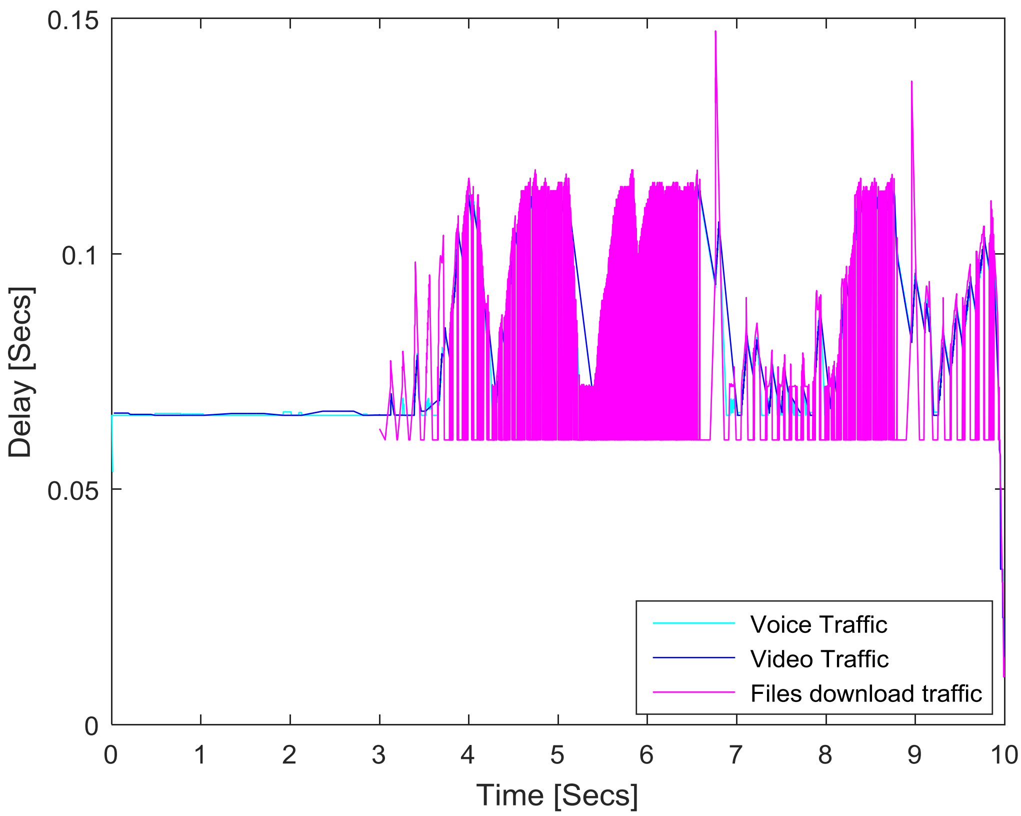

Figure 4.13: Delay in conventional IP for files download traffic handover

Figure 4.13 showed packet delay in conventional IP for files download traffic handover. The voice traffic and video traffic end to end delay ranged between 65 milliseconds to 67 milliseconds for the first three seconds and then increased to range between 65 milliseconds to 115 milliseconds for the rest of the simulation while the files download traffic end to end ranged between 60 milliseconds to 120 milliseconds with some spikes in between. The noticeably small voice traffic and video traffic end to end delay in the first three second before handover was due to enough bandwidth that served both voice traffic and video traffic. After handover the voice traffic and video traffic end to end delay rose due to congestion in the bottleneck link although files download traffic end to end delay was slightly more due to TCP protocol being slow in sending traffic as traffic load increased. Thanks to its bidirectional properties where the receiver acknowledges the traffic sent. The files download traffic showed high burstiness as evidenced by the clustering together of line hence high end to delay between consecutive packets during that time.

Figure 4.14 showed packet delay in adaptive bandwidth allocation for files download traffic handover. The voice traffic and video traffic end to end delay ranged between 65 milliseconds to 67 milliseconds before handover and then later varied between 65 milliseconds to 72 milliseconds after handover. At the start of handover files download traffic end to end delay shot up to 115 milliseconds while voice traffic and video traffic end to end delay shot up to 95 milliseconds due warm up period. Files download traffic ranged between 60 milliseconds to 80 milliseconds for most of the simulation.

The adaptive bandwidth allocation has an admission control mechanism incorporated in it. It admits traffic that can comfortably be served after the offered load is up to the bottleneck limit. This explains the lesser end to end delay range that was observed in adaptive bandwidth allocation compared to conventional IP after handover. After handover video and voice traffic got a significantly low packet end to end delay whereas

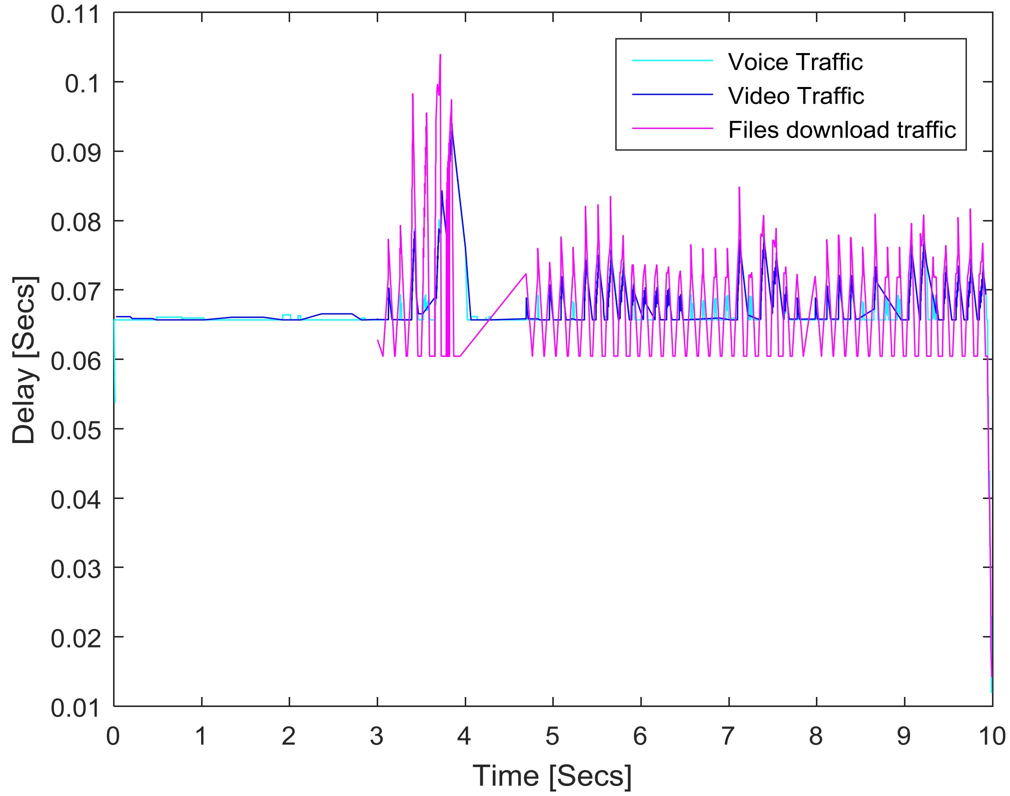

Figure 4.14: Delay in adaptive bandwidth allocation for files download traffic handover

files download traffic got affected dramatically in adaptive bandwidth allocation than in conventional IP because low priority traffic is non-pre-emptive in adaptive bandwidth allocation. Also it should be taken into account that end to end delay for files download traffic between consecutive packets was low in adaptive bandwidth allocation than in conventional IP due to regulation of the traffic that can be allowed into the network in adaptive bandwidth allocation. Generally, end to end delay increased after handover as traffic rose in queues making them to wait longer.

4.2.4 Jitter

Figure 4.15 showed packet jitter in conventional IP for files download traffic handover. The voice traffic and video traffic jitter ranged between 1 milliseconds to -1 milliseconds for the first three seconds. After handover voice traffic jitter ranged

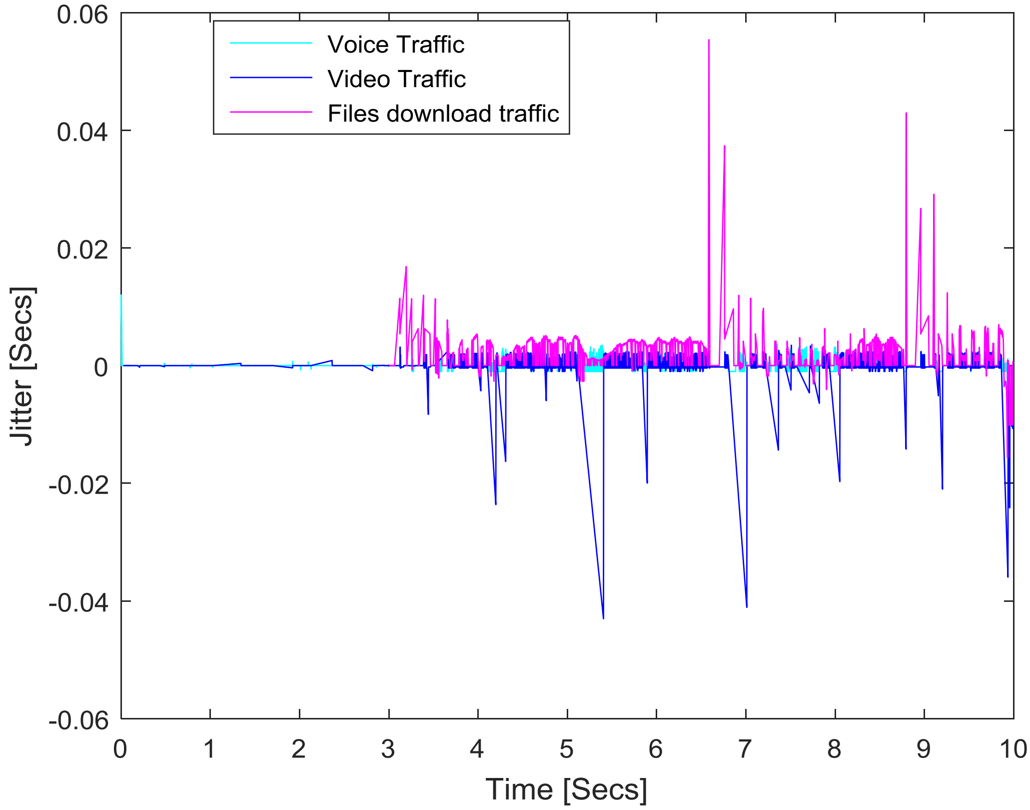

Figure 4.15: Jitter in conventional IP for files download traffic handover

between 5 milliseconds to -2 milliseconds for the rest of the simulation period while video traffic jitter ranged between 5 milliseconds to -42 milliseconds. Files download traffic jitter ranged between 55 milliseconds to -5 milliseconds.

In figure 4.16,the voice traffic and video traffic jitter ranged between 1 milliseconds to -1 milliseconds for the first three seconds. After handover voice traffic jitter ranged between 4 milliseconds to -1 milliseconds for the rest of the simulation period while video traffic jitter ranged between 4 milliseconds to -10 milliseconds. Files download traffic jitter ranged between 17 milliseconds to -17 milliseconds. It was seen that jitter range for the three traffics in adaptive bandwidth allocation was less than in conventional IP.

In figure 4.16, it was inferred that the packet jitter range in adaptive bandwidth allocation for files download traffic handover was highest in files download traffic followed by video traffic while voice traffic jitter range was the lowest. The reason

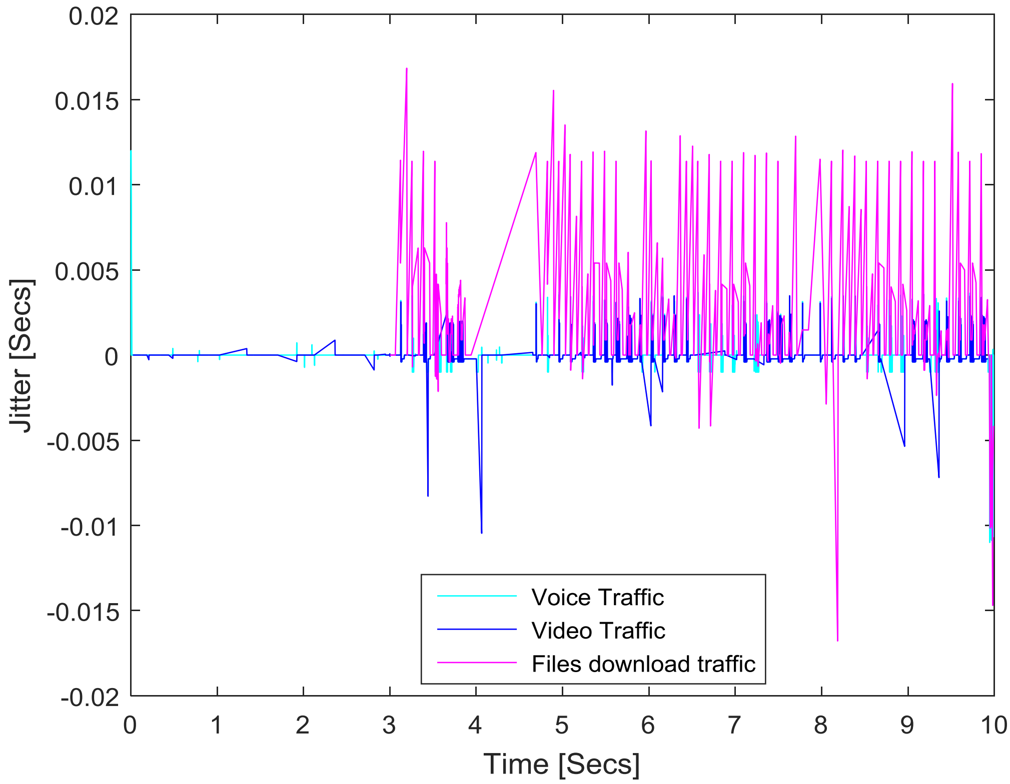

Figure 4.16: Jitter in adaptive bandwidth allocation for files download traffic handover

for lowest jitter range in adaptive bandwidth allocation for the three traffics than in conventional IP was due to admission control in adaptive bandwidth allocation that limit amount of files download traffic into the bottleneck link. Non-real time traffic had highest jitter range due to it waiting for real time traffic to be served first. Jitter also increased after handover due to increase in traffic load in the network causing traffic to wait in queues.

4.3 Validation of Results

Among the standardization groups that have covered QoS include International telecommunication Union (ITU), by European Telecommunication Standards Institute (ETSI) or 3rd Generation Partnership Project (3GPP). Table 4.1 gives a summary of the mapping of multimedia services and end-to-end QoS requirements for users as defined ITU,ETSI or 3GPP [76] [77] [78].

Table 4.1: A summary of the mapping of multimedia services and end-to-end QoS requirements

Medium	**Delay**	**Jitter**	**Capacity**	**CoS UMTS**	**PHP**
Audio	<150 ms	<1 ms	DBW	Conversational	EF
Video	<10 s	<1 ms	VBW	Streaming	AF
Web	<15 s(<60 s A)	U	VBW	Background	BE

Where

- A: Acceptable
- DBW: Dedicated Bandwidth
- VBW: Variable Bandwidth
- U: Unspecified

Table 4.2 gives a summary of a list the value ranges of the UMTS bearer service attributes [79]

Table 4.2: The value ranges of the UMTS bearer service attributes

Traffic class	**Conversational**	**Streaming**	**Interactive**	**Background**
Maximum bitrate(kbps)	<2048	<2048	<2048	<2048
Guaranteed bit rate(kbps)	<2048	<2048	U	U
Transfer delay(ms)	<100	<250	U	U

The adaptive bandwidth allocation that was formulated showed to dedicate a bandwidth of 2Mb/s to voice traffic while video traffic bit rate varied and peaked at 2Mb/s. The files download traffic also varied and was below 2Mb/s. This results are in agreement with the standardization groups as can be observed in table 4.1 and table 4.2. It should be noted that packet loss affect throughput as high packet loss will lead to a low transmission rate.

In adaptive bandwidth allocation the traffics had end to end delays below 100ms, this value is generally acceptable by ITU,ETSI or 3GPP. Jitter in adaptive bandwidth allocation for files download traffic handover had an absolute value of 1ms for both voice and video traffics which is acceptable by the standardization groups but during congestion that was introduced by handover, it depends on priority with voice traffic that was favoured having the least jitter. With voice traffic handover, jitter for video traffic had an absolute value of 1ms but with spikes that that could go up to 4ms. After handover as traffic goes beyond the link capacity jitter was according to priority with high priority traffic having the least absolute value.

Chapter 5

CONTRIBUTIONS, CONCLUSIONS AND FUTURE WORK

5.1 Contributions

The following are the main contributions of this thesis:

- It leads to the design and simulation of a novel adaptive bandwidth allocation algorithm to share bandwidth according to priority to support QoS provisioning to end users of multimedia applications in cellular networks during handover. The algorithm provide efficient use of available bandwidth through service level agreements (SLAs) contracted between mobile service providers and Mobile users. It policies multimedia traffic according to its agency to provide bandwidth on demand in a constrained bandwidth cellular network.

- The scheme formulated achieves scalability by aggregating multimedia traffic into categories.

- The scheme administers an admission control scheme that works in conjunction with an adaptive bandwidth allocation scheme to quantitatively control the level of QoS provisioned in a bandwidth limited cellular network through a bandwidth broker manager.

5.2 Conclusions

In this thesis, an Adaptive bandwidth allocation scheme is formulated that guarantee different QoS levels for multimedia traffic in constrained bandwidth cellular networks. The scheme takes the DiffServ approach to guarantee QoS without needing bandwidth over-provisioning to the end users. The scheme also incorporates admission control for QoS support. The functioning of this scheme is that, if there is enough bandwidth in a cell, the handed-over traffic is admitted without any problem. But if the available bandwidth in that cell is scarce then some bandwidth from non-real time traffic in that cell is borrowed to serve real time traffic. Hence, this strategy gives high priority to handed-over voice traffic then video and finally files download with the lowest priority.

Four performance metrics; throughput, packet loss, end to end delay and jitter are used to evaluate performance. NS2 release 2.35 simulator is used to test and validate this scheme. Through the simulation results it is observed that adaptive bandwidth allocation yields better performances than conventional IP. It has also been proven that conventional IP does not guarantee service when the network experience congestion due to increased traffic in a cell during handover. 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.

5.3 Future Work

Attention should be given to the following areas in future for further research:

- The effects of adaptive power control schemes on coverage issues and how these issues affects handovers in relation to multimedia traffic in cellular network.

- Attempts should be made to improve the QoS for Multimedia handover traffic in cellular through Multi-protocol Label Switching (MPLS)

- A thought should be given to design a comprehensive framework for bandwidth allocation of multimedia handover services in heterogeneous network for QoS improvement for end user (mobile subscriber)
- Designing an adaptive bandwidth allocation model that maximizes revenue

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Appendix A

Published Work

The following are the publications in the course of the research period:

1. E. Omosa, K. Lang'at and S. Musyoki, " Dynamically Adaptive Bandwidth Allocation for Handoff Multimedia Services for Quality of Service Performance in Mobile Cellular Networks," The 2015 JKUAT Scientific Conference, vol. 10, pp. 338-352, November 2015.

2. E. Omosa, K. Lang'at and S. Musyoki, "Performance of an Admission Control Scheme for Bandwidth in Multimedia Handover Services in UMTS mobile Cellular Network," Proceedings of the 2016 Annual Conference on Sustainable Research and Innovation, Nairobi Kenya, pp. 94-103, May 2016.

3. E. Omosa, K. Lang'at and S. Musyoki, " Performance of Adaptive Bandwidth Allocation for Multimedia Handover Services in UMTS mobile Cellular Networks," Journal of Sustainable Research in Engineering, vol. 2,no. 4 pp. 148-157, May 2015.

Appendix B

Conventional IP Network Code

```
set ns [new Simulator]
set testTime 10.0
# colors for traffic flows
$ns color 1 Blue
$ns color 2 Green
$ns color 3 Red
$ns color 4 Purple
$ns color 5 Gray
$ns color 6 Yellow
set tracefile [open packetLoss.tr w]
$ns trace-all $tracefile
set nf [open packetLoss.nam w]
$ns namtrace-all $nf
# the network topology structure
set n0 [$ns node]
set n1 [$ns node]
set n2 [$ns node]
set n3 [$ns node]
set n4 [$ns node]
set n5 [$ns node]
set n6 [$ns node]
set n7 [$ns node]
set n8 [$ns node]
```

```
set n9 [$ns node]
set n10 [$ns node]
set n11 [$ns node]
set n12 [$ns node]
set n13 [$ns node]
set n14 [$ns node]
$ns duplex-link $n0 $n6 2Mb 10ms DropTail
$ns duplex-link $n1 $n6 2Mb 10ms DropTail
$ns duplex-link $n2 $n6 2Mb 10ms DropTail
$ns duplex-link $n3 $n7 2Mb 10ms DropTail
$ns duplex-link $n4 $n7 2Mb 10ms DropTail
$ns duplex-link $n5 $n7 2Mb 10ms DropTail
$ns duplex-link $n6 $n8 6Mb 10ms DropTail
$ns duplex-link $n7 $n8 6Mb 10ms DropTail
$ns duplex-link $n8 $n9 12Mb 10ms DropTail
$ns duplex-link $n9 $n10 6Mb 10ms DropTail
$ns duplex-link $n10 $n11 6Mb 10ms DropTail
$ns duplex-link $n12 $n11 4Mb 10ms DropTail
$ns duplex-link $n13 $n11 4Mb 10ms DropTail
$ns duplex-link $n14 $n11 4Mb 10ms DropTail
# node position
$ns duplex-link-op $n0 $n6 orient right-up
$ns duplex-link-op $n1 $n6 orient right-center
$ns duplex-link-op $n2 $n6 orient right-down
$ns duplex-link-op $n3 $n7 orient down-right
$ns duplex-link-op $n4 $n7 orient down
$ns duplex-link-op $n5 $n7 orient down-left
$ns duplex-link-op $n6 $n8 orient right-up
```

```
$ns duplex-link-op $n7 $n8 orient down
$ns duplex-link-op $n8 $n9 orient right
$ns duplex-link-op $n9 $n10 orient right
$ns duplex-link-op $n10 $n11 orient right
$ns duplex-link-op $n11 $n12 orient right-up
$ns duplex-link-op $n11 $n13 orient right
$ns duplex-link-op $n11 $n14 orient right-down
set udp1 [new Agent/UDP]
$ns attach-agent $n0 $udp1
set udp2 [new Agent/UDP]
$ns attach-agent $n1 $udp2
#set tcp1 [new Agent/UDP]
set tcp1 [new Agent/TCP/Reno]
$ns attach-agent $n2 $tcp1
set udp4 [new Agent/UDP]
$ns attach-agent $n3 $udp4
set udp5 [new Agent/UDP]
$ns attach-agent $n4 $udp5
# set tcp2 [new Agent/UDP]
set tcp2 [new Agent/TCP/Reno]
$ns attach-agent $n5 $tcp2
set trafic01 [new Application/Traffic/CBR]
# $trafic01 set interval_ 0.0025
$trafic01 set rate_ 1.8Mb
$trafic01 set packetSize_ 500
$trafic01 attach-agent $udp1
$udp1 set class_ 1
# The Poisson process is set by configuring burst_time_ to 0 and rate_ to a large value.
```

```
set trafic02 [new Application/Traffic/Exponential]
$trafic02 set rate_ 1.8Mb
$trafic02 set packetSize_ 500
$trafic02 set burst_time_ 0ms
$trafic02 set idle_time_ 100ms
$trafic02 attach-agent $udp2
$udp2 set class_ 2
set trafic03 [new Application/FTP]
$trafic03 attach-agent $tcp1
$trafic03 set type_ FTP
$tcp1 set class_ 3
set trafic04 [new Application/Traffic/CBR]
# $trafic04 set interval_ 0.0025
$trafic04 set rate_ 1.8Mb
$trafic04 set packetSize_ 500
$trafic04 attach-agent $udp4
$udp4 set class_ 4
set trafic05 [new Application/Traffic/Exponential]
$trafic05 set rate_ 1.8Mb
$trafic05 set packetSize_ 500
$trafic05 set burst_time_ 0ms
$trafic05 set idle_time_ 100ms
$trafic05 attach-agent $udp5
$udp5 set class_ 5
# FTP
set trafic06 [new Application/FTP]
$trafic06 attach-agent $tcp2
$trafic06 set type_ FTP
```

```
$tcp2 set class_ 6
set recep4 [new Agent/LossMonitor]
$ns attach-agent $n12 $recep4
set recep5 [new Agent/LossMonitor]
$ns attach-agent $n13 $recep5
# set recep6 [new Agent/LossMonitor]
set recep6 [new Agent/TCPSink]
$ns attach-agent $n14 $recep6
set recep7 [new Agent/LossMonitor]
$ns attach-agent $n12 $recep7
set recep8 [new Agent/LossMonitor]
$ns attach-agent $n13 $recep8
# set recep9 [new Agent/LossMonitor]
set recep9 [new Agent/TCPSink]
$ns attach-agent $n14 $recep9
$ns connect $udp1 $recep4
$ns connect $udp2 $recep5
$ns connect $tcp1 $recep6
$ns connect $udp4 $recep7
$ns connect $udp5 $recep8
$ns connect $tcp2 $recep9
proc finish {} { global ns tracefile nf
$ns flush-trace
close $nf
close $tracefile
exec awk -f packetLoss1.awk packetLoss.tr > Non-adaptiveBandwidthforVoice.xg &
exec awk -f packetLoss2.awk packetLoss.tr > Non-adaptiveBandwidthforVideo.xg &
exec awk -f packetLoss3.awk packetLoss.tr > Non-adaptiveBandwidthforWeb.xg &
```

```
exec awk -f packetLoss4.awk packetLoss.tr > Non-adaptiveBandwidthforVoiceHO.xg
&
exec awk -f packetLoss5.awk packetLoss.tr > Non-adaptiveBandwidthforVideoHO.xg
&
exec awk -f packetLoss6.awk packetLoss.tr > Non-adaptiveBandwidthforWebHO.xg
&
exec nam packetLoss.nam &
exit 0
} $ns at 0.0 ”$trafic01 start”
$ns at 0.0 ”$trafic02 start”
$ns at 0.0 ”$trafic03 start”
$ns at 0.0 ”$trafic05 start”
$ns at 0.0 ”$trafic06 start”
$ns at 3.0 ”$trafic01 stop”
$ns at 3.0 ”$trafic04 start”
$ns at $testTime ”$trafic02 stop”
$ns at $testTime ”$trafic03 stop”
$ns at $testTime ”$trafic04 stop”
$ns at $testTime ”$trafic05 stop”
$ns at $testTime ”$trafic06 stop”
$ns at [expr $testTime + 1.0] ”finish”
$ns run
```

Appendix C

Adaptive Bandwidth allocation in IP Network Code

```
set ns [new Simulator]
set testTime 10.0
# colors for traffic flows
$ns color 1 Blue
$ns color 2 Green
$ns color 3 Red
$ns color 4 Purple
$ns color 5 Gray
$ns color 6 Yellow

set tracefile [open packetLoss.tr w]
$ns trace-all $tracefile
set nf [open packetLoss.nam w]
$ns namtrace-all $nf
# network topology structure
set n0 [$ns node]
set n1 [$ns node]
set n2 [$ns node]
set n3 [$ns node]
set n4 [$ns node]
```

```
set n5 [$ns node]
set n6 [$ns node]
set n7 [$ns node]
set n8 [$ns node]
set n9 [$ns node]
set n10 [$ns node]
set n11 [$ns node]
set n12 [$ns node]
set n13 [$ns node]
set n14 [$ns node]
$ns duplex-link $n0 $n6 2Mb 10ms DropTail
$ns duplex-link $n1 $n6 2Mb 10ms DropTail
$ns duplex-link $n2 $n6 2Mb 10ms DropTail
$ns duplex-link $n3 $n7 2Mb 10ms DropTail
$ns duplex-link $n4 $n7 2Mb 10ms DropTail
$ns duplex-link $n5 $n7 2Mb 10ms DropTail
#$ns duplex-link $n6 $n8 6Mb 10ms DropTail
$ns simplex-link $n6 $n8 6Mb 10ms dsRED/edge $ns simplex-link $n8 $n6 6Mb 10ms
dsRED/core
#$ns duplex-link $n7 $n8 6Mb 10ms DropTail $ns simplex-link $n8 $n7 6Mb 10ms
dsRED/core $ns simplex-link $n7 $n8 6Mb 10ms dsRED/edge
#$ns duplex-link $n8 $n9 12Mb 10ms DropTail $ns simplex-link $n9 $n8 12Mb 10ms
dsRED/edge $ns simplex-link $n8 $n9 12Mb 10ms dsRED/core
#$ns duplex-link $n9 $n10 6Mb 10ms DropTail $ns simplex-link $n10 $n9 6Mb 10ms
dsRED/core $ns simplex-link $n9 $n10 6Mb 10ms dsRED/edge
#$ns duplex-link $n10 $n11 6Mb 10ms DropTail $ns simplex-link $n11 $n10 6Mb
10ms dsRED/edge $ns simplex-link $n10 $n11 6Mb 10ms dsRED/core
$ns duplex-link $n12 $n11 4Mb 10ms DropTail
```

```
$ns duplex-link $n13 $n11 4Mb 10ms DropTail
$ns duplex-link $n14 $n11 4Mb 10ms DropTail

# Set node position in nam
$ns duplex-link-op $n0 $n6 orient right-up
$ns duplex-link-op $n1 $n6 orient right-center
$ns duplex-link-op $n2 $n6 orient right-down
$ns duplex-link-op $n3 $n7 orient down-right
$ns duplex-link-op $n4 $n7 orient down
$ns duplex-link-op $n5 $n7 orient down-left
$ns duplex-link-op $n6 $n8 orient right-up
$ns duplex-link-op $n7 $n8 orient down
$ns duplex-link-op $n8 $n9 orient right
$ns duplex-link-op $n9 $n10 orient right
$ns duplex-link-op $n10 $n11 orient right
$ns duplex-link-op $n11 $n12 orient right-up
$ns duplex-link-op $n11 $n13 orient right
$ns duplex-link-op $n11 $n14 orient right-down
set qE1C1 [[$ns link $n6 $n8] queue ]
set qC1E1 [[$ns link $n8 $n6] queue ]
set qE2C1 [[$ns link $n7 $n8] queue ]
set qC1E2 [[$ns link $n8 $n7] queue ]
set qE3C1 [[$ns link $n9 $n8] queue ]
set qC1E3 [[$ns link $n8 $n9] queue ]
set qE3C2 [[$ns link $n9 $n10] queue ]
set qC2E3 [[$ns link $n10 $n9] queue ]
set qE4C2 [[$ns link $n11 $n10] queue ]
set qC2E4 [[$ns link $n10 $n11] queue ]
```

```
#source 1 2 3 dest 12 13 14
# Set DS RED parameters from Edge1 to Core1:
$qE1C1 meanPktSize 500
$qE1C1 set numQueues_ 3
$qE1C1 setNumPrec 1
$qE1C1 addPolicyEntry [$n0 id] [$n12 id] TokenBucket 46 2000000 50000
$qE1C1 addPolicyEntry [$n1 id] [$n13 id] TokenBucket 20 2000000 50000
$qE1C1 addPolicyEntry [$n2 id] [$n14 id] TokenBucket 0 2000000 50000
$qE1C1 addPolicerEntry TokenBucket 46 47
$qE1C1 addPolicerEntry TokenBucket 20 21
$qE1C1 addPolicerEntry TokenBucket 0 1
$qE1C1 addPHBEntry 46 0 0
$qE1C1 addPHBEntry 20 1 0
$qE1C1 addPHBEntry 0 2 0
$qE1C1 configQ 0 0 20 40 0.000020
$qE1C1 configQ 1 0 11 20 0.20
$qE1C1 configQ 2 0 1 5 0.40
# Set DS RED parameters from Edge3 to Core1: The problem starts
$qE3C1 meanPktSize 500
$qE3C1 set numQueues_ 3
$qE3C1 setNumPrec 1
$qE3C1 addPolicyEntry [$n12 id] [$n0 id] TokenBucket 46 2000000 50000
$qE3C1 addPolicyEntry [$n13 id] [$n1 id] TokenBucket 20 2000000 50000
$qE3C1 addPolicyEntry [$n14 id] [$n2 id] TokenBucket 0 2000000 50000
$qE3C1 addPolicerEntry TokenBucket 46 47
$qE3C1 addPolicerEntry TokenBucket 20 21
$qE3C1 addPolicerEntry TokenBucket 0 1
$qE3C1 addPHBEntry 46 0 0
```

```
$qE3C1 addPHBEntry 20 1 0
$qE3C1 addPHBEntry 0 2 0
$qE3C1 configQ 0 0 20 40 0.000020
$qE3C1 configQ 1 0 11 20 0.20
$qE3C1 configQ 2 0 1 5 0.40
# Set DS RED parameters from Edge3 to Core2:
$qE3C2 setSchedularMode PRI
$qE3C2 addQueueRate 0 2000000
$qE3C2 addQueueRate 1 1500000
$qE3C2 addQueueRate 2 1000000
$qE3C2 meanPktSize 500
$qE3C2 set numQueues_ 3
$qE3C2 setNumPrec 1
$qE3C2 addPolicyEntry [$n0 id] [$n12 id] TokenBucket 46 2000000 50000
$qE3C2 addPolicyEntry [$n1 id] [$n13 id] TokenBucket 20 2000000 50000
$qE3C2 addPolicyEntry [$n2 id] [$n14 id] TokenBucket 0 2000000 50000
$qE3C2 addPolicerEntry TokenBucket 46 47
$qE3C2 addPolicerEntry TokenBucket 20 21
$qE3C2 addPolicerEntry TokenBucket 0 1
$qE3C2 addPHBEntry 46 0 0
$qE3C2 addPHBEntry 20 1 0
$qE3C2 addPHBEntry 0 2 0
$qE3C2 configQ 0 0 20 40 0.000020
$qE3C2 configQ 1 0 11 20 0.20
$qE3C2 configQ 2 0 1 5 0.40
# Set DS RED parameters from Edge4 to Core2:
$qE4C2 setSchedularMode PRI
$qE4C2 addQueueRate 0 2000000
```

```
$qE4C2 addQueueRate 1 1500000
$qE4C2 addQueueRate 2 1000000
$qE4C2 meanPktSize 500
$qE4C2 set numQueues_ 3
$qE4C2 setNumPrec 1
$qE4C2 addPolicyEntry [$n12 id] [$n0 id] TokenBucket 46 2000000 50000
$qE4C2 addPolicyEntry [$n13 id] [$n1 id] TokenBucket 20 2000000 50000
$qE4C2 addPolicyEntry [$n14 id] [$n2 id] TokenBucket 0 2000000 50000
$qE4C2 addPolicerEntry TokenBucket 46 47
$qE4C2 addPolicerEntry TokenBucket 20 21
$qE4C2 addPolicerEntry TokenBucket 0 1
$qE4C2 addPHBEntry 46 0 0
$qE4C2 addPHBEntry 20 1 0
$qE4C2 addPHBEntry 0 2 0
$qE4C2 configQ 0 0 20 40 0.000020
$qE4C2 configQ 1 0 11 20 0.20
$qE4C2 configQ 2 0 1 5 0.40
# source 3 4 5 dest12 13 14
# Set DS RED parameters from Edge2 to Core1:
$qE2C1 meanPktSize 500
$qE2C1 set numQueues_ 3
$qE2C1 setNumPrec 1
$qE2C1 addPolicyEntry [$n3 id] [$n12 id] TokenBucket 46 2000000 50000
$qE2C1 addPolicyEntry [$n4 id] [$n13 id] TokenBucket 20 2000000 50000
$qE2C1 addPolicyEntry [$n5 id] [$n14 id] TokenBucket 0 2000000 50000
$qE2C1 addPolicerEntry TokenBucket 46 47
$qE2C1 addPolicerEntry TokenBucket 20 21
$qE2C1 addPolicerEntry TokenBucket 0 1
```

$qE2C1 addPHBEntry 46 0 0

$qE2C1 addPHBEntry 20 1 0

$qE2C1 addPHBEntry 0 2 0

$qE2C1 configQ 0 0 20 40 0.000020

$qE2C1 configQ 1 0 11 20 0.20

$qE2C1 configQ 2 0 1 5 0.40

Set DS RED parameters from Edge3 to Core1: The problem starts $qE3C1 meanPktSize 500 $qE3C1 set numQueues_ 3 $qE3C1 setNumPrec 1 $qE3C1 addPolicyEntry [$n12 id] [$n3 id] TokenBucket 46 2000000 50000

$qE3C1 addPolicyEntry [$n13 id] [$n4 id] TokenBucket 20 2000000 50000

$qE3C1 addPolicyEntry [$n14 id] [$n5 id] TokenBucket 0 2000000 50000

$qE3C1 addPolicerEntry TokenBucket 46 47

$qE3C1 addPolicerEntry TokenBucket 20 21

$qE3C1 addPolicerEntry TokenBucket 0 1

$qE3C1 addPHBEntry 46 0 0

$qE3C1 addPHBEntry 20 1 0

$qE3C1 addPHBEntry 0 2 0

$qE3C1 configQ 0 0 20 40 0.000020

$qE3C1 configQ 1 0 11 20 0.20

$qE3C1 configQ 2 0 1 5 0.40

Set DS RED parameters from Edge3 to Core2:

$qE3C2 setSchedularMode PRI

$qE3C2 addQueueRate 0 2000000

$qE3C2 addQueueRate 1 1500000

$qE3C2 addQueueRate 2 1000000

$qE3C2 meanPktSize 500

$qE3C2 set numQueues_ 3

$qE3C2 setNumPrec 1

```
$qE3C2 addPolicyEntry [$n3 id] [$n12 id] TokenBucket 46 2000000 50000
$qE3C2 addPolicyEntry [$n4 id] [$n13 id] TokenBucket 20 2000000 50000
$qE3C2 addPolicyEntry [$n5 id] [$n14 id] TokenBucket 0 2000000 50000
$qE3C2 addPolicerEntry TokenBucket 46 47
$qE3C2 addPolicerEntry TokenBucket 20 21
$qE3C2 addPolicerEntry TokenBucket 0 1
$qE3C2 addPHBEntry 46 0 0
$qE3C2 addPHBEntry 20 1 0
$qE3C2 addPHBEntry 0 2 0
$qE3C2 configQ 0 0 20 40 0.000020
$qE3C2 configQ 1 0 11 20 0.20
$qE3C2 configQ 2 0 1 5 0.40
# Set DS RED parameters from Edge4 to Core2:
$qE4C2 setSchedularMode PRI
$qE4C2 addQueueRate 0 2000000
$qE4C2 addQueueRate 1 1500000
$qE4C2 addQueueRate 2 1000000
$qE4C2 meanPktSize 500
$qE4C2 set numQueues_ 3
$qE4C2 setNumPrec 1
$qE4C2 addPolicyEntry [$n12 id] [$n3 id] TokenBucket 46 2000000 50000
$qE4C2 addPolicyEntry [$n13 id] [$n4 id] TokenBucket 20 2000000 50000
$qE4C2 addPolicyEntry [$n14 id] [$n5 id] TokenBucket 0 2000000 50000
$qE4C2 addPolicerEntry TokenBucket 46 47
$qE4C2 addPolicerEntry TokenBucket 20 21
$qE4C2 addPolicerEntry TokenBucket 0 1
$qE4C2 addPHBEntry 46 0 0
$qE4C2 addPHBEntry 20 1 0
```

```
$qE4C2 addPHBEntry 0 2 0
$qE4C2 configQ 0 0 20 40 0.000020
$qE4C2 configQ 1 0 11 20 0.20
$qE4C2 configQ 2 0 1 5 0.40
# Set DS RED parameters from Core1 to Edge1:
$qC1E1 meanPktSize 500
$qC1E1 set numQueues_ 3
$qC1E1 setNumPrec 1
$qC1E1 addPHBEntry 46 0 0
$qC1E1 addPHBEntry 20 1 0
$qC1E1 addPHBEntry 0 2 0
$qC1E1 configQ 0 0 20 40 0.000020
$qC1E1 configQ 1 0 11 20 0.20
$qC1E1 configQ 2 0 1 5 0.40
# Set DS RED parameters from Core1 to Edge2:
$qC1E2 meanPktSize 500
$qC1E2 set numQueues_ 3
$qC1E2 setNumPrec 1
$qC1E2 addPHBEntry 46 0 0
$qC1E2 addPHBEntry 20 1 0
$qC1E2 addPHBEntry 0 2 0
$qC1E2 configQ 0 0 20 40 0.000020
$qC1E2 configQ 1 0 11 20 0.20
$qC1E2 configQ 2 0 1 5 0.40
# Set DS RED parameters from Core1 to Edge3:
$qC1E3 meanPktSize 500
$qC1E3 set numQueues_ 3
$qC1E3 setNumPrec 1
```

```
$qC1E3 addPHBEntry 46 0 0
$qC1E3 addPHBEntry 20 1 0
$qC1E3 addPHBEntry 0 2 0
$qC1E3 configQ 0 0 20 40 0.000020
$qC1E3 configQ 1 0 11 20 0.20
$qC1E3 configQ 2 0 1 5 0.40
# Set DS RED parameters from Core2 to Edge3:
$qC2E3 meanPktSize 500
$qC2E3 set numQueues_ 3
$qC2E3 setNumPrec 1
$qC2E3 addPHBEntry 46 0 0
$qC2E3 addPHBEntry 20 1 0
$qC2E3 addPHBEntry 0 2 0
$qC2E3 configQ 0 0 20 40 0.000020
$qC2E3 configQ 1 0 11 20 0.20
$qC2E3 configQ 2 0 1 5 0.40
# Set DS RED parameters from Core2 to Edge4:
$qC2E4 meanPktSize 500
$qC2E4 set numQueues_ 3
$qC2E4 setNumPrec 1
$qC2E4 addPHBEntry 46 0 0
$qC2E4 addPHBEntry 20 1 0
$qC2E4 addPHBEntry 0 2 0
$qC2E4 configQ 0 0 20 40 0.000020
$qC2E4 configQ 1 0 11 20 0.20
$qC2E4 configQ 2 0 1 5 0.40
set udp1 [new Agent/UDP]
$ns attach-agent $n0 $udp1
```

```
set udp2 [new Agent/UDP]
$ns attach-agent $n1 $udp2
#set tcp1 [new Agent/UDP]
set tcp1 [new Agent/TCP/Reno]
$ns attach-agent $n2 $tcp1
set udp4 [new Agent/UDP]
$ns attach-agent $n3 $udp4
set udp5 [new Agent/UDP]
$ns attach-agent $n4 $udp5
# set tcp2 [new Agent/UDP]
set tcp2 [new Agent/TCP/Reno]
$ns attach-agent $n5 $tcp2
set trafic01 [new Application/Traffic/CBR]
# $trafic01 set interval_ 0.0025
$trafic01 set rate_ 1.8Mb
$trafic01 set packetSize_ 500
$trafic01 attach-agent $udp1
$udp1 set class_ 1
# The Poisson process is set by configuring burst_time_ to 0 and rate_ to a large value.
$trafic02 set rate_ 1.8Mb
$trafic02 set packetSize_ 500
$trafic02 set burst_time_ 0ms
$trafic02 set idle_time_ 100ms
$trafic02 attach-agent $udp2
$udp2 set class_ 2
set trafic03 [new Application/FTP]
$trafic03 attach-agent $tcp1
$trafic03 set type_ FTP
```

```
$tcp1 set class_ 3
set trafic04 [new Application/Traffic/CBR]
# $trafic04 set interval_ 0.0025
$trafic04 set rate_ 1.8Mb
$trafic04 set packetSize_ 500
$trafic04 attach-agent $udp4
$udp4 set class_ 4
set trafic05 [new Application/Traffic/Exponential]
$trafic05 set rate_ 1.8Mb
$trafic05 set packetSize_ 500
$trafic05 set burst_time_ 0ms
$trafic05 set idle_time_ 100ms
$trafic05 attach-agent $udp5
$udp5 set class_ 5
# FTP
set trafic06 [new Application/FTP]
$trafic06 attach-agent $tcp2
$trafic06 set type_ FTP
$tcp2 set class_ 6
set recep4 [new Agent/LossMonitor]
$ns attach-agent $n12 $recep4
set recep5 [new Agent/LossMonitor]
$ns attach-agent $n13 $recep5
# set recep6 [new Agent/LossMonitor]
set recep6 [new Agent/TCPSink]
$ns attach-agent $n14 $recep6
set recep7 [new Agent/LossMonitor]
$ns attach-agent $n12 $recep7
```

```
set recep8 [new Agent/LossMonitor]
$ns attach-agent $n13 $recep8
# set recep9 [new Agent/LossMonitor]
set recep9 [new Agent/TCPSink]
$ns attach-agent $n14 $recep9
$ns connect $udp1 $recep4
$ns connect $udp2 $recep5
$ns connect $tcp1 $recep6
$ns connect $udp4 $recep7
$ns connect $udp5 $recep8
$ns connect $tcp2 $recep9
proc finish {} { global ns tracefile nf
$ns flush-trace
close $nf
close $tracefile
exec awk -f packetLoss1.awk packetLoss.tr > Non-adaptiveBandwidthforVoice.xg &
exec awk -f packetLoss2.awk packetLoss.tr > Non-adaptiveBandwidthforVideo.xg &
exec awk -f packetLoss3.awk packetLoss.tr > Non-adaptiveBandwidthforWeb.xg &
exec awk -f packetLoss4.awk packetLoss.tr > Non-adaptiveBandwidthforVoiceHO.xg
&
exec awk -f packetLoss5.awk packetLoss.tr > Non-adaptiveBandwidthforVideoHO.xg
&
exec awk -f packetLoss6.awk packetLoss.tr > Non-adaptiveBandwidthforWebHO.xg
&
exec nam packetLoss.nam &
exit 0
} $ns at 0.0 ”$trafic01 start”
$ns at 0.0 ”$trafic02 start”
```

```
$ns at 0.0 "$trafic03 start"
$ns at 0.0 "$trafic05 start"
$ns at 0.0 "$trafic06 start"
$ns at 3.0 "$trafic01 stop"
$ns at 3.0 "$trafic04 start"
$ns at $testTime "$trafic02 stop"
$ns at $testTime "$trafic03 stop"
$ns at $testTime "$trafic04 stop"
$ns at $testTime "$trafic05 stop"
$ns at $testTime "$trafic06 stop"
$ns at [expr $testTime + 1.0] "finish"
$ns run
```

Appendix D

Packet Loss Code

```
BEGIN {
packDrops = 0;
timeInterval = 1;
currentTime = 0;
nxtTime = currentTime + timeInterval;
}
{ act = $1;
time = $2;
nod1 = $3;
nod2 = $4;
source = $5;
trafficFlowId = $8;
nod1Addres = $9;
nod2Addres = $10;
seqNum = $11;
packetId = $12;
if (time < nxtTime)
{ if (act == "d" && trafficFlowId ==1)
{ packDrops+=packDrops;
} } else {currentTime = nxtTime;
nxtTime += timeInterval;
packDrops+=packDrops;
printf("% d % d \n",currentTime, packDrops);} } END { }
```

Appendix E

Throughput Code

```
BEGIN { byteReceived = 0;
thruput = 0;
timeInterval = 1;
currentTime = 0;
nxtTime = currentTime + timeInterval;
} { act = $1;
time = $2;
from = $3;
to = $4;
trafficType = $5;
packetSize = $6;
trafficFlowId = $8;
source = $9;
destination = $10;
sequenceNum = $11;
packetId = $12;

if (time < nxtTime)
{ if (act == "r" && trafficFlowId ==1)
{ byteReceived = byteReceived + packetSize;
} } else { currentTime = nxtTime;
nxtTime += timeInterval;
thruput += byteReceived / currentTime;
```

```
printf("% d % d \n",currentTsime, thruput/1000000);} }
END { }
```

Appendix F

End to End Delay Code

```
# This program is used to calculate the end-to-end delay for CBR
BEGIN { highestPacketId = 0;
} { act = $1;
time = $2;
from = $3;
to = $4;
trafficType = $5;
packetSize = $6;
trafficFlowId = $8;
source = $9;
destination = $10;
sequenceNum = $11;
packetId = $12;
if ( packetId > highestPacketId ) highestPacketId = packetId;
if ( startTime[packetId] == 0 )
startTime[packetId] = time;
if ( tafficFlowId == 1 && act != ”d” ) {
if ( act == ”r” ) {
endTime[packetId] = time;
} }
else { endTime[packetId] = -1;
} } END {
for ( packetId = 0; packetId <= highestPacketId; packetId++ ) {
```

```
start = startTime[packetId];
end = endTime[packetId];
packetDuration = end-start;
if (start< end )
printf("% f % f \n" ,start , packetDuration);
} }
```

Appendix G

Jitter Code

```
BEGIN { # Init
highestPacketId = 0;
} { act = $1;
time = $2;
from = $3;
to = $4;
trafficType = $5;
packetSize = $6;
trafficflowId = $8;
source = $9;
destination = $10;
sequenceNum = $11;
packetId = $12;
if ( packetId > highestPacketId ) {
highestPacketId = packetId;
} # transmission time calculation
if ( startTime[packetId] == 0 ) {
# account for sequence number
packetSequenceNum[packetId] = sequenceNum;
startTime[packetId] = time;
} # receiving time for traffic
if ( trafficFlowId == 1 && act != "d" ) {
if ( act == "r" ) {
```

```
endTime[packetId] = time;
} } else { endTime[packetId] = -1;
} } END { lastSequenceNum = 0;
lastDelay = 0;
sequenceNumDiff = 0;
for ( packetId = 0; packetId <= highestPacketId; packetId++ ) {
start = startTime[packetId];
end = endTime[packetId];
packetDuration = end - start;
if ( start < end ) {
sequenceNumDiff = packetSequenceNum[packetId] - lastSequenceNum;
delayDiff = packetDuration - lastDelay;
if (sequenceNumDiff == 0) {
jitter =0;
} else { jitter = delayDiff/sequenceNumDiff;
} printf("% f % f \n", start, jitter);
lastSequenceNum = packetSequenceNum[packetId];
lastDelay = packetDuration;
} } }
```

Appendix H

Video Handover and its Effects on other Multimedia Traffic Graphs

H.1 Packet Loss

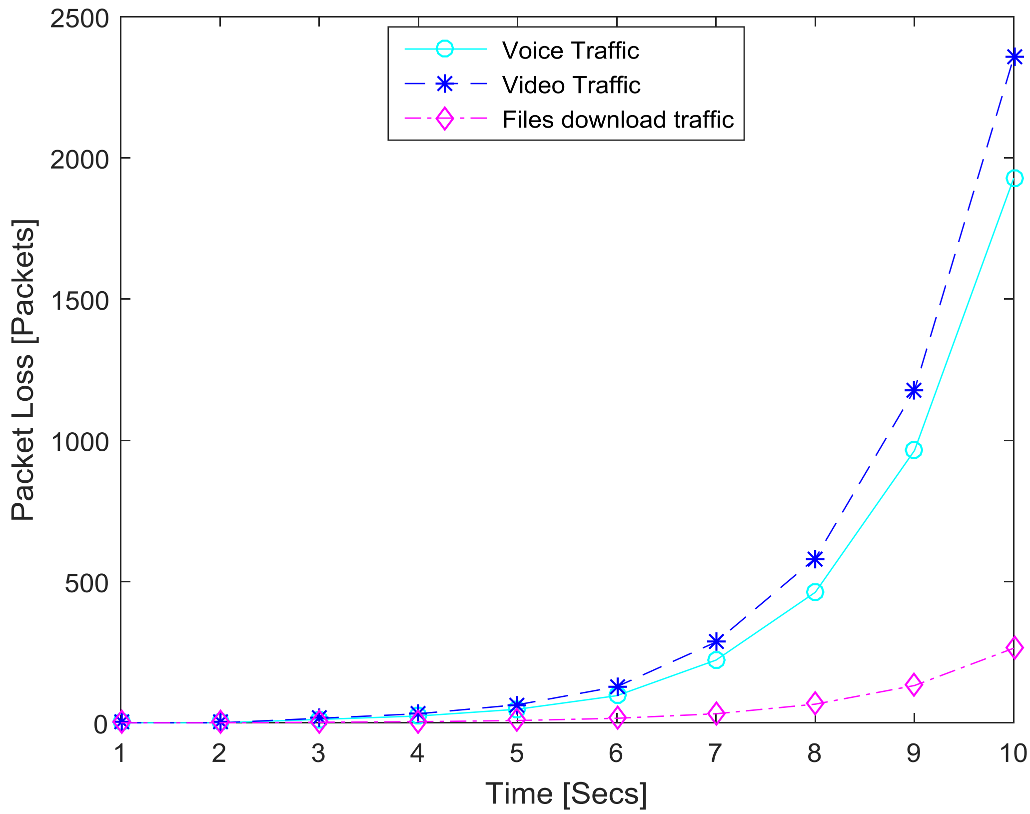

Figure H.1: Packet loss in conventional IP for video traffic handover

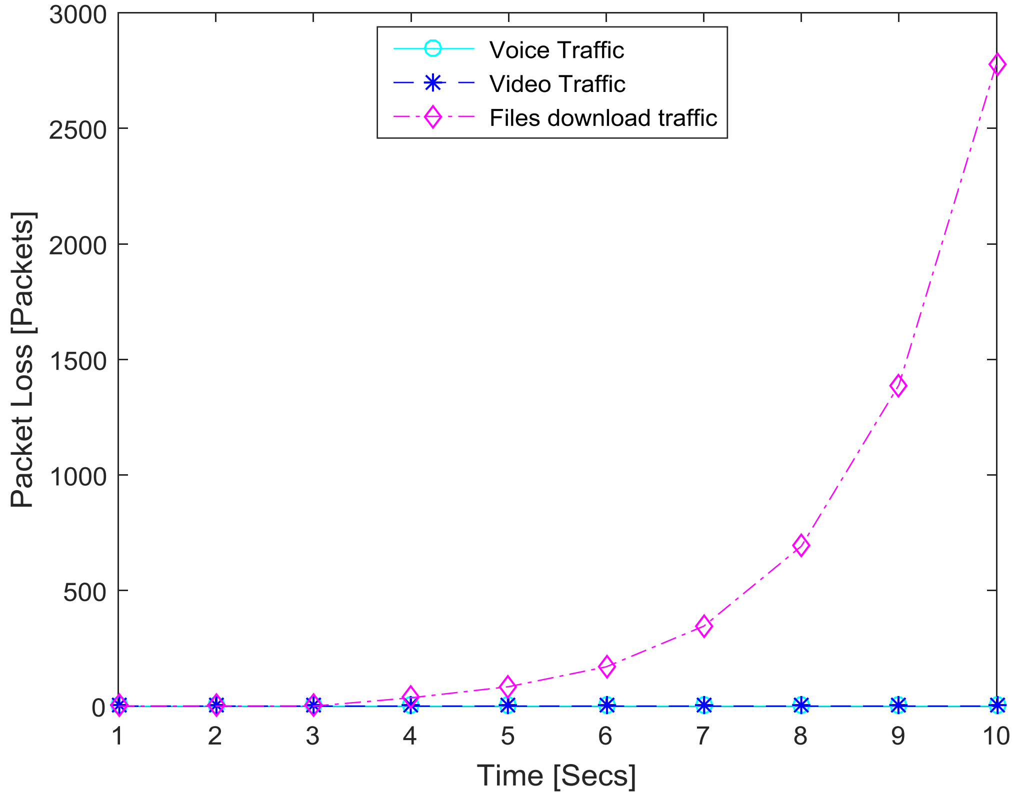

Figure H.2: Packet loss in adaptive bandwidth allocation for video traffic handover

H.2 Throughput

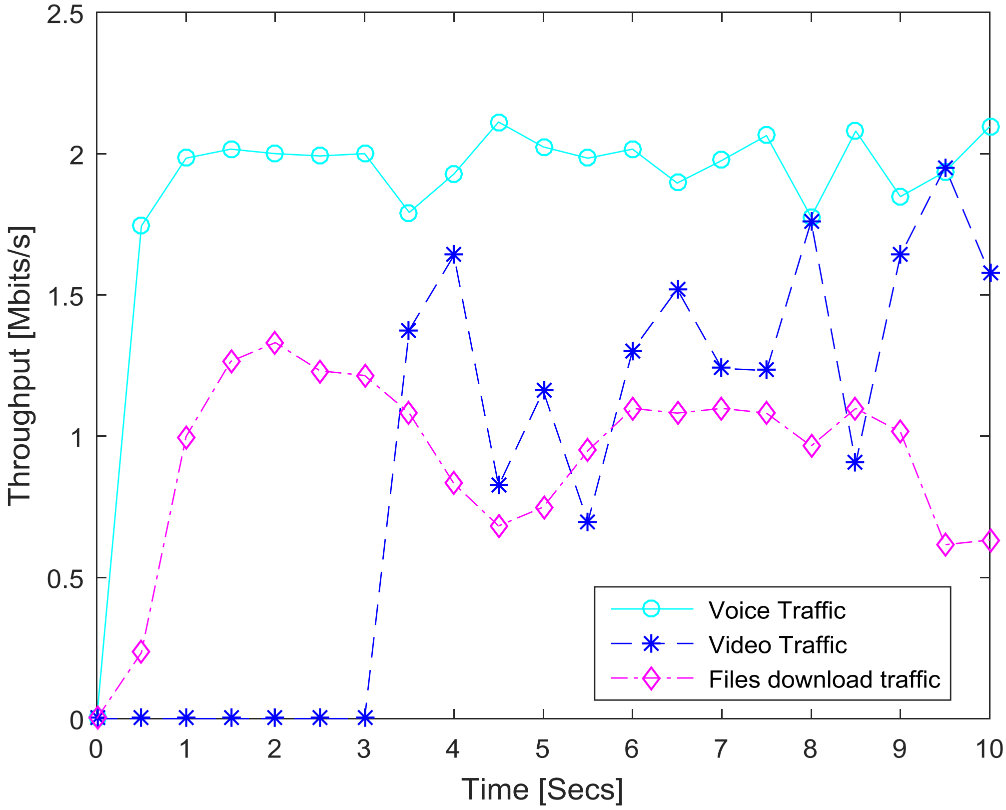

Figure H.3: Throughput in conventional IP for video traffic handover

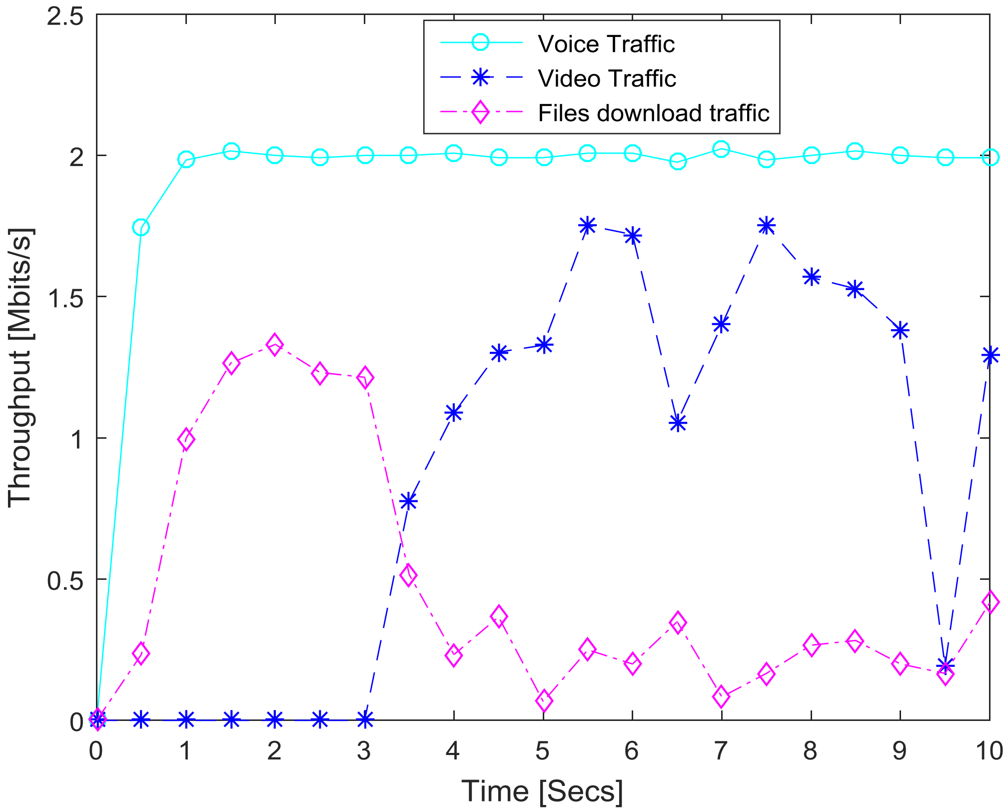

Figure H.4: Throughput in adaptive bandwidth allocation for video traffic handover

H.3 End to End Delay

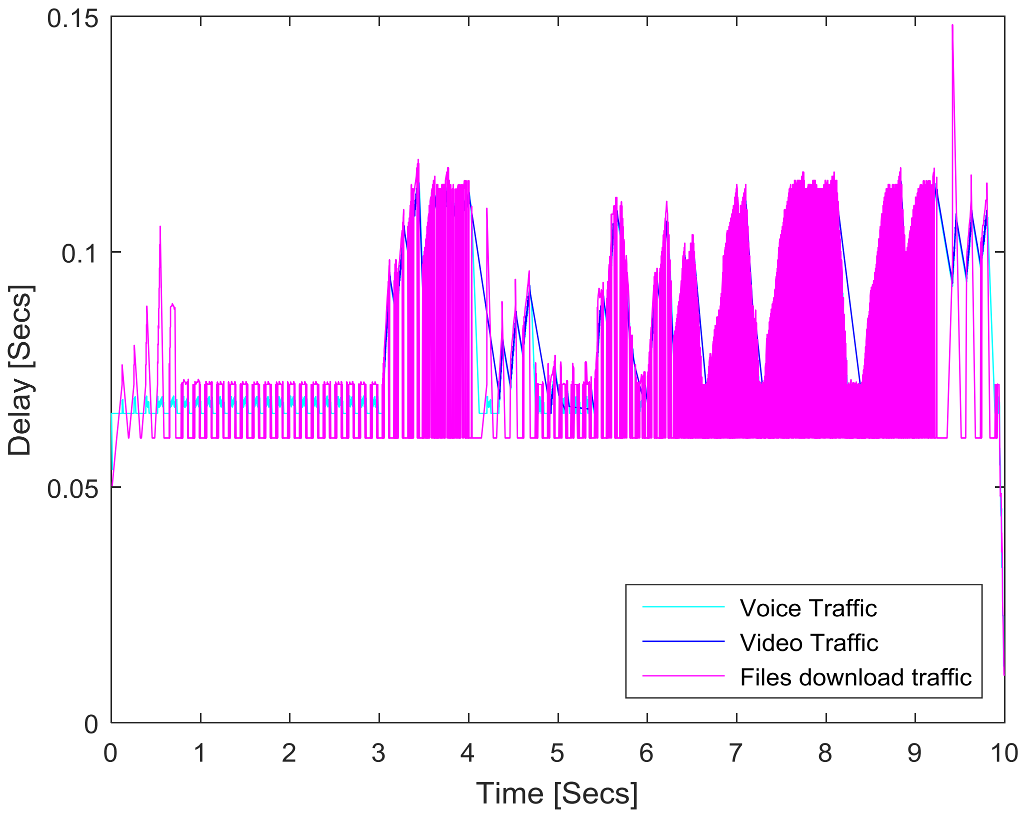

Figure H.5: Delay in conventional IP for video traffic handover

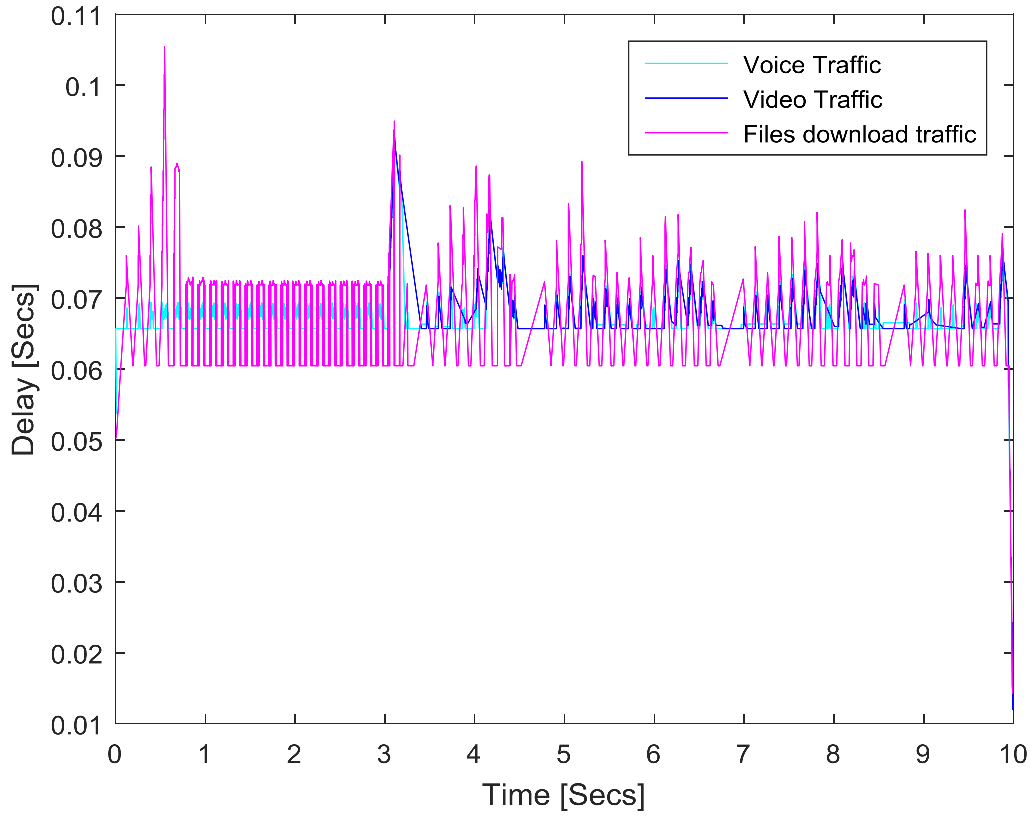

Figure H.6: Delay in adaptive bandwidth allocation for video traffic handover

H.4 Jitter

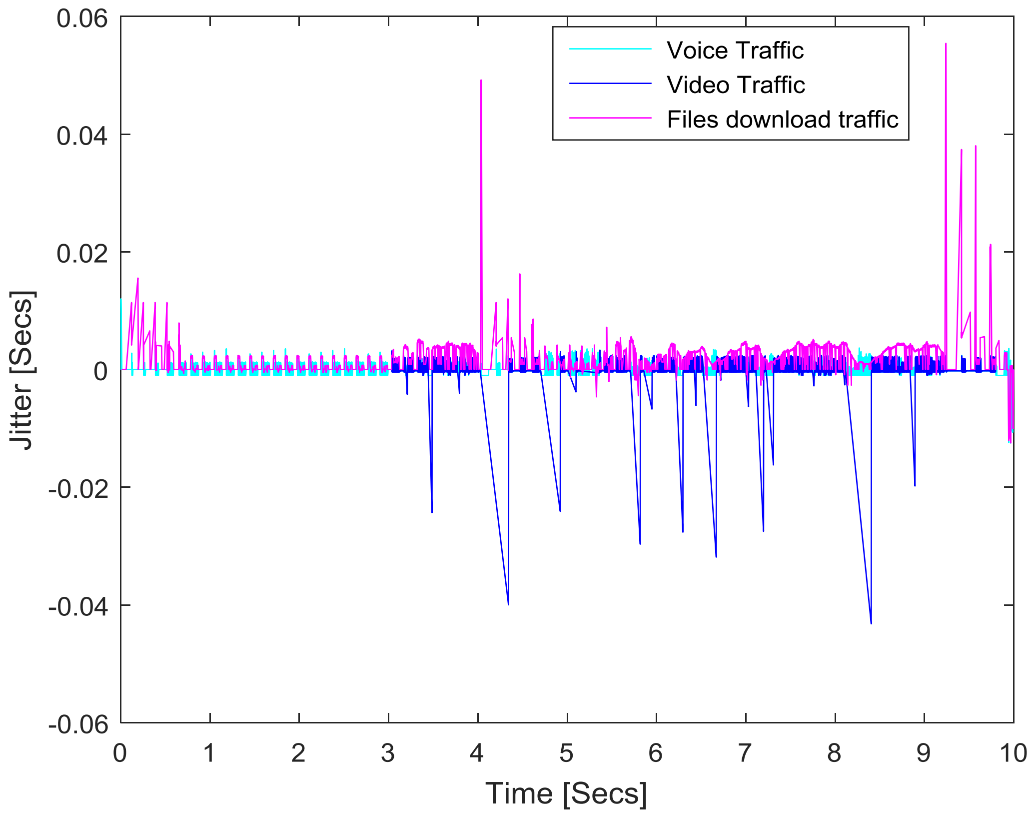

Figure H.7: Jitter in conventional IP for video traffic handover

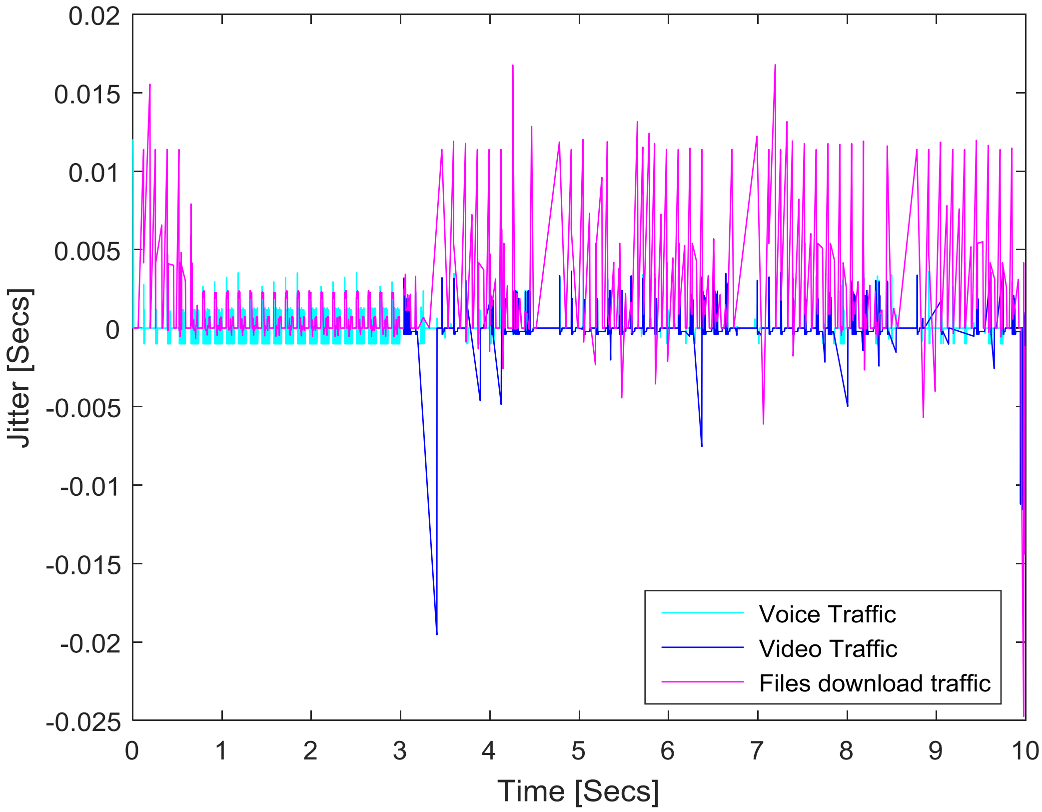

Figure H.8: Jitter in adaptive bandwidth allocation for video traffic handover