QoS OPTIMIZATION THROUGH CAPACITY AGGREGATION OF MULTIPLE LINKS IN HETEROGENEOUS WIRELESS NETWORKS

by

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A thesis submitted to the
Department of Computer Science
in partial fulfillment of the requirements for the degree of
DOCTORATE OF SCIENCE IN COMPUTER SCIENCE

Faculty of Engineering and Applied Sciences
Mohammad Ali Jinnah University
Islamabad
March 2011

ACKNOWLEDGEMENTS

First of all, I present my humble thankfulness to ALLAH for his forgiveness, mercy and guidance, without that this thesis work would have not been possible.

I would like to express my sincere gratitude to the leadership of Dr. Mansoor Ahmed, Executive Vice President, Mohammad Ali Jinnah University, Islamabad; for his whole hearted support during my PhD work. He has been highly influential and supportive in promoting research culture in the campus and provided moral, administrative and social support in this research work.

I would like to say special thanks to my thesis supervisor, Dr. Muhammad Abdul Qadir, Dean, Mohammad Ali Jinnah University, Islamabad; who provided me highly skillful guidance, technical support, motivation and feedback that helped me in converging and completing this thesis work.

I am especially thankful to Professor Abdelaziz Bouras, Member LIESP Laboratory at University of Lumiere, Lyon and Head of the CERRAL knowledge transfer center of the Lumiere Technology Institute in Lyon, France. He provided highly motivating research environment at Lyon which helped me a lot in concluding my research. The members of his team in CERRAL also provided thoughtful feedback during seminar session and general discussions. I am thankful to all of them for their assistance.

Special thanks for the members of ENS Lyon, network research team for their technical assistance and valuable comments in refining and improving the quality of work of this thesis.

I am also thankful to Dr. Noor Mohammad Khan, Associate Professor at Mohammad Ali Jinnah University, Islamabad for his valuable suggestions and guidance in improving presentation of this thesis.

I owe a special thanks to the members of distributed and semantic computing group at Mohammad Ali Jinnah University, Islamabad, who provided useful feedback throughout the work and also created interactive and lively research environment that abridged academic isolation and motivated competitive work.

I am also thankful to Higher Education of Pakistan (HEC) for providing patronage to my PhD work. The support provided by HEC during my studies, conference travel and publications provided high motivation for me to put more effort in my research work.

Last but not least, my special thanks to my family; my mother, wife and my two daughters who showed remarkable patience during the whole period of work. Without their silent support this thesis would have not been possible.

ABSTRACT

Wireless and mobile networking is considered as the most appreciated technological innovation that has stormed into the life-styles of the people and has found applications in business, education, health, social networking and all other major areas of day-to-day work. Inspired by the enormous potential of mobile computing, wireless communication and mobile networking technologies have emerged as major research disciplines within the domain of communication systems. Most of the communication based applications have been developed and executed on robust wired communication networks and when they are ported to mobile devices, face serious challenges in meeting quality of service (QoS) requirements over the mutable wireless media. The QoS enabling for mobile networks has, therefore remained a major research area in this discipline. The presence of multiple communication interfaces in modern multi-mode mobile devices has enabled a new dimension for improving QoS solution during mobility but the heterogeneity of these networks pose serious challenges for delay bounded QoS aware applications. In this thesis, capacity aggregation (CAG) of multiple available wireless links has been investigated to quantify its suitability in providing QoS during mobility.

There are numerous challenges faced by the designers and developers in synchronizing flows, transported over multi-path. One such issue is the out-of-sequence (OOS) reception of packet at the receiver due to transport of these packets on multiple heterogeneous paths. The larger extent of OOS receptions; generally cause serious performance degradation in delay bounded real-time applications, and is also a source of expensive buffer management for in-order delivery of packets to their respective destinations. The problem gets adverse during mobility, as the end-to-end path repeatedly changes during mobility and accompanies with changes in characteristics of E2E path that enhances probability of OOS reception. Therefore, maximization of QoS through minimization of OOS reception in a CAG environment is the main problem investigated in this thesis. The problem has been persuaded with novel multi-server scheduling schemes that minimize the OOS reception at the receiver. The proposed multi-server scheduling schemes provide a general service model that accommodates multiple types of traffic flows belonging to different classes-of-service. This approach is in contrast to the QoS maximization of a single flow at upper layers of TCP/IP protocol stack.

The approach of multi-server scheduling to minimize OOS reception has been investigated with the help of novel deterministic and stochastic analytical models to provide quantified solution of the above given problem. The fundamental mathematical structure of novel deterministic and stochastic models has been elaborated to enumerate multi-path transport dynamics of a mobile flow. These models provide appropriate formulae for the dimensioning questions of the QoS during mobility. Hence, the analysis of the multi-path transport of mobile flows remains as the main focus of this research work and constitutes core of this dissertation. Since the deterministic and stochastic framework for the analysis of E2E multi-path dynamic are not fully developed, my research focused on efficient estimations of E2E delay variations and its impact on OOS reception of packets. The tight deterministic and stochastic performance bounds have been derived through proposed models. The main application of these models has been exercised in development of suitable scheduling strategies that minimize E2E delay variation and OOS reception in CAG during mobility. The results of proposed models have been validated through simulations as well. The results have shown robust performance of proposed scheduling schemes in achieving acceptable QoS levels for real-time flows during mobility, with minimized OOS reception and reduced buffer occupancy.

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ASM	Actively Scalable Multi-server
BAG	Bandwidth Aggregation
BVNC	Bounded Variance Network Calculus
CAG	Capacity aggregation
CoA	Care-of-Address
CoS	Class-of-Service
CN	Corresponding Node
DRR	Deficit Round Robin
DiffServ	Differentiated Services
EDPF	Earliest delivery path first
E2E	End-to-End
GR	Guaranteed Rate
GPS	Generalized Processor Scheduling
HWN	Heterogeneous wireless network
HBI	Highest bandwidth interface
HA	Home Agent
HHO	Horizontal handover
IntServ	Integrated Services
ICMP	Internet Control Message Protocol
IPv4	Internet Protocol version 4
LR	Latency Rate
MIHF	Media Independent Handover Function
MH	Mobile Host
MIPv4	Mobile IPv4
MR	Mobile Router
MA	Mobility Agent
MMD	Multi-mode Mobile Devices
MRR	Multi-Server Round Robin
NC	Network Calculus
OTT	One-way Trip Time
OOS	Out-of-Sequence
PoA	Point-of-Attachment
QoS	Quality-of-service
RSVP	Resource Reservation Protocol
RTT	Round-trip time
SCTP	Stream Control Transport Protocol
SRR	Surplus Round Robin
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
VHO	Vertical Handover
WLAN	Wireless LAN

Chapter 1

INTRODUCTION

The infusion of wireless technologies in the modern digital communication infrastructure and its impact on our day-to-day life has been phenomenal in the recent time. The inception of wireless data services with their extensive support for legacy applications, such as e-mail, web browsing, file transfer etc., complemented with more sophisticated applications like geo-positioning system(GPS), mobile games, mobile multimedia services, mobile commerce, social networking etc., has given new dimensions to work, leisure, information access and social interactions. In fact, once dreamed paradigm of ubiquitous computing (Lyytinen Kalle 2002) has just started to deliver part of its ultimate promise. It would not be wrong assertion that high speed, reliable and dynamically scalable wireless data communication services work at the core to enable true ubiquitous access for end users. There are numerous wireless access technologies, more commonly termed radio access technologies (RATs), already deployed and contending to provide data services to end users for their specific needs, within constraints of each technology. These include IEEE standards 802.11 a/b/e/g/n based Wireless Local Area Networks (WLANs) (IEEE-802.11 2007), IEEE 802.16e based Wireless Metropolitan Area Networks (WMANs) (802.21-2008 2009), and series of generations of Cellular Networks (2G, 2.5G, 3G, etc) (Lawton 2005).

Each of the above mentioned technologies has its own advantages as well as some limitations. For instance, WLAN is good for moderate data rates with restricted mobility at lower cost; whereas, WMAN provides higher data rates with low mobility support at higher cost (S. K. Leung 2008). Similarly; cellular networks provide intermediate data rates at higher mobility with high cost but are constrained by user density and ambient conditions (T. Huang 2009). Despite their momentous flexibility, wireless media lacks consistency due to various factors; such as, mobility events, channel fading, fluid user density and traffic load that cause service quality variations and generally result in less than optimum throughput to mobile sessions (Jamalipour, Wada and Yamazato 2005). Such perturbations result in unpredictable performance metrics (reduced throughput, extended variations in end-to-end latencies, higher drop rates), and occasionally, the network may become completely incapable of meeting the requirements of ongoing sessions, particularly in case of real-time applications (Kameswari. C 2006). Generally, for the quality-aware sessions, the resources are reserved for the whole duration of the session during call admission control (CAC) process but mobility and varying signal conditions may not comply with such agreements due to physical constraints (Devarapalli 2005).

At the same time, processing capabilities of mobile communication devices have also hoisted from simple phones to very sophisticated pocket personal computers (PCs) and mobile routers (MRs) (Lucian Suciu 2005). We find today, mobile devices equipped with powerful processors, large memories and multiple wireless interfaces that provide redundant choices of connecting to any of available wireless networks (Chen Yiping 2007). The enhanced computing power of mobile devices has also brought much heavier applications on these devices, enabling them to be used as mobile servers and routers. These services are heavy in terms of both processing complexity as well as communication volume. Further, real-time applications like video streaming and video conferencing require consistent service guarantees; more commonly referred to as Quality of Service (QoS), in terms of data rates and bounded latencies. Keeping in view the critical nature of all these services; the guaranteed service provisioning in such environments is challenged by the mutable service of wireless data networks and has received significant interest of researchers in recent years (Devarapalli 2005) (X. Z. Chen 2008).

The availability of communication resources, like output buffers in intermediate routers and communication link's time-slots, is crucial for providing service guarantees to QoS-aware sessions. In the wired subnets, these resources are used more efficiently due to the higher predictability and consistency of link. In mobility using Mobile Internet Protocol (MIP), the link availability changes significantly in time and space and causes greater degree of uncertainty about the next point-of-attachments (PoAs), that also leads to major change in the path of session/flow (Perkins 2002). It is quite unlikely that the same resource set availed at the previous PoA is also available at all subsequent PoAs. It is therefore, desirable that some level of over-provisioning of resources is maintained in the mobility supporting routers and associated paths to overcome expensive signaling cost of re-negotiations with multiple PoAs (S. De Vuyst 2008). This approach results in non-scalable systems that may not accommodate even modest user loads despite being quite expensive. There are quite a few parameters, such as user density, network capacity, mobility profiles and patterns, channel access mechanisms that influence the service consistency of mobile data networks and make related decision making more complex during mobility events. The QoS provisioning during mobility is therefore, a question that requires multi-facet analysis before a comprehensive solution may be possible (Sharma 2008).

In the backdrop of finding comprehensive solution for QoS during mobility, the mobile devices equipped with multiple wireless interfaces; experiencing a heterogeneous wireless network (HWN) environment, provide a window for enhancing QoS by opportunistic use of any or more than one networks simultaneously (X. Z. Chen 2008). The multimode mobile device (MMD); the term found commonly in literature for devices equipped with multiple wireless interfaces, not only can improve service guarantees for their application and services, but can also provide sufficient leverage to other single interfaced mobile devices by choosing network with lower traffic load. Therefore, simultaneous use of multiple interfaces of MMDs has been keenly studied as bandwidth aggregation (BAG) techniques in recent past (X. Z. Chen 2008). Though initial studies indicate considerable potential of BAG techniques (sometimes also referred to as capacity aggregation; abbreviated as CAG of multiple wireless channels), the usefulness of any such configuration is dependent on many parameters; such as the degree of asymmetric characteristics of wireless channels, the congestion state of each path of these wireless channels, the architectural placement of such BAG/CAG techniques in the protocol stack, number of hops on each path, ratio of load distribution on each path, and overhead of managing BAG/CAG (Kameswari. C 2006). In worst cases, the cumulative impact of all these factors can be catastrophic for end-to-end (E2E) performance metric (retransmissions due to out-of-sequence reception, higher packet drop rate, complex congestion control mechanisms). Even in best case scenarios, the performance of BAG/CAG techniques may not coincide with the theoretical sum of all available capacities due to its operational and maintenance overheads. Mobility of such MMDs adds another major constraint. It is therefore, important to quantitatively analyze the concerned parameters to make BAG/CAG techniques more useful and workable. The accuracy of such analysis can greatly facilitate process of developing suitable scheduling strategies and distribution of traffic load on available interfaces. The purpose of these scheduling strategies may be focused on subsiding counter-productive impact of heterogeneity of multiple paths and to achieve one or more desirable performance metrics; either for the individual flows or for the networks (Chebrolu K. 2005). This thesis investigates the possibility of maximizing QoS of individual flows during mobility, through quantification of impact of multi-path heterogeneity.

1.1. Background

One of the possible consequences of multi-path traversal of a single flow is the out-of-sequence (OOS) reception of packets at the receiver or at intermediate proxy, deployed to harmonize these flows (X. Z. Chen 2008). The OOS reception could seriously increase complexity of congestion and flow control mechanisms of transport protocols such as transmission control protocol (TCP) and Stream Control Transport Protocol (SCTP) (Janardhan R. Iyengar 2006). The other QoS enabled protocols such as Real-time Transport Protocol (RTP) and Real-time Stream Transport Protocol (RSTP) that require stringent time and rate constraints may also face higher buffer occupancy and packet drop due to OOS reception of packets at the receiver (Piratla N M. 2008). The problem gets sterner during mobility as episodes of various mobility events, such as handovers cause repeated disconnections at layer 2 (S. K. Leung 2008). In a multiple care-of-address (CoA) registration scenario using Mobile IPv4 (MIPv4), multiple tunnels are available between MMD and home agent (HA) to simultaneously use these tunnels and transparently route backlogged packets over any of available tunnels (R. Wakikawa 2009).

The major challenge in such scenarios is to transport multiple flows with distinct QoS requirement, on these multiple tunnels in a manner that minimizes the probability of OOS reception for each flow (Devarapalli 2005). Figure 1-1 depicts one such scenario where there are multiple flows, such that flow i has a data rate r_i, are queued in multiple buffers before being transmitted over the multiple tunnels through a multi-server scheduler[1], working at MMD. The scheduled packets then traverse through any of the M, MIPv4 tunnels, as per deployed scheduling strategy. The major challenges in devising suitable scheduling strategy depends on identification and estimation of each path characteristics accurately, to support scheduling process in a way that E2E delay variations and OOS reception are minimized at the tunnel ends. A comprehensive model for analysis of such scenarios is therefore important before adopting any scheduling strategy. This thesis work de-

[1] A multi-server scheduler is a terminology used in queuing system where multiple service points are available. In packet switched communication networks, the multiple network interfaces act as the service points.

velops foundations for QoS aware multi-server scheduling paradigm for mobile sessions. Under the assumption of sufficient availability of communication capacity of multiple networks, it minimizes OOS reception through proactively adaptive scheduling strategies that accurately estimate the characteristics of multiple tunnels and emulates multiple MIPv4 tunnels as a single virtual channel with known probability of OOS arrival. It develops both stochastic and deterministic analytical models to signify E2E path dynamics of such tunnels to minimize impact of E2E path asymmetries of these tunnels.

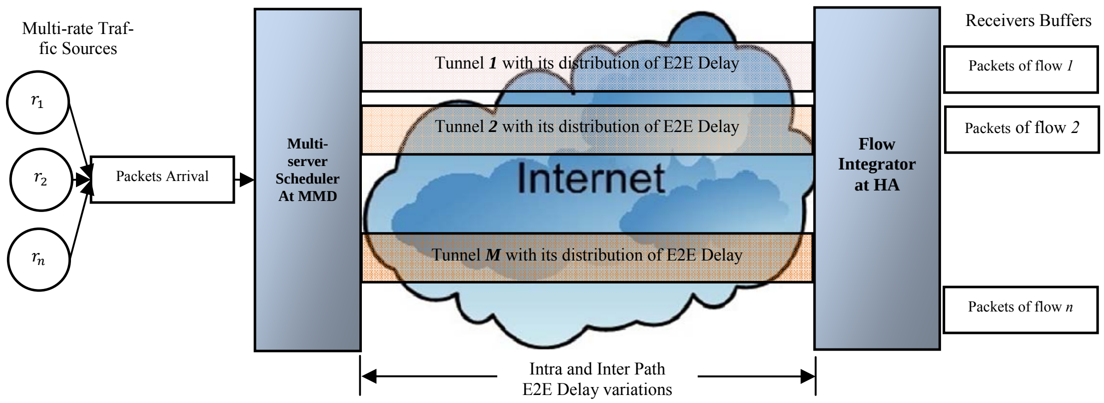

Figure 1-1: A simplified model of Multipath E2E flow management during mobility

1.2. State-of-the-Art in Bandwidth Aggregation

The simultaneous use of multiple available wireless interfaces has gained considerable interest of research community in recent years. The majority of researches attempted to achieve BAG to tap maximum service for specific set of applications, such as video streaming, online streaming. The upper layers are considered as more convenient means of service maximization for individual flows due to Application Programming Interface (API) support of operating systems. In such approaches, individual applications open multiple sockets over multiple ports and transmit multiple streams of a single flow to achieve desired service rates (T. Huang 2009). It is obvious that in such arrangement, large volumes of data may be transported in short time, but due to varying E2E dynamics of multiple networks, service may not be guarantee, particularly in terms of latencies. Similarly, transport layer BAG approaches use single or multiple congestion windows to maintain unified E2E stream through multiple paths (Chebrolu K. 2005), (Sharma 2008) & (Janardhan R. Iyengar 2006). The above mention approaches achieve desirable service guarantees at the cost of service fairness, as multiple congestion windows may reduce the share of other flows considerably. The multiple congestion windows add significant overhead of congestion and flow control operations and may also induce congestions in the host networks. It is important to note that despite heavy overheads and congestion vulnerabilities, service guarantees in terms of bounded latencies and in-order reception of flow may not be guaranteed. In (Kameswari. C 2006), a network layer solution has been proposed that sends video traffic to an MMD using a proxy server at the base station (BS). The BAG is accomplished through transmission of packets, arrived at the proxy server over any one of available links that promises the fastest delivery of traffic to the MMD. Since it is a single hop multipath traversal, the OOS arrival is nominal but in case, packets are already arriving OOS at the proxy server, the in-order delivery to MMD cannot be guaranteed. In (K. R. Evensen 2009), a network layer, proxy server based approach of BAG is presented that ranks a link according to its service quality and transmits packets on the link with best link quality. This approach has significant benefits provided path characteristics don't change significantly over the time. In mobility based scenarios, the OOS arrival may be much higher due to frequent path changes (Ming Yang 2004). The uncertainty in acquiring the desired QoS parameters at the next PoA has also motivated for pre attachment resource reservation for the mobile sessions at the probable PoA (Xiao 2004). In this approach, resources are reserved for a mobile session while it is still in the previous PoA. This approach seriously hampers the scalability of system and also results in poor resource utilization. Table 1.1, 1.2 and 1.3 summarize major contributions in this area. The review of above mentioned state-of-the-art techniques of BAG/CAG highly motivates thorough investigation of QoS maximization potential anticipated through BAG/CAG techniques, especially in presence of OOS reception due to multi-path traversal of a flow during mobility. The same is thus, the core research question investigated in this thesis.

1.3. Hypothesis and Contributions of the Thesis

The hypothesis of the thesis is the provision of QoS[2] while in mobility, through multi-path transport of flows under the constraints of minimum OOS reception of packets at the receiver end. In multi-path flows, QoS may be compromised by the E2E delay variations of constituent paths that cause OOS reception of packets, as discussed before. Therefore, we

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[2] Though QoS has diverse meanings in communication networks, but in the scope of this thesis, it has been defined in terms of desired data rates within a bounded delay constraint.

need to minimize OOS in order to achieve required QoS. The OOS arrival is mainly attributed to some parameters; including E2E delay and its variations. In this thesis work, we have formalized the fundamental E2E parameters that influence the multi-path traversal of QoS enabled flows during mobility in HWN environment and contribute significantly in OOS reception. The fundamental question of interest is whether or not CAG of heterogeneous paths can provide QoS guarantees during mobility under the constraint of minimum OOS reception? The subsequent questions that arise include what parameters can be most crucial in quantifying heterogeneity of the multiple paths and the impact of heterogeneity on QoS guarantees of traffic flows? Can we upper-bound the impact of these parameters to guarantee desired QoS levels? How can we make such upper-bounds to be tight enough so as they don't hinder the scalability of the system without compromising desired QoS levels? What kind of scheduling strategies can best absorb the impact of path heterogeneity? How closer we can get to the optimization of QoS guarantees during mobility, after answering the above questions?

The thesis answers most of the questions of the research hypothesis. The focal point in this work is to analyze the probable causes of OOS reception, and the impact of such reception. We have adapted an incremental approach in which asymmetric characteristics of collocated subnets are analyzed through both deterministic and stochastic models to estimate intra-path and inter-path E2E delay variations and then combined it into a model that minimizes OOS reception. These models provide closed form upper bounds for the probable OOS reception of packets of the same flow. The deterministic model provides useful insight for achieving upper bounds for the two key metrics i.e. Intra-path and inter-path E2E delay variation and corresponding extent of OOS reception. The deterministic model, though is useful for basic system design, provides only best or worst case statistics for the above mentioned E2E metrics. In order to derive complete spectrum of the distribution of these metrics, stochastic models are developed to achieve tighter bounds and improve scalability of installed capacity in the system.

To our best knowledge, both deterministic as well as stochastic models are not available in literature for QoS-enabled multi-server scheduling over heterogeneous networks. Similarly, the proactive description of OOS reception of traffic over the multiple asymmetric paths is also a novel contribution of this thesis. The bounds achieved through the modeling process are used to devise novel scheduling algorithms to distribute traffic on multiple paths. The validation of these models and accompanying scheduling algorithms is performed through simulations in ns-2 simulator (The Network Simulator - ns-2 n.d.).

Table 1-1: BAG/CAG proposals for the network layer of protocol stack

BAG Proposal	Aggregation Technique	Impact
(Kameswari, Bhaskaran and R., 5 2005)	Capacity Estimation over the E2E path and Weighted Traffic Distribution	**Contribution:** Basic E2E multi-path solution for TCP **Drawback:** Capacity estimation is expensive in terms of bandwidth
(Evensen, et al. 2009)	Intermediate Proxy deployment based traffic integration and distribution	**Contribution**: General solution for multi-class traffic an better resource utilization **Drawback**: Location of proxy not evaluated as the multi-hop parallel path adds complex E2E delay distribution
(Kameswari and R, Bandwidth Aggregation for Real-time Applications in Heterogeneous Wireless Networks 2006)	EDPF based scheduling integrated at the Proxy server	**Contribution:** Lesser OOS arrival and improved throughput. **Drawback:** Restricted to a single hop wireless section solution
(Jui-Tang, Yuan-Ying and Chien-Chao 2008)	Redundant resource reservation on available paths	**Contribution:** QoS is ensured through reservations at multiple probable PoA **Drawback:** Redundant reservation results in poor resource utilization
(Phatak, Goff and Plusquellic 2003)	FIFO scheduling on multiple IP-in-IP encapsulated Tunnels	**Contribution:** Higher bandwidth for TCP applications **Drawback:** Path asymmetries not handled causing poor E2E delay performance for real-time applications
(Jiang and Yao 2003)	FIFO based Scheduling	**Contribution:** E2E delay variation is accounted for service guarantees **Drawback:** The distribution of E2E delay variation not studied

The results of analytical models and simulations indicate significantly reliable QoS provisioning during mobility through multi-server scheduling while demonstrating low order OOS reception at the tunnel end. It has been found through the proposed deterministic model that the OOS reception at the tunnel ends is primarily based on the variations in server latencies that eventually lead to intra-path as well as inter-paths E2E delay variations. It has been further investigated that delay variations are also directly proportional to the number of servers (nodes) in the paths. The variations in packet length also contributed significantly in overall E2E delay variations through variable transmission delay. The variations in propagation delays of heterogeneous wireless channels also contributed in OOS reception of packets. The simulations of proposed scheduling scheme has revealed that about 80% of traffic experience delay variations of fewer than 10% of mean, and the accompanying OOS receptions are even lower than 8% of all received packets. It was further investigated that QoS guarantees were accomplished with lesser buffer overhead at the tunnel ends for holding OOS reception that helps in maintaining in-order forwarding of packets to the application or the next subnet. The QoS guarantees were maintained up to a very high reliability of 90%. The results of proposed models were validated through simulations and were found in close binding with analytical model results within a narrow range of 10% deviations.

Table 1-2: BAG/CAG proposals at the network layer of protocol stack

BAG Proposal	Aggregation Technique	Impact
(Puneet, et al. 2009)	Collaborative sharing of community resources	**Contribution:** Scalable solution for end-users through higher communication resource availability **Drawback:** Group membership is essential which is difficult to keep up-to-date during mobility
(T. Huang 2009)	API and socket level supported link aggregation	**Contribution:** Easy to use aggregation **Drawback:** Control at application that may be unfair for other applications
(Fernandez, et al. 2009)	QoS Negotiation through Multiple Interfaces	**Contribution:** QoS provision is more scalable **Drawback:** Negotiation cost is generally high
(Yao, Guo and Bhuyan 2008)	FIFO Scheduling over homogeneous channels	**Contribution:** Maximum resource utilization **Drawback:** Generally good for only one hop or over a well defined tunnel

Table 1-3: BAG/CAG proposals for Transport layer of protocol stack

BAG Proposal	Aggregation Technique	Impact
(Kalyanaraman, Ramakrishnan and V. 2008)	Weighted Congestion window approach	**Contribution:** Higher throughput through multi-stream transmission **Drawback:** oscillation in congestion window based on historic data
(Ken and Hiroshi 2006)	Congestion Control tuned to fast retransmission problem	**Contribution:** Reduces fast retransmission issues **Drawback:** Mobility issues not discussed
(Magalhaes and Kravets 2001)	Congestion control based on differentiation of link and network loss	**Contribution:** Improved stability of consistent throughput **Drawback:** Complexity of Congestion control and capacity estimation is high
(Al, Saadawi and Lee 2004)	Modified Congestion Control of SCTP	**Contribution:** Higher Throughput with multi-streaming **Drawback:** The congestion control mechanism is tuned for specific applications.
(Iyengar, Amer and Stewart 2006)	Multi-streamed, multiple congestion window based link aggregation	**Contribution:** Improved throughput **Drawback:** The behavior of multiple congestion windows adds unpredictable retransmission scenarios

1.4. Outline of Thesis

The flow of the thesis is given in the following text. Chapter 2 describes the system model along with underlying assumptions of the multi-tunnel mobility using multiple CoA registration at HA. The symbols and notation used in the stochastic and deterministic models are also described here. In chapter 3, a novel deterministic analytical model for multipath traversal of a flow is presented. The basic motive is to quantify E2E delay variation and bounded OOS reception. The deterministic model upper-bounds E2E delay variation on the basis of server latencies, transmission delays, propagation delays and packet arrival time at the scheduling server. The guaranteed rate service model is used to describe E2E path delay. The major source of delay variation is cause by the size of the packets and numbers of hop on each path. Under proposed model intra-path delay variation are much lesser than the inter-path delay variations. In the second model, the strict upper bounds of the E2E delay and its variations were relaxed to lower levels at significantly realistic thresholds to improve scalability of the service. In this model, it is identified that the delay variations are rooted in variable packet sizes and average queue lengths on each server in the E2E path. It is proven that the proposed model has some salient properties that make it suitable for estimation of E2E delay variations and associated OOS reception. We also provide proof of theorems that enable characterization and quantification of E2E delay related metrics.

In chapter 4, a novel bounded variance network calculus (BVNC) based stochastic model for E2E delay variations, OOS receptions and buffer overflow bounds is presented. The stochastic study of these properties helps in optimizing the buffer management used to handle OOS reception. The stochastic model provides two important metrics, i.e., the probability of violation of a delay threshold at any server, and probability of violation of a given E2E delay threshold. These two metrics help in designing the appropriate strategy of buffer management to cope with OOS reception problem. The proof of theorems for violation of thresholds for E2E delay variations, OOS reception, and buffer occupancy are also given.

In chapter 5, we present multi-server scheduling algorithms that are based on the outcome of modeling exercise given in chapter 3 & 4. The proposed scheduling algorithms use E2E metrics to rank the available path, and schedule multi-class traffic, accordingly. It also provides simulation results of the cross-layer algorithms developed to minimize E2E delay variation and OOS reception. The results are compared with some of the existing schemes for comparison.

Chapter 6 summarizes the overall contribution of the thesis and findings of the research. Some future dimensions of research in this domain are also discussed. The importance of BAG/CAG is discussed with the support of some useful results as discussed in chapter 6.

Chapter 2

SYSTEM MODEL

In this chapter, the system model with underlying assumptions is described with reference to the proposed multi-server scheduling environment. The proposed system model is constrained by mobility events, QoS and asymmetric conditions of the heterogeneous network. In the beginning section, some key terms used in this thesis are defined, particularly in lieu of analytical and stochastic models. The notations and symbols used in this thesis are also presented at the end of this chapter.

2.1. Key Terms and Definitions

The terms defined in this section are closely related to multi-server scheduling and its analytical models. The scope of these terms in this thesis will strictly remain restricted as defined below.

System Busy Time: The k^{th} System Busy Time, $\mathcal{H}_k^S(t, t + \tau)$ of a multi-server scheduler during any backlogged interval $(t, t + \tau)$ is the duration of time when server leaves idle state to start servicing backlogged traffic over all or any of available multiple interfaces till its returns back to the idle state after completion of service.

The system busy time is a direct measure of performance of any server as it contributes to server latencies experienced by the packets while getting service. The system busy time of a multi-server scheduler is generally less than its single-server counterpart.

Server Latency: The Server Latency $\theta_S(p_f^k)$, of any server is the amount of delay experience by any backlogged packet p_f^k at the server before it is completely transmitted. Essentially it is caused by the different factors such as its committed data rate, packet length, its scheduled transmission time, and wait due to busy media. In fact, it implicitly encapsulates queuing delay at the server as well.

The server latency plays a key role in determining the E2E delay as in scope of this thesis, the E2E delay comprises of server latencies and transmission delays of all the intermediate servers and associated communication links. In case of reserved communication resources, the role of server latency in determining E2E delay may restrict to only the first server in the path. The same is discussed in more details in chapter 3.

Out-of-Sequence Reception: Out-of-Sequence reception φ_f^k of a packet p_f^k is defined as the late reception of this packet as compared to some of it successor packets of the same flow. In a QoS-aware multi-path traversal scenario, the OOS reception is primarily attributed to the differences in E2E delay of constituent paths along with length of packets.

In the scope of this thesis, OOS reception is rooted back in the server latencies and the path lengths. The path differences are randomly occurring events and may be caused by only multi-server scheduling or mobility events such as handovers.

Buffer Occupancy: The buffer occupancy ϑ_f^k is defined as the amount of buffer space (in bits) occupied by some of the successor packets of p_f^k due to its late arrival.

Assuming flow-based servers, the in-order forwarding is essential at each server. In the scope of this thesis, the in-order forwarding is considered as the responsibility of the destination or the mobility agent working at the tunnel ends.

2.2. Description of System Model

The proposed system model for multi-server scheduling consists of three main components and technologies. First of all, it is designed over IP mobility infrastructure supported through MIPv4 (Perkins 2002) by maintaining multiple tunnels with home agent (HA) through foreign Agents (FA), if available. Further, it provides multiple class-of-services (CoS), supporting different data rates and latencies requirements. The cumulative communication resources of multiple tunnels are used to fulfill service requirement of multiple flows. It intends to provide minimize OOS reception at the receiver end of the tunnels to hide asymmetric path characteristics of multiple tunnels. The proposed model also uses event services of Media Independent Handover (MIH) proposed by IEEE (IEEE Standard 802.21 2009). The QoS has been defined in terms of known data rate and upper bounded E2E latency, as outlined in (J. Gozdecki 2003). The characteristics of multiple MIPv4 tunnels operating between the MMD and HA keep on changing during the mobility, as horizontal handover (HHO) events take place at the layer 2. It is worth mentioning here that there is no essential requirement of vertical handover (VHO) in the proposed system. The already established tunnels through other links take the load of lost connectivity of link till the HHO is complete. The QoS parameters are not renegotiated with the new PoA and the QoS is managed through other links though it may suffer from reduced service levels during handover events. The main advantage of this approach is to save signaling cost of multiple negotiation messages that may be needed in search of desired service levels from any of probable PoA. The traffic is scheduled through the proposed scheduling algorithm in a way that the QoS guarantees are maintained within a certain known degree of tolerance in violation of pre-determined threshold.

The proposed model serves a set of flows belonging to different class-of-services (CoS) which are differentiated by their data rate and latency requirements. The mobility is abstracted by the changes in available capacity of the system, as described above. Figure 2-1 sketches key component of the multi-server scheduling model in association with the Internet that includes the unified arrival queue of traffic flows constraint by token-bucket regulator (Blake 1998), the multi-server scheduler, and the output queues of multiple wireless links. A unified queue at the input holds arrived packets, sorted in accordance with their departure schedule. The multi-server scheduler schedules these packets on any of available link using E2E delay estimates of the links. The E2E path selection metrics work independently to estimate E2E delay of each tunnel. The available capacity metric provides support for admission control in association with the MIH mobility events (Ahmad, Qadir and Akbar, Towards Dependable wireless networks a QoS constraint resource management scheme in heterogeneous environment 2008). The estimation of E2E delay provides useful assistance in improving efficiency of multi-server scheduling in term of QoS guarantees and minimization of OOS reception at the receiver.

It is noticeable in Figure 2.1 that multiple flows are represented solely by their data rate $r_i, i = 1,2,..,n,$ for n classes-of-service (CoS), each bounded by different data rate and delay constraints. The data rate and delay constraints for each CoS are predefined and each flow acquires its desired service through mapping its requirements over nearest available CoS during the Call Admission & Control (CAC) process (Fang and Zhang 2002). The CAC process shall deny any request in case of insufficient communication resources. In proposed system model, a virtual CAC works at the application layer to admit new flows in accordance with the available set of resources (Ahmad, Qadir and Akbar 2008). The encapsulation and tunneling process is not highlighted in this figure to focus only at the multi-server scheduling and its processing components.

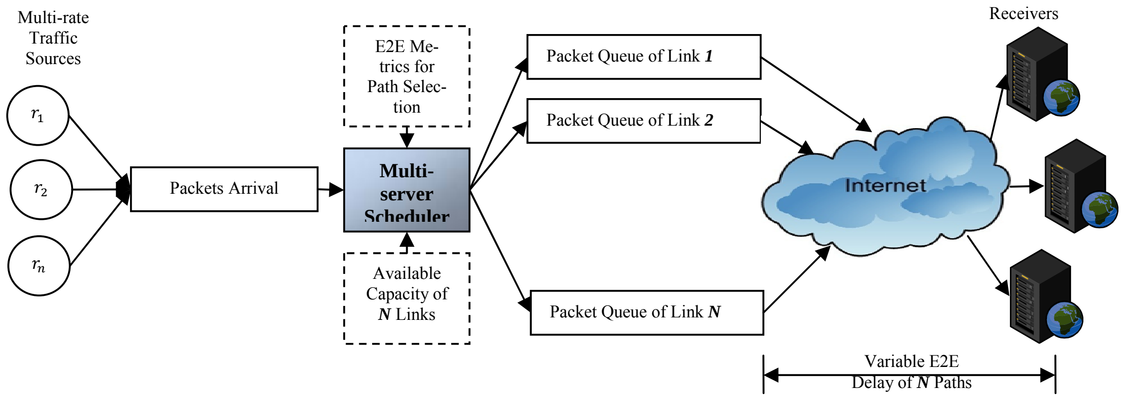

Figure 2-1: A basic multi-server scheduling model for multi-mode mobile devices (MMDs)

The operation of multiple tunnels between the MMD and HA are described in Figure 2.2. In this figure, the E2E paths are presented as a scheduling server only view, hiding all other processing complexity. The MMD and HA are shown here as actively scalable, multi-server (ASM) schedulers. In this model, each tunnel comprises of a finite numbers of servers (schedulers), deployed at the routers on the E2E path of the tunnel, hence each server represents a router on the path. The queuing delay experienced by each packet at the routers is upper-bounded by the transmission schedule given by the proposed scheduler discussed in chapter 3. The guaranteed service scheduling over a single server is performed by per flow state maintenance at each server, as specified in latency-rate ($\mathcal{LR}$) service paradigm (Stiliadis and D.Varma 1998). The $\mathcal{LR}$ servers quantify service guarantees in terms of server latency and flow-rate. Alternatively, guaranteed rate (GR) servers quantify their service in terms of flow-rate and transmission rate parameters (Goyal 1996). The GR server also upper-bound maximum delay that a packet may experience before it is transmitted. It has been shown (Y. Jiang 1998) that both $\mathcal{LR}$ and GR servers are equivalent. The proposed (ASM) service model has some salient advantages of service guarantees during mobility, adaptability to link layer (layer-2) changes and multi-server based concurrent transmission of multiple packets that enables much shorter server busy time and reduced queuing delays.

The proposed ASM scheduler comprises of N, GR servers, where each GR server represents a wireless link of the MMD, i.e. $ASM_N^U = \{GR_1^0, GR_2^0, .., GR_N^0\}$. The symbol U indicates up-link direction and is replaced by D in the downlink direction. Each of the GR servers in ASM leads a path comprising of GR servers, connected in series, as shown in the Figure 2.2. The uplink direction specifies MMD to HA traffic, where as downlink direction is specifies traffic from HA to MMD. There is also a provision of direct parallel path communication between the sender and receiver. The QoS guarantees may be further ensured on static path during mobility by using *DiffServ* (Blake 1998) path between the mobility agent and the corresponding node (CN[3]). The CN is also represented as a *GR* server to enable it as QoS aware sender. The scope of the thesis, shall however, be restricted to schedule the multi-class traffic over multiple tunnels between MMD and HA. The state of the system is maintained at the tunnel ends by synchronizing the aggregated capacity(C_A^S), and list of active flows as a function of there committed rates $\mathcal{F}(r_i)$, where $\mathcal{F}(r_i) = \bigcup_{i=1}^{all} f_i(r_i)$.

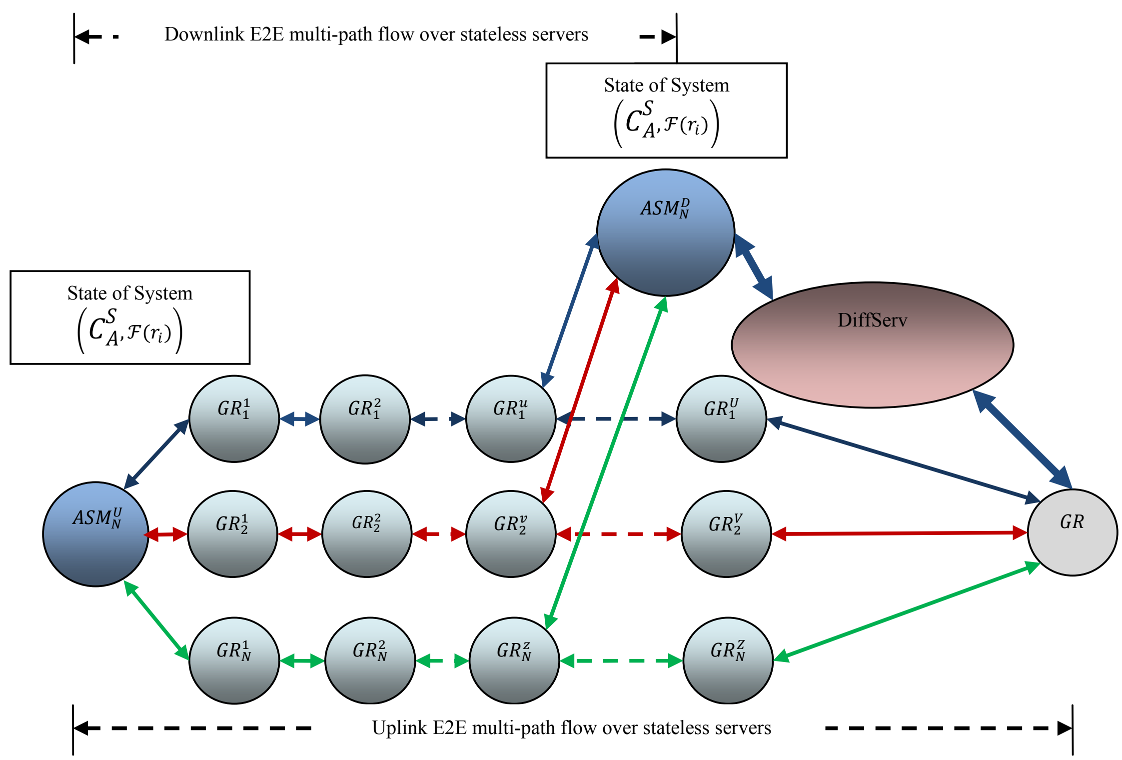

Figure 2-2: System model of multi-server scheduling during network mobility

2.3. Symbols and Notations

The symbols and notations used in this thesis are summarized in Table 2-1. The symbols and notations given here are generic in general and are redefined where ever scope

[3] The corresponding node is a standard terminology used in MIP to represent a node that has established a session with a mobile node. The corresponding node may also be mobile, but in our scope it is a static node with out mobility support.

changes. In case of any deviation from the description in the table, the derived terms are specified accordingly at the point of their reference. These notations are used in both deterministic and stochastic model given in chapter 3 & 4 and are also applicable in chapter 5 where different scheduling schemes are proposed.

Table 2-1: List of notations and symbols

$x_f(t)$	The cumulative traffic generated by flow f during time interval t
$b_{f,j}^{in}(t)$	The input traffic envelop for source $x_f(t)$ during time t at node j[4]
$b_{f,j}^{out}(t)$	The output traffic envelop for source $x_f(t)$ during time t at node j
$S_{f,j}(t)$	Service envelop of flow f at node j
F	Number of flows passing through any server/node
$\lambda_f(s^{-1})$	Transition rate of flow f from OFF state to ON state, in a two state Markov modulated traffic source
$\mu_f\ (s^{-1})$	Transition rate of flow f from ON state to OFF state, in a two state Markov modulated traffic source
C_A^S	Aggregated capacity of ASM, measured in bytes
B_A^S	Aggregated backlog of ASM, measured in bytes
$ASM(p_f^k)$	Deadline for packet p_f^k for server $s \in S$ belonging to ASM
$a^{ASM}(p_f^k)$	Arrival time of packet p_f^k of flow f at ASM server
l_f^k	Length of packet k of flow f
r_f	Data-rate of flow f
l_s^{max}	Length of Maximum length packet served at $s \in S$
θ_S	Latency of server S
π_j	Propagation delay between nodes j & $j+1$
C_s	Capacity of server $s \in S$
$\delta_f^{k,k+1}$	Delay variation of packet k, and $k+1$of flow f
φ_f^k	The upper bound for wait of OOS arrival of packet k of flow f
ϑ_f^k	Buffer occupancy upper bound due to late arrival of packet k of flow f
$\Upsilon_{d_{f,E2E}^K}(t)$	Distribution of E2E delay
$\Upsilon_{\vartheta_f^k}(t)$	Distribution of buffer occupancy
$\Upsilon_{\varphi_f^k}(t)$	Distribution of OOS arrival
$\mathcal{H}_k^S(t, t+\tau)$	System Busy Time as define in the Section 2-1.

[4] In case of general description of node we skip the second subscript j to be applicable for all servers of similar configurations

Chapter 3

DETERMINISTIC MULTI-SERVER SCHEDULING MODEL

In this chapter, we propose a novel analytical model of ASM scheduling of mobile network traffic, based on deterministic quantification of service guarantees over the multiple interfaces of an MMD. The purpose of this model is to quantify the impact of asymmetric path characteristics on multi-path flow behavior. This model provides some essential quantification of certain key parameters like E2E delay, OOS reception, and buffer occupancy that are important for performance evaluation and system design of any mobile multipath traffic flow scheme. Since we intend to develop a suitable scheduling approach under HWN environment, a basic scheduling model is devised to assure transmission of backlogged traffic within its data rate and delay constraints. This scheduling model has been kept open with respect to scheduling approaches, such as round-robin (RR), or any other fair queuing scheduling approach with the intention that the scheduling approach (with respect to link selection) shall be adopted in accordance with the capacity, E2E delay estimates and availability of individual links. It is assumed that the transmission deadline assigned for each packet is within the link resource constraints and the scheduler shall be able accommodate all backlogged traffic within transmission deadlines. In this chapter, we have answered first two major questions of the thesis hypothesis through analytical model and have interlinked some key parameters of multi-path traversal, like server delay, E2E delay, E2E delay variation, OOS arrival process, Buffer occupancy extent and packet drop to maximize performance gains in HWN environment.

3.1. Guaranteed Rate Single-server Scheduling

Virtual Clock and its numerous variants provide self-clocking mechanism to ensure bounded delays at the servers (L. Zhang 1990). Guaranteed rate clock (GRC) is one such scheme for per-flow guaranteed service based scheduling algorithms and is defined in (Goyal 1996) as:

$$GRC^j\left(p_f^1\right) = a^j\left(p_f^1\right) + \frac{l_f^1}{r_f}, \qquad \textit{(Eq 3-1)}$$

$$GRC^j(p_f^k) = max\left(a^j(p_f^k), GRC^j(p_f^{k-1})\right) + \frac{l_f^k}{r_f} \quad for\ k > 1, \qquad \textit{(Eq 3-2)}$$

where $GRC^j(p_f^k)$ represent the deadline for start of transmission of k^{th} packet of flow f at server j. The GRC service model guarantees that a packet p_f^k shall be transmitted by server j, no later than $GRC^j(p_f^k) + \frac{l_j^{max}}{C_j}$, provided server capacity is not exhausted (Goyal 1996). These guarantees are quite useful under the assumption of consistent channel availability and strictly constant-bit-rate (CBR) traffic source, but may not be effective in situations where channel availability is constraint by mobility and mutable link conditions (Ahmad & Qadir, 2009). Further in burst arrival cases, it adds predetermined latencies for each packet of burst sub-stream, irrespective of channel availability or otherwise that not only increases inter-arrival delay of the sub-stream packets but also under-utilizes link resource, in case capacity was available. We will show how this situation can be effectively rectified using our proposed ASM scheduling. The delay bound of packet k at j^{th} GRC server is evaluated in (Goyal 1996) as,

$$GRC^j(p_f^k) \leq GRC^{j-1}(p_f^k) + max_{i\in[1..k]}\frac{l_f^i}{r_i} + \pi_{j-1} + \frac{l_{j-1}^{max}}{C_{j-1}}, \qquad \textit{(Eq 3-3)}$$

where $max_{i\in[1..k]}\frac{l_f^i}{r_f^i}$, be the maximum $\frac{l}{r}$ ratio up to the packet number k, and π_{j-1} is the propagation delay between nodes $j-1$ and j and it represents the largest value of $\frac{l}{r}$, observed so far in the list of inter-arrived packets. It is noticeable that all the values on right hand side of inequality (3-3) are known at server $j-1$. Therefore, $GRC^j(p_f^k)$ given in (3-3) may be pre-encoded for the next server within packet header at each server, to release succeeding server on the path from state maintenance overhead between the MMR and HA. In such cases, the right hand side of the (3-3) is used as the upper bound of the GRC scheduler at the server j.

3.2. Scalable Guaranteed Rate Multi-server Scheduling

Let there be an ASM server $S = \{s_1, s_2, \dots, s_N\}$, comprising of N distinct servers, and each of the servers s_i belonging to S is a GR server. This implies that $s_i = GR_i^0$, *w.r.t.* Figure 2.2 of Chapter 2. For the sake of simplicity, we will use notation S for ASM server, s_i for any of its elements and ASM for the deadline of transmission of a specific packet.

The ASM server also assumes a set of GR servers in each of its forward paths starting from s_i. Each path may comprise of distinct numbers of GR servers. The ASM server virtually sees the available aggregated capacity as $C_A^S = \sum_{i=1}^{M} C_{s_i} \geq C_{s_i}, \forall\ s_i \in S,$ where $M \leq N$represents number of active servers and $C_{s_i}$is the capacity of i^{th} server i.e. s_i (Ahmad, Akbar & Qadir, 2008). This eventually means that the backlogged traffic can be served on multiple servers. The ASM server schedules a packet according to the following definition.

$$ASM(p_f^1) = a^{ASM}(p_f^1) + \left(\frac{l_f^1}{r_f^1}\right)\left(\frac{B_A^S}{C_A^S}\right), \quad \textit{(Eq 3-4)}$$

$$ASM(p_f^k) = max\left(a^{ASM}(p_f^k), ASM(p_f^{k-1})\right) + \left(\frac{l_f^k}{r_f^k}\right)\left(\frac{B_A^S}{C_A^S}\right), for\ k > 1, \quad \textit{(Eq 3-5)}$$

where $ASM(p_f^k)$ is the deadline for packet p_f^k over any of the server $s_i \in S$, and B_A is the total backlogged bytes in the system. It is noticeable in (Eq 3-4) and (Eq 3-5) that the deadlines calculated for each packet is scaled by a factor of $\left(\frac{B_A^S}{C_A^S}\right)$, that helps in providing a sharper deadline for each packet, provided capacity is more than total number of backlogged bytes. This ensures that the deadlines shall be adaptable with changes in available capacity in the system. The scaling factor is also instrumental in accommodating burst sub-stream on access bandwidth available during any arbitrary time period. The scaling factor accommodates the availability of $M \leq N$ servers and any increase in the number of available servers shall automatically result in accompanying decrease in the deadline of each scheduled packet. Similarly; in case connectivity of one or more servers is lost, deadline for each packet increases accordingly to gracefully scale impact of reduced capacity. The following two cases describe the scalability of the proposed analytical model during mobility events.

Case 1: $C_A^S \geq B_A^S$

In this case, since the available capacity is more than the backlogged traffic, all the backlogged traffic shall be served as before the mobility event that resulted in reduced capacity in the system. The minor difference that may obviously result is a slight increase in the ratio$\left(\frac{B_A^S}{C_A^S}\right)$**.** The corresponding impact of such scenario is slightly escalated deadline for each packet during current backlogged period. Hence the system remains within stable state after the disconnection of a link due to mobility event.

***Case 2:* $C_A^S < B_A^S$**

In this case, since capacity is less than the backlogged traffic, the ratio$\left(\frac{B_A^S}{C_A^S}\right)$**,** goes greater than 1, causing an increase in the deadline of the packet beyond the desired service rate. The increase in delay reduces service guarantees to certain degree; depending on how large is the $\left(\frac{B_A^S}{C_A^S}\right)$ratio. However, the service continues at a reduced rate till the restoration of the lost connectivity. Although such scenario may not provide desired service guarantees but it still ensures service continuity. The burst arrival during this time may face more serious impair, as it may get longer deadlines and may have a higher probability of being dropped due to capacity depletion.

The concurrent transmission of backlogged traffic through multi-server scheduling reduces the system busy time considerably. Figure 3.1 demonstrates the arrival, and departure processes in case of single server and multi-server scheduling modes. Figure 3.1(a) depicts the arrival process of three flows with different data-rates. The packets are scheduled as per definition given in (3-4) and (3-5). Figure 3.1(b) shows single server scheduling with transmission order in a sequential manner. Figure 3.1(c) shows the transmission of the same three flows on three different servers. It may be noticed that the system busy time in single server mode is much longer as compared to multi-server mode. It is also important to note that the scheduling of a specific packet on any available server is based on some E2E statistics such as E2E delay (discussed in Chapter 5), its mean and variance which are not essentially round-robin (RR) allocations. The Figure 3.1(c) tries to emulate this aspect of scheduling.

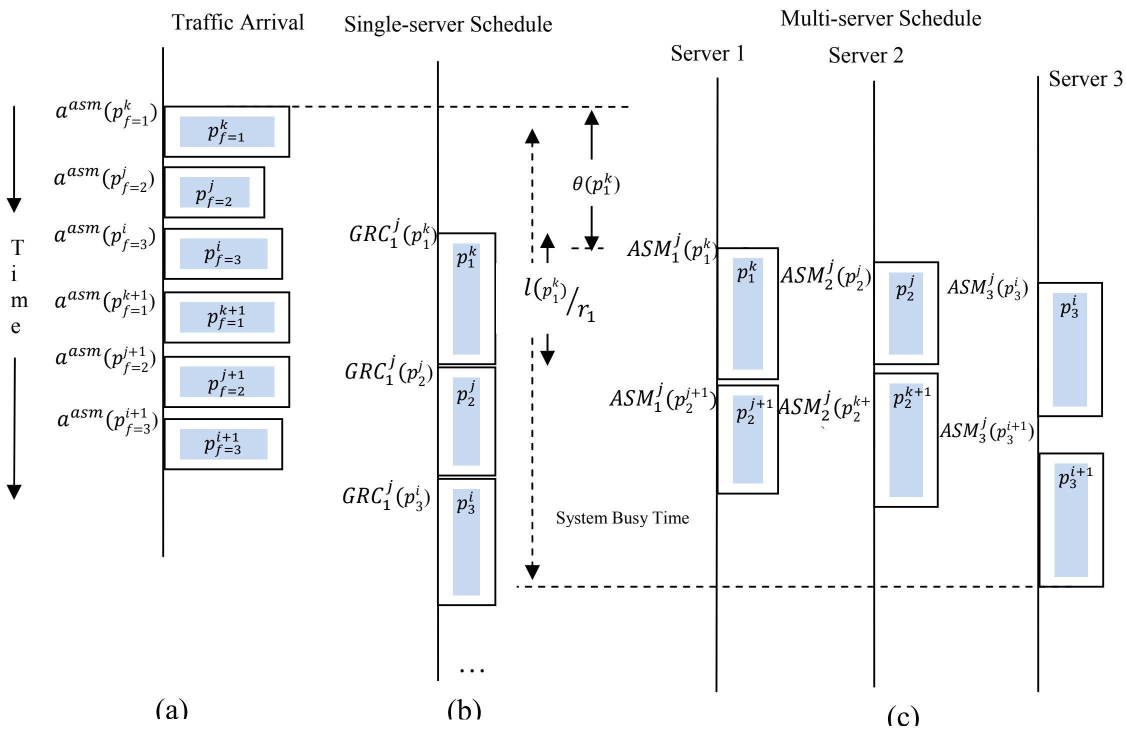

Figure 3-1: Simplified arrival and departure mechanism in single and multi-server scheduling modes

3.3. The Analytical Modeling

In this sub-section we present analytical models to derive latency of the ASM server, its E2E delay bounds, delay variation bounds, buffer occupancy and OOS arrival bounds.

Lemma 3.1: The latency of ASM server is upper-bounded by $\theta_S(p_f^k) \leq ASM(p_f^k) + \frac{L_S^{max}}{C_S}$, where $\theta_S(p_f^k)$ represents the latency faced by packet p_f^k, at ASM server, before it is completely transmitted.

Proof: In the proof of above stated lemma, we use the definition of ASM as given in (3-4) and (3-5). Since $ASM(p_f^k)$ is the deadline for the packet transmission at any of the er $s_i \in S$, the maximum delay before server s_i starts transmitting the packet is bounded by $ASM(p_f^k)$. Since the maximum transmission time needed would not be more than transmitting maximum sized packet over server $s_i \in S$, the latency directly depends on the capacity of server it is assigned for transmission. For simplicity, we use more general notation s for s_i to represent the transmission latency of any of the server as $\frac{L_S^{max}}{C_S}$, hence the delay of the ASM is upper-bounded as given in (Eq 3-6).

$$\theta_S(p_f^k) \leq ASM(p_f^k) + \frac{L_S^{max}}{C_s}, \qquad \textit{(Eq 3-6)}$$

This also completes the proof of lemma. ■

This lemma 3.1 provides a useful performance metric of the proposed scheduling scheme. The delay bound of the ASM is sharper than any of GRC scheduling schemes. Firstly, it is using the full capacity of the server $s \in S$ for the transmission delay, and is guaranteed to be holding the inequality $\frac{l_S^{max}}{C_s} \leq \frac{l_S^{max}}{r_f}$. Secondly; in case of multiple packets waiting for service, they are scheduled in parallel on any of the available servers through ASM. It ensures that the service time will be much sharper for the subsequent packets waiting for service. Analytically, the same is justified through allocation of full capacity of server which is numerically faster than the rate r_f of the flow. This shows that the delay of ASM is lesser than any of the guaranteed rate server.

Lemma 3.2: The k^{th} System Busy period of ASM server, during a backlogged interval $(t, t+\tau]$, is given by,

$$\mathcal{H}_k^S(t, t+\tau) \leq \left(Max_{i\in[1..J]} ASM\left(p_k^i\right) - Min_{i\in[1..J]} a^{ASM}\left(p_k^i\right) + Max_{s\in[1..N]} \frac{L_S^{max}}{C_s}\right),$$

where $\mathcal{H}_k^S(t, t+\tau)$, is k^{th}system busy period of backlogged traffic during the interval $(t, t+\tau]$.

Proof: Let there be J backlogged packets in the k^{th} system busy period i.e., the maximum time after the service has started for a packet before server becomes idle (Stiliadis and D.Varma 1998). In case of ASM, service is spread over multiple parallel channels of available servers. These channels though are not symmetric but predominantly overlap in time and the frame duration $F^k = \sum(F_s^j - F_e^j), for\ all\ j\ in\ T_j^k$, is mostly describing a single largest slot time in T_j^k. (Note: F_s^j and F_e^j are the starting and ending time of each slot in the virtual frame T_j^k).

In case of overlapping channels all the backlogged traffic will be served within F^k, and the last served packed shall be the one with farthest departure schedule. Hence the service of the last packet shall start no later than $\left(Max_{i\in[1..J]} ASM\left(p_k^i\right) - Min_{i\in[1..J]} a^{ASM}\left(p_k^i\right)\right)$. The second parameter for service completion is the length of the largest packet and the capacity of the server over which it is scheduled. This can be given by $Max_{s\in[1..N]}\frac{L_s^{max}}{C_s}$. Therefore, the system busy period can be given as follows in (Eq 3-7),

$$\mathcal{H}_k^S(t,t+\tau) \le \left(Max_{i\in[1..J]}ASM\left(p_k^i\right) - Min_{i\in[1..J]}a^{ASM}\left(p_k^i\right) + Max_{s\in[1..N]}\frac{L_s^{max}}{C_s}\right), \quad \text{(Eq 3-7)}$$

Hence also completes the proof. ■

Lemma 3.3: The service rate, $\mathcal{R}_k^S(t,t+\tau)$ of ASM server, during the interval $(t,t+\tau]$ is lower-bounded by $\mathcal{R}_k^S(t,t+\tau) \ge \frac{Min\left(\sum_{i=1}^{J}\rho_i,\sum_{i=1}^{M\le N}c_{s_i}\right)}{\mathcal{H}_k^S(t,t+\tau)}$,

Proof: The service rate of ASM is primarily constraint by the peak traffic arrival and the corresponding busy period of the server. The service provided by the server during any time interval may not exceed C_A^S, the total capacity available in the system and can be given by (Eq 3-8)

$$\mathcal{R}_k^S \ge Min(B_A^S, C_A^S), \quad \text{(Eq 3.8)}$$

Since backlog traffic is constrained by the token bucket regulator (TBR), and C_A^S is the aggregate capacity of all links, (Eq 3-8) can also be represented as (Eq 3-9)

$$\mathcal{R}_k^S \ge Min\left(\sum_{i=1}^{J}\rho_i, \sum_{i=1}^{M\le N}c_{s_i}\right), \quad \text{(Eq 3-9)}$$

where (ρ_i,σ_i) describe token-bucket conformant arrival with an average rate of ρ_i and maximum burst size of σ_i, during time interval $(t,t+\tau]$. Lemma 3.2 provides the system busy time during any backlogged interval. Therefore, the service rate during the system busy period can be given using (Eq 3-7) and (Eq 3-9) as follows.

$$\mathcal{R}_k^S(t,t+\tau) \ge \frac{Min\left(\sum_{i=1}^{J}\rho_i,\sum_{i=1}^{M\le N}c_{s_i}\right)}{\mathcal{H}_k^S(t,t+\tau)}, \quad \text{(Eq 3-10)}$$

that also completes the proof of Lemma. ■

It is noticeable in Lemma 3.3 that the service rate of ASM scheduler depends on two factors; namely, the duration of system busy time and aggregated capacity available to the server. Since system busy time increases as the backlogged traffic increases, the performance of the proposed algorithm relies on maximizing the aggregated capacity. Further, the system busy time in concurrent transmission of multiple packets is much shorter, the proposed ASM model has higher server rate.

The ASM server schedules a packet on any of available server after which the path may comprise of a maximum of Q servers (other paths may have lesser server count). Using the methodology given in (J. K. Vin 2001) and (Ion Stoica 1999), we evaluate the E2E delay of a packet scheduled through ASM over a path, comprising of GRC servers in theorem 3.1.

Theorem 3.1: The E2E delay $D^L(p_f^k)$ of the packet p_f^k, scheduled through ASM over the set of GRC servers is upper-bounded by

$$D^L(p_f^k) \leq \left(ASM(p_f^k) + \frac{L_s^{max}}{C_s}\right) - a^{ASM}(p_f^k) + (L-1) \max_{i\in[1..L]} \frac{l_f^i}{r_i} + \sum_{i=1}^{(L-1)} \left(\frac{l_i^{max}}{r_f} + \pi_i\right),$$

where $L \leq Q$ is the number of servers in the E2E path and l_i^{max} is the delay of the largest packet served by server i.

Proof: A simple E2E delay model for packets scheduled through ASM can be based on three parameters namely, arrival of the packet, service time at the last server and the segmented E2E propagation delay after transmission at L^{th} server (last link), in the path (Goyal 1996). This can be given (Eq 3-11) as,

$$D^L(p_f^k) \leq GRC^L(p_f^k) - a^{ASM}(p_f^k) + \alpha^L, \qquad \textit{(Eq 3-11)}$$

where $\alpha^L = \frac{l_L^{max}}{rf} + \pi_L$ and $a^{ASM}(p_f^k)$ is the arrival time of packet p_f^k at ASM.

The arrival time of the packet p_f^k, at any succeeding server in the E2E path can be given in-terms of its transmission schedule at the preceding along-with transmission and propagation delays between the two servers. The same can be given by (Eq 3-12) as follows.

$$a^L(p_f^k) \cong GRC^{L-1}(p_f^k) + \frac{l_{L-1}^{max}}{r_f} + \pi_{L-1}, \qquad \textit{(Eq 3-12)}$$

The back-substitution procedure using (Eq 3-12) can be further extended to trace it back into first server in the path, i.e., ASM server. Hence, the $GRC^L(p_f^k)$ can be given as in (Eq 3-13).

$$GRC^L(p_f^k) \leq \left(ASM(p_f^k) + \frac{L_S^{max}}{C_S}\right) + + \sum_{i=1}^{L-1} \left(\frac{l_i^{max}}{r_f} + \pi_i\right), \qquad \textit{(Eq 3-13)}$$

Considering the non-preemptive scheduling, the transmission on a scheduled packet may not start before the completion of current packet. Therefore, we can add a transmission delay of maximum length packet to account for the possible delay before transmission actually starts and get (Eq 3-14).

$$GRC^L(p_f^k) \leq \left(ASM(p_f^k) + \frac{L_S^{max}}{C_S}\right) + (L-1)\, max_{n\in[1..L]} \frac{l_f^n}{r_f} + \sum_{i=1}^{L-1} \left(\frac{l_i^{max}}{r_f} + \pi_i\right), \qquad \textit{(Eq 3-14)}$$

By substitution of (Eq 3-14) in (Eq 3-12) we get,

$$D^L(p_f^k) \le \left(ASM(p_f^k) + \frac{L_s^{max}}{C_s}\right) - a^{ASM}(p_f^k) + (L-1) \max_{n\in[1..L]} \frac{l_f^n}{r_f} + \sum_{i=1}^{(L-1)} \left(\frac{l_i^{max}}{r^f} + \pi_i\right)$$

(Eq 3-15)

Since $D^L(p_f^k)$, is the last server in the path hence proof of theorem 3.1 completes. ■

The theorem 3.1 provides useful information about the E2E delay and possible sources of intra-path variation in delay. It can be noticed that the variable lengths of packets and possible one maximum packet transmission delays at each of the servers on the path are two main causes of variations in delay. The servers on the path, except the first one are assumed to be the GRC servers and need not maintain the state of the packet at each server for scheduling it on the E2E path (J. K. Vin 2001). Therefore, E2E delay bounds are guaranteed with stateless configuration. Theorem 3.1 also provides foundation for quantifying inter-path E2E delay variations that contribute significantly in the successful multi-path flow management.

Theorem 3.2: The E2E delay variations of two successive packets p_f^k and p_f^{k+1}, belonging to same flow f and scheduled on two different paths through ASM server is upper-bounded by,

$$\delta_f^{k+1,k} = D^K(\,p_f^k) - D^L(\,p_f^{k+1})$$
$$\le a^{ASM}(p_f^{k+1}) - a^{ASM}(p_f^k) + (K-L) \max_{n\in[L..K]} \frac{l_f^n}{r_f} + \sum_{i=L}^{K} \left(\frac{l_i^{max}}{r_f} + \pi_i\right),$$

Proof: Let we assume that two consecutive packets p_f^k and $p_f^{k+1}$of flow f are scheduled by ASM on two different servers $s_s \& s_t \in S, \exists\, s\,, t \le M \le N$, leading two different E2E paths to the same destination. The two servers map on two distinct paths, characterized by different number of intermediate servers, K and L; respectively. We assume that $K > L$ and the two paths are moderately congested. The E2E delay of one path is bounded through theorem 3.1, and we can directly achieve the following delays of two packets, as given in (Eq 3-16) and (Eq 3-17).

$$D^K(p_f^k) \leq \left(ASM(p_f^k) + \frac{L_{s_s}^{max}}{C_{s_s}}\right) - a^{ASM}(p_f^k) + (K-1) \max_{n\in[1..K]} \frac{l_f^n}{r_f} + \sum_{i=1}^{(K-1)} \left(\frac{l_i^{max}}{r_f} + \pi_i\right)$$

(Eq 3-16)

$$D^L(p_f^{k+1}) \leq \left(ASM(p_f^{k+1}) + \frac{L_{s_t}^{max}}{C_{s_t}}\right) - a^{ASM}(p_f^{k+1}) + (L-1)\, max_{n\in[1..L]} \frac{l_f^n}{r_f} + \sum_{i=1}^{(L-1)} \left(\frac{l_i^{max}}{r_f} + \pi_i\right), \quad \textit{(Eq 3-17)}$$

Since the two servers $s_s$and s_t are asymmetric in their characteristics, their transmission and propagation delays may be different. We argue that since flow f has a guaranteed service rate and the entire set of servers in the two paths are GRC servers, the overall delay variations of all the servers shall restrict within a tight range-bound. It is also noticeable that the latency of the ASM server shall be same, as depicted by Lemma 3.1. On the basis of this, we take two terms of (Eq 3-16) and (Eq 3-17) to be approximately equal as shown in (Eq 3-18),

$$\left(ASM(p_f^K) + \frac{L_{s_s}^{max}}{C_{s_s}}\right) \cong \left(ASM(p_f^{k+1}) + \frac{L_{s_t}^{max}}{C_{s_t}}\right), \quad \textit{(Eq 3-18)}$$

Using (Eq 3-18), the difference of (Eq 3-16) & (Eq 3-17), yields the following,

$$D^K(p_f^k) - D^L(p_f^{k+1}) \leq a^{ASM}(p_f^k) - a^{ASM}(p_f^{k+1}) + (K-L)\, max_{n\in[L..K]} \frac{l_f^n}{r_f} + \sum_{i=L}^{K} \left(\frac{l_i^{max}}{r_f} + \pi_i\right), \quad \textit{(Eq 3-19)}$$

The (Eq 3-19) provides a tight bound on the E2E delay variation of two consecutive packets. Using $\delta_f^{k,k+1} = D^K(p_f^k) - D^L(p_f^{k+1})$, to represent delay variation the proof of the theorem follows. ■

The (Eq 3-19) describes that the E2E delay variations of the two packets are primarily dependent on the difference of number of servers on the two paths and the length of packets. The length of packet may produce quite serious impairment in case transmission delay is larger than propagation delay. Theorem 3.2 provides basis for quantifying delay variations and accordingly; helps in developing a specific buffering strategy. Alternatively, E2E delay variation metric can also help in devising proactive strategies for controlling E2E delay variation at the sender side, provided sufficient time space is available for reordering the transmission schedule according to the different link characteristics.

Theorem 3.3: The deterministic bound for OOS arrival extent φ_f^k, of any two successive packets p_f^k and p_f^{k+1} in case of multi-path traversal of a single flow f is upper-bounded by

$$\varphi_f^k \leq a^{ASM}\left(p_f^k\right) + (K-L) \max_{n\in[L..K]} \frac{l_f^n}{r_f} + \sum_{i=L}^{K} \left(\frac{l_i^{max}}{r_f} + \pi_i\right),$$

Proof: The OOS arrival is mainly caused by the E2E delay variations of different path followed by the packets. The upper bound for the E2E delay variation has been quantified in Theorem 3.2. Let's assume an arbitrary packet p_f^k that traverses $(K-L)$ servers in addition to the L server of its successor. The OOS reception definitely means late arrival of predecessor packet. Hence, the OOS receiving of packet p_f^k can be given by the difference of maximum E2E delay variation of p_f^k and arrival of packet p_f^{k+1} at the first server. This is shown in (Eq 3-20) as follows,

$$\varphi_f^k \leq +\delta_f^{k,k+1} - a^{ASM}\left(p_f^{k+1}\right), \qquad \textit{(Eq 3-20)}$$

Substituting the value of $\delta_f^{k,k+1}$ in (Eq 3-20) from (Eq 3-19), we find the maximum extent of OOS reception of a predecessor packet as given in (Eq 3-21).

$$\varphi_f^k \leq a^{ASM}\left(p_f^k\right) + (K-L)\, max_{n\in[L..K]} \frac{l_f^n}{r_f} + \sum_{i=L}^{K} \left(\frac{l_i^{max}}{r_f} + \pi_i\right), \qquad \textit{(Eq 3-21)}$$

Hence the proof of theorem follows. ■

It is noticeable that E2E delay variation is directly dependent on the difference in count of number of servers in each path. There are two major components of the OOS arrival; namely, the arrival of the predecessor packet, and maximum delay variation between the two successive packets. Hence, the OOS arrival extent given in (3-21) provides a tight bound on OOS arrival of any packet.

Corollary 3.1: The OOS extent φ_f^k is a suitable metric of maximum allowed delay before a packet p_f^k may be considered as dropped to reduce complexity of buffer management. The consecutive successors packets, already received and waiting for in-order forwarding may immediately be forwarded to release the occupied buffers. In such cases φ_f^k may be treated as timeout value for the ending of possible wait of an OOS arrival.

The Theorem 3.3 and its corollary 3.1, provide a useful tight-bound for the OOS arrival and an upper-bound for the wait of such arrivals. The timeout φ_f^k can be very effective in buffer management that may be essential for the asymmetric multi-channel transmission of real-time traffic. While serving real-time traffic on such channels, the bounded time arrival is also convoluted with in-order arrival and buffers are essential to achieve this added constraint. The following theorems analytically range-bound the occupancy of such buffers and minimizes it to achieve scalable and cost effective buffering model.

Theorem 3.4: The extent of Buffer Occupancy ϑ_f^k, caused by the OOS reception of packet p_f^k is upper-bounded by,

$$\vartheta_f^k \leq \frac{\sum_{i=k+1}^{j} l_i^{max}}{a^{ASM}\left(p_f^k\right) + (K-L)\max\limits_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{K}\left(\frac{l_i^{max}}{r_f} + \pi_i\right)},$$

Proof: Let there are j packets of flow f, arrived just before the occurrence of OOS timeout event for packet p_f^k, *i.e.* φ_f^k. In such case, maximum possible buffer occupancy value for the stream started from packet p_f^{k+1} and ending at p_f^j. The number of packets arrived till the timeout event is $j-k$. Therefore, the buffers occupied can be given by the number of bytes occupied by $j-k$ packets and can be given by the following,

$$\vartheta_f^k \leq \frac{\sum_{i=k+1}^{j} l_i^{max}}{\varphi_f^k}, \qquad \textit{(Eq 3-22)}$$

In case of in-order arrival, packets are removed from the buffers and result in release of occupied buffers. The occurrence of timeout interval indicates a packet drop event and the next in-order sub-stream is forwarded. Therefore, the length of buffer occupied divided by the maximum time over which such occupancy persist can provide the upper-bound of buffer occupancy. Hence substitution of φ_f^k from (Eq 3-21) in (Eq 3-22), we get (Eq 3-23) as follows,

$$\vartheta_f^k \leq \frac{\sum_{i=k+1}^{j} l_i^{max}}{a^{ASM}\left(p_f^k\right) + (K-L)\max_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{(K)}\left(\frac{l_i^{max}}{r_f} + \pi_i\right)}, \qquad (3\text{-}23)$$

Hence the proof of theorem completes. ■

Theorem 3.5: The probability of dropping of packet p_f^k, scheduled through ASM server, due to buffer occupancy is upper bounded by

$$P[\psi_f^k] \leq P\left[\varphi_f^k \geq a^{ASM}(p_f^k) + (K-L)\max_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{K}\left(\frac{l_i^{max}}{r_f} + \pi_i\right)\right]$$

where $P[\psi_f^k]$ is the probability of dropping packet p_f^j.

Proof: Though packet drop event may be caused by various events; such as, link capacity depletion or handover event, but we are mainly concerned by the packet drop due to the buffer occupancy at the receiver due to OOS arrival of any packet. The packet drop event, in case of an infinite buffer assumption turns out to be zero. But in any real scenario, this assumption may not hold. Hence, we assume that the receiver is equipped with buffers, equivalent to the holding packets till the timeout duration i.e. φ_f^kexceeds. The same is given in (Eq 3-24) as follows,

$$\begin{cases} P[\psi_f^k] = 0, if\ \varphi_f^k \leq a^{ASM}(p_f^k) + (K-L)\,max_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{K}\left(\frac{l_i^{max}}{r_f} + \pi_i\right) \\ P[\psi_f^k] \geq 0 \leq 1, if\ \varphi_f^k \geq a^{ASM}(p_f^k) + (K-L)\,max_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{K}\left(\frac{l_i^{max}}{r_f} + \pi_i\right) \end{cases}, \quad \textit{(Eq 3-24)}$$

The same can be expressed in more compact form as in (Eq 3-25)

$$P[\psi_f^k] \leq P\left[\varphi_f^k \geq a^{ASM}(p_f^k) + (K-L)\,max_{n\in[L..K]}\frac{l_f^n}{r_f} + \sum_{i=L}^{K}\left(\frac{l_i^{max}}{r_f} + \pi_i\right)\right], \quad \textit{(Eq 3-25)}$$

Hence the proof of theorem completes. ■

3.4. Source Traffic Specification

The proposed analytical model deeply links with the source traffic arrival process. Let $a^{ASM}(t, t+\tau)$, be the arrival process of flow f during the interval $(t, t+\tau]$, at er ASM. The arrival event is considered only at the completion of arrival of all bits of the packet at ASM server. Hence, it can be modeled through an on-off system; also known as Poisson process. In this scenario, the arrival process of any flow f, can be given as in (Eq 3-26),

$$a_f^{ASM}(t, t+\tau) = \{n\ | n > 0\ and\ ASM(p_f^{k-1}) \leq a^{ASM}(p_f^k)\}, \quad \textit{(Eq 3-26)}$$

Based on (Eq 3-26), let there be n packets arrived during the interval $(t, t+\tau]$, from the flow f. The total traffic arrived at ASM from flow f can be given by (Eq 3-27) as follows,

$$a_f^{ASM}(t, t+\tau) = \sum_{j=k}^{n+k} l_f^j, \qquad \textit{(Eq 3-27)}$$

Since traffic source is assumed to be regulated through TBR (Karandikar 2003), which regulates the arrival with parameters (σ_f, r_f), and can be given by (Eq 3-28),

$$a_f^{ASM}(t, t+\tau) \leq \sigma_f + r_f(t, t+\tau), \qquad \textit{(Eq 3-28)}$$

Therefore, the total arrival from flow f is strictly constraint by the following inequality,

$$\sum_{j=k}^{n+k} l_f^j \leq \sigma_f + r_f(t, t+\tau), \qquad \textit{(Eq 3-29)}$$

Using (Eq 3-26) and (Eq 3-29), we quantify the E2E delay of ASM server for TBR traffic source in the following theorem.

Theorem 3.6*:* A flow f conformant to a TBR with parameters (σ_f, r_f), scheduled through ASM with each of the server on the multiple paths for f is a GRC server, and then the E2E delay D_f^k of packet p_f^k, is given by

$$D^L(p_f^k) \leq \frac{L_s^{max}}{c_s} + \frac{\sigma_f + (K-1)\max\limits_{n\in[1..L]} l_f^n}{r_f} + \sum_{n=1}^{K-1}\left(\frac{l_i^{max}}{r^f} + \pi_i\right),$$

where K is the number of servers on the path of flow f.

Proof: Let $k \leq j$ be the largest integer belonging to set S_f^1. Clearly, such a k must exist. Since flow f conforms to leaky bucket specification, we get,

$$a_f^{ASM}\left(a^{ASM}(p_f^k), a^{ASM}(p_f^j)\right) \leq \sigma_f + r_f\left(a^{ASM}(p_f^j) - a^{ASM}(p_f^k)\right), \qquad \textit{(Eq 3-30)}$$

From (Eq 3-26) we know that in case of more than one packet arrival during any given time period, the transmission of second packet cannot be started before the first one. Hence, (Eq 3-31) reduces to the following,

$$ASM(p_f^k) \leq \frac{\sigma_f}{r_f} + a^{ASM}(p_f^j), \qquad \textit{(Eq 3-31)}$$

The (Eq 3-31) can be further rearranged to get,

$$ASM(p_f^k) - a^{ASM}(p_f^j) \leq \frac{\sigma_f}{r_f}, \qquad \textit{(Eq 3-32)}$$

Substituting the above value in (Eq 3-15) of theorem 3.1, we get,

$$D^L(p_f^k) \leq \frac{L_s^{max}}{c_s} + \frac{\sigma_f + (K-1)\max_{n\in[1..L]} l_f^n}{r_f} + \sum_{n=1}^{K-1}\left(\frac{l_i^{max}}{r^f} + \pi_i\right), \qquad \textit{(Eq 3-33)}$$

This also completes the proof of theorem. ■

(Eq 3-33) describe a tight bound for the E2E delay that in turn ensures reduced E2E delay variations due to known transmission latencies, hop count difference, and propagation delays. The more predictable E2E delay behavior is a key contributor in providing service guarantees in presence of multiple competing flows. A reduced E2E delay variation (due to tighter E2E delay bound) ensures reduced OOS reception, reduced buffer occupancy and lesser packet drop. In the following section, we present some results of the proposed analytical model.

3.5. Results

The numerical results of proposed analytical model highlight some salient performance gain in terms of bounded E2E delays and related metrics. The foremost metric of interest is the E2E delay distribution which is given in Figure 3-2. The plot shows a tight range of E2E delay distribution, reducing the extent of E2E delay variation to about 60 ms. This indicates that a video source generating a packet every 10 ms mean with an exponential bounded burst (EBB) source may need about 6 to 10 buffers to completely ensure in-order reception within the allowed delay threshold. The plots for the different number of flows shows a slight drift towards the higher E2E delay mean, but the variance remain more or less same. It is noticeable that the distribution of delay is quite range bound majority of the packet reach the destination (tunnel end) in less than 150 ms time. It is also noticeable that E2E delays mean increases with the increase in number of competing flows. This result answers the fundamental question of the thesis regarding usefulness of CAG to ensure QoS guarantees during mobility. It also proves that the E2E delay is upper-bounded to ensure service guarantees.

Figure 3-3 shows the buffer occupancy due to OOS reception. The plot indicates that a great majority of packets face buffer occupancy of very low numbers in the range of 3 to 4 buffers. The buffer occupancy is reaching values of 6 to 8 for a very low number (less than 3% of total transmitted packets) of packet counts. Plots shows these values for 8 concurrent flows and the values are even lower with lesser completing flows. This plot also highlights the fact that cumulative buffer is more likely to produce higher utilization as the traffic sources deviate with the exponential distribution of traffic generation. The plot also validated the delay distribution curves of Figure 3-2.

Figure 3-4 plots the buffer occupancy behavior during burst traffic generation from 8 flows. It may be noticed that the behavior of the plot shows consistent performance with respect to low buffer occupancy. Only a few packets reach late enough that buffer occupancy level reaches values in double digit. It is noticeable that the buffer occupancy does not converge to a single number and some of the packets face buffer occupancy more than 10 buffers. Since the worst case scenario in a 10 ms mean traffic source under EBB distribution may generate packets back to back with time gap of 1-5 ms, the higher buffer occupancy is an expected result in case of traffic bursts. It is also important to recall that the probability of such fast traffic generation is very low and the higher buffer occupancy may be rarely faced.

Figure 3-5 shows a plot of packet drop at different percentages of OOS reception. The plot shows that with the increasing percentage of OOS reception, the packet drop rate also increase linearly. It is important characteristic of the proposed model that it does not exhibit exponential growth of the packet drop. This indicates that we can substantially increase service quality by ensuring OOS reception to lower percentages. This also shows that the OOS reception may be a direct source of packet drop. The plot also shows that the OOS reception under 10% has a significantly good packet drop rate. In case OOS reception is guaranteed to less than 10 %, the QoS guarantees can be easily achieved. Similarly, Figure 3-6 plots packet drop at different buffer occupancy levels. This plot shows that at higher buffer occupancy levels packet drop rate is also higher. The reason behind this behavior is rooted in the fact that at higher buffer occupancy, the timeout events of the delay upper bound is violated more frequently and consequently; it contributes to more packet drop events.

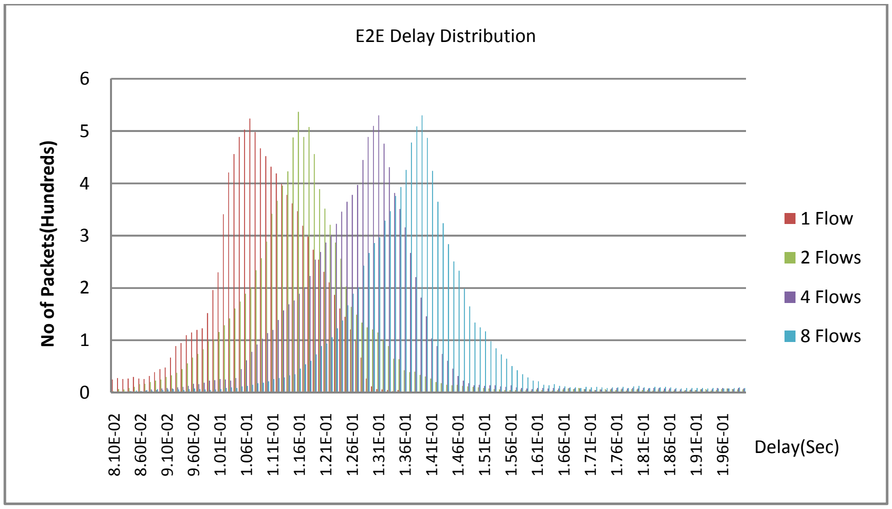

Figure 3-2: The E2E delay distribution for proposed ASM model

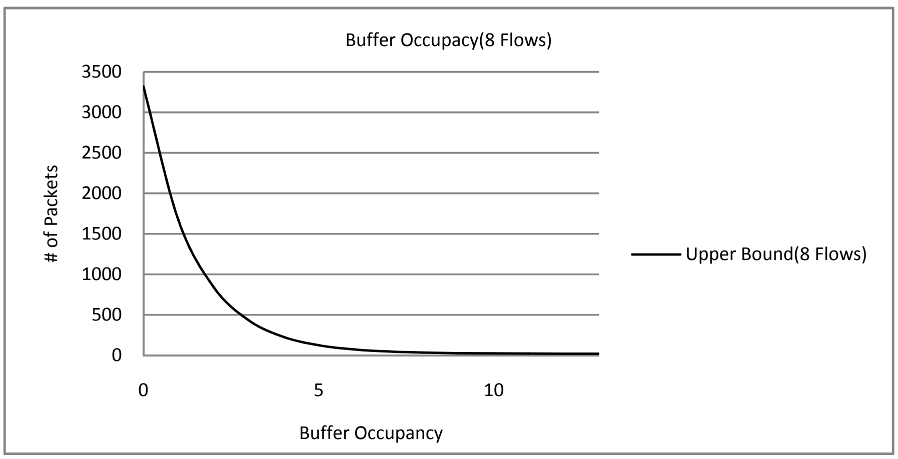

Figure 3-3: Buffer Occupancy of proposed ASM model

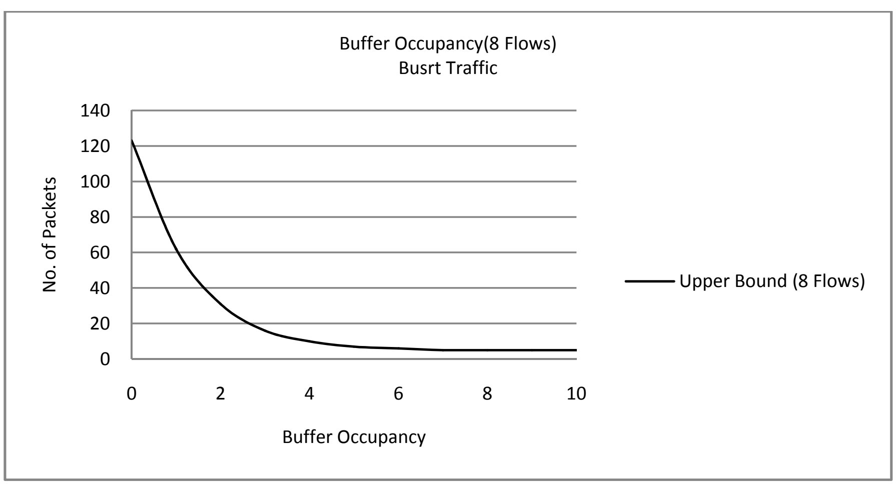

Figure 3-4: Buffer Occupancy during traffic bursts

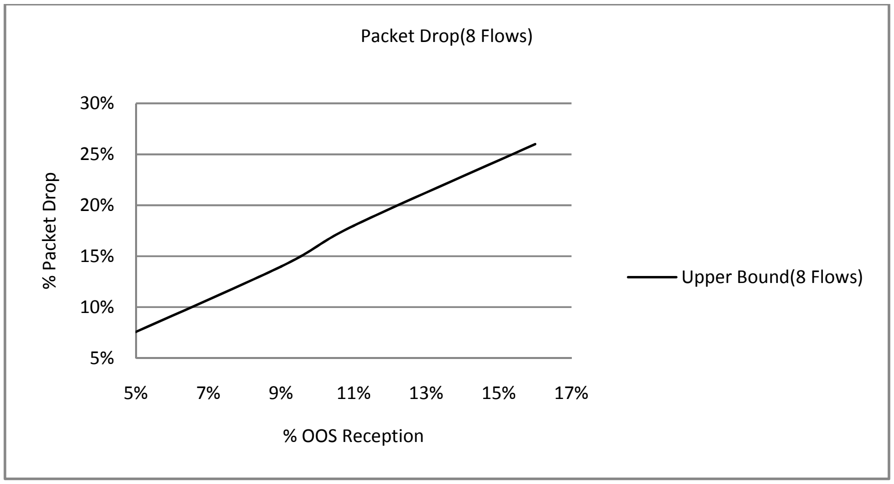

Figure 3-5: Packet drop at different percentage of OOS reception

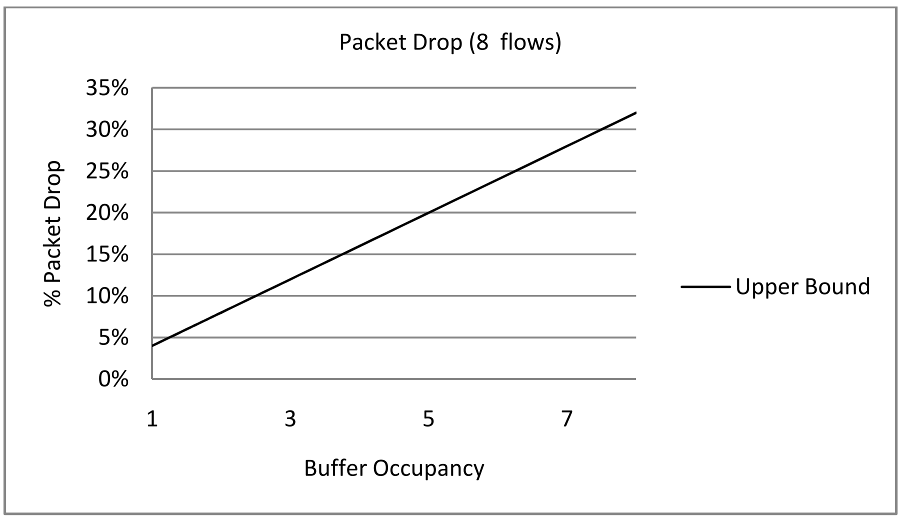

Figure 3-6: Packet drop at different levels of buffer occupancy

3.6. Summary

In this chapter we have derived a comprehensive deterministic analytical model of the proposed multi-server scheduling for the mobile sessions running over mutable wireless links. The model has been instrumental in identifying the challenges of the multi-path flow management and provides useful insight of the complexity of multi-server scheduling in the mutable wireless environment. The bounds achieved in this model are tight with respect to maximum possible variation in the concerned parameters. Since it is a deterministic model, based on the *GR* server philosophy, it has lesser possibility of large deviations from the given bounds, as the number of flows don't exceed beyond the capacity depletion of the system. The proposed model answers two main questions of thesis hypothesis. First of all we are able to achieve guaranteed QoS provision over multi-path transport in the presence of OOS arrival. Secondly; we are also successful in quantifying E2E delay and associated parameters that contribute in OOS arrival over the multi-path transport. Further, the upper-bounds for E2E delay, E2E delay variations, OOS reception and buffer occupancy are also derived.

The derived upper-bounds for E2E delay variations and consequently OOS arrival, buffer occupancy etc. indicate that the possible variations are based on packet length, data rate and the number of nodes on each path as the propagation delays are considered negligible. The length of packets from each flow directly influence the transmission time and cause certain degree of uncertainty in the service of other backlogged packets. This signifies the fact that the E2E metrics are heavily dependent on traffic sources which exhibit certain stochastic properties and can provide sufficient leverage to accommodate more flows under the assumption of Gaussian sources. For more concrete bounds, we present statistical bound in chapter 4.

Chapter 4

STOCHASTIC MULTI-SERVER SCHEDULING MODEL

4.1. Network Calculus Approach

The deterministic analytical model of capacity aggregated, multi-path flow analysis discussed in chapter 3 provides useful upper bounds for the E2E delay, its variations, OOS reception and buffer occupancy. These upper-bounds help in estimation and allocation of sufficient resources to realize desired QoS levels for flows that belong to different CoS. One obvious limitation of such model relates to the static allocation of these resources that may, sometimes lead to under utilization of communication resources. For example, in case of shaped and regulated traffic sources, there is a limit of peak traffic rates but the sources may generate traffic at lower than peak rate. The resource allocation with peak rate may therefore lead to poor scalability and under utilization problems. In most of the practical situations, the aggregated flow pattern of a group of flows exhibit Gaussian characteristics (Jarma Kilpi 2002). This eventually leads to the hypothesis of two moment stochastic analysis of flows to squeeze upper-bounds to lower levels with a certain allowed probability of violation of these bounds (Jinwoo Choe 1998). This approach can provide more economical resource allocation scheme and help in improving scalability of the solution (P. Giacomazzi 2009).

In this chapter, we extend our proposed deterministic model and develop a novel stochastic model of E2E delay and its variations associated with multipath flows. The primary purpose of this model is to realize tighter bounds for crucial E2E metrics for efficient use of communication resources. This model is based on bounded-variance network calculus, as discussed in (Paolo Giacomazzi 2008). Traditionally, min-plus algebra has been used for such E2E delay analysis (J. Boudec 2001). Despite its effectiveness in computing tighter delay bounds, the major problems associated with min-plus algebra include lack of scalability and higher computation complexity (P. Giacomazzi 2009). One of the recent innovations in stochastic network analysis is the bounded variance network calculus that provides tighter bounds at lower computational complexity (Paolo Giacomazzi 2008). The resultant E2E delay is a tight-approximation rather than a bound, which is an obvious trade-of in case of simplification of problem. The marginal reduction of accuracy and reliability provide significant gain in simplification of network traffic analysis. Therefore, we evaluate our stochastic analysis model on the theory of bounded variance network calculus and provide useful approximations for E2E delay and its variations. These approximation help in determining probabilistic OOS arrival and associated issues like buffer occupancy and packet drop. In the following sections, we first describe the basic foundation of bounded variance network calculus, and then we evaluate the multi-path flow characteristics.

In multi-path flow scenario, there are multiple E2E delay distributions that are convoluted to produce a single delay distribution. Figure 4.1 describes the basic schematic for such scenario. The schematic highlights three main components of the system. The first one on the left is the traffic source, which is a two-state Markov modulated Poison process (MMPP) with ON and OFF states. Secondly; there are multiple tunnels comprising of distinct number of network nodes. Finally; there is a flow sink which represents either a final destination or the HA. At the source, the transition rate from OFF state to ON state is $\lambda(s^{-1})$, and from ON state to OFF state is μ (s^{-1}). In this model, each tunnel carries both through and cross[5] traffic.

4.2. Bounded Variance Network Calculus

Bounded variance network calculus (BVNC) is a simplified approach to achieve approximation of E2E delay that is inspired by the maximum variance asymptotic (MVA) technique (Jinwoo Choe 1998). The great majority of researches in network domain use min-plus algebra for network traffic analysis (A. Burchard 2001). Min-plus algebra though provides good estimates of network node delay, it suffers from scalability problem and delay bounds derived through it significantly degrade as the number nodes increase. This limitation has been reduced in (F. Ciucu 2006), but the complexity of calculating these bounds through numerical optimization still ruins high. The MVA implicitly works under the assumption that all traffic sources are Gaussian and calculates upper-bound for the probability of exceeding a given delay threshold at any node (Paolo Giacomazzi 2008).

[5] The through flow is one which traverses all the nodes of tunnel; whereas, a cross flow is one which passes over some of the nodes on the E2E path and essentially does not traverse all the nodes of E2E path.

The MMPP being a random process is also a special case of Gaussian distribution. Inspired by the simplicity and efficiency of the MVA bounds, the framework for BVNC was presented in (Paolo Giacomazzi 2008). The framework outlines a simplified approach of estimating E2E delay of a flow with a certain known probability of delay violation. In this thesis, we extend this approach to derive tighter upper-bounds with known probability of violation of these bounds over the multi-path flow operation. The stochastic upper-bounds for E2E delay variations and OOS arrival provide leverage for more efficient resource utilization.

The BVNC is based on two inequalities that relate variance of minimum and maximum of two bi-variate normal random variables. The inequalities state that given two bi-variate normal random variables x and y, the variance of $a = min(x, y)$ and $b = max(x, y)$ are bounded as given in (Eq 4-1) and (Eq 4-2).

$$Var(a) \leq \max\big(var(x), var(y)\big), \qquad \textit{(Eq 4-1)}$$

$$Var(b) \leq \max\big(var(x), var(y)\big), \qquad \textit{(Eq 4-2)}$$

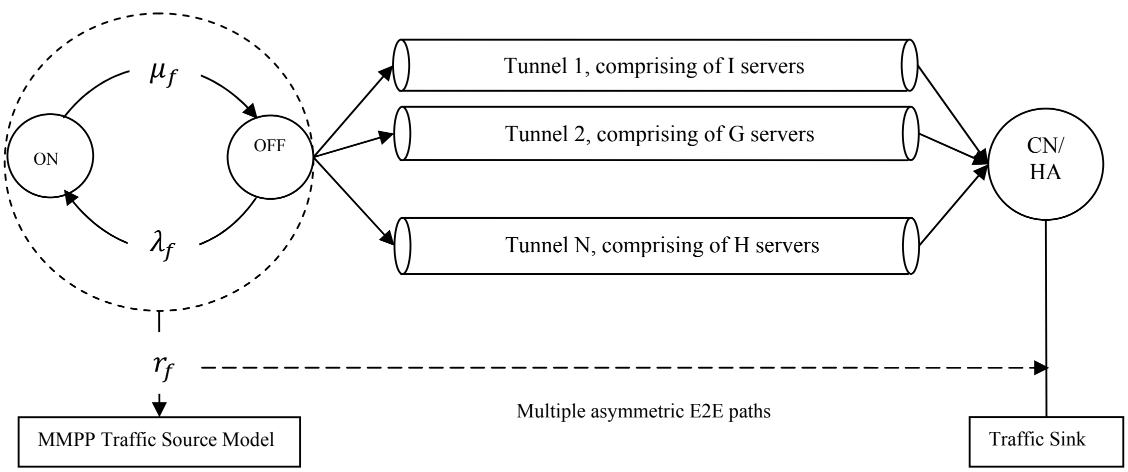

Figure 4-1: Reference model of stochastic modeling of MMPP traffic source over multiple E2E

The inequalities given in (Eq 4-1) and (Eq 4-2) can be used in conjunction with traffic and service envelops to approximate node delay, as described in (J.-Y. Qiu 1999). The inequalities (Eq 4-1) and (Eq 4-2) also provide foundation for the achieving statistical bounds for delay at k^{th} server. The Table 2-1 describes various symbols used in this model. The traffic $x_f(t)$, generated by a source during time interval of length t, is constrained by traffic envelops $b_f^{in}(t)$, which is a random process and can be described as in (Eq.4-3).

$$\forall z, t: Pr\{x_f(t) > z\} \leq Pr\{b_f^{in}(t) > z\}, \qquad \textit{(Eq 4-3)}$$

where z is the upper bound for the arrival process during time t. Similarly, the statistical service envelop $S_f(t)$, describes the service offered to flow f at a node with a bounded probability of exceeding the desired service delay at that node. If $\theta(p_f^k)$ is the actual delay faced by a packet belonging to flow f, with d_f as its delay threshold and p is the probability of exceeding this threshold at a given node than statistical delay bound can be given by,

$$Pr\{\theta(p_f^k) \ > d_f\} \leq p, \qquad \textit{(Eq 4-4)}$$

In accordance with the min-plus algebra and network calculus (J. Boudec 2001), the probability of exceeding a delay threshold $d_{f,}$ at any server is upper bounded by:

$$Pr\{\theta(p_f^k) \ > d_f\} \leq Pr\{\max_{t\geq 0}\{b_f^{in}(t) - S_f(t+d_f)\} > 0\}, \qquad \textit{(Eq 4-5)}$$

where $b_f^{in}(t)$ and $S_f(t)$ are the input traffic and service envelops; respectively, for the flow f. The complex inequality given in (Eq 4-5) has been solved by Giacomazzi in (P. Giacomazzi 2009), using MVA upper-bound with following useful approximation.

$$\sigma^2(t) = var[b_f^{in}(t) - S_f(t+d_f)], \qquad \textit{(Eq 4-6)}$$

The (Eq 4-6) describes the variance of delay in terms of traffic and service envelops. The (Eq 4-6) also yields following relationship.

$$\alpha(t) = -\frac{E[b_f^{in}(t) - S_f(t+d)]}{\sigma(t)}, \qquad \textit{(Eq 4-7)}$$

The (Eq 4-6) & (Eq 4-7) yield the following useful result.

$$Pr\{\theta(p_f^k) \ > d_f\} = e^{-\frac{\alpha_{min}^2}{2}} \leq p, \qquad \textit{(Eq 4-8)}$$

The (Eq 4-8) shows that it is possible to find the approximated maximum probability of a delay bound violation in a node, if the traffic source and service envelops are known. Giacomazzi et al. have shown a recurrence to express traffic envelops of flow f at the $(j+1)^{th}$ server, based on the traffic and service envelops of the j^{th}server (Paolo Giacomazzi 2008). The same is given as in (Eq 4-9),

$$var\left(b_{f,j+1}^{in}(t)\right) \leq max\left(var\left(b_{f,j}^{out}(t)\right), var\left(S_f(t)\right)\right), \qquad \textit{(Eq 4-9)}$$

The recurrence equation (Eq 4-9) is iterated through all nodes on the E2E path to find the probabilistic upper-bound for the variance in traffic envelop for flow f at each server. The variance in traffic envelop is then used in (Eq 4-7) to find $\alpha(t)$, that yields the probability of violation of delay threshold d_f for each flow, at each server. The probability distribution function (PDF) for delay violation at each server is convoluted to get the PDF for E2E delay violation. If $\Upsilon_{d_f,j}(t)$ represents PDF of delay violation at server j, then PDF of E2E delay violation can be given by (Eq 4-10).

$$\Upsilon_{d_f,E2E}(t) = \Upsilon_{d_f,1}(t) * \Upsilon_{d_f,2}(t) * \ldots * \Upsilon_{d_f,K}(t), \qquad \textit{(Eq 4-10)}$$

where K is the number of server in the E2E path. In case of multipath flow, the convolution of multiple paths can provide delay PDF for the multiple E2E paths. If we represent E2E delay probability distribution function for m^{th} path as $\Upsilon^{m}_{d_f,E2E}(t)$, than the convolution of M path can be given as follows.

$$\Upsilon^{Mul}_{d_f,E2E}(t) = \Upsilon^{1}_{d_f,E2E}(t) * \Upsilon^{2}_{d_f,E2E}(t) * \ldots * \Upsilon^{M}_{d_f,E2E}(t), \qquad \textit{(Eq 4-11)}$$

The probability of delay violation can then be calculated using the (Eq 4-12).

$$Pr\,(D_{f,E2E} > d_f) = \int_{d_f}^{\infty} \Upsilon^{Mul}_{d_f,E2E}(t)\,dt. \qquad \textit{(Eq 4-12)}$$

The (Eq 4-3) through (Eq 4-12) provide a framework for calculating stochastic delay bounds for a single node as well as the E2E path (P. Giacomazzi 2009). In the subsequent section, we extend this framework to multi-path flow analysis and derive service and traffic envelops upper-bounds for E2E delay variations, and OOS arrival.

4.3. Stochastic Multi-server Scheduling Model

Lemma 4.1: The stochastic service envelop for any arbitrary packet k of flow f, $S^{k}_{f,ASM}(t)$, at the ASM scheduler is given by

$$S^{k}_{f,ASM}(t) \le \max\left(0, C^{S}_{A}t - \sum_{j=1, j\# f}^{pos\left[d\left(p^{k}_{f}\right)\right]} b^{in}_{j}\left(t - \max\left(0, \theta'_{S}\left(p^{k}_{j}\right) - d_j\right)\right)\right),$$

Where $pos\left[d\left(p_f^k\right)\right]$, is the position of packet p_f^k in a sorted departure schedule list of total F flows contending for service, $\theta_S'(p_j^k)$ is the actual delay experienced by the packet p_j^k before getting service, and d_j is the delay threshold for flow j.

Proof: The deterministic delay associated with the service of packet $p_j^k$was calculated in Lemma 3.2 in Chapter 3. It was discussed in Chapter 3 that the service time of different packets depends on their packet lengths and the transmission latencies of predecessor packets. Therefore, the service latency of packet p_f^k depends on the following aggregated delay variance of predecessor packets.

$$d_{total} \leq \sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} \left(\theta_S'\left(p_j^k\right) - d_j\right), \qquad \textit{(Eq 4-13)}$$

It is noticeable that in (Eq 4-13), the transmission unit is a packet, reflecting arrival of a complete packet. Since the source of traffic is two state ON-OFF systems that randomly switch between states and the outcome of the system is of varying length packets, the (Eq 4-13) may be modified to reflect communication resource consumption as follows.

$$\sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} S_{j,ASM}^k(t) \leq \sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} b_j^{in}\left(t - \max\left(0, \theta_S'\left(p_j^k\right) - d_j\right)\right), \qquad \textit{(Eq 4-14)}$$

Finally; the above given resource is depleted before the packet p_f^k, the service available for this packet can be no more than the difference of total capacity in the system i.e. C_A^S, and already depleted capacity, as given by (Eq 4-15).

$$S_{f,ASM}^k(t) \leq \max\left(0, C_A^S t - \sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} b_j^{in}\left(t - \max\left(0, \theta_S'\left(p_j^k\right) - d_j\right)\right)\right), \qquad \textit{(Eq 4-15)}$$

The service available in (Eq 4-15) is a residual service, available for all successor packet of p_f^k. It is obvious that these packets will get service from the residual service available after the transmission of p_f^k. The proof of Lemma follows from (Eq 4-15). ■

Lemma 4.2: The variance in service envelop for any arbitrary packet k of flow f, $VAR\left[S_{f,ASM}^k(t)\right]$, at the ASM scheduler is upper-bounded by

$$VAR\left[S_{f,ASM}^k(t)\right] \leq \sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} b_j^{in}\left(t - \max\left(0, \theta_S'\left(p_j^k\right) - d_j\right)\right)$$

Proof: The service envelop given in (Eq 4-15) takes both service as well as delay as maximum terms. This indicates that the maximum variance in the service is essentially based on the inner capacity depletion term and hence the variance in service is upper-bounded by it as given in (Eq 4-16).

$$VAR\left[S_{f,ASM}^{k}(t)\right] \leq \sum_{j=1,j\#f}^{pos\left[d\left(p_f^k\right)\right]} b_j^{in}\left(t - max\left(0, \theta_S\left(p_j^{k\prime}\right) - d_j\right)\right), \quad \text{(Eq 4-16)}$$

Hence the proof of Lemma follows. ■

Lemma 4.3: The mean traffic arrival for the token bucket (TB) regulated source, using a two state Markov chain is given by

$$E\left[b_f^{in}(t)\right] \leq \left(r_f l_f \left(\frac{\lambda_f}{\lambda_f + \mu_f}\right)\right) t,$$

Proof: The deterministic mean arrival of traffic through TB regulated source with parameters (σ_f, r_f), is described by the (Eq 4-17).

$$b_f^{in}(t) \leq \sigma_f + r_f t, t \geq 0 \quad \text{(Eq 4-17)}$$

where σ_f is the depth of the bucket to reflect controlled burstiness in traffic and r_f is the average rate of arrival. Considering Gaussian source, the cummulative avarage arrival $X(t)$, at time t can be given by the following Equation.

$$E\big(X(t)\big) = r_f t, \quad \text{(Eq 4-18)}$$

In (P. Giacomazzi 2009), it is shown that an M state Markov process has a mean given by the following Equation,

$$E\big(X(t)\big) = \sum_{i=1}^{M}(r_{f,i}\, l_i p_i)t, \quad \text{(Eq 4-19)}$$

where $\lambda_i$is the rate of transition to state i, l_i is the length of packet arriving in state i, and $p_i$is the probability of transition into state i. Since we are dealing with a 2-State ON-OFF model where the generation of traffic is possible only in ON state, and the probability of arrival of traffic in OFF state is Zero, the summation of (Eq 4-19) can be reduced to the following.

$$E\big(X(t)\big) = (r_f l_f p_{ON})t, \quad \text{(Eq 4-20)}$$

The p_{ON}, the probability of transition to state ON, can be represented as $\frac{\lambda_f}{\lambda_f+\mu_f}$, as per model shown in Figure 4-1, the (Eq 4-20) can be written as

$$E\big(X(t)\big) = \left(r_f l_f \left(\frac{\lambda_f}{\lambda_f+\mu_f}\right)\right)t, \qquad \textit{(Eq 4-21)}$$

$$b_f^{in}(t) \leq \left(r_f l_f \left(\frac{\lambda_f}{\lambda_f+\mu_f}\right)\right)t, \qquad \textit{(Eq 4-22)}$$

The proof of Lemma follows. ■

Lemma 4.4: The variance of traffic envelop of 2-state Markov modulated source, serviced by ASM server is upper-bounded by,

$$var\big(b_f^{in}\big) \leq \left(2\frac{\lambda_f\mu_f}{\big(\lambda_f+\mu_f\big)^3} {r_f}^2 {l_f}^2 + \frac{\mu_f}{\big(\lambda_f+\mu_f\big)} r_f\big({l_f}^2 + var(l_f)^2\big)\right)t,$$

Proof: The deterministic variance in the cumulative arrival of TB regulated source $X(t)$ is given by

$$var\big(X(t)\big) = r_f\sigma_f t, \qquad \textit{(Eq 4-23)}$$

In (P. Giacomazzi 2009), it is shown that the variance

$$var\big(b_f^{in}\big) = \left(\sum_{i=1}^{M} p_i r_{f,i}\left({l_i}^2 + var\big[{l_i}^2\big]\right) + 2\sum_{i=1}^{M} p_i l_i r_{f,i} \sum_{j=1}^{M} l_j r_{f,j} \sum_{k=1}^{M-1}\frac{\gamma_{ij,k}}{\omega_k}\right)t - 2\sum_{i=1}^{M} p_i l_i r_{f,i} \sum_{j=1}^{M} l_j r_{f,j} \sum_{k=1}^{M-1}\frac{\gamma_{ij,k}}{{\omega_k}^2}\left(1-e^{\omega_k t}\right), \qquad \textit{(Eq 4-24)}$$

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where ω_k is the kth Eigen value & $\gamma_{ij,k}$ is a real constant. The variance is upper-bounded as follows,

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$$var\big(b_f^{in}\big) \leq \left(\sum_{i=1}^{M} p_i r_{f,i}\left({l_i}^2 + var\big[{l_i}^2\big]\right) + 2\sum_{i=1}^{M} p_i l_i r_{f,i} \sum_{j=1}^{M} l_j r_{f,j} \sum_{k=1}^{M-1}\frac{\gamma_{ij,k}}{\omega_k}\right)t, \qquad \textit{(Eq 4-25)}$$

In case of a 2-state model, the above Equation reduces to the following,

$$var(b_f^{in}) \leq \left(p_{ON} r_f \left({l_f}^2 + var\left[{l_f}^2\right]\right) + 2{l_f}^2{r_f}^2 p_{ON}\right)t, \qquad \text{(Eq 4-26)}$$

The same is reduce to following

$$\left(2\frac{\lambda_f \mu_f}{\left(\lambda_f+\mu_f\right)^3}{r_f}^2{l_f}^2 + \frac{\mu_f}{\left(\lambda_f+\mu_f\right)} r_f \left({l_f}^2 + var\left(l_f\right)^2\right)\right)t, \qquad \text{(Eq 4-27)}$$

This completes the proof. ■

Theorem 4.1: The probability of violating a service delay threshold $\boldsymbol{d_f^{th}}$ of packet p_f^k of flow f, in presence of F competing flows at ASM server is upper-bounded by,

---

$$p\left(\theta_S\left(p_f^k\right) > \boldsymbol{d_f^{th}}\right) \leq exp\left(-\frac{2C_S^A\left(C_S^A - \beta F\left(\frac{\lambda_f}{\lambda_f+\mu_f}\right) r_f l_f\right)}{\left(2\beta F\frac{\lambda_f \mu_f}{\left(\lambda_f+\mu_f\right)^3}{r_f}^2{l_f}^2 + \frac{\mu_f}{\left(\lambda_f+\mu_f\right)} r_f \left({l_f}^2 + var\left(l_f\right)^2\right)\right)} t\right),$$

where $\boldsymbol{d_f^{th}}$ is delay threshold for packets of flow $\boldsymbol{f}$.

Proof: The procedure for determining probability of violation of a delay threshold in a server is discussed in the introduction section of this chapter. It was discussed there that the variance of traffic and service envelops greatly reduce the complexity of finding above said probability. Lemma 4.2 & 4.4 provide traffic and service envelops of the $\boldsymbol{ASM}$ server. Using recurrence (Eq 4-9), it is obvious that the variance of service to any flow may not exceed its arrival envelop. In case of sufficient capacity, the service is bound to be maximum under the traffic envelop. Hence, recurrence (Eq 4-9) can be represented as follows,

$$var\left(b_{f,2}^{in}(t)\right) \leq max\left(\begin{array}{l}\left(2\beta F\frac{\lambda_f\mu_f}{(\lambda_f+\mu_f)^3}r_f{}^2l_f{}^2+\frac{\mu_f}{(\lambda_f+\mu_f)}r_f\left(l_f{}^2+var(l_f)^2\right)\right)t,\\\left(2\beta F\frac{\lambda_f\mu_f}{(\lambda_f+\mu_f)^3}r_f{}^2l_f{}^2+\frac{\mu_f}{(\lambda_f+\mu_f)}r_f\left(l_f{}^2+var(l_f)^2\right)\right)t\end{array}\right),$$

(Eq 4-28)

Where F represent both through and cross flows. The ratio of the through flows to cross flows $\beta \ni, 0 \leq \beta < 1$ may be made constant to evaluate different traffic. Since both terms are same in (Eq 4-28), hence the arrival at the next server is given by (Eq 4-29).

$$var\left(b_{f,2}^{in}(t)\right) \leq \left(2\beta F\frac{\lambda_f\mu_f}{(\lambda_f+\mu_f)^3}r_f{}^2l_f{}^2+\frac{\mu_f}{(\lambda_f+\mu_f)}r_f\left(l_f{}^2+var(l_f)^2\right)\right)t, \quad \text{(Eq 4-29)}$$

Now using (Eq 4-7), we find

$$\alpha_{min}{}^2 = \frac{2C_S^A\left(C_S^A-\beta F\left(\frac{\lambda_f}{\lambda_f+\mu_f}\right)r_fl_f\right)}{\left(2\beta F\frac{\lambda_f\mu_f}{(\lambda_f+\mu_f)^3}r_f{}^2l_f{}^2+\frac{\mu_f}{(\lambda_f+\mu_f)}r_f\left(l_f{}^2+var(l_f)^2\right)\right)}t, \quad \text{(Eq 4-30)}$$

The (Eq 4-30) yields the desired delay violation probability by putting it in (Eq 4-8).

$$p\left(\theta_S(p_f^k) > \boldsymbol{d_f^{th}}\right) \leq exp\left(-\frac{2C_S^A\left(C_S^A-\beta F\left(\frac{\lambda_f}{\lambda_f+\mu_f}\right)r_fl_f\right)}{\left(2\beta F\frac{\lambda_f\mu_f}{(\lambda_f+\mu_f)^3}r_f{}^2l_f{}^2+\frac{\mu_f}{(\lambda_f+\mu_f)}r_f\left(l_f{}^2+var(l_f)^2\right)\right)}t\right), \quad \text{(Eq 4-31)}$$

The proof of the theorem follows. ■

Corollary 4.1: The probability of delay at *ASM* server, as specified in (Eq 4-31) has may take any of the values from a complete range of distribution. The integration of the (Eq 4-31) over a range $\boldsymbol{d_f^{th}}$ to ∞ can provide such distribution. Therefore, the delay distribution $\Upsilon_{\theta_S(p_f^k)}(t)$, of any arbitrary packet p_f^k, of flow $\boldsymbol{f}$ at the $\boldsymbol{ASM}$ server is given by (Eq 4-32).

$$\Upsilon_{\theta_S\left(p_f^k\right)}(t) \leq \kappa\, e^{-\kappa t}, \tag{4-32}$$

Where $\kappa = \frac{2C_S^A\left(C_S^A - \beta F\left(\frac{\lambda_f}{\lambda_f+\mu_f}\right)r_f l_f\right)}{\left(2\beta F\frac{\lambda_f\mu_f}{\left(\lambda_f+\mu_f\right)^3}r_f^{\,2}l_f^{\,2}+\frac{\mu_f}{\left(\lambda_f+\mu_f\right)}r_f\left(l_f^{\,2}+var\left(l_f\right)^2\right)\right)}.$

Theorem 4.2: The distribution of E2E delay of a path comprising of ***K*** number of servers, starting from ASM server is upper-bounded by,

$$\Upsilon_{f,E2E}(t) \leq e^{-\kappa t}\sum_{i=1}^{K}\frac{(\kappa t)^i}{i!},$$

Proof: It is well known fact in network calculus (J. Boudec 2001) that the E2E delay distribution is the convolution of all the individual delay distribution of individual servers in the path. The same is also described in (Eq 4-10). Since κ in (Eq 4-32) comprises of all constants, the convolution integral has only one variable t, and can be represented as follows.

$$\Upsilon_{d_{f,E2E}}(t) = \int_{d_{f,E2E}}^{\infty} \Upsilon_{\theta_{S,\left(p_f^k\right)}}(t)\, dt.$$

$$\Upsilon_{d_{f,E2E}}(t) = \int_{d_{f,E2E}}^{\infty} \kappa\, e^{-\kappa t}\, dt. \tag*{(Eq 4-33)}$$

The above integrals yields the following,

$$\Upsilon_{d_{f,E2E}}(t) = \frac{\kappa^{\,K} t^{\,K-1} e^{-\kappa t}}{(K-1)!}, \tag*{(Eq 4-34)}$$

The (Eq 4-34) describe the whole spectrum of E2E delay distribution of path with K servers. Therefore, the violation of E2E delay threshold is upper-bounded by the following.

$$\Upsilon_{d_{f,E2E}}(t) \leq e^{-\kappa t} \sum_{i=1}^{K} \frac{(\kappa t)^i}{i!}.$$

The proof of theorem completes. ■

Theorem 4.3: The distribution of E2E delay variations $\Upsilon_{\delta_{f,E2E}}(t)$, of a flow $\boldsymbol{f}$ traversing multiple paths en-route destination is upper-bounded as**,**

$$\Upsilon_{\delta_{f,E2E}}(t) \leq e^{-\kappa t} \sum_{i=0}^{K-L} \frac{(\kappa t)^i}{i!}.$$

Proof: Let there be multiple paths between the source and destination; such that the path with largest number of server has hop count of K, where as the one with lowest number of servers has L hops. As per theorem 4.2, the E2E delay distributions of two paths can be given by (Eq 4-35) & (Eq 4-36) as follows; respectively.

$$\Upsilon_{d_{f,E2E}^{K}}(t) \leq e^{-\kappa t} \sum_{i=0}^{K} \frac{(\kappa t)^i}{i!}. \qquad \textit{(Eq 4-35)}$$

$$\Upsilon_{d_{f,E2E}^{L}}(t) \leq e^{-\kappa t} \sum_{i=0}^{L} \frac{(\kappa t)^i}{i!}. \qquad \textit{(Eq 4-36)}$$

The delay variation of two paths is obviously given by the difference of the two terms as the E2E distribution is dependent directly on number of server in the path. The proof of the theorem follows. ■

Theorem 4.3: The distribution of time duration for probable OOS arrival $\Upsilon_{\varphi_f^k}(t)$, in case of multi-path traversal of a single flow f is upper-bounded by

$$\Upsilon_{\varphi_f^k}(t) \leq e^{-\kappa\varphi t} \sum_{i=0}^{K-L} \frac{(\kappa\varphi t)^i}{i!},$$

Proof: The stochastic behavior of OOS arrival is essentially a bi-variate distribution, comprising of distribution of both arrival as well as delay variation distribution. The arrival process is controlled through Poison random process with exponential distribution having some mean. In the two worse case scenarios, a latest arrival may travel over the lowest delay path or an earliest arrival may find a longest path. Since both the processes are exhibit exponential distribution, a shifted convolution of the two distributions provides the distribution of OOS arrival.

$$\Upsilon_{\varphi_f^k}(t) \le e^{-\kappa\varphi t} \sum_{i=0}^{K-L} \frac{(\kappa\varphi t)^i}{i!}, \qquad \textit{(Eq 4-37)}$$

The proof of the theorem completes. ■

Theorem 4.4: The distribution of Buffer Occupancy $\Upsilon_{\vartheta_f^k}(t)$, caused by the OOS reception of packets of flow f is upper-bounded by,

$$\Upsilon_{\vartheta_f^k}(t) \le \frac{\mathrm{P}\left[\left(W - \sum_{i=k+1}^{j} l_i^{max}\right) > 0\right]}{e^{-\kappa\varphi t} \sum_{i=1}^{K-L} \frac{(\kappa\varphi t)^i}{i!}},$$

Proof: The possible of buffer occupancy rises with probable arrival of multiple packets other than the expected packet, during a delay distribution of expected packet. Hence, the buffer occupancy is depends on two stochastic processes, i.e. the traffic source arrival PDF and the PDF of E2E delay variation. Let there be a packet p_f^k expected at the receiver but $q, \exists\, q \in N$ number of packet arrive (not essentially in sequence) without arrival of the expected packet. If W is the total buffer size (in bytes) then the total buffers occupied till the end of upper-bound of delay variation is represented by (Eq 4-38)

$$\Upsilon_{\vartheta_f^k}(t) \le \frac{\mathrm{P}\left[\left(W - \sum_{i=k+1}^{j} l_i^{max}\right) > 0\right]}{\Upsilon_{\varphi_f^k}(t)}, q > k \qquad \textit{(Eq 4-38)}$$

where l_i, represents the length of each out-of-sequenced arrived packet.

The length of each packet also follows Gaussian PDF but for simplicity we use only the mean length of packet. In case of in-order arrival, packets are removed from the buffers and result in release of occupied buffers. The occurrence of timeout interval indicates a packet drop event and the next in-order sub-stream is forwarded. Therefore, the length of buffer occupied divided by the maximum time over which such occupancy persist can provide the upper-bound of buffer occupancy. Hence substitution of $\Upsilon_{\varphi_f^k}(t)$ from (Eq 4-37) we get (Eq 4-39) as follows,

$$\Upsilon_{\vartheta_f^k}(t) \le \frac{\mathrm{P}\left[\left(W - \sum_{i=k+1}^{j} l_i^{max}\right) > 0\right]}{e^{-\kappa\varphi t} \sum_{i=1}^{K-L} \frac{(\kappa\varphi t)^i}{i!}}, \qquad \textit{(Eq 4-39)}$$

Theorem 4.5: The probability of dropping of packet p_f^k, scheduled through ***ASM*** server, due to buffer occupancy is upper-bounded by

$$P\left[\psi_f^k\right] \leq P\left[\vartheta_f^k(t) \geq e^{-\kappa\varphi t} \sum_{i=1}^{K-L} \frac{(\kappa\varphi t)^i}{i!}\right]$$

where $P\left[\psi_f^k\right]$ is the probability of dropping packet p_f^j.

Proof: The packet drop event may be caused due to the buffer depletion and exceeding of delay violation from given upper bound. In case of an infinite buffer, the sole cause of packet drop is due to the timeout event that indicates the end of wait for the concerned packet. In case, the timeout event is define at the upper end of the PDF of delay variation, the chances of packet drop reaches near to zero. Practically, a timeout threshold is defined not exactly near to the tail to accommodate more traffic; the chances of violation of time-out persist. The assumption of an infinite buffer is also not practical and for scalable service, a finite buffer is necessary to maintain in-order delivery of traffic of any flow. Therefore, packet drop depends on the degree of E2E delay variations and accompanying extent of OOS arrival. The same is given in (Eq 4-40) as follows,

$$\begin{cases} P\left[\psi_f^k\right] = 0, & if\ \vartheta_f^k(t) < e^{-\kappa\varphi t} \sum_{i=1}^{K-L} \frac{(\kappa\varphi t)^i}{i!} \\ 0 \leq P\left[\psi_f^k\right] < 1, & if\ \vartheta_f^k(t) \geq e^{-\kappa\varphi t} \sum_{i=1}^{K-L} \frac{(\kappa\varphi t)^i}{i!} \end{cases} \qquad \textit{(Eq 4-40)}$$

That also completes the proof of theorem. ■

4.4. Results

The proposed stochastic model for determining some thresholds for E2E metrics in a multi-server scheduling produces some interesting results. The mobility based characterization of multiple-path E2E delay metrics provide useful inside for providing flow based service guarantees. The major emphasis is on devising thresholds for each desired metric and use the communication resource more efficiently during mobility. Figure 4-2 plots E2E delay violation probability at different traffic loads in terms of number of flows, with different number of nodes on each path. The plot shows a high compliance of E2E delay threshold for wide range of traffic load and path diversities. The path diversity has been enumerated for more precise quantification in terms of path length (in terms of number of hops). It is observable from the plots that the variation in path length and increased traffic load increases the probability of violation of E2E delay. This result provides useful reference to device an E2E delay threshold that increases resource utilization without seriously hampering service guarantees.

In Figure 4-3, the plot of the OOS reception against E2E delay violation at different traffic loads has been recorded. The plot shows exponential rise in OOS reception up to initial stages of delay threshold violation and then saturating quickly at low traffic loads. It however, has a higher OOS reception at high traffic loads. The main cause of this behavior lies in the higher buffer occupancy levels at the higher traffic loads. The more important characteristic of this behavior lies in the fact that the OOS arrival saturates that highlight convergence of the proposed model with respect to OOS arrival. This result highlights the tightness of the upper bounds as the violation of the threshold has very low probability.

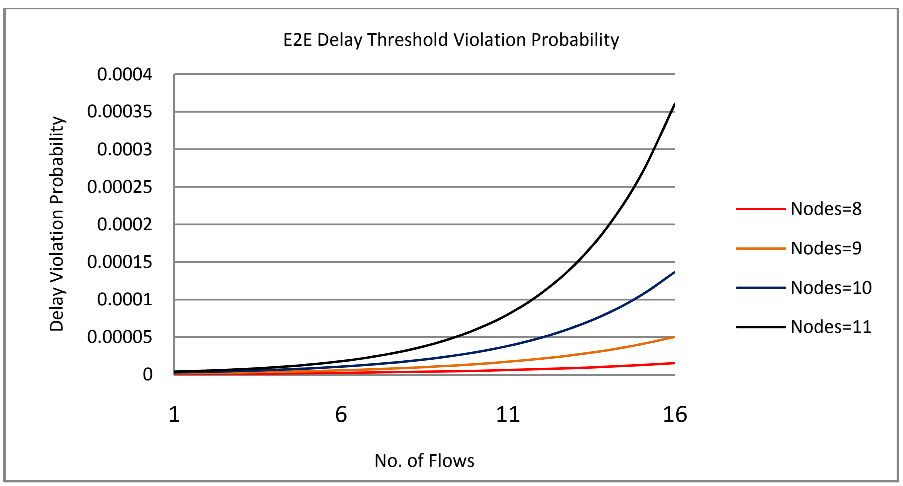

Figure 4-2: Probability of E2E delay violation at different traffic loads and length of path

In Figure 4-4, the probability of buffer occupancy is plotted against the buffer occupancy values in bytes. It is noticeable that the probability of higher buffer occupancy quickly drops to low values for higher occupancy levels. Buffer occupancy is high for the higher traffic loads of 16 flows. It is important to note that the traffic sources are all TB regulated EBB sources. It also signifies that the buffer occupancy shows convergence at lower levels of buffers to enable efficient buffer design.

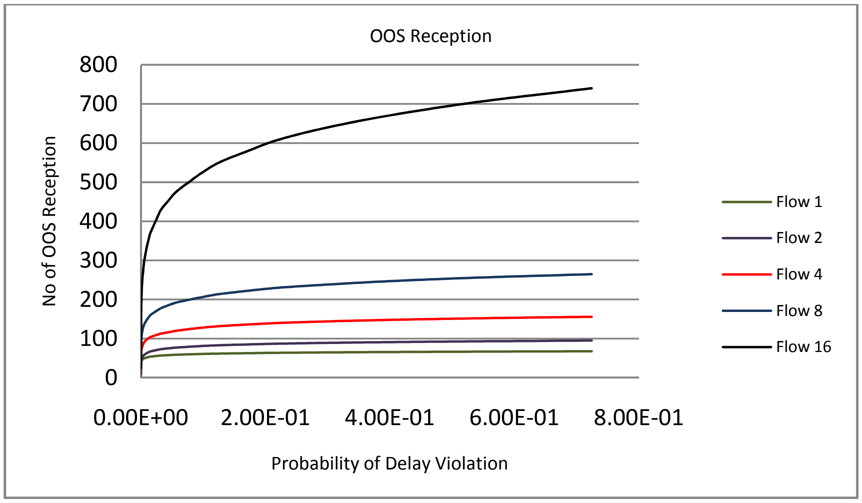

Figure 4-3: OOS reception at different E2E delay violation probability and different path length

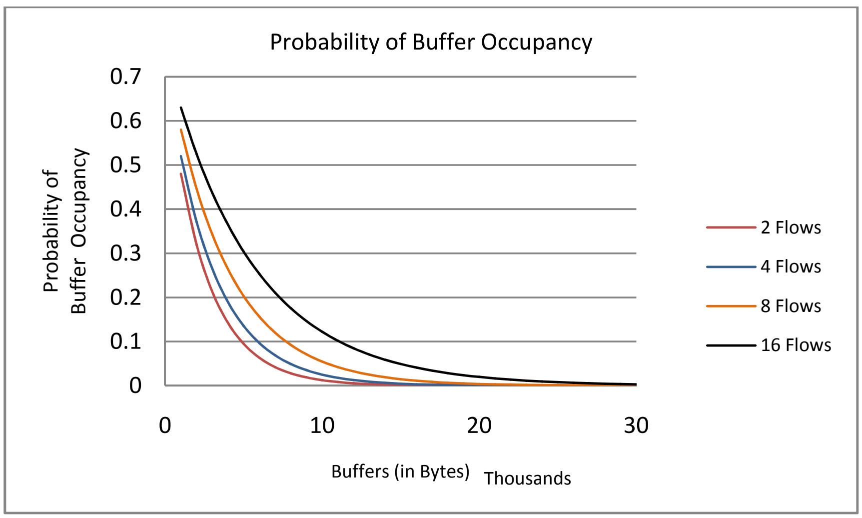

Figure 4-4: Probability of Buffer occupancy at different buffer levels

4.5. Summary

In this chapter, we have developed a stochastic model for the evaluation of E2E delay variations of multiple paths, each comprising of at-least one wireless hop. In fact this model is stochastic extension of the model presented in chapter 3. The PDFs of the E2E parameters provide useful tool in devices threshold for different metrics of interest and accordingly; help in developing more scalable multi-path communication channels. The proposed stochastic model answers fourth question of the thesis hypothesis in terms of achieving tighter upper bounds for the metrics of interest and help in better control over the design of threshold to select desired service levels. The analysis exercise of this chapter also provides useful parameters to improve performance of scheduling algorithms for the multi-path transmission of flows. The same is discussed in the next chapter.

Chapter 5

MULTI-SERVER SCHEDULING

The proposed deterministic and stochastic models of multi-server scheduling, as presented in chapter 3 and 4, provide basic quantification of key parameters that influence performance of flows transmitted through multiple E2E paths in mobility. These models also identify that the E2E delays can significantly vary in such flows and result in larger E2E delay variations. Therefore, conventional round robin scheduling approaches for using multiple interfaces of MMDs produced higher OOS reception added with higher buffer occupancy due to transmission over these paths. In fact, it requires added intelligent to adapt according to E2E parameters and available capacity. In this chapter, some scheduling strategies are proposed to enhance the performance of multi-path flows during mobility. It is evident from the analytical models that the main contributors of E2E delay variations and OOS reception include path length in term of number of intermediate nodes. The second most important contributor is the service delay at each node. The variation in service delay is modeled through service envelops, as discussed in chapter 4, and it is noticeable in the analysis of chapter 3 &4 that the service delay distribution is primarily caused by the transmission delays of varying size packets. Due to the underlying assumption of right sized admission control, it is obvious that each packet will be transmitted within its transmission deadline with a minor probability of violation of these deadlines. The multi-server scheduling of QoS aware traffic is discussed in lieu of the finding of proposed analytical models. Firstly, multi-server fluid flow (MFF) scheduling is proposed. It is assumed that the granularity of transmission units is divisible up to bit level. Secondly, the CAG and link ranking function are given that are used for adaptive multi-server scheduling. Finally, multi-server scheduling algorithm is proposed that is based on two-moment E2E delay metrics. The results of proposed models of chapter 3 and 4 are validated through simulation of proposed scheduler.

5.1. Multi-server Fluid Flow Scheduling

Let at any time t, there be N_p^t number of bytes back-logged in p queues such that $Q = \{q_1, q_2, \dots, q_p\}$ with lengths in bytes of each queue be given by l_i^t i.e. $\{l_1, l_2, \dots, l_p\}$. These queues are prioritized such that the priority of $q_1 > q_2 > .. > q_m$. Further, there are M links available with individual link capacities $C_A^S = \{c_1, c_2, \dots, c_m\}$, represented by c_i^t and effective transmission time[6] (ETT) units of $d = \{d_1, d_2, \dots, d_m\}$, represented by d_i^t. Assuming a work-conserving model, we use product of capacity and ETT of a link to access the fitness of link as a deciding parameter for transmitting highest priority queue. The process of capacity estimation of each link is based on MIHF event services and is given in detail in the following reference (Ahmad, Akbar, Qadir, 2008). Assuming τ, a small time interval during which backlogged traffic status does not change, and proposed algorithm completes it scheduling, algorithm shown in Figure 5-1 schedules traffic on available links, according to the queue priority and minimum packet reordering.

The algorithm after calculating c_i^t & ETT product of each link enters in two main loops. The first loop controls complete scheduling of back-locked queues according to their priorities, with highest priority queue served first, on best possible links or depletes all the capacity of the link, in case there are more back-locked bytes then capacity of the link. The second loop iterates over the multiple queues for sending batch of traffic on the best available link. The best link is found as H_k^{C}, where k is the link index and C is its capacity. This link is used to send i^{th} queue, if it possess sufficient capacity to send the whole queue. The $\boldsymbol{if}$ block is used in case k^{th} link has more capacity then the length of i^{th} queue, whereas, $\boldsymbol{else}$ block is used when the capacity of k^{th} link is less then i^{th} queue.

The MFF scheduling is a non-round robin scheduling approach in which the order of the link selection is based on the status of link regarding its capacity and transmission delay. The MFF does not iterate the link use unnecessarily as it keeps on transmitting on the same link till its capacity is depleted. This approach proactively reduces the chances of OOS reception as in case of a single good link, all backlogged traffic may be transmitted

[6] The effective transmission time is defined with respect to each wireless interface and it describes the physical transmission delay of a minimum transmission unit. i.e. a packet.

on a single link. In case, single best quality link does not possess sufficient link capacity, the next best quality link is selected for transmission of remaining backlogged traffic.

Input: $Q, N_p^t, l_i^t, c_i^t, d_i^t$
Output: *Schedule for all links*
For i *=1 to* m *;; No. of links*
$C_i^t = c_i^t * d_i^t$ *;; Capacity, ETT product of each link*
Next i
$H_k^C = max\ (C_i^t)$ *;; Find link with best capacity*
$While\ (H_k^C > 0\ and\ N_p^t > 0)$ *;; Proceed while link capacity and backlogged byte are non-zero*
$for\ i = 1\ to\ Q$ *;; loop for all queues*
$(if\ (\ C_k^t \geq l_i^t)$
$S_k^t = Schedule\ l_i^t\ on\ Link\ k$ *;;Complete Queue scheduled*
$N_p^t = N_p^t - l_i^t$ *;;Backlogged bytes reduced*
$C_k^t = C_k^t - l_i^t$ *;;Link capacity reduced*
$Empty\ l_i^t$ *;; ith queue is empty now.*
$Else$
$S_k^t = Schedule\ C_k^t\ bytes\ of\ l_i^t on\ link\ k$ *;;*
$l_i^t = l_i^t - C_k^t$ *;;*
$N_p^t = N_p^t - C_k^t$ *;;*
$C_k^t = 0$ *;; Link capacity depleted*
$End\ if$
$H_k^C = max\ (C_i^t)$ *;; Find new best capacity link*
$End\ For\ (\ Next\ i)$
$End\ While$

Figure 5-1: Multi-server Fluid Model scheduling algorithm

5.2. ASM Scheduler

The ASM scheduler is based on the proposed deterministic model, discussed in chapter 3. The scheduler first determines deadline for each arrived packet and then assigns an available physical link slot to backlogged packets in ascending order of their transmission deadline. The output of ASM server is a sorted list of departure schedules for the backlogged packets. The choice of a particular S_i for transmission of this list is decided by ASM, and is based on an ordered list of available transmission slots $T = \{t_1, t_2, \dots, t_N\}$, over a server busy time. Each of the t_i in T, contains information of physical channel number and time of its availability.

The operation of ASM is pictorially shown in Figure 5-2, where we find three logical queues of traffic. The queue shown in the top of Figure 5-2 contains traffic in its order of arrival. This queue is used by ASM to calculate the deadlines of each arrived packet, in accordance with the guaranteed rate of the flow. The deadlines for each arrived packet result in a sorted list of departure schedule, in ascending order, which is shown at the middle of the Figure 5-2. Finally using information available in T, sorted traffic queue is mapped onto the physical channels according within the constraints of deadlines for each packet.

In $TDMA$ based networks the frame and slot allocation information is provided by the base stations (BS) as per session/call admission agreement and is accurate in terms of time. The effectiveness of time division duplex (TDD) based channel allocation schemes in packet based communication that generally has high transmission in one direction and very low in the other, has also motivated non-TDMA based wireless technologies to make provisions for TDD. Generally, the time-slots allocated for the multiple links shall be in accordance with the guaranteed rate of all admitted sessions. The above mentioned scenario persists without mobility events and the ASM attains a stable state. In case of a mobility event, the reduced aggregate capacity due to disconnected link is accommodated through increased deadline for each packet. There could be following two possibilities in such scenario.

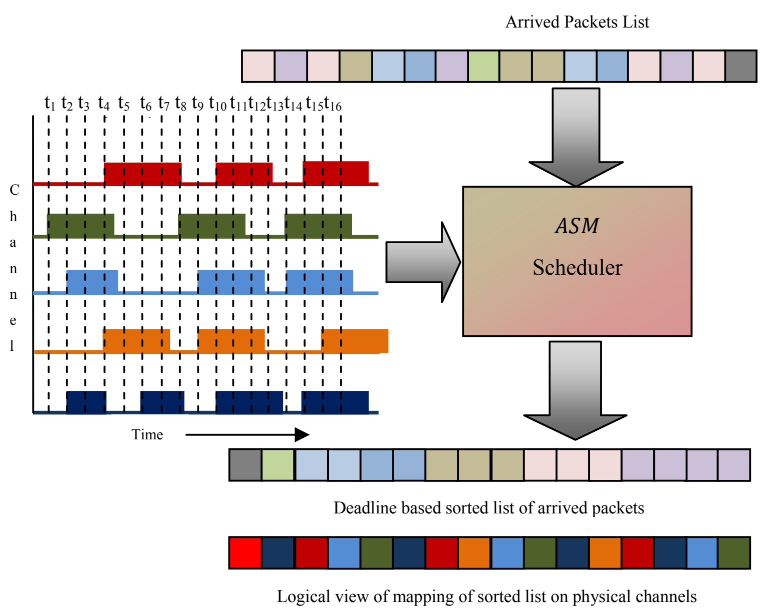

Figure 5-2: A graphical representation of ASM scheduler

In the downlink direction, the anchor point (HA or any other mobility agent, such as mobility anchor point) where an ASM is deployed, splits traffic in similar fashion as discussed for the uplink direction with a remote capacity exchange procedure. Since there is no sensors available at that end to measure link capacity at each of the wireless links, it is communicated through a packet exchange during the registration update process (Kyung-Joon Parka 2006). It may be recalled that multiple care-of-addresses are registered at the HA and these multiple tunnels are used to carry forward traffic destined to MMD. The work conserving nature of the scheme also helps it in serving burst sub-stream as it would generate sharper schedules for packets of such sub-streams provided capacity is not exhausted. Figure 5-3 shows the flow of scheduling algorithm in pseudo-code form.

5.3. E2E Delay: Maximum Likelihood Approach

In this section, proposed active multi-server delay-budget ordered ($AMDO$) scheduling scheme to aggregate capacity of multiple wireless interfaces of an MMD is described. This approach is specifically tuned to satisfy QoS needs of ongoing sessions with higher degree of availability and dependability at layer 3. The stochastic models of chapter 4 suggested a tighter delay bound at a slight degraded service criteria violation, i.e. the delay threshold for better service scalability. The $AMDO$ uses the delay threshold as delay budget for each flow during its E2E transportation. The practical E2E delay logs, along with MIHF triggers are transformed into a useful decision parameter for link ranking. The E2E delay log is maintained as one-way trip time (OTT), in contrast to the round trip time (RTT) used in TCP. The RTT has some proven in accuracies due to different paths traversed by packets and their acknowledgements. The basic transformation model of heterogeneous channel conditions over $E2E$ delay variations is been proposed to take care of inter path differential. The link ranking function (LRF) is also described in continuation of this transformation. The role of LRF is to strangulate delay variations and improved reliability of link selection process.

5.3.1 Link Ranking Function (LRF)

The LRF consists of two components; namely link-layer event transformation model, and effective OTT estimation model. The purpose of link layer events transformation is to quickly update link information to the upper-layers, where the OTT is estimated. The role of effective OTT estimation is to know about the OTT variations within link and accordingly incorporate it into the scheduling process. The services of two are described in more detail in the following text.

Input: $d(p^i)$, Sorted list of packets transmission schedules

T_j^k, A list of temporally ordered slots in the k^{th} logical frame during system busy time.

Output: Schedule of distribution of backlogged traffic as per transmission deadline on multiple links

i=1; index of departure schedule in $d(p^i)$, initialized to 1

j=1; index of the list of slots available in T_j^k, initialized to 1

While (TRUE) ;; Waiting for the start of next system busy time

$\boldsymbol{While}$! $\boldsymbol{Empty}\left(\boldsymbol{d}(\boldsymbol{p^i})\right)$; If any backlogged packets, start service,

Schedule p^ion T_j^k

increment j ;; Select next service slot

Increment i ;; Select next packet in backlogged queue

$\boldsymbol{End\ While}$;; Service next packet

$\boldsymbol{End\ While}$;; End of system busy period

Figure 5-3: The ASM scheduling algorithm

5.3.2 Link Layer Event Transformation Model

During mobility, the link status continuously changes and can be quickly detected by the link layer triggers (Ng 2007). These triggers describe the general status of the link to be active, inactive or any change in link quality. The statistical readings of OTT_j^i, the OTT for i^{th} session on j^{th} path, is historical accrual of large number of OTT values and a link status change event may take some time before it is accumulated in the prevailing OTT_j^i. This may cause wrong information about the link status, causing inappropriate schedules for the backlogged traffic. Some threshold definition is, therefore essential to simplify the link ranking task through some initialization process. This transformation is given below,

$$\mathcal{H}_j^i(\mathrm{e}) \stackrel{\text{def}}{=} \begin{cases} if\ Link\ i\ Goes\ Down,\ OTT_j^i = OTT_j^{\infty} \\ if\ Link\ i\ Becomes\ active, OTT_j^i = OTT_j^{\mu} \\ if\ Link\ i\ Quality\ Change, OTT_j^i = OTT_j^i \pm F_j \end{cases}, \qquad (Eq\ 5\text{-}1)$$

where $\mathcal{H}_j^i(\mathrm{e})$, represents transformation of a link layer event occurring on link j on the OTT_j^i values on i^{th}session using that link. It is important to mention here that the impact of $\mathcal{H}_j^i(\mathrm{e})$ is only restricted to the occurrence of event. The definition given in (Eq 5-1), simplifies the LRF service by re-initializing the OTT_j^i value on a specific link layer trigger to quickly inhale link status change that may otherwise take at-least a timeout value to be updated.

5.3.3. Effective OTT Estimation Model

The estimated value of effective OTT for session i over link j, $\left(OTT_{j\ eff}^i\right)$ has been derived using mean and variance of OTT_j^i. The two-moment analysis of the E2E delay has been described in detail in chapter 4. The complexity of estimation of such moments for multiple flows has been reduced by using some known techniques. These include moving averages, linear and polynomial regression etc. that estimate probable future value(s) on the basis of historical data. The complex nature of Internet traffic convolved with distributed control over multiple hops in a path makes these techniques less effective due to lack of stationary behavior. We have, therefore used stochastic model to improve the estimation accuracy of $OTT_{j\ eff}^i$. There may be multiple readings for each links for multiple sessions running for diverse destinations, but for simplicity we use only one such value.

The estimated $OTT_{j\ eff}^i$values are based on the recent past characteristics w.r.t. E2E delay variations. The E2E delay variation is recorded locally through the DB status of each arriving packet; as well as remotely through a UDP packet-pair exchange, for cumulative set of departed packets. The mean $(OTT_{j_\mu}^i)$ and variance $OTT_{j_{VAR}}^i$of $E2E$ delay are mathematically related in algorithmic function to assist evaluate $OTT_{j\ eff}^i$ of link i that is given in (Eq 5-2).

$$OTT_{j\ eff}^i = Max\left(OTT_{j_\mu}^i * \left(1 - \frac{log\left(Abs(OTT_{j_\mu}^i - OTT_{j_{VAR}}^i)\right)}{2}\right), \mathcal{H}_j^i(\mathrm{e})\right), \qquad \textit{(Eq 5-2)}$$

The skew $\left(1-\frac{log\left(Abs\left(OTT_{j_{\mu}}^{i}-OTT_{j_{VAR}}^{i}\right)\right)}{2}\right)$, in the delay variation is accrued accordingly, in the mean value of the session $OTT_{j_{\mu}}^{i}$.The inclusion of $\mathcal{H}_{j}^{i}(\mathrm{e})$ provides a means for quick realignment of $OTT_{j_{eff}}^{i}$, in case a link layer trigger occurs, as discussed in the previous sub-section.

5.4. AMDO Scheduling Algorithm

The prime objective of Active multi-server delay-budget ordered ($AMDO$) scheduling algorithm is to device a unified virtual path comprising of multiple asymmetric physical paths that adheres to the QoS requirements of sessions running through it and is robust enough to scale-down gracefully in case mobility events occur; such as handoffs on a subset of active events. The main challenge to this objective lies in inter-link delay variations causing OOS arrival at the receiver. The proposed algorithm uses DB of backlogged packets and $OTT_{j\ eff}^{i}$ of each path to the destination for scheduling traffic on available link. A brief summary of the pre-scheduling scenario is described now.

Let there are n ongoing sessions, each with a possibly distinct DB value d_i. It is assumed that scheduling process is fast enough so as the backlogged traffic of each queue never exceed one maximum length packet. It is also assumed that the token allocated to each flow is equal to the minimum packet length, on the worst. A concurrent queue handling process maintains backlogged queues in order of their DB value (in ascending order). The queues are served on parallel $\mathcal{LR}$ servers in their defined order irrespective of their current token value. This means that a queue shall be served irrespective of its token counter (provided it has non-negative value) to minimize processing complexity. In case, backlogged packet is larger than the token value, it is still served at its turn and the token value in all cases, is subtracted by the packet length. This ensures that an over serviced queue shall not be served in sub-sequent scheduling round(s) unless token counter attains non-negative integer value. This particular feature is similar to multi-server round robin (MRR) scheduling. Figure 5-4 shows pseudo-code of the proposed scheduling algorithm.

The decision regarding allocation of a particular $\mathcal{LR}$ server to a backlogged packet is based on the minimum $OTT_{j\ eff}^{i}$amongst all active links. Hence, smallest DB valued packet is scheduled on the best $OTT_{k_{eff}}^{i}$valued link and the $OTT_{k_{eff}}^{i}$value of the link is incremented by the Frame duration F_i of the link. The purpose of this addition is to indicate that the next probable schedule on this link shall be expensive in terms of E2E delay by a minimum of F_i time units. The time elapsed after the completion of scheduling round automatically removes the added F_i factor in the next round.

The token generation process for each backlogged queue is based on TBR with constraint parameters (ρ_i, σ_i), where ρ_i is token generation rate (equivalently; traffic arrival rate), in accordance with the service requirements of the queue and constraint by the available data rate. The σ_i represent the maximum bucket size to constraint burst transmission. The token generation, $OTT_{j_{eff}}$ estimation, DB based queue sorting and clock management processes are kept separate from the scheduling algorithm and run concurrently. The main loop of scheduling algorithm completes transmission of a complete packet in a single iteration. This means that the complexity of the algorithm will be constant. This eventually reduces the complexity of $AMDO$ to a constant *O(1)*.

Input: $P^t, d_i^t, T_i^t, F_i,\ OTT_{j_{eff}}^{i}$

Output: Schedule for all backlogged packets

${k_{OTT}}^{best} = Min(OTT_{1_{eff}}^{i}, OTT_{2_{eff}}^{i}, .., OTT_{n_{eff}}^{i})$*;choose index of best OTT link*

$i = 0$*; The first backlogged queue(Highest priority)*

$While\ (P^t\ is\ Not\ Empty)$ *;Repeat until all queues served*

  $if\ (\ T_i^t > 0)$ *; serve queues with positive* T^t

    $S_k^t = S_k^t \cup\ p_i^L$; $Schedule\ p_i^L\ on\ Link\ k$

    $T_i^t =\ T_i^t - L$

    $OTT_{k_{eff}}^{i} =\ OTT_{k_{eff}}^{i} + F_k$*;Increase the OTT value of this link*

    $p_i^l = 0;\ Empty\ p_i^L$

  $end\ if$

  Increment i; Service next queue

  ${k_{OTT}}^{best} = Min(OTT_{1_{eff}}^{i}, OTT_{2_{eff}}^{i}, .., OTT_{n_{eff}}^{i})$*;choose index of best OTT link*

$Loop\ While$

Figure 5-4: $AMDO$ scheduling algorithm

5.5. Simulation Results

The performance of the proposed ASM and $AMDO$ scheduling schemes was tested in Matlab environment as well as ns-2 simulator for validation. The proposed analytical models of chapter 3 and 4 were implemented in Matlab environment whereas the impact of topological and nodal parameters, traffic load and mobility was monitored in *ns-2* simulator. The following section describes the simulation environment.

5.5.1 Simulation Setup

For simulation purposes, the system model of Figure 2-2 was mimicked in ns-2. The two MMD nodes were re-designed in terms of node configurations for the proposed ASM and $AMDO$ multi-server scheduling schemes. The intermediate nodes were used as GRC servers with same queuing capacity. These modifications were also based on some configuration changes in the traffic shaper and regulator services. The capacity estimation block were implemented in the ASM through a capacity estimation technique, proposed in one of our previous work. The GRC servers provide service guaranteed on the basis of predefined traffic patterns controlled through shapers and regulators. The traffic sources were shaped and controlled through TBR, using differentiated services module of ns-2 (Karandikar 2003). Four different sets of flows {1, 2, 4, & 8} were generated to test the scalability of the proposed model. New admissions were not allowed to enhance the impact of capacity depletions and additions during handover initialization and completion operations, respectively. Three simultaneous tunnels between MMD and HA were created through Mobile IP (Perkins 2002). These tunnels form the multiple paths for the traffic between the MMD and HA. For simplicity the measurement were recorded at the tunnel ends. It is assumed that in case of satisfactory performance over these tunnels, the ultimate E2E guarantees can be easily achieved. The normal schedulers of the MMD were replaced by ASM and $AMDO$ server to impersonate multi-server scheduling. The availability of physical link was accomplished through the MIH event service that provides timely information for the link layer changes (IEEE Standard 802.21 2009). The capacity was also modified through link quality change events to have a realistic state of capacity availability during mobility. All the intermediate nodes between the MMD and HA were configured to GRC servers through modification in scheduling algorithm.

The VBR traffic sources were chosen for testing the usefulness of proposed scheme. The VBR sources closely model the dynamics of audio and video traffic. Video traces were used to model realistic EBB traffic source. The average rate of video transmission was assigned at 3Mbps. The actual traces of Jurassic Park I was best matching this capacity with an average frame size of 1500 bytes. This capacity was used to play high quality MPEG-4 coded video. The overall capacity of the system of three paths through 3 802.11 WI-Fi links was integrated to be sufficient to handle 8 simultaneous flows of high quality. The traffic bursts were additionally generated through exponentially bounded burstiness (EBB) sources with much faster inter-arrival time of packets as compared to the mean of the distribution (D. Starobinski 2000). This experiment was used to study tail-end distribution of VBR traffic sources. The flow rate was determined by the number of packet of a given maximum length. In case of inter-arrival rate of 10 ms, it comes out to be around 80 to 120 packets of maximum length of 1500 bytes. The inter-arrival time of the packet at source was modeled through exponential distribution with a mean value of 10 milliseconds (ms).The metrics of interest during the simulation study were the E2E delay, its variation over the life cycle of a flow, OOS reception with corresponding buffer occupancy density and packet drop events with fixed number of buffers for each flow at the receiver. These metrics have close coupling for QoS guarantees since we need lower packet drop rate, lesser computing and buffering complexity which is caused by OOS arrival. The extent of OOS arrival, caused by the higher E2E delay variations not only raises buffer occupancy but is also probable source of higher packet drop rate.

Figure 5-5 shows a plot of cumulative distribution of E2E delay variations in multi-path flows. The ASM server schedules traffic from batches of 1, 2, 4 and 8 distant flows, independently over multiple available interfaces. The mean values of E2E delay observed for the given set of flows are 130, 133, 135 and 140 ms respectively. It can be noticed in the plot for a single flow that the variations in E2E delay are range bounded between -25 to +30 ms, with 80% of packets reaching the destination within ranges of ±10 ms delay variation. In case of 2, 4 and 8 flows, the delay variation ranges between -40 to +45, -60 to +70 and -70 to +85 ms; respectively. However, the above mentioned ranges are the worst case readings with majority of the packets reaching the destination in a much narrower ranges. The analytical bound for 8 flows also indicate an overwhelmingly high number of packets reaching below the upper bound. It can also be noticed that the upper-bound is range-bounded between ±50 ms; and more than 90% of packet reaches within this range.

The tail distribution of remaining 10% packets is studied with respect to EBB (D. Starobinski 2000) and discussed later in this section. It is important to note that *ASM* server schedules packets on the basis of match making between the steepest deadline and the earliest available channel. Hence no additional processing is carried out to find the best path for a specific packet. A significantly high number of packets following the normal ranges of E2E delay also highlight effectiveness of proposed multi-server scheduling algorithm.

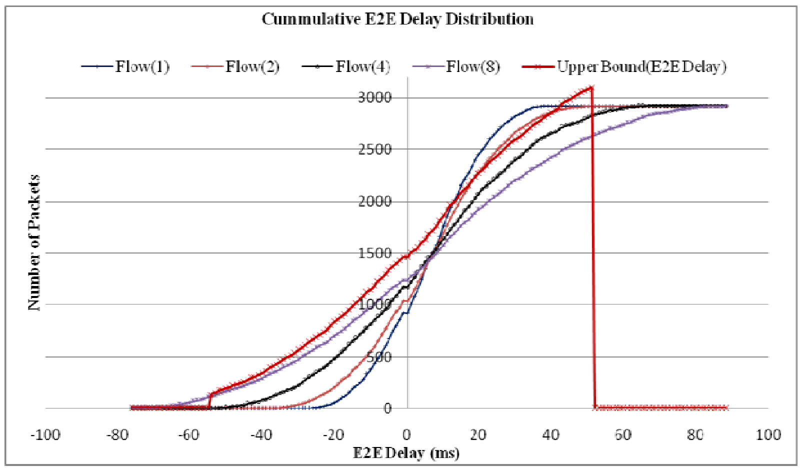

Figure 5-5: Distribution of delay variation during multi-path scheduling of flows through

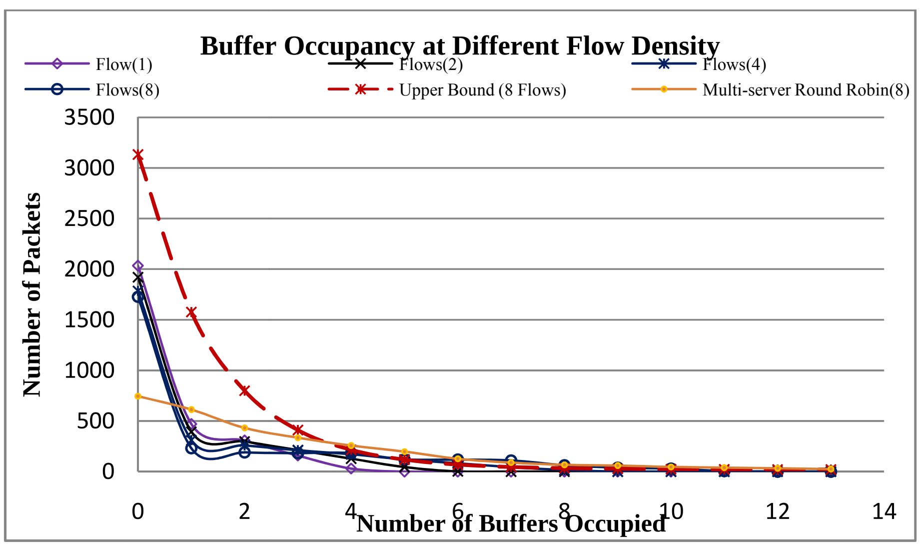

Figure 5-6: Distribution of buffer occupancy during multi-path scheduling of flows through

In Figure 5-6, we plot the distribution of buffer occupancy at the receiver at different sets of simultaneous flows. The buffer occupancy describes the impact of delay variations on managing the buffers for in-order delivery of packet to the destination. The *ASM* is plotted in comparison with *MRR* scheduling. The plot indicates that the significant number of packets for all the four sets of flows, exhibit low buffer occupancy; in the range of 3 to 6 buffers. This leads us to use a buffer size of 6 to take care of in-order delivery for 99%, 95%, 90% and 86% of the packets transmitted with simultaneous load of 1, 2, 4 & 8 flows respectively. The MRR slows a low density of packets at lower level of buffer occupancy. The higher buffering levels may definitely reduce packet-drop rate but increases the cost of buffering and may also not be scalable due to exponential increase of buffering with the increase in number of flows. The plot also reflects the usefulness of *ASM* server in scheduling packets irrespective of their upper layer binding to achieve a more pragmatic virtual channel comprising of multiple heterogeneous channels.

Figure 5-7 plots the packet drop at different buffer sizes per flow. Two buffer sizes of 4 and 6 were used to monitor the impact of buffering on packet drop in scenarios where OOS arrival may be high. The four sets of flows i.e. 1, 2, 4 and 8 flows were tested separately to assess scalability of system as well. The scalability is monitored through increased load of traffic flows with negligible increase in performance indicators. The results of the *ASM* simulations are compared with the *MRR* and Upper Bounds of the *ASM* deterministic model. It can be seen that the impact of increase in buffering level reduces packet drop significantly. The packet drop for 4 simultaneous sessions is around 10 % at buffering level of 4, which could be considered fine for video and audio traffic. The packet drop reduces considerably low to around 6% for the 4 sessions at the buffering level of 6. Similarly the packet drop is also under 10% for 8 concurrent sessions at buffering level of 6. The upper bound for the possible packet drop, acquired through the analytical model shows strict alignment with the results. A minor anomaly can also be seen at the buffer level of 4 with four simultaneous flows, where experimental results just exceed the upper bound. The trend shows a further decrease of packet drop at the buffering levels of 8 and 10, but the trade-off between the buffering cost and the added benefit favors lesser buffering to keep mobile routers more scalable with lesser processing complexity. It is also important to note that the packet drop rate for MRR is higher than the ASM upper-bound as well as simulations.

Figure 5-8 plots the buffer occupancy during EBB arrival. We take some burst arrivals in the range of 50 to 100 packets to quantify worst case performance limits of the *ASM* scheduling scheme. It can be noticed that the buffer occupancy for 4 flows is around 4 buffers for more than 90% of the traffic. For 8 flows, buffer occupancy rarely exceeds value of eight. This signifies the fact that an increase of buffering level to 8 can reduce the packet drop to almost zero but the 8 buffers will remain mostly vacant and reduce the scalability of the scheme. Further, higher buffering levels are also not permissible due to the time critical nature of packet. The packet is useful only if it comes within allowed time for the real-time applications. The upper bound exceeds forecasted value achieved through analytical model for 4 and 8 flows at the tail distribution. This behavior is primarily based on the fact that the analytical model is based on TBR shaped traffic that has relatively steady inter-arrival. Hence a higher buffering may not be very useful despite adding significant cost and complexity in the system.

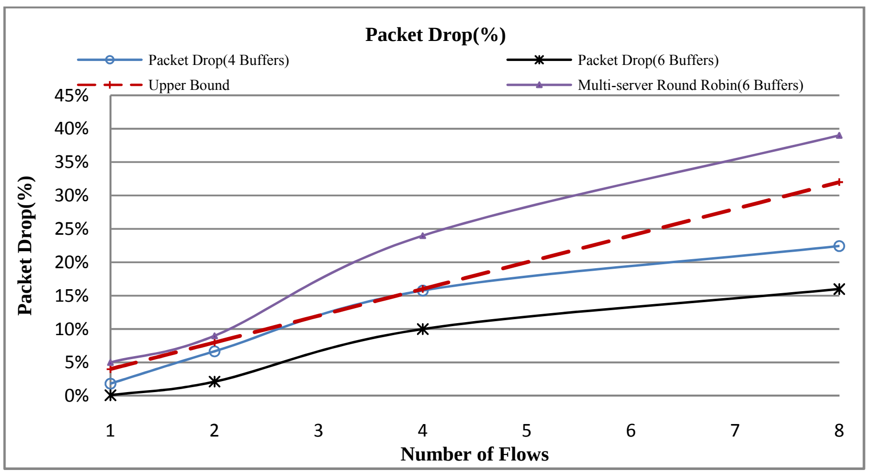

Figure 5-7: Packet drop (%) behavior during multi-path scheduling of flows through ASM

Figure 5-9 plots the packet drop behavior during burst arrival. It can be noticed that the packet drop for 8 flows is 14% and 12% at buffering levels of 4 and 6 respectively. It is noteworthy to recall here that the behavior depicted in the graph is a tail distribution and it occurs very few times as compared to overall behavior of the system. The upper bound graph indicates anomaly at lower buffers that is perceivable under the assumption that the higher buffer occupancy persisted for EBB arrival due to less time to wait for the late arrivals. A dynamic management of buffers could be one possible solution to reduce the packet drop and increase the buffer utilization. The plot shows a consistently stable behavior of packet drop at higher number of buffers i.e. 6 in this case. All these results presented above have shown tight coupling of E2E delay variation with OOS reception and correspondingly, with buffer occupancy and packet drop rate. A higher E2E delay variation raises chances of higher OOS reception. Similarly; a higher OOS reception causes higher buffer occupancy for flows that are constraint by in-order delivery. Finally, in a finite buffering system the chances of packet drop increases due to the depletion of buffer (caused by higher buffer occupancy) that necessitate flushing of buffers for subsequent arrivals. In comparison to ASM, the MRR scheduling shows much lower number of packets at lower buffer occupancy.

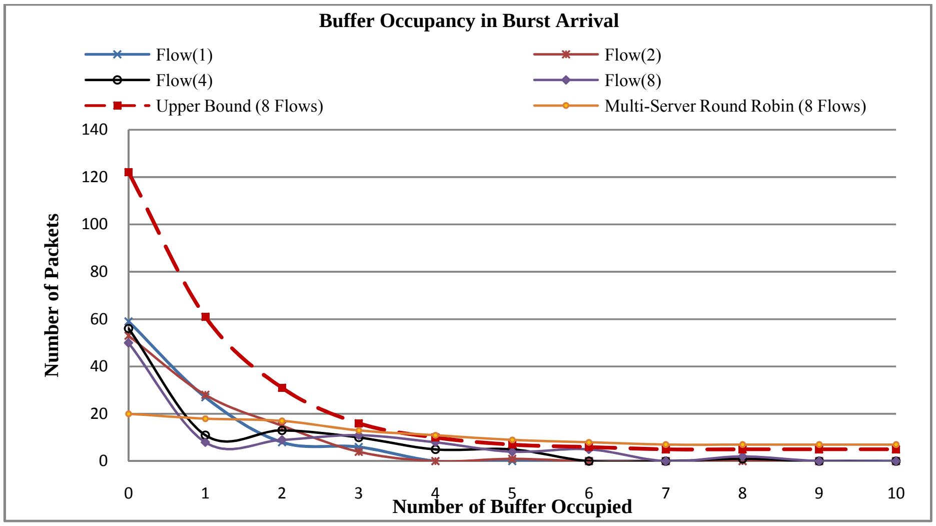

Figure 5-8: Distribution of buffer occupancy during service of burst-arrival at $\boldsymbol{ASM}$

Figure 5-10 shows a plot of E2E delay distribution of $AMDO$ scheduling in comparison with earliest delivery path first $(EDPF)$ and multi-server round robin (MRR) schemes (Xiao 2004). It can be seen that the $AMDO$ has packet delivery time of less than 160 msec for about 90% of traffic it schedules. In comparison, $EDPF$ and MRR achieve same delivery statistics at 170 milli-seconds (ms), and 177 ms; respectively. This provides a significant performance edge in favor of $AMDO$ as a lower delivery time produces higher throughput as well as lower inter arrival delay variation. This result is also useful for the interactive video sessions which require a consistently low $E2E$ delay of around 150 msec. The major reason for this performance edge is mainly due to the multi-hop delay monitoring and estimation capability of $AMDO$ whereas, $EDPF$ uses a single-hop (wireless section) best link selection policy and MRR does not take into account any link characteristics while scheduling traffic.

Figure 5-11 presents a plot of reorder complexity at the receiver against the extent of E2E delay variation. By extent of delay variation, we mean the maximum delay a packet experiences in access of its expected arrival. The OOS arrival of packets raise repeated reorder before servicing packets to applications and in case of longer extent of such pattern, there can be multiple inner reorders, within the scope of a packet delayed arrival. The results show that the cumulative reorder complexity of the $AMDO$ is much lesser as compared to the $EDPF$ and MRR. The reorder complexity is a simple measure of number of reorder needed in the overall session run. This result indicates about 40% and 120% lesser reorder for $AMDO$ in comparison to $EDPF$ and MRR respectively. The reduced reorder complexity is a definite performance edge for the QoS aware sessions.

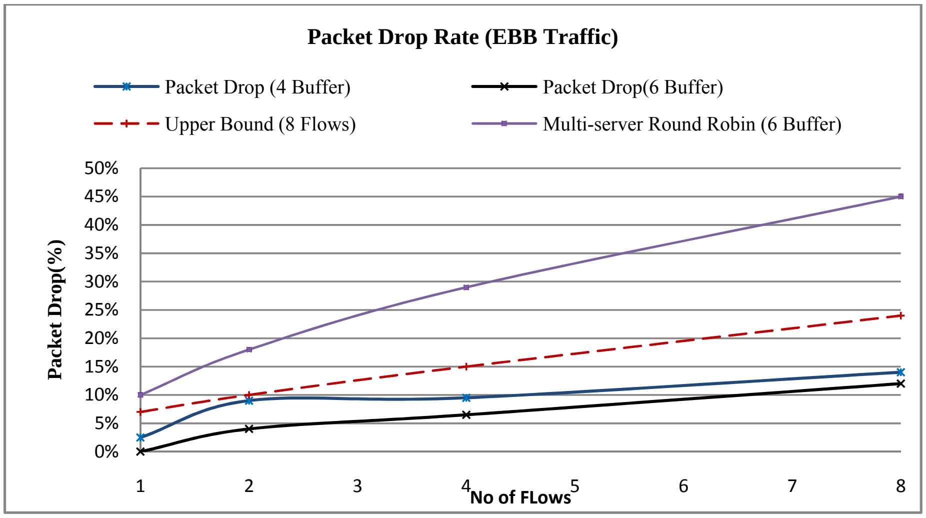

Figure 5-9: Packet Drop behavior during burst arrival

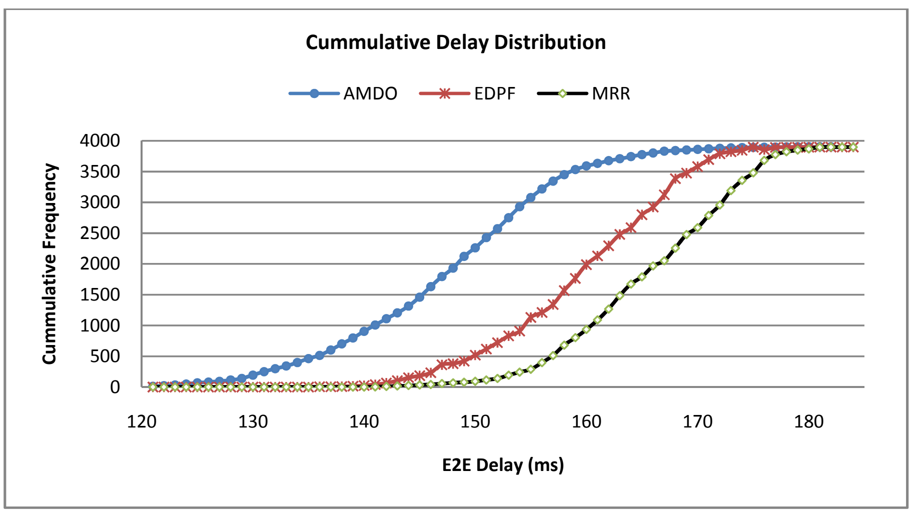

Figure 5-10: Cumulative distribution of E2E delay using AMDO, EDPF and MRR scheduling schemes

Figure 5-12 compares packet drop against E2E delay variations at different buffer sizes. The buffer size is measured in terms of numbers of max-sized packets. It is noticeable that the packet drop rate rises more or less linearly with increase in delay variations. In case of a buffer size of 20 packets, the packet drop rate is about 5%; even at very high delay variations of upto 15%. Even at 20% delay variations, drop rate is not more than 5%, which is highly acceptable in stored video streaming applications. On contrary, at small buffer sizes the packet drop rate is higher, and it reaches to highest levels of about 27% at the buffer size of 1 packet. The main reason behind this high drop rate is the occupancy of an OOS packet arrival which cannot be served at present to make room for new arrival. Such kind of behavior may exist in interactive sessions which required fast in-time arrival without buffer support and cannot afford longer buffer waits and reader delays. Intermediate buffer sizes of 5 and 10 packets perform well up to 8% and 17% delay variation that makes a good trade-off between performance and buffering cost. It's important to note that smaller buffer sizes also reduce the reorder overhead.

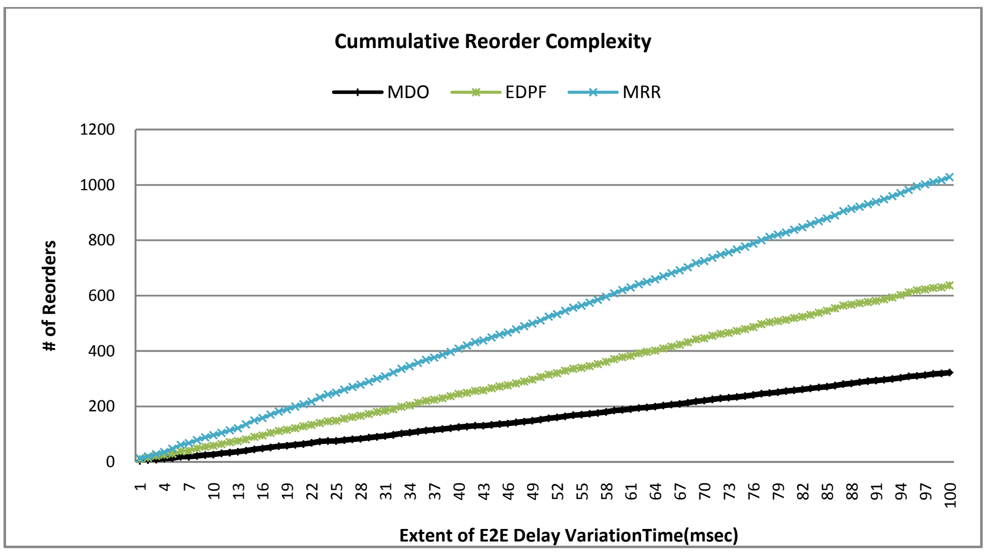

Figure 5-11: Reorder complexity against extent of delay variations

Figure 5-13 plots cumulative throughput achieved through the proposed scheduling scheme at different session loads. The plot shows a consistent increase with added load of sessions. The throughput of 2.8 MB/s is a significant indicator of high throughput virtual channel/path comprising of three heterogeneous networks. The linear rise of cumulative throughput is a sound indicator of stability of the system at higher load conditions. It also shows that proposed scheme is highly scalable and added load does not strain the overall throughput of the system.

The E2E delay behavior of the proposed ASM scheduling scheme is also measured at varying traffic load. Figure 5-14 plots the mean E2E delay behavior against increased number of flows, at different path lengths. It is noticeable in this plot that in case the path length is the same i.e. when maximum hop distance is zero, the performance of the system shows very slow increase in mean E2E delay and rarely reaches at 150 ms. Even in case the path length differs by one and two node, the mean E2E delay varies slowly with increased number of flows. The E2E delay shows slight divergent behavior when the path length is more than two hops. The delay experienced in this scenario, at additional number of flows not only increases but also increases the buffer-hold time for the low E2E delay packets waiting for in-order forwarding. This result indicates the proposed scheme can be more effective in situations where path length is equal or does not vary by more than two nodes. The plot supports the suitability of ASM multi-server scheduling for delay-bounded QoS provisions during mobility. This supported favorably to another important question of the thesis hypothesis.

Figure 5-15 plots a mean E2E delay at increased traffic load measured at different hop lengths for AMDO scheduling scheme. The behavior here is slightly different as the mean E2E delay is slightly higher in this case but does not show diverging trends even when the hop distance between the paths is more than 2. Since the major decision for scheduling is based on the E2E delay of each path, the traffic load on longer paths or the paths with longer E2E delay is minimized. The E2E delay tolerance for different applications may be different and situations where the E2E delay of 180ms is acceptable, AMDO may be a better performer. It is important to note that the mean E2E delay is taken for each flow to depict general per flow behavior. This plot also supported favorably to the successful design of suitable scheduling strategies for QoS guarantees during mobility, as stated in thesis hypothesis and investigated through analytical models.

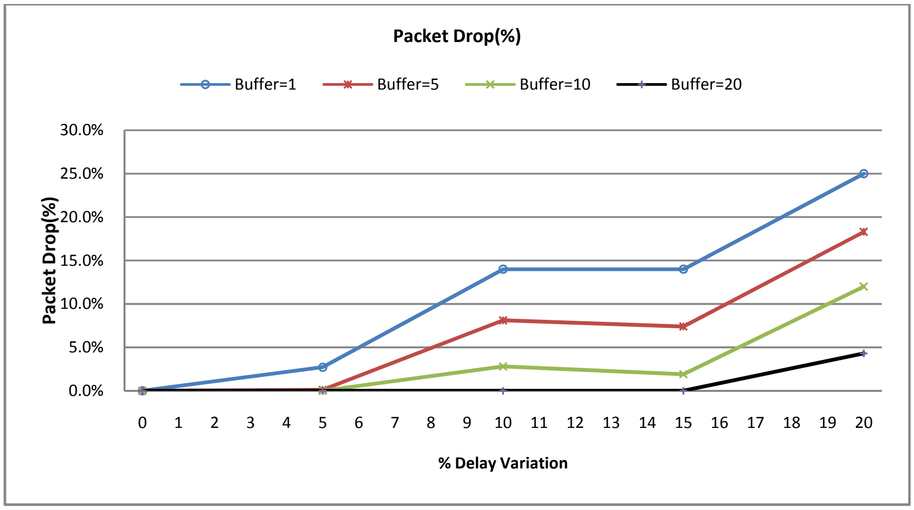

Figure 5-12: Packet drop percentage at different levels of delay variation

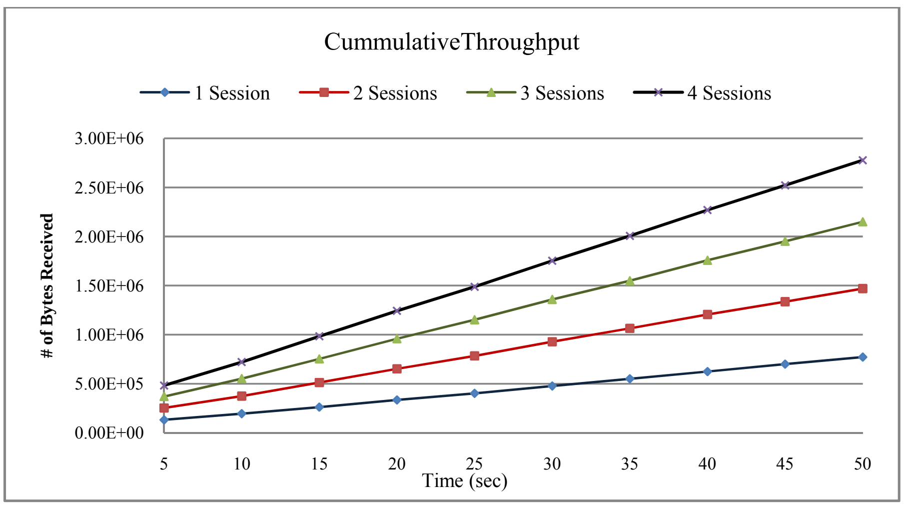

Figure 5-13: Cumulative throughput during simulation runs at different session load

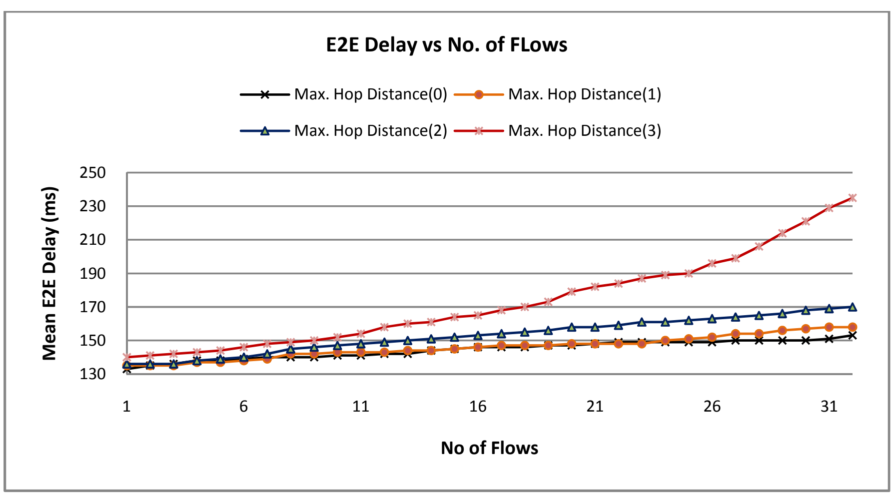

Figure 5-14: Mean E2E delay behavior at different traffic load in terms of no. of flows for ASM

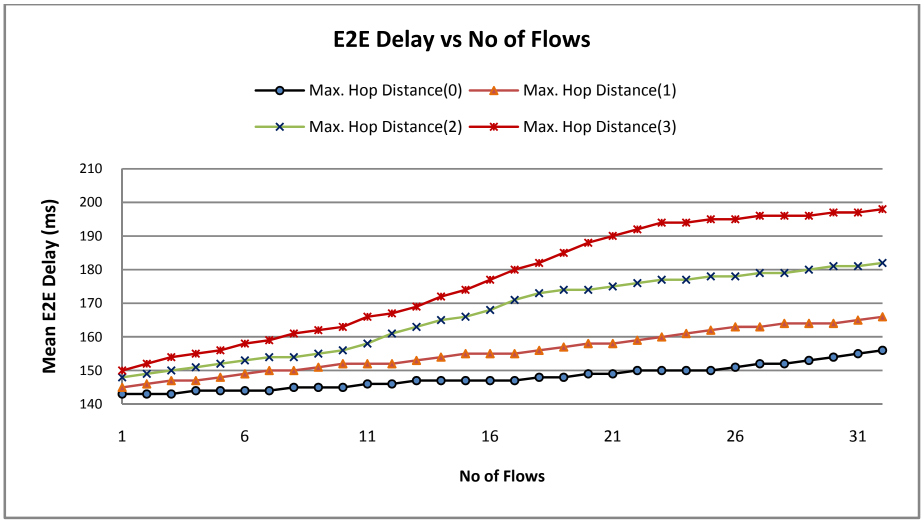

Figure 5-15: Mean E2E delay behavior at different traffic load in terms of no. of flows for AMDO

Chapter 6

CONCLUSIONS

This thesis proposes some techniques for meeting QoS demands of mobile sessions over heterogeneous wireless and mobile networks. An MMD experiences such heterogeneous environment through its multiple wireless interfaces. The most common use of such environment is in the 3G and beyond wireless networks that provide alternate strategies of vertical handovers during mobility for better service guarantees and traffic load management. Instead of vertical handovers, we investigated simultaneous use of more than one available interfaces of MMDs to aggregate their capacities. It has been generally perceived that the MMDs can improve their performance gain through simultaneous use of multiple available networks. This gain is assumed to be much higher than any of alternate strategies; such as, VHO. In this thesis, the space of perceived performance gain has been investigated through analytical and stochastic modeling. Based on these models, adaptive scheduling strategies are proposed to maximize service guarantees during mobility. The main focus of this thesis is on estimating possible OOS reception of packet of a flow, in case of multipath traversal to reach destination. The causes of OOS reception and its possible implications on individual flows as well as communication services have been studied. The proposed models have been fueled to evolve suitable scheduling strategies to enhance QoS provision for individual flows during mobility.

The simultaneous use of multiple interfaces has assorted dynamic at different architectural layer of TCP/IP network protocol. In a broader sense, it has been inferred that the simultaneous use of multiple interfaces can either be controlled by reactive or proactive flow control strategies. The reactive strategies are generally adopted at upper layers; such as, transport and application layers, whereas, proactive approaches are more appropriate at layer-2 or layer-3. The reactive strategies are found friendlier to the individual flows, where a flow can gain more share of communication resource, in case it is available due to lesser congestion in the core network. In contrast, proactive strategies are more controlled in terms of resource utilization, as these approaches allocate desired resources to each flow and manage these requirements over the multiple available links in a controlled admission control environment. This thesis has sought a proactive approach to maximize service requirements in a multi-path flow management scenario at layer 3, with the help of mobility event triggers from layer-2.

In this thesis, multi-path flow is analyzed under the constraint of QoS provision during mobility with minimized OOS reception at the destination. The deterministic and stochastic analytical modeling has been proposed to analyze BAG/CAG applications. The proposed models are novel in multi-server scheduling environment and have found useful upper bounds for the important multi-path flow metrics, like E2E delay, E2E delay variations, OOS reception, and buffer occupancy to ensure in-order forwarding of packets to the applications. The core metrics of interest is OOS reception caused by path diversity. Since path diversity contributes to E2E delay variation, the key focus includes analysis of E2E delay variations and other metrics are direct derivation from it. The stochastic analytical model is based on bounded variance network calculus which has proven strength of simplicity in handling complex E2E scenarios. The bounds achieved through stochastic models provide thresholds of violation of given metric and providing sufficient cushion for maximizing service guarantees during mobility. The results of analytical models are validated through simulation in ns-2 environment.

It has been found that under guaranteed rate service paradigm, the continued provision of service to mobile flows is possible through simultaneous use of multiple available wireless interfaces. The OOS reception of packets from same flow is imminent due to E2E delay variations which in turn are caused by transmission delay variation of different sized packets and the path length. The delay at individual hops has been upper-bounded but the convolution of this delay with delay experience at other hops is the major contributor of E2E delay variations. It has been found that the delay variations are generally with-in inter-arrival time of the packet at source which in turn moderates the impact of OOS reception as buffer occupied by the OOS received packet are quickly disposed off. It has also been found that by allowing slight leniency in violation of QoS based threshold in favor of network (about 10%), the scalability of service provision increases significantly (about 25%). It has been found that static scheduling strategies in such environment may not provide service guarantees due to mobility and need adaptive approach to take care of physical link changes. The ASM and $AMDO$ scheduling strategies, devised to take care of link layer changes are more efficient in term of service maximization and absorbing multi-path asymmetries. The proposed scheduling schemes are compared with the multi-server round robin approaches. Despite higher control, the complexity of the both algorithm is very low; i.e. $O(1)$. We have also evaluated major characteristics of ASM scheduling algorithm with respect to server latencies, OOS arrival and fairness and have proved that the proposed scheme adheres to tight delay and OOS arrival bounds. It is also fair for all competing sessions and unfairness can only extend up to the difference of packet sizes of contending packets/flows. The simulation results show that the performance of the scheme is significantly higher with other similar proposals. The results achieved have shown that the efficient design of some of the core components used in this mechanism, such as real-time capacity estimation, optimum scheduling algorithms and tunneling mechanism greatly enhances its utility in IP based mobile networks.

The thesis hypothesis was favorably supported by the proposed models as well as scheduling strategies. All the questions raised in the hypothesis were successfully tested and supported by the results. The last question of the hypothesis regarding the optimization of such approach needed further investigation, hence is not answered in this thesis and proposed as the future research work. The major challenge in optimization lies in parameterization of optimization parameters and complexity of gaining optimized performance in more than one parameter.

6.1. Future work

The multi-server scheduling of traffic is a relatively new paradigm in packet switching networks and the studies so far conducted can be improved in many aspects. Following is a brief summary of planned future work in this domain.

The stochastic modeling of E2E metrics is a highly complex model and we have only reduced its complexity by considering special case scenarios of bounded variance network calculus. A more comprehensive analysis model is possible with higher degree of computing power. In future we plan to develop such model on the basis of MVA to further optimize the performance of scheduling algorithms.

In this thesis, we have used delay deadline model of service guarantees to reduce the complexity of nodal delay analysis process. The same could also be done through queuing analysis with higher complexity but with more accurate service upper-bounds. The changing states with the service of each backlogged packet can be modeled through Markovchains at node level. The implication of E2E analysis of such model has very high number of states that can be solved through higher computing power.

The current approach of the network service model can also be extended and compared analytically with any of transport layer protocol supporting multi-homing. The E2E model at transport layer reduces the possible states of the system to much lower level provided congestion window is evaluated separately.

LIST OF PUBLICATIONS/PROCEEDINGS

Journal Publications

J-1. Syed Zubair Ahmad, Muhammad Abdul Qadir, Muhammad Saeed Akbar and Abdul-Aziz Boras; Analysis of Multi-Server Scheduling Paradigm for Service Guarantees during Network Mobility**,** Published in Wireless Personal Communications, DOI: 10.1007/s11277-010-0114-5. Online Version, Publisher Springer. Oct 2010

J-2. M. S. Akbar, S. Z. Ahmad, M. A. Qadir; Optimization of Transmission Control Protocol Performance in Heterogeneous wireless networks, Published in International Journal of Computer Science and Network security,South Korea, ISSN 1738-7906, Vol. 8. No. 7, July 2008

Conference Proceedings

P-1. Syed Zubair Ahmad, Muhammad Abdul Qadir, and Abdelaziz Bouras; An Opportunistic Multi-server Scheduling Approach for Service Guarantees through Mobile Routers, In proc. of IERIA conference held in Venice, Italy (18-25 July, 2010)

P-2. Syed Zubair Ahmad, Muhammad Abdul Qadir, Muhammad Saeed Akbar; Service optimization of in-elastic flows stripped over multiple asymmetric mobile channels, In proc of 13th INMIC 2009, Print ISBN: 978-1-4244-4872-2, DOI: 10.1109/INMIC.2009.5383124, 14-15 Dec. 2009.

P-3. Syed Zubair Ahmad, Muhammad Abdul Qadir, Muhammad Saeed Akbar; Dependable Quality-of-Service Provisioning through Aggregate Scheduling of Multiple Service Classes on Heterogeneous Wireless Channels, In proc. of International Conference on Future Information Networks, ICFIN 2009. Page(s): 167 -172, DOI: 10.1109/ICFIN.2009.5339576, Beijing, China, Oct. 2009.

P-4. S. Z. Ahmad, M.S. Akbar, M.A. Qadir; A distributed resource management scheme for load-balanced QoS provisioning in heterogeneous mobile wireless networks, In proc. of 4th ACM symposium on QoS and security for wireless and mobile networks, held in Vancouver, British Columbia, Canada, October 2008, Pages 63-70.

P-5. S. Z. Ahmad, Akbar, M.S. Qadir, M.A.; Towards Dependable wireless networks- a QoS constraint resource management scheme in heterogeneous environment, In proc. of 4th IEEE International Conference on Emerging Technologies. ICET, Held in Rawalpindi, October 2008, page(s): 182-186.

P-6. S. Z. Ahmad, M. A. Qadir, M. S. Akbar; Intellegent Resource Management through Reliable Transport of Mobility related Services for Vertical Handovers, In proceedings of 11th International Symposium on Wireless Personal Multimedia Communications, 08-11 September, 2008, Finland.

P-7. M. S. Akbar, S. Z. Ahmed, and M. A. Qadir; Mobility Aware Transmission Control Protocol (MA-TCP); In proceedings of 4th ICMU 2008 Tokyo, Japan

P-8. S. Z. Ahmad, M. S. Akbar, M. A. Qadir, A Cross-Layer Vertical Handover Decision Model for Heterogeneous Wireless Networks, 4th IEEE, International Conference on Innovations in Information Technology, Innovations 2007, 18-20 November 2007, Dubai UAE

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