CAPITAL UNIVERSITY OF SCIENCE AND TECHNOLOGY, ISLAMABAD

ADAPTIVE SIMULTANEOUS MULTIPATH TRANSMISSION SCHEMES

by

Samiullah Khan

A thesis submitted in partial fulfillment for the degree of Doctor of Philosophy

in the

Faculty of Computing
Department of Computer Science

October 2017

Copyright © 2017 by Samiullah Khan

All rights reserved. No part of this thesis may be reproduced, distributed, or transmitted in any form or by any means, including photocopying, recording, or other electronic or mechanical methods, by any information storage and retrieval system without the prior written permission of the author.

This is for you, Adaey(Mom) and Abo Je(Dad). Thanks for always being there for me.

CAPITAL UNIVERSITY OF SCIENCE & TECHNOLOGY
ISLAMABAD
Expressway, Kahuta Road, Zone-V, Islamabad
Phone:+92-51-111-555-666 Fax: +92-51-4486705
Email: info@cust.edu.pk Website: https://www.cust.edu.pk

CERTIFICATE OF APPROVAL

This is to certify that the research work presented in the thesis, entitled "**Adaptive Simultaneous Multipath Transmission Scheme**" was conducted under the supervision of **Dr. Muhammad Abdul Qadir**. No part of this thesis has been submitted anywhere else for any other degree. This thesis is submitted to the **Department of Computer Science, Capital University of Science and Technology** in partial fulfillment of the requirements for the degree of Doctor in Philosophy in the field of **Computer Science.** The open defence of the thesis was conducted on **25 October, 2017.**

Student Name : Mr. Samiullah Khan (PC083003)

The Examination Committee unanimously agrees to award PhD degree to Mr. Samiullah Khan in the field of Computer Science.

Examination Committee :

(a) External Examiner 1: Dr. Muhammad Sher, Professor, IIU, Islamabad

(b) External Examiner 2: Dr. Ghulam Abbas, Associate Professor, GIKI, Topi

(c) Internal Examiner : Dr. Muhammad Azhar Iqbal, Assistant Professor, CUST, Islamabad

Supervisor Name : Dr. Muhammad Abdul Qadir, Professor, CUST, Islamabad

Name of HoD : Dr. Nayyer Masood, Professor, CUST, Islamabad

Name of Dean : Dr. Muhammad Abdul Qadir, Professor, CUST, Islamabad

AUTHOR'S DECLARATION

I, **Mr. Samiullah Khan (Registration No. PC083003)**, hereby state that my PhD thesis titled, '**Adaptive Simultaneous Multipath Transmission Scheme** is my own work and has not been submitted previously by me for taking any degree from Capital University of Science and Technology, Islamabad or anywhere else in the country/ world.

At any time, if my statement is found to be incorrect even after my graduation, the University has the right to withdraw my PhD Degree.

Samiullah

(Mr. Samiullah Khan)

Dated: October, 2017 Registration No : PC083003

PLAGIARISM UNDERTAKING

I solemnly declare that research work presented in the thesis titled "**Adaptive Simultaneous Multipath Transmission Scheme**" is solely my research work with no significant contribution from any other person. Small contribution/ help wherever taken has been duly acknowledged and that complete thesis has been written by me.

I understand the zero tolerance policy of the HEC and Capital University of Science and Technology towards plagiarism. Therefore, I as an author of the above titled thesis declare that no portion of my thesis has been plagiarized and any material used as reference is properly referred/ cited.

I undertake that if I am found guilty of any formal plagiarism in the above titled thesis even after award of PhD Degree, the University reserves the right to withdraw/ revoke my PhD degree and that HEC and the University have the right to publish my name on the HEC/ University Website on which names of students are placed who submitted plagiarized thesis.

Samiullah

(Mr. Samiullah Khan)
Registration No. PC083003

Dated: October, 2017

Acknowledgements

Foremost, I would like to express my sincere gratitude to my advisor Prof. Dr. Mohammad Abdul Qadir for the continuous support of my PhD study and research, for his patience, motivation, enthusiasm, and immense knowledge. His guidance helped me in all the time of research and writing of this thesis. I could not have imagined having a better advisor and mentor for my PhD study.

Besides my advisor, I would like to thank the rest of my PhD thesis committee: Prof. Dr. Amir Qayyum, Prof. Dr. Mohammad Sher, Prof. Dr. Ghulam Abbas and Dr. Muhammad Azhar Iqbal for their encouragement, insightful comments, and hard questions. My sincere thanks also go to Dr. Muhammad Tanvir Afzal and Engr. Khalid Mahmood (Director, Gradate Studies) for their guidance and sincere support in completion of the requirements for completion of my PhD.

Last but not the least, I would like to thank my family: my parents for supporting me spiritually throughout my life. Without their continuous support, it was not possible for me to achieve a lot of milestone in my career.

I am also especially thankful to all other members of Center for distributed and Semantic Computing (CDSC) especially Dr. Munir Ahmed, Dr. Syed Zubir and Mr. Tariq Ali for their motivation and suggestions. Many thanks to Higher Education Commission (HEC) for indigenous PhD grant, which enabled me to pursue my studies. Not to forget the anonymous reviewers of our papers, who gave us the direction to move forward, and to improve our research work.

List of Publications

Journal Papers

1. Khan, S. and Qadir, M. A (2017). Deterministic Time Markov Chain Modeling of Simultaneous Multipath Transmission Schemes, IEEE Access, DOI: 10.1109/ACCESS.2017.2701769, (Impact factor - 3.244).

2. Khan, S., Qadir, M. A, Khan F. A. and Rehman, E. (2017). Adaptive fast retransmission (AFR) with respect to receiver buffer (Rbuf) space in simultaneous multipath transmission (SMT) , Malaysian Journal of Computer Science, (Impact factor - 0.6).

3. Khan S., Qadir, M. A. (2015). Inter-Path OOS Packets Differentiation Based Congestion Control for Simultaneous Multipath Transmission, Accepted in International Arab Journal of Information Technology, (Impact factor - 0.582).

Conference Papers

1. Khan, S., Ahmed, S. Z., and Qadir, M. A. (2011). Throughput Enhancement of Simultaneous Multipath Communication Using Modified Fast Retransmit (MFR) Scheme. International Conference on Computer Networks and Information Technology 2011, Department of Computer Science, University of Peshawar-Pakistan, (pp. 9-12).

Abstract

The increase in the availability of multimode devices for ubiquitous network access and the need for larger bandwidth create thrust for utilization of simultaneous network connections. Unfortunately, the standard transport layer protocols like the transmission control protocol (TCP) and user datagram protocol (UDP) have architectural constraints due to which an Internet application can use only one interface at a time. The stream control transmission protocol (SCTP) provides support for concurrent multipath transfer (SCTP-CMT). Aggregated performance is analyzed with a number of experiments to measure the aggregated throughput of SCTP-CMT by using a very popular network simulator, NS-2. It is observed that the aggregated throughput is about 20% of the available aggregated bandwidth. The significant reduction in the aggregated throughput demands a careful scrutinization of its reasons.

After carefully analyzing and carrying out some further experiments, it is diagnosed that non-differentiation of missing packets into intra and inter-path, usage of traditional congestion window management for these missing packets and using static fast retransmit threshold which is independent of available receiver buffer space, are the main reasons for the aggregated throughput degradation. Simultaneous multipath transmission (SMT) schemes are proposed to handle the above mentioned issues with the intention to increase aggregated throughput by avoiding Rbuf blocking problem and efficient utilization of available Rbuf space. SMT-modified fast retransmit (SMT-MFR) and SMT-adaptive modified fast retransmit (SMT-AMFR) schemes are formulated for SCTP. To analyze the SMT-MFR in realistic network environments, a number of simulation scenarios are carried out. The initial results revealed that SMT-MFR has overcome Rbuf blocking with improvement in aggregated throughput ranging from 164% to 72.4% (from normal to worst scenario respectively). SMT- MFR is composed of two sender side modules, i.e., inter-path missing packet differentiation (IMPD) and multihomed congestion control (MCC). The IMPD module differentiates the missing packets according to its cause of missing such as, packet missing due to network congestion or due to multiple path effects. The MCC mechanism triggers the fast retransmit event with respect to the cause of the missing packet. The SMT-MFR has successfully overcome the Rbuf blocking problem, abnormal congestion window (cwnd) reduction and has improved the aggregated throughput.

Like the traditional congestion control mechanisms, the proposed SMT-MFR uses static fast retransmit threshold which is independent of available Rbuf space for intra and inter-path missing packets in order to enhance the redundant data transmission. This aggressive nature of SMT-MFR is tackled by proposing SMT-adaptive fast retransmit (SMT-AMFR) mechanism. SMT-AMFR categorizes available Rbuf space into critical, substantial and moderate zones. In case of inter-path missing packets; the SMT-AMFR uses three dynamic fast retransmit threshold values for these zones. This provides the arrival opportunity to the delayed packet, if the receiver has enough available Rbuf space. This increases the efficient utilization of available Rbuf space by decreasing the redundant data retransmission and thus increasing the aggregated throughput. The simulation results revealed that SMT-AMFR outperformed the SMT-MFR by having an average throughput gain of 19.8% and 16.9% in bandwidth and delay based disparity scenarios respectively. Deterministic-Time Markov Chain (DTMC) model is used to mathematically verify simulation results of SMT-MFR and SMT-AMFR. The analytical model results revealed that SMT-MFR outperformed SCTP-CMT by the 8.54% gain in average throughput. SMT-AMFR outperformed the SCTP-CMT and SMT-MFR by 19.65% and 11.15% gain in average throughput.

Contents

List of Figures

List of Tables

List of Algorithms

Abbreviations

Ack	Acknowledgment
a_rwnd	Advertised Receiver Window
AMCC	Adaptive Mulithomed Congesetion Control
AMFR	Adaptive Modified Fast Retransmit
BDP	Bandwidth Delay Product
C_ACK	Cumulative Acknowledgment
CMT	Concurrent Multipath Transmission
CMT-CQA	QoS-aware Adaptive CMT
cwnd	Congestion Window
Cwnd-MPs	Congestion Window Management Policies
DCCP	Datagram Congestion Control Protocol
dup_Ack	Duplicate Acknowledgment
DTMC	Deterministic Time Markov Chain
E2E Delay	End to End Delay
F_{RT}	Fast Retransmit
FTP	File Transfer Protocol
FPS	Forward Prediction Scheduling
IAP	Intra-path Missing Packet
HIP	Host Identity Protocol
IETF	Internet Engineering Task Force
IMPD	Inter-path Missing Packets Differentiation
IP	Internet Protocol
LS-SCTP	Load Sharing SCTP
MCC	Multihomed Congestion Control

mCMT	Mobile CMT
MFR	Modified Fast Retransmit
MHS	Multihomed Sender
MHR	Multihomed Receiver
MTU	Maximum Transmission Unit
NMT	Naive Multipath Transmission
NAT	Network Address Translation
NS-2	Network Simulator-2
OOS	Out Of Sequence
ODS	On Demand Scheduler
PID	Path Identity
PSNs	Path Sequence Numbers
Rbuf	Receiver Buffer
RTT	Round Trip Time
SACK	Selective Acknowledgment
Sbuf	Send Buffer
SCTP	Stream Control Transmission Protocol
SFR	Split Fast Retransmit
SSN	Stream Sequence Number
SSThreshold	Slow Start Threshold
SMT	Simultaneous Multipath Transmission
TCP	Transmission Control Protocol
TSN	Transmission Sequence Number
UDP	User Datagram Protocol
W-SCTP	Westwood SCTP
WM^2-SCTP	Wireless multipath multiflow SCTP
WCMT-SCTP	Wireless Concurrent Multipath Transfer

Symbols

ω	Congestion Window
τ	Slow Start Threshold
γ	Packet Loss
υ	Available Receiver Buffer
Q	Transition Probability
π	Steady-state Probability Distribution
ε	Sent Packets
Z	Positive Integer
d	Propagation Delay

Chapter 1

Introduction

1.1 Background

Recent advancements in ubiquitous networks and the proliferation of multimode devices have increased the connectivity in terms of both, the availability and increased throughput, and through simultaneous use of multiple network interfaces. These network interfaces can be used to provide bandwidth aggregation by simultaneous data transmission over multiple network interfaces by using multihoming. The multihoming is the facility where the end nodes with multiple interfaces are capable of connecting multiple networks, as shown in Figure 1.1. A multihomed sender (MHS) communicates using multiple paths (Path 1 to N) with a multihomed receiver (MHR). The multiple paths pass through the Internet represented by a cloud. This enables a multihomed device to select any path for data transmission, thereby providing load balancing and decreasing the congestion on a link. Multihoming also provides the reliability in a scenario where one path fails; the other companion path is still available to transmit the data traffic. Proper use of multihoming also adds an extra complexity. One of the complexities is the efficient utilization of these multiple network interfaces to maximize the overall throughput.

The multihoming capability can be provided at the application layer [2] at the transport layer [3],[4],[5] , [6] ,[7] ,[8] or at the network layer [9] of the protocol stack. Furthermore, new layers may be introduced to support multihoming as available in some proposals such as, the Host Identity Protocol (HIP) [10] or site multihoming by IPv6 intermediation (SHIM6). The newly introduced layers perform specific functionalities, and are aimed at reducing the resultant complexity due to multihoming mechanisms in the original protocol stack [11] .

The transport layer is preferred for multihoming due to maximum utilization of multipath benefits such as, reliability, congestion control, robustness, optimal network resource utilization and fast deployment [12]. The transport layer bandwidth aggregation requires only the end hosts to handle the processing in this regard. The intermediate nodes have to act only as relay machines which support the Simple Core, Smart Edges technology perspective of the Internet. The transport layer has end to end information of each path, such as, delay, packet drop etc., which are helpful for load sharing, fault tolerance, and congestion control. Transport layer manages congestion by shifting load to non-congested paths. In the worst case, when an Internet link fails, it is the transport layer which can take a decision about switching to the next feasible path in a best possible way. The information about these path features is essential to maximize the multipath benefits, which are not available at the network layer or at the application layer.

The Internet has a number of interlinking paths with intermediate nodes such as, router and gateways, etc. These interlinking paths have different path features such as, bandwidth, delay, and packet loss rate. If a multihomed sender and receiver select the best interlinking paths for data transmission, still these multiple paths may have a disparity in bandwidth, delay, and packet loss features. The simultaneous transmission of data on such multiple paths generates out of sequence packet (OOS). This study focuses on the efficient handling of OOS packet to optimally utilize the bandwidth of multiple paths.

Now the question crops up that which transport layer protocol should be used for multihoming. The best known transport layer protocols are user datagram

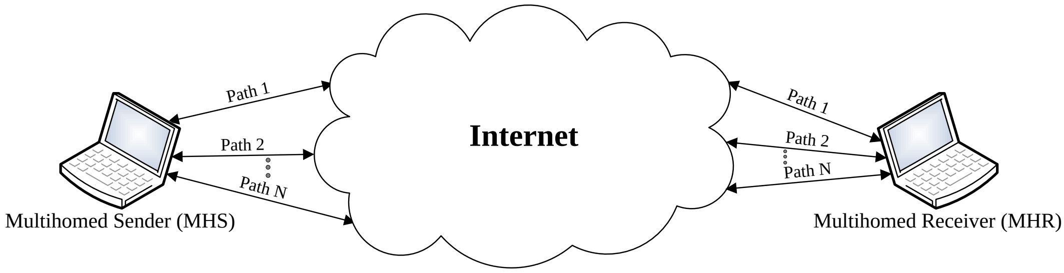

FIGURE 1.1: A network model of multihomed devices with forward and reverse direction

protocol (UDP) [13] and transmission control protocol (TCP) (Postel, 1981). One may think that these protocols should be used for multihoming. The UDP is a transport layer protocol, mostly used for real time data streaming. The UDP being a lightweight, connectionless, unreliable protocol, avoids packets retransmission and reordering. In case of multipath transmission scenarios, UDP is considered to be a good choice for multipath transmission due to the fact that UDP does not bind or connect to an Internet Protocol (IP) address. This enables UDP to transmit data on multiple paths without bothering the IP address. UDP assumes that all the data arrives at the destination is going to be insequence. There do not exist the out-of-sequence packets. If there are out-of-sequence packets, then these are ignored. Similarly, UDP has no congestion control mechanism to maintain network status information at endpoints such as, packet loss, congestion occurrence and reordering of the datagram. Hence, UDP may underutilize available bandwidth or worsen the congestion in the network, which makes UDP unsuitable for multipath transmission for a reliable communication.

On the other hand, the standard transmission control protocol (TCP) is a connection oriented protocol and has a congestion control mechanism. Congestion control mechanism helps TCP in packet retransmission, reordering and maintaining the network status of each path. The main hurdle in using TCP for multipath transmission is that a single TCP connection binds to only one IP host address on both sides; changing the IP address will kill any active connection. On the other hand, TCP does not prevent congestion by moving traffic away from congested paths. TCP only spreads out its traffic over time on the same path. In short, both

TABLE 1.1: Comparison of SCTP features with TCP and UDP [1]

Service/Features	SCTP	TCP	UDP
Full-duplex data transmission	Yes	Yes	Yes
Connection-oriented	Yes	Yes	No
Reliable data transfer	Yes	Yes	No
Partially reliable data transfer	Optional	No	No
Ordered data delivery	Yes	Yes	No
Unordered data delivery	Yes	No	Yes
Flow and congestion control	Yes	Yes	No
Explicit congestion notification support	Yes	Yes	No
Selective Acknowledgment SACK	Yes	Optional	No
Preservation of message boundaries	Yes	No	Yes
Path maximum transmission unit discovery	Yes	Yes	No
Application data fragmentation/bundling	Yes	Yes	No
Multistreaming	Yes	No	No
Multihoming	Yes	No	No

TCP and UDP are single path transport protocols which cannot use multiple paths between multihomed devices equipped with multiple network interfaces.

The multihoming capability is provided by a relatively young transport protocol called stream control transmission protocol (SCTP) [14], developed by Internet Engineering Task Force (IETF). SCTP contains best feature of two legacy transport protocols, i.e., TCP and UDP. SCTP provides stable, ordered delivery of data between two endpoints like TCP and also preserves data message boundaries like UDP as mentioned in table 1.1.

1.2 Basics of Stream Control Transmission Protocol (SCTP)

SCTP establishes an association between the sender and receiver, in which multiple connections are created. SCTP association is a combination of IP address and port number of end nodes.IP version 4 (IPv4)used as Internet Protocol in this research work. In case of multi-homing, SCTP association comprises of endpoints with more than one IP addresses but a single port number. Hence, the de-multiplexing converges at the given port number where the OOS arrival of segment needs careful handling. Currently, SCTP uses multihoming only as a backup service; using one primary IP address at a time while keeping the remaining IP address as secondary resources to ensure reliability. In case of a link failure, secondary IP address is used to reach the destination. SCTP lacks capability to use more than one available interface at a time for transmission, called simultaneous multipath transmission (SMT). The research community also refers SMT with other names in various contexts such as, concurrent multipath transfer (CMT) [15], bandwidth aggregation [16], resource pooling [12], load sharing [17] and striping [18].

SCTP assigns a stream sequence number (SSN) to each message within a stream or a connection for in-sequence delivery of messages. It also assigns a transmission sequence number (TSN) to each users data message; fragmented or unfragmented, independent of SSN. The TSN is used to uniquely identify the data packets. The receiver end acknowledges all in-sequence TSNs using acknowledgments while SACK (selective acknowledgment) is used to acknowledge TSN with gaps, as shown in Figure 1.2.

SACK carries important variables of receivers such as, cumulative TSN acknowledgment (CACK), advertised receiver window (a_rwnd), gap acknowledgment block start and end of each gap block [14]. The receiver side informs the sender about the last insequence TSN received using CACK. The receiver maintains the flow control by limiting the sending capability of the sender to its advertised receiver window (a_rwnd). The gap Ack block start and end are generated

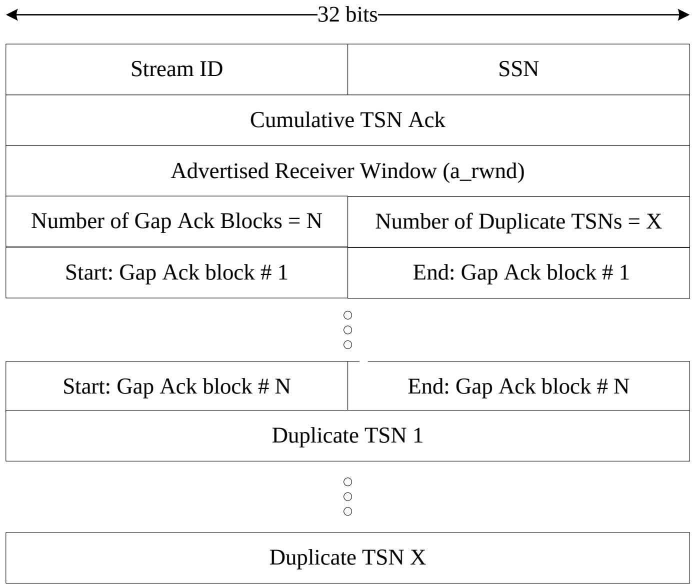

FIGURE 1.2: Selective acknowledgment (SACK) packet format of SCTP

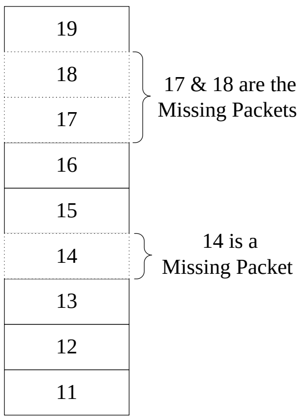

FIGURE 1.3: Packets received at the receiver side

due to the missing packet at the receiver side. The TSN of the first packet received after the first missing packet is indicated by gap Ack block start, while the gap Ack block end points out the last insequence TSN received in this block. The situation would be more feasible to understand with the help of an example. Let us assume that the receiver receives the data packets having missing packets, as shown in Figure 1.3.

These missing packets enable the receiver to transmit SACK to the sender. Here the packets received at the receiver side with TSN are greater than the CACK i.e., 15,16 and 19 are out of sequence packets (OOS). The packets that fail to reach the destination i.e., 14, 17 and 18 are considered as missing packets. Let us assume

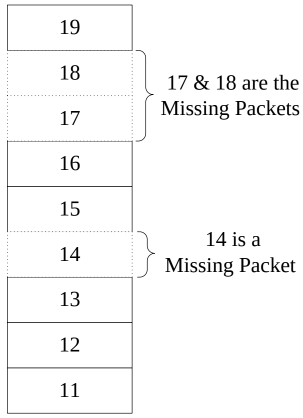

FIGURE 1.4: Packets received at the sender side

that a_rwnd is mentioned as 4660 bytes by receiver side. The numerical values represented by SACK packet to represent the gap Ack blocks are shown in Figure 1.4.

In this case, the receiver sends a SACK with the arrival of each data packet until the missing packets are retransmitted. The sender side multihomed congestion control mechanism triggers the retransmission of missing packets after the number of SACK received as mentioned in fast retransmit threshold. The sender maintains an independent congestion window (cwnd) for each destination in multihoming. Only one a_rwnd is kept for the whole association, regardless of the fact that peer may be multi-homed or has a single address. SCTP endpoint uses a_rwnd and congestion window (cwnd) to adjust the transmission rate between the sender and receiver.

To conclude this section, SCTP is considered interesting for simultaneous multipath transmission (SMT) due to the lack of mature multihoming mechanism in any other deployed transport layer protocol. Datagram Congestion Control Protocol (DCCP) also provides multihoming for mobility while SCTP multihoming is comparatively more mature and most investigated protocol and has support for the fault tolerance and mobility. SCTP can be configured with general transport layer features (such as, using a single stream of data instead of multistreaming). Such generalized SCTP configuration makes it feasible to incorporate multihoming related solutions in other transport layer protocols [19].

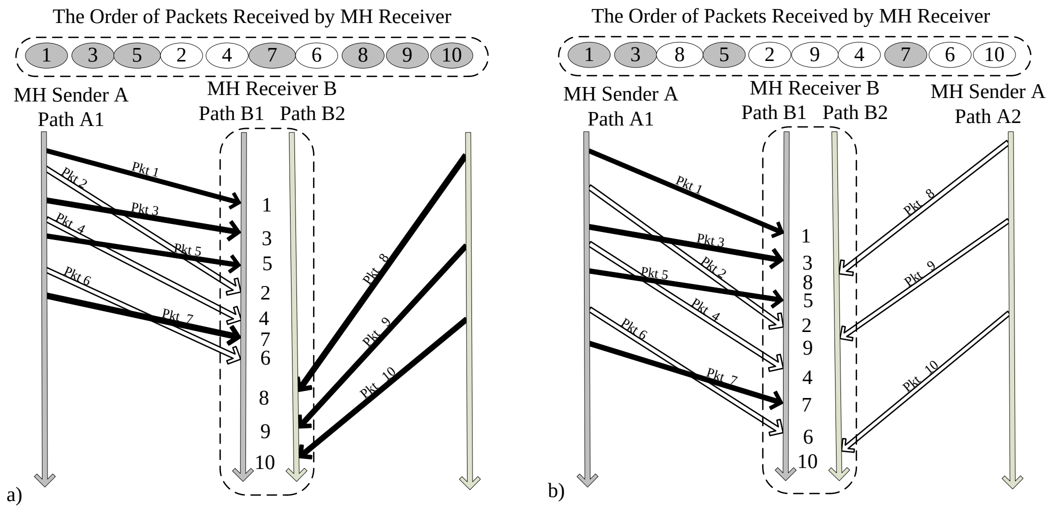

FIGURE 1.5: Pattern of a) In sequence packets arrival b) Intra-path No missing packets and Inter-path missing packets

1.3 Intra and Inter-path Missing Packets

Dynamic path characteristics such as capacity, delay, loss rate and congestion make it difficult to estimate network recourses precisely, particularly during the end to end communication. The devices have to infer the network congestion from an event such as, packet getting delayed or even lost in the worst case scenario. Traditionally, single homed congestion control mechanism estimates the network capacity by a gradual increase in transmission rate of sequenced packets in flight. The missing packets result in the OOS packet arrival, which consequently enforces the CC mechanism to cut half of its transmission rate (of packets) to reflect the occurrence of congestion. This sense of congestion detection (based upon OOS packet arrival) in network creates confusion for multihomed devices. The OOS packet arrival is by default a consequence of multihomed devices communicating through multiple network paths at a time. Traditional congestion control mechanism infers this OOS packet arrival as lost event, which ultimately slows down the transmission unnecessarily, even when no congestion exists at all. This problem of spurious losses becomes severe when multihomed devices transmit single data stream over multiple paths. Transmission of a single stream of data over multiple paths creates the problem of OOS packet arrival at the receiver side due to intra and inter-path missing packets, as shown in Figure 1.5 and 1.6.

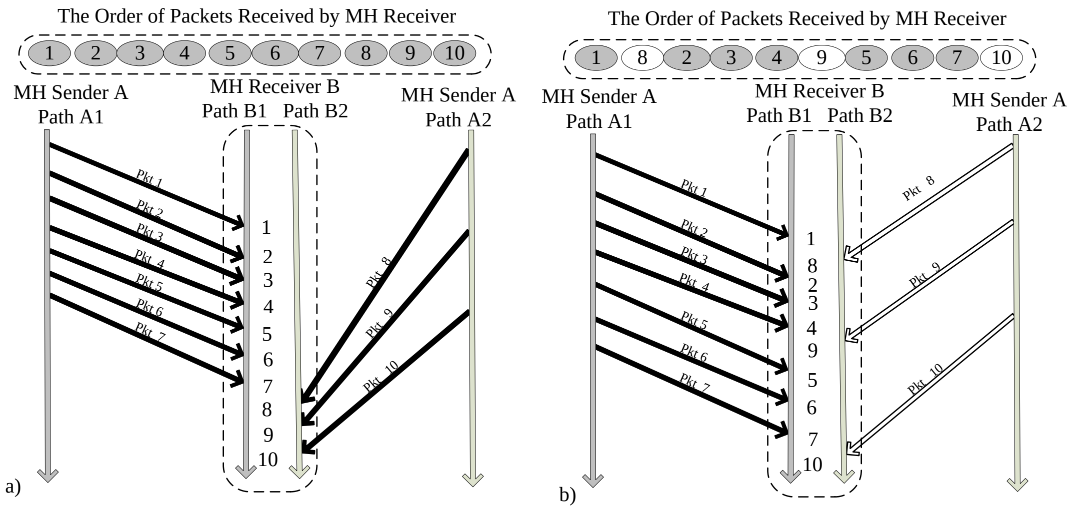

FIGURE 1.6: Pattern of a) Inter-path No missing packets and Intra-path missing packets b) Both Intra-path & Inter-path missing packet

Suppose a multihomed (MH) sender uses paths A1 and A2 to communicate with a multihomed receivers Paths B1 and B2 respectively. MH sender schedules packets 1-7 on the 1st path (A1-B1) and 8-10 on 2nd path (A2-B2) based upon the advertise receiver window and congestion window (cwnd). Single finite size receiver buffer (Rbuf) is maintained by the multihomed receiver to accommodate received packets from both the paths with the assumption that there is no cross traffic in the network.

In an ideal situation, an MH receiver receives intra and inter-path insequence packets, as shown in Figure 1.5(a). In reality, the multiple paths have disparities in bandwidth and delay due to which, packets are received out of order at the receiver side. One of the situations is shown in Figure 1.5(b), where a multihomed receiver B receives insequence packets from each path with an indication that there is no congestion in the network. But the combined effect of multiple paths creates OOS packet at Rbuf such as, reception of packet 1, 8, 2, 3 & 4. On the arrival of packet 4, the intermediate packets of 4 and 8 are considered as inter-path missing packets i.e., 5, 6 and 7. The OOS packet arrival at the receiver side due to multipath effect creates the effect of the inter-path missing packet. The OOS packets block Rbuf by waiting for the arrival of missing packets, causing Rbuf blocking problem. The probability of inter-path missing packets increases with the disparities in bandwidth and end to end delay of multiple paths.

On the other hand, the occurrence of traditional packet loss within the same path causes OOS packets arrival at the receiver side and is called intra-path missing packet as indicated by packet 2, 4 and 6 in Figure 1.6 (a). Intra-path missing packet is a sign of congestion in the network. The probability of intra-path missing packet increases with increase in congestion in the network and path failure. The intra-path missing packets demand immediate congestion control mechanism action to cope with network congestion. In reality, the MH receiver receives both intra-path and inter-path missing packets due to network congestion and disparities in multipath features, as shown in Figure 1.6 (b) [20].

1.4 Research Hypothesis and Research Questions

The research hypothesis explored in this thesis is stated below:

The degradation of aggregated throughput in simultaneous multipath transmission in SCTP can be improved by managing the congestion window based on the differentiation in the intra and inter-path missing packets and making fast re-transmit threshold value adaptive with respect to available Rbuf space.

This thesis is going to answer the following research questions:

1. What are the major reasons for the degradation in throughput in a simultaneous multipath transmission system of SCTP?
2. Is the role of OOS packets significant in the degradation of the throughput of SCTP?
3. Is there a need to make the fast retransmit threshold value according to the receiver buffer size in order to increase the aggregated throughput of SCTP?
4. How can the missing packets in selective acknowledgments (SACK) be classified into intra and inter-path packets?

5. How can the multihomed congestion control (MCC) mechanism of SCTP be modified by using differentiation of the missing packets in selective acknowledgments (SACK)?

1.5 Significance of Research Hypothesis

The traditional congestion control mechanism considers the missing packet as congestion in the network and handles it by reducing the congestion window. In multipath transmission, the packet gets missed either due to congestion in the network or due to multipath effect. This research work focuses on the differentiation of missing packets on behalf of its causes (i.e., intra and inter-path) and triggered multihomed congestion control mechanism accordingly. In order to further optimize the proposed work, the fast retransmit threshold value is kept adaptive with respect to available Rbuf space. This improves the aggregated throughput by efficiently utilizing the available Rbuf space and by providing an arrival opportunity for missing packet.

1.6 Research Contributions

The contributions of this research work are mentioned below.

1. Utilization of single sequence number space for data transmission.
2. Identification of reasons for the degradation of aggregated throughput of multipath transmission in SCTP.
3. Differentiation of missing packet into intra and inter-path.
4. Provision of intra and inter-path aware multihomed congestion control mechanism to overcome the abnormal throughput degradation.
5. Provision of adaptive fast retransmit threshold with respect to available Rbuf space in order to efficiently utilize the limited Rbuf space.

6. Overcome the throughput degradation of high capacity link due to low capacity companion link.

7. Deterministic-Time Markov Chain model for SCTP-CMT and SMT-Schemes (SMT-MFR and SMT-AMFR) has been adapted to verify the simulation results.

1.7 Thesis Organization

This thesis is based on the idea that involves the critical scrutinization of the concurrent data transmission on multiple paths using a single stream of data in SCTP and the causes of aggregate throughput degradation. This thesis presents SMT schemes (SMT-MFR and SMT-AMFR schemes) to enhance the aggregate throughput by using a congestion window based on the differentiation in the intra and inter-path missing packets and making fast re-transmit threshold value adaptive concerning available Rbuf space.

The chapter wise thesis organization is shown in Figure 1.7 for better overview. The chapter 1 is about the critical analysis the transmission of data concurrently on multiple paths and presents the causes of its throughput degradation. This helped us to formulate the hypothesis. The chapter 2 presents the background of this research study and detailed analysis of state-of-the-art multipath transmission schemes. The present and past research studies are discussed and cited. The proposed SMT-MFR and SMT-AMFR scheme is presented in chapter 3, which differentiates the missing packet into intra and inter-path and optimizes the solution further.

In chapter 4, various simulation multihomed scenarios are discussed to compare and evaluate the performance of SMT schemes (SMT-MFR & SMT-AFMR) with SCTP-CMT which acts as a benchmark multipath transmission scheme. The performance analysis parameters used for evaluation of SMT schemes are presented theoretically and mathematically. Chapter 5 presents the analytical models of

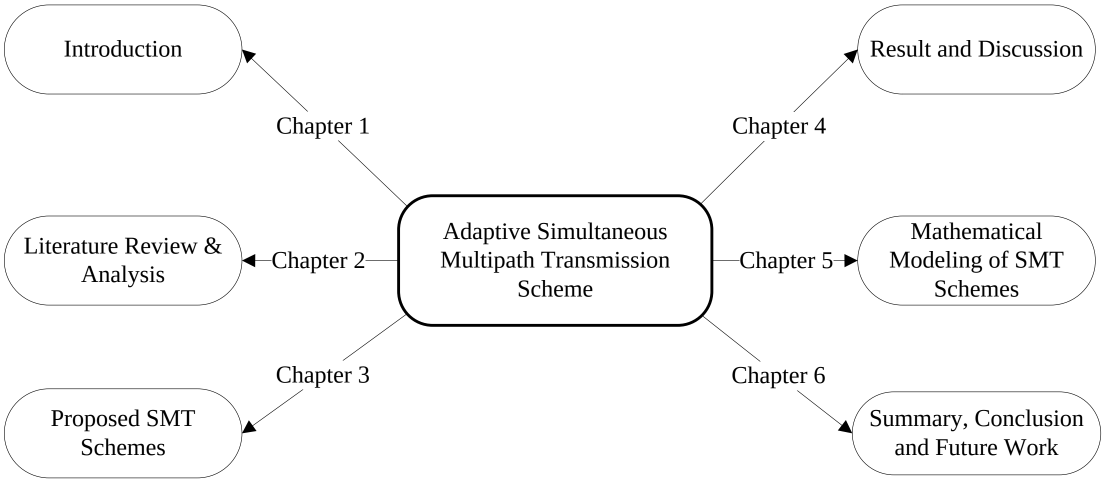

Figure 1.7: Thesis Organization

SMT-MFR, SMT-AMFR and SCTP-CMT schemes. Deterministic-Time Markov Chain (DTMC) model is used to mathematically verify simulation results of SMT-Schemes. The summary, critical analysis and conclusion of this research study are depicted in chapter 6 with potential limitations of the SMT schemes.

Chapter 2

LITERATURE REVIEW & ANALYSIS

This chapter presents the critical analysis of the contemporary state-of-the-art research studies related to multihoming in SCTP, with the intention to identify the existing research gaps in the light of our research questions. The focus of this analysis is on the multihomed congestion control schemes to deal with missing packets and the role of Rbuf space management to handle the out of sequence (OOS) packets.

2.1 SCTP-Based Concurrent Multipath Transmission Protocols

Soon after the standardization of SCTP in 2000, the multihoming domain grabbed the attention of research community. Researchers started to explore its issues such as, OOS packet arrival, Rbuf blocking, load sharing, fault tolerance and congestion issues [21],[22], [1]. To date, various SCTP based concurrent multipath transmission protocols are proposed in the literature, as shown in Figure 2.1.

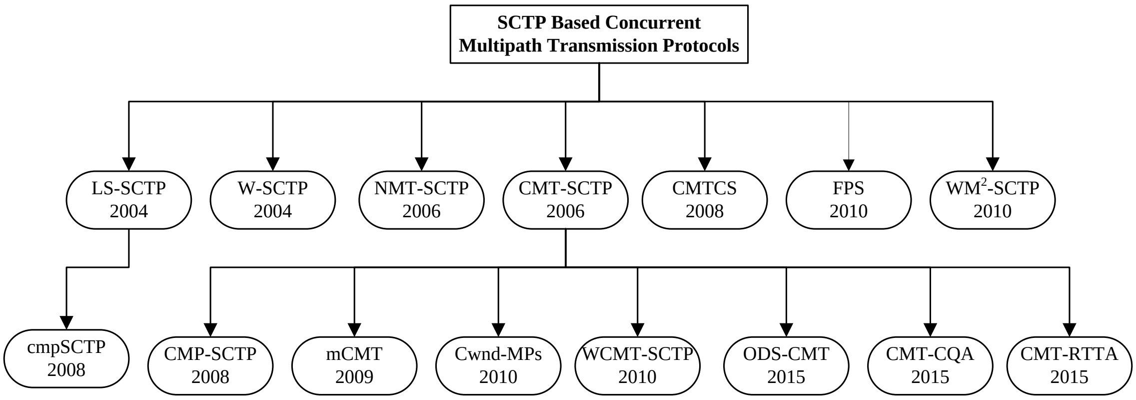

FIGURE 2.1: SCTP based concurrent multipath transmission protocols

Westwood SCTP (W-SCTP) has extended the SCTP to include parallel transmission on multiple paths [23]. W-SCTP manages each data connection independently by retaining its cwnd and send buffer independently. Selective acknowledgment (SACK) is modified to work with multiple send buffers of multiple paths. The sender side is modified in which bandwidth aware scheduler is configured to balance the data transmission load across the multiple paths. W-SCTP estimates the delivery time of a packet on all available paths and then sends the packet to a path with shortest delivery time to ensure the sequenced delivery. This process repeats until the congestion window for this path is fully utilized. Single Rbuf is maintained to accommodate the incoming packets from multiple paths. W-SCTP has no mechanism to deal with the OOS packet arrival at the receiver side. Rbuf blocking is handled by the bandwidth aware scheduler who has missed the important path feature such as, packet losses in best path selection. The traditional congestion control mechanism of a single path is not good enough to be used for handling the congestion and missing packet information during multipath transmission.

Load sharingSCTP has modified the SCTP by decoupling the congestion control from flow control during multipath data transmission [24]. LS-SCTP has used flow control on the basis of an association where congestion control is utilized on the basis of per path; in such a way that separates cwnd is maintained for each path. Two additional sequence numbers are used, i.e., path identity (PID) and path sequence numbers (PSNs) for load balancing and in sequence delivery of

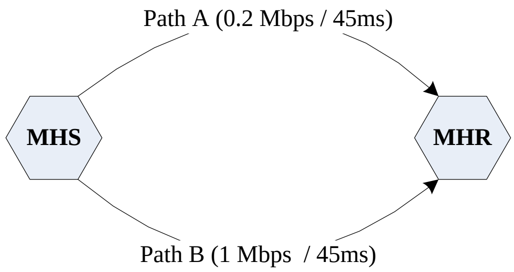

FIGURE 2.2: Simulation scenarios having two multiple paths

data which creates additional processes overhead. The cwnd based scheduler is used instead of nave round robin approach. LS-SCTP has increased the end to end delay by using a large receiver buffer to handle the out of sequence packets. The similar multipath transmission approach is adopted by concurrent multipath-SCTP (cmp-SCTP) with the modification that a separate send buffer is utilized for each connection to limit the head of line blocking to a path while the performance of remaining companion remains unaffected [25].

Nave multipath transmission (NMT-SCTP) is the simplest form of a multipath transmission scheme which transmits packets on each path in a round-robin fashion without being bothered about paths bandwidth, delay and packet loss features. NMT-SCTP experiences the OOS packet arrival, abnormal fast retransmission, and frequent cwnd collapse [26]. Although, NMT-SCTP experiences no path losses till the cwnd collapses frequently and triggers abnormal fast retransmission. The abnormal collapse of cwnd is a hindrance in achieving a full performance gain in the utilization of parallel paths.

CMT-SCTP focuses on the abnormal collapses of cwnd in NMT-SCTP. This issue is handled by SCTP-CMT using split fast retransmit (SFR) algorithm. The intention of SFR algorithm is to ignore the spurious fast retransmission, which triggers due to multipath effect. SCTP-CMTs cwnd status is improved, but its throughput has degraded a lot due to Rbuf blocking. In order to analyze the scheme, a simple simulation scenario is used where a multihomed sender (MHS) transmits a data parallel on two disjoint paths to a multihomed receiver (MHR) as shown in Figure 2.2.

Table 2.1: Simulation configuration parameters for testing the causes of misinterpretation of the missing packets

Parameters	**Values**
Traffic source	File transfer protocol (FTP)
Stream	(single stream) 1
Transport protocol	CMT-SCTP
Packet size	1500 Bytes
Receiver window (rwnd)	65536 Bytes
Path A bandwidth & delay	0.2 Mbps & 45 milliseconds(ms)
Path B bandwidth & delay	1 Mbps & 45 milliseconds(ms)

Both paths have the same end to end delay and have a disparity in bandwidth to generate the effect of missing packet, and OOS packet at the sender and receiver side respectively.

A data file is simultaneously transmitted on the two paths using file transfer protocol (FTP) at the application layer. Limited standard Rbuf size is used in MHR and standard packet size is configured as mentioned in table 2.1. Congestion window (cwnd), advertised receiver window (a_rwnd) and throughput are used as performance analysis parameters. Congestion and flow controls are the two important factors that define the sending capability of a sender over a path. Path A is a slow link that creates the missing packet effect on path B, that is why, the path A behavior is found to be a normal. The path B features are affected by missing packets, and worthy of discussion here.

SCTP-CMT focuses on the abnormal collapses of cwnd in the nave multipath transmission (NMT-SCTP) and improves the cwnd, but its throughput degrades, as shown in Figure 2.3. At the start of multipath transmission, the whole Rbuf size is advertised as a_rwnd to the sender. The initial a_rwnd is large enough to queue the OOS packet up to few early seconds of multipath transmission. Congestion window (cwnd) of SCTP-CMT also reaches to its peak value during this time (in 2 seconds).

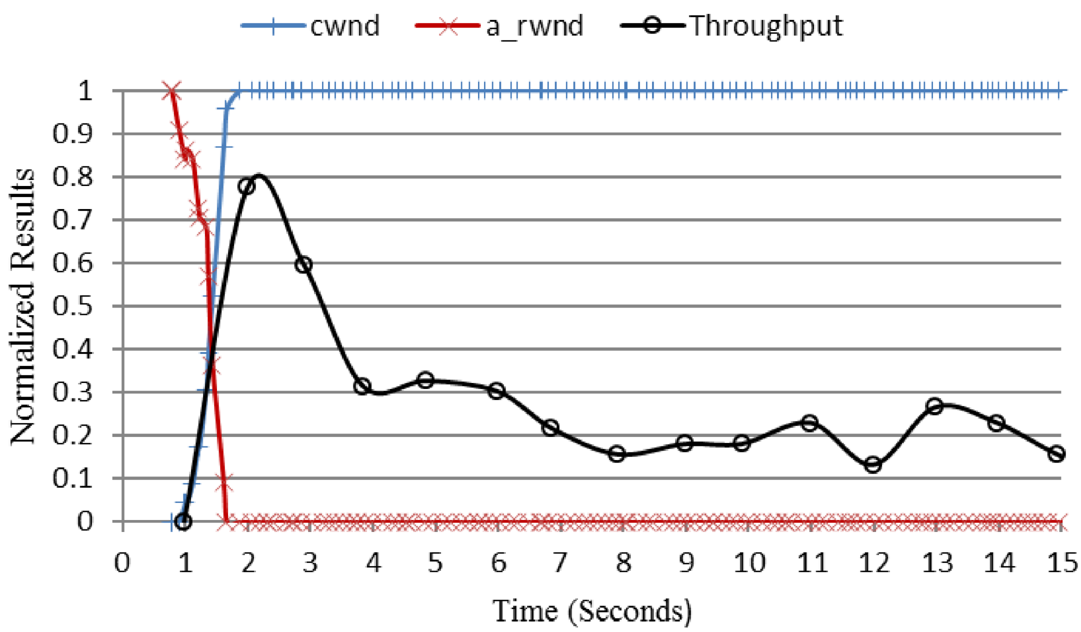

FIGURE 2.3: The normalized cwnd, a_rwnd, and throughput of SCTP-CMT in path B

This enables the sender to transmit maximum data in the first 2 seconds and results in high throughput at this stage. SCTP-CMT ignores the inter-path missing packet, i.e., the missing packet notification due to multipath effect, which is helpful in avoiding abnormal fast retransmission and to stop spurious collapses of cwnd.

Rbuf has to transmit insequence data packets to the application. The receive buffer (Rbuf) blocking is a phenomenon in which, very early OOS packet blocks the receiver buffer by making it wait for the arrival of missing packets to an extent that the entire receiver buffer is consumed by incoming packets. This forces the receiver to advertise very low Rbuf and hence, restricting the sending rate to the very low [19] [27].

OOS packets are generated as a result of these missing packets cause Rbuf to overflow. This result show a drastic decrease in the size of a_rwnd for the rest of the simulation duration. The throughput remained comparatively high at a time interval of 5th, 9th ,11th and 14th second and continued to swing between 0.1 to

0.3 due to the arrival of delayed inter-path missing packets at these intervals (5th, 9th, 11th and 14th second).

In short, SCTP-CMT has failed to find that the Rbuf overflows by waiting for OOS packets due to multiple paths effect, not due to congestion in the network. SCTP-CMT has ignored the spurious fast retransmission for the inter-path missing packet, which stimulates the OOS packet arrival, resulting into Rbuf blocking. This forces the receiver to advertise minimum rwnd in order to limit the sending rate of the sender and hence, degrades the throughput. Such sort of degrading situation is handled by differentiating the missing packets into intra and inter-path and the usage of respective multihomed congestion control mechanism as proposed by inter-path missing packet differentiation (IMPD) algorithm, which is presented chapter 3.

Moreover, the five retransmission policies are evaluated by SCTP-CMT. The intention of these five policies is to quickly retransmit the missing packet to a destination using path features such as cwnd, slow start threshold (SSThresh) and loss rate. Another study utilizes a combination of larger SSThresh, cwnd and low loss rate in the selection of retransmission path and is known as compound parameters retransmission policy [28]. If all destinations have same above mentioned features then a retransmission path is selected randomly.

Probably the phenomena of OOS packet arrivals due to the missing packet and its causes are misunderstood and the focus has been on retransmission of the missing packet using traditional congestion control mechanism. The SCTP-CMT has failed in handling the OOS packets according to its cause of occurrence i.e., OOS packets arrive due to congestion in the network or due to multipath effect. SCTP-CMT has assumed infinite buffer which is unrealistic due to the usage of finite buffer space in the network. The Rbuf blocking is observed using limited buffer size [29]. SCTP-CMT is still in developing phase related to load sharing and congestion control mechanism to handle the OOS packets [30] [31]. OOS packet arrival is an inherited issue with the simultaneous multipath transmission, which causes an abnormal fast retransmission, frequent cwnd collapses, increased packet losses and reduced aggregated throughput. This situation becomes worse with the increase in desperately of multiple path features such as, bandwidth and propagation delay. There is a need to modify the multihomed congestion control mechanism to handle the missing packets with respect to its cause. This study focuses on the differentiation of missing packets according to its cause and modifies the multihomed congestion control mechanism to handle the missing packet accordingly.

CMP SCTP is another distinct approach of SCTP-CMT that uses the multi-buffer, multi-state management and additional sequence number [25]. Still, some schemes utilize the single buffer for easier packet scheduling. The additional sequence number also increases the complexity in congestion control mechanism. This thesis focuses on using the single sequence number space to minimize the extra processing overhead and may be acceptable to the intermediated nodes during the end to end communication.

Mobile CMT (mCMT) has extended SCTP-CMT to overcome the receiver buffer blocking problem in the wireless network due to packet losses [32]. The cross-layer approach is used in mCMT, where the application layer is responsible for defining the streams. The packet from same streams is allotted the same path to minimize packet reordering at the receiver side. This creates the overhead of packet reordering and reliability requirement on the application layer. Mobile CMT has no support for transmission of a single stream of data over multiple connections. In true sense, the multipath transmission protocols should transmit the single stream of data simultaneously over multiple connections and should provide reliability at the transport layer by transmitting the sequence data to the application layer.

Forward prediction scheduling (FPS) has modified the SCTP for the sequence arrival of the packet at receiver side using the concurrent multipath transmission [33]. FPS has a scheduling module that estimates the end to end latencies of multiple paths and schedules data on these paths in such a way that minimizes the OOS packet arrival. FPS scheduler uniqueness determines the amount of data to be sent on the fast path before the arrival of data on the slow path. FPS is unable to achieve the optimal multipath transmission with varying path conditions due to the negligence of other path features such as, loss rates. FPS focuses on in sequent delivery of packets in multipath transmission; still, there is OOS packet arrival at the receiver side, which results in degradation of aggregated throughput. The missing packet should be treated with the multihomed congestion control mechanism at the sender side to minimize the performance degradation due to arrival of the OOS packet at the receiver side.

In addition to this, a number of congestion window management policies (Cwnd-MPs) have been introduced in the past decade, mostly designed for single path end to end communication [34]. These Cwnd-MPs have no concept of OOS packet arrival at the receiver side due to multipath effect, causing a collapsed performance in concurrent multipath transmission. Researchers have introduced multihomed cwnd management schemes for efficient bandwidth aggregation of multiple paths.

Wireless multi-path multi-flow stream control transmission protocol (WM2 -SCTP) has migrated the congestion control from association level to sub-flow level [35]. The sub-flows have a fair share of bandwidth in between them, by disturbing the fairness with other application traffic on the internet. The usage of multi-buffer, three-level sequence number and packet pair capacity estimation has introduced network overhead and complexity for handling OOS packets. The active packet pair technique is used for bandwidth estimation. The multiple sub-flows have a similar effect on multiple associations for the same traffic as compared to the single association, which is a violation of fair bandwidth sharing on the internet. The drop rate is increased with an increase in the number of sub flows due to lack of handling inter-path OOS packet. Similarly, the relative delay estimator algorithm has provided a retransmission solution by ignoring inter-path missing packet differentiation [36].

Sender-based multipath out-of-order scheduler (SMOS) is proposed to handle the OOS packet arrival due to bandwidth based disparity at the receiver side

[37]. The SMOS tries to send the more packet on a path having higher bandwidth. The SMOS uses probability for distribution of packet among multiple paths based on their bandwidth. So a path with high bandwidth is assigned higher probability as compared to a path having low bandwidth. SMOS tries to reduce the out of order arrival packet at the receiver side. SMOS does not differentiate the missing packets into intra and inter-path, nor it modifies the multihomed congestion control mechanism accordingly.

Most of the studies employ the scheduling algorithm by using path heuristic approaches for estimation of control parameters [38] [25] [39]. These studies have used intelligent, optimized multipath scheduler to minimize the packet reordering using various parameters such as, Rbuf space, cwnd, slow start threshold (SSThresh), path losses and bandwidth delay product (BDP) of each destination. These parameters are used for scheduling policies for data transmission and retransmission of missing packets on multiple paths. These schedulers have tried their best to make the sequenced delivery of packets at the receiver side. Still, there are OOS packet arrivals at the receiver side. These schemes do not have any information about the nature of missing packets, OOS packet generation and no information about how to deal with the OOS packet at the receiver side and missing packet at the sender side. Moreover, the complexity of the multipath scheduler increases with an increase in the number of parameters used for scheduling purpose.

On-Demand Scheduler (ODS) uses SCTP-CMT to transmit data simultaneously over multiple paths using smartphones with limited battery power and memory space [40]. ODS makes the scheduling decision on the basis of each path reception index. The reception index is calculated using path features such as, the ratio of the current size of scheduled data and unacknowledged data in flight (cwnd) to estimate bandwidth. ODS creates another inefficiency by sending data to a destination with lowest round-trip time (RTT) and larger cwnd size. This allows one destination to have a high proportion of shared resources. The processing delay involved in the recursive search of a suitable packet in send buffer (Sbuf) increases with an increase in a number of destinations and size of Sbuf. This makes

ODS to be very expensive for smartphones with limited battery power. ODS ignores the OOS packet arrival by using an assumption that all outstanding packets reach their destinations successfully.

OSI communication follows the waterfall model with virtually strict boundaries between the layers. On the other hand, the cross-layer solution has provided the flexibility of getting feedback from any layer with the incentive of performance optimization. QoS-aware adaptive CMT (CMT-CQA) is a cross-layer approach, in which slowest trouble maker path is removed from multipath transmission to avoid aggregated bandwidth degradation [41]. The choices of best paths are made using cross layer paths history and MAC layer QoS information. The bandwidth of the inactive path is estimated before including it into multipath transmission to avoid slow start and jitters in delay. CMT-CQA is unable to handle the OOS packets, and missing packets due to multipath effect. Therefore, the CMT-CQA has simply removed the slow companion path to solve the Rbuf blocking issue. The removal of slow companion paths wastes the available network resources, especially in case of multiple slow companion paths. The slow start phase in congestion control mechanism is used to cure the aggressive data transmission in order to control the congestion. On contrary, the CMT-CQA ignores the slow start phase for reactivating path which will cause more congestion due to aggressive data transmission.

The wireless concurrent multi-path transfer stream control transmission protocol (WCMT-SCTP) is proposed to solve the Rbuf blocking problem [42]. WCMT-SCTP has solved the OOS packet and missing packet issue by binding a stream of data for its rest of a lifetime to a specific path. WCMT-SCTP has no support for parallel transmission of a single stream of data over multiple paths and hence, cannot handle the OOS packets and missing packets generation in this scenario. WCMT-SCTP does not optimally utilize the bandwidth resources where the short stream finished earlier, while the long stream of data continues to use one path for the rest of its lifetime.

The CMT-RTTA has proposed buffer splitting technique based on the round trip time (RTT) of a path [43]. The path with less RTT occupies more Rbuf space

Table 2.2: Critical analysis of SCTP based multipath extensions Protocols

Protocol	**Contribution**	**Shortcoming**
W-SCTP	• Has bandwidth-aware multipath scheduler,	• Has mismanagement issue of separate send buffers for each connection. • Has lack of support for handling OOS packets. • Has not utilized all important path features in bandwidth aware scheduler.
LS-SCTP	• Has decouple congestion control from flow control. • Has cwnd based scheduler.	• Has created overhead due to two additional sequence number spaces. • Has usage of large receiver buffer • Has increased the end to end delay.
cmpSCTP	• Same as LS-SCTP with the modification that a separate send buffer is used for each connection. • Limit the head of line blocking	• Has created overhead due to two additional sequence number spaces. • Has usage of large receiver buffer. • Has increased the end to end delay. • Has issue of managing of multiple sends buffers.
NMT-SCTP	• Has nave round robin transmission of data packets on multiple paths	• Has frequent cwnd collapses. • Has abnormal fast retransmission.
SCTP-CMT	• Has solved the abnormal collapses of cwnd due to packet reordering and five retransmission policies to increase throughput.	• Has performance degradation with limited Rbuf and increase in disparities of companion multiple paths features. • Has Rbuf blocking problem. • Has immature load sharing and multihomed congestion control mechanism.
mCMT	• Has overcome Rbuf blocking by allocating packet from the same stream to the same path.	• Has a cross-layer approach where packet reordering and reliability overhead lies at the application layer.

Protocol	Contribution	Shortcoming
FPS	• Has a delay based multipath scheduler to minimize packet reordering.	• Has Ignored packet losses in multipath scheduling and failed to achieve optimal multipath transmission due disparities in multipath features.
Cwnd-MPs	• Has introduced policies for multi-homed congestion control.	• Has ignored the OOS packets cause performance degradation.
WM^2-SCTP	• Has congestion control per sub-flow, a fair share of bandwidth among the flows.	• Has network overhead and complexity due to the usage of multi-buffer and three levels of the sequence number. • Has an increase in packet losses with increase in the number of sub-flows.
ODS-CMT	• Has scheduling based on a ratio of current schedule data and unAck data in flight with the estimated bandwidth.	• Has unfairness due to a path with low RTT and larger cwnd gets a high proportion of shared recourses. • Has processing delay due to increase in a number of destinations and the size of the send buffer. • Very expensive for smartphones with a limited buffer and energy resources.
CMT-CQA	• Has QoS aware Scheduler where slow path is removed to avoid performance degradation.	• Has a cross-layer approach where an efficient utilization of Rbuf space decreases with increase in number of paths.
WCMT-SCTP	• Has avoided Rbuf blocking by constraining specific data streams to specific paths.	• Has throughput degradation for long flow in high capacity networks.
CMT-RTTA	• Has Rbuf splitting on the basis of RTT features of multiple paths.	• Has failed due to high fluctuation in delay and jitter in the network. • Has ignored packet losses.

and has a higher priority as compared to longer RTT in data transmission. The low Rbuf space with a traditional congestion control mechanism causes performance degradation in CMT-RTTA due to OOS packet arrival and non-differentiation of the missing packet. CMT-RTTA is good enough for multiple paths with a bandwidth based disparity, while its performance may degrade in the presence of delay and loss based disparities due to lack of a mechanism to differentiate the cause of missing packets.

Content-aware multipath transmission scheme (CA-CMT) is an application layer scheme to transmit the high definition (HD) video over heterogenous networks [44]. CA-CMT provide a priority based scheduler to decrease the distortion in video. Congestion control mechanism and data scheduler are both used at same time to improve the performance in term of signal-to-noise ratio, the end to end delay and goodput. CA-CMT has used the cross-layer approach and need both sender and receiver side modifications. CA-CMT does not differentiate the OOS packet. The congestion control mechanism of CA-CMT is not able to respond missing packet notification based on intra and inter-path.

Adaptive concurrent multipath transfer (A-CMT) is used to overcome the arrival of out of order packets due to bandwidth and delay based disparities [45]. The path features such as bandwidth and delay are used to schedule the packets on multiple paths. The difference of delay between slow and fast path is used to distribute packets over multiple paths. A-CMT has outperformed CMT by having higher throughput. However, A-CMT has only used 30% of the network bandwidth. The remaining 70% bandwidth is not being utilized. The out of order packets arrival is an inherited issue with the multipath transmission. A-CMT does not provide any mechanism to handle the intra and inter-path OOS packet independently.

Selective retransmission based concurrent multipath transmission (CMT-SR) scheme is proposed to transmit simultaneously high-quality video traffic over multiple paths [46]. CMT-SR scheme transmits the high priority data on a fast, reliable path. The lost packet is retransmitted on a path having less end to end delay to decrease the waiting time at the receiver side. Decoding time is associated with each data packet to decide either to retransmit or drop the packet. A packet with high priority is retransmitted first as compared to low priority data packets. For this purpose, the video traffic is transmitted our multiple UDP links while the path related feedback control information is sent through TCP. The disadvantage of CMT-SR is that it requires sender and receiver side modification. The video traffic is transmitted using multiple UDP interfaces and TCP connections which incurs overhead for multiple connection establishment.

2.2 Conclusions

This chapter has presented the research studies that describe the state-of-the art multihoming schemes in the SCTP in light of support of the research hypothesis. The multihoming schemes are critically analyzed by mentioning their shortcoming with reasons that support the research problem and helped in finding the alternate solutions.

This in-depth analysis of this chapter concludes that two major reasons are diagnosed for the degradation of the aggregated throughput:

1. In multipath transmission protocols, the management of the congestion window and fixation of fast retransmit threshold value does not take into consideration, the intra and inter-path missing packets in selective acknowledgments (SACK).
2. There may be a situation when no congestion occurs, but OOS packets cause the reduction in Rbuf free space (Rbuf blocking); the sending capability of the sender will be reduced and hence, will reduce the aggregated throughput.

Hence, this literature review supports the hypothesis and research questions as mentioned in chapter 1 and enables us to identify the core issues of performance degradation of multihoming in SCTP and helps us to understand the possible solutions provided by others and their shortcomings. This facilitates us to identify the research gap, which is present in other research studies in order to address the above mentioned issues of parallel transmission of a single stream of data over multiple paths.

2.3 Summary

This chapter describes the background of this thesiss research and details the state-of-the-art multipath transmission schemes. The contemporary research studies are discussed in detail that led to the conclusion that the aggregate throughput degradation is caused by the OOS packet arrival, non-differentiation of missing packets into intra and inter-path and traditional use of single homed congestion window mechanism for multipath transmission.

Chapter 3

PROPOSED SIMULTANEOUS MULTIPATH TRANSMISSION (SMT) SCHEMES

The quintessential findings from the literature review and in-depth analysis have helped us to propose a multihoming scheme named SMT-modified fast retransmit (SMT-MFR I & II). SMT-MFR differentiates the missing packets and triggers a multihomed congestion control mechanism according to its causes. In order to further optimize the SMT-MFR scheme, a number of simulation scenarios are simulated and analyzed with various sizes of available Rbuf space and different fast retransmit threshold values. This chapter presents the proposed schemes.

3.1 SMT-Modified Fast Retransmit-I

Initially, modified fast retransmit (SMT-MFR-I) scheme is proposed to solve the missing packets differentiation issue and activation of respective multihomed congestion control mechanism. The complete process of the proposed scheme is shown in a self-explanatory Figure 3.1. This scheme is published in 2011 (khan et al., 2011). On the reception of a SACK at the receiver side, the SMT-MFR-I first finds out the highest acknowledged transmission sequence number (TSN) with the condition that it is acknowledged (Acked) for the first time and is noticed as highest TSN. This highest TSN helps in the classification of missing packets into intra and inter-path packets. The simulation results of the MFR-I show that the inclusion of the highest TSN only limits the performance. This suggests another scheme with the highest as well as the lowest TSN with the name of MFR-II.

3.2 SMT- Modified Fast Retransmit-II

The proposed IMPD algorithm (shown in Figure 3.2) is used to trigger the specific multihomed congestion control (MCC) mechanism with respect to the causes of a missing packet, due to real network congestion or multipath path effect. In this solution, only the sender side modification is required while receiver remains unaffected as mentioned in the next subsection 3.2.1.

3.3 Inter-path Missing Packet Differentiation (IMPD)

The MH sender differentiates the missing packets into intra and inter-path using information conveyed by the selective acknowledgment (SACK). SACK is a gap report, sent from the receiver to inform the sender about missing packets. The sender keeps the copy of sent packets in sender buffer and waits for the arrival of their acknowledgment. Those packets are removed from the sender buffer whose transmission sequence number (TSN) is equal to or less than TSN mentioned in cumulative acknowledgment (C_A).

The multi-homed devices maintain the cumulative acknowledgment (C_A), highest_in_sack_for_dest(H_{D_i}) and saw_new_ack (S_{NAck_i}) variables for each destination (D_i). Two fast retransmit counters, i.e.,$Count_{IP}$ and $Count_{IAP}$ are maintained for inter-path and intra-path missing packets with TSN (T_i). The sender differentiates the missing packet by using the information conveyed by the SACK packet as shown in Algorithm 1.

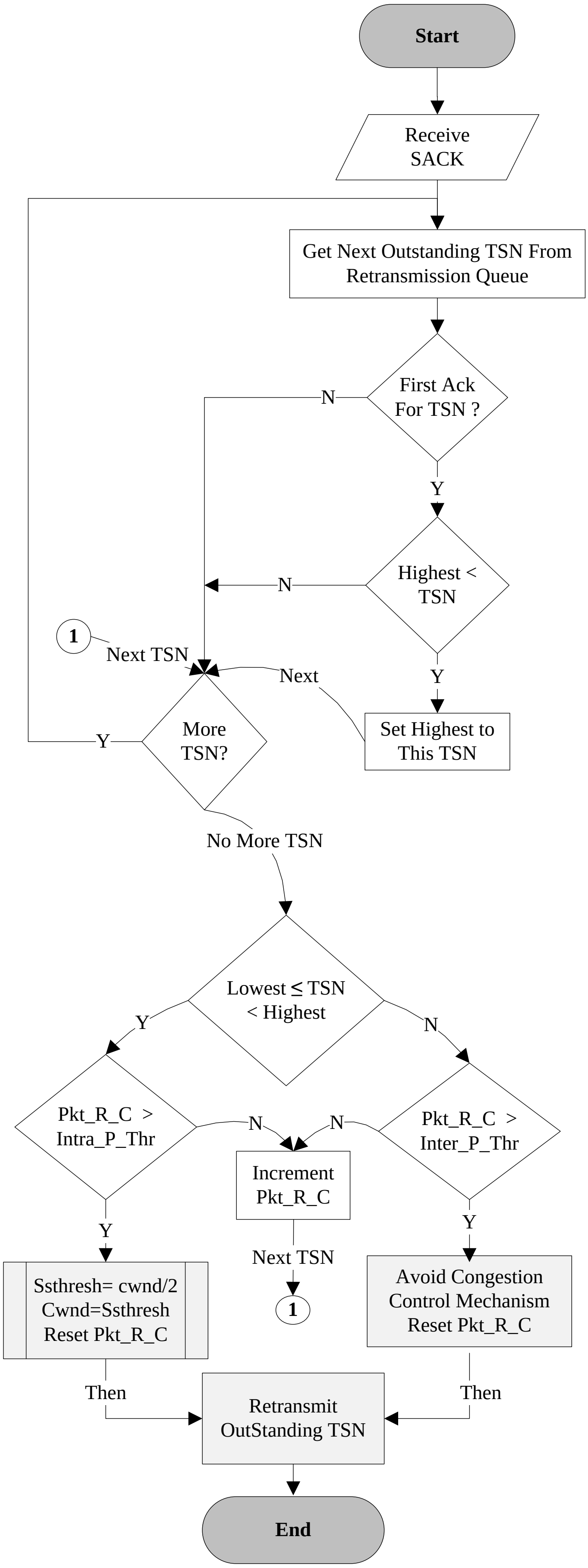

Figure 3.1: SMT- Modified fast retransmit scheme - I

IMPD maintains the following variables per destination in multihomed devices.

1. Transmission sequence number (TSN) represented by T_a, is a unique sequence number assigned to data packets transmitted between MH sender and receiver.

2. Highest_in_sack_for_des (H_{D_i}): The highest TSN acked per destination (D_i) using the SACK, is stored in this variable.

3. saw_new_ack (S_{NAck_i}): This variable stores the boolean status of each destination interface to find out the causative TSNs: causative TSNs for a SACK are those TSNs which cause the SACK to be sent.

4. Low_TSN (Low_{D_m}) and High_TSN ($High_{D_m}$): These variables have maintained a pointer for lowest and highest TSN in a sender queue for each destination.

Algorithm 1 Inter-path missing packet differential (IMPD)

Input: $\{ SN_{Ack}, T_a, D_n, H_{D_i}, Count_{IP}, Count_{IAP}, High_D, Low_D \}$
Output: $\{ Count_{IP}, Count_{IAP} \}$
$\forall$, $initialize\ SN_{Ack_i} = False$
$\exists!$ T_a being acked that is not acked in any SACK
Let D_a be the destination to which T_a is sent
S_{NAck_a} = True; {Missing packet or Gap Noticed}
$\forall$, D_i , Set H_{D_i} to the highest TSN being newly acked on i^{th} destination D.
{To determine whether missing report counts for a TSN should be incremented for inter-path OR for intra-path missing packet.}
Let T_k be the missing packet T with k^{th} TSN reported by sack (whose copy is still maintained in outstanding packets queue of sender buffer).
if S_{NAck_a} == True && S_{NAck_a} == True **then**
  $Counter_{IAP}++$ {Intra-path missing packet}
else
  $Counter_{IP}++$ {Inter-path missing packet}
end if

IMPD maintains C_A , H_{D_i} and S_{NAck_i} variables for each destination. In IMPD algorithm, the missing packet differentiation into intra and inter-path is decided on the basis of following three conditions.

1. The first condition is to verify that the gap report (SACK) for a particular destination is noticed.

2. The second one is that the TSN of the missing packet (T_k) must be less than the highest TSN Acked for a particular destination.

3. The third condition is to verify whether the missing packets belong to the same path or not. For this purpose, if the TSN reported in gap block by SACK lies between Low_{D_m} and $High_{D_m}$, then this TSN is treated as an intra-path missing packet, otherwise reported as inter-path missing packet.

3.4 Multihomed Congestion Control (MCC) Mechanism

IMPD algorithm helps us to infer the following observations.

1. The transport layer is responsible for the transmission of insequent data packets to the application layer. The transport layer protocols hold all received packets irrespective of their order in the anticipation of slightly delayed arrival of the missing packets. These protocols wait for the timeout to retransmit the inter-path missing packets. In order to minimize extra delay and buffering cost, these missing packets may be retransmitted by some alternate eager strategy after getting feedback from IMPD algorithm instead of waiting. This will increase the Rbuf space utilization that ultimately enhances the data transmission from a sender at the cost of some retransmission. To minimize the retransmission cost, the threshold is made adaptive to the Rbuf space.

2. The inter-path missing packet may not indicate any kind of congestion in the network. Hence, this logic is implemented in a multihomed congestion control mechanism by avoiding cwnd reduction while retransmitting delayed inter-path missing packets.

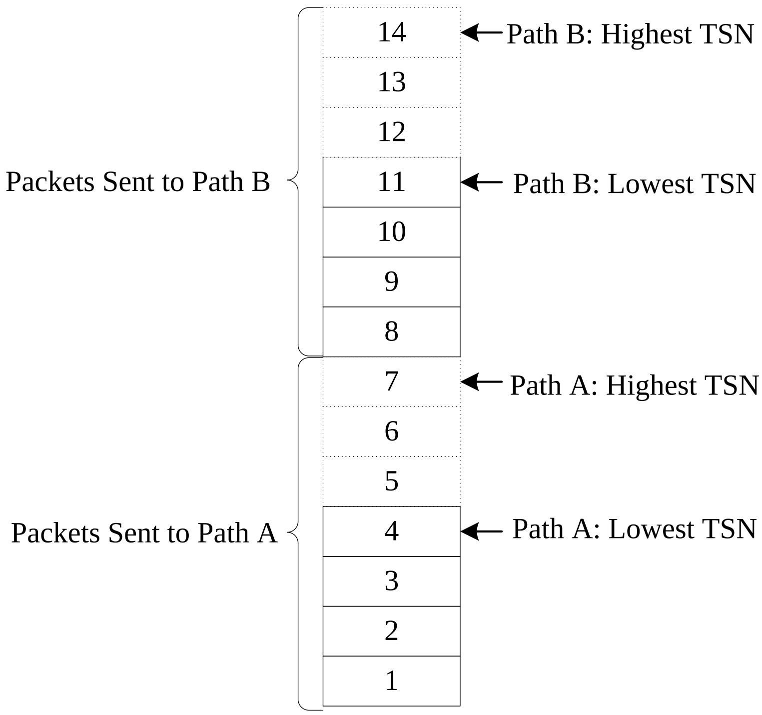

FIGURE 3.2: Marking of highest and lowest TSN at the sender side according to IMPD algorithm

3. SMT needs adaptive cwnd management for two types of missing packets (such as intra and inter-path) based upon IMPD algorithm.

SCTP is a reliable transport protocol, where the duplicate acknowledgments (dup_acks) are sent for each missing packet as a notification to the multihomed sender. The multihomed sender decides the retransmission of the missing packets on receiving these dup_acks. One such approach is named as a fast retransmit, which dictates retransmission of the packet whose three dup_acks are received, without lapse of a retransmission timeout. The underlying assumption in such retransmission is that the segment may be lost due to congestion as the three segments have already reached, as indicated by three consecutive dup_acks. The fast retransmit event is a quicker approach as compared to a timeout event. One major consequence of a fast retransmit event is the readjustment of cwnd of the stream that may be slashed down called fast recovery.

The strategy of IMPD algorithm is to classify the missing packets into inter-path or intra-path using some variables maintained on the sender side for each destination as discussed in the previous section 3.2.1. In the next phase, the highest and lowest valued variables are updated, which are used in missing packet classification. In the very next phase, this classification is used to increment the fast retransmit threshold counter for inter-path or intra-path missing packet. This process of IMPD algorithm is repeated for each outstanding packet (identified by unique TSN) as first time notified in the SACK and then moves toward next packet (Next TSN) as shown in Figure 3.3. For example, the packets with TSNs 1 to 7 are sent to destination A and packets with TSNs 8 to 14 are sent to destination B as shown in Figure 3.2. Then, according to IMPD algorithm, the lowest and the highest TSNs for destination A are 4 and 7. Similarly, the lowest and the highest TSNs for destination B will be 11 and 14 respectively. The missing packet notifications received at path B for a packet with TSN 5 to 7 are considered to be an inter-path missing packet.

Multihomed congestion control (MCC) mechanism is the second part of an SMT-MFR-II scheme, where two fast retransmit thresholds are maintained. The received SACK has cumulative acknowledgments for TSNs 4 and 11 i.e., intra-path fast retransmit threshold and inter-path fast retransmit threshold. By default, the static value of both fast retransmit thresholds for intra-path and inter-path is 3 in the SMT-MFR-II scheme. SMT-Adaptive Modified Fast Retransmit scheme has used the dynamic values of these fast retransmit thresholds, which kept on varying with respect to the available size of Rbuf as discussed in next section 3.5.

MCC mechanism is designed with two types of congestion control mechanisms, as shown in Figure 3.3. The first one is the standard congestion control mechanism, which is triggered in case of intra-path missing packet arrival. In this case, the missing packet is retransmitted on the fast link that will be helpful in avoiding the Rbuf blocking problem. At the same time, the cwnd of this destination decreases to its half value with the intention to avoid the congestion in the network by reducing the sending rate.

The second type of congestion control mechanism is used for inter-path missing packet arrival, where the missing packet is retransmitted without reducing the cwnd of that destination. As in this case, the packet is received out of order

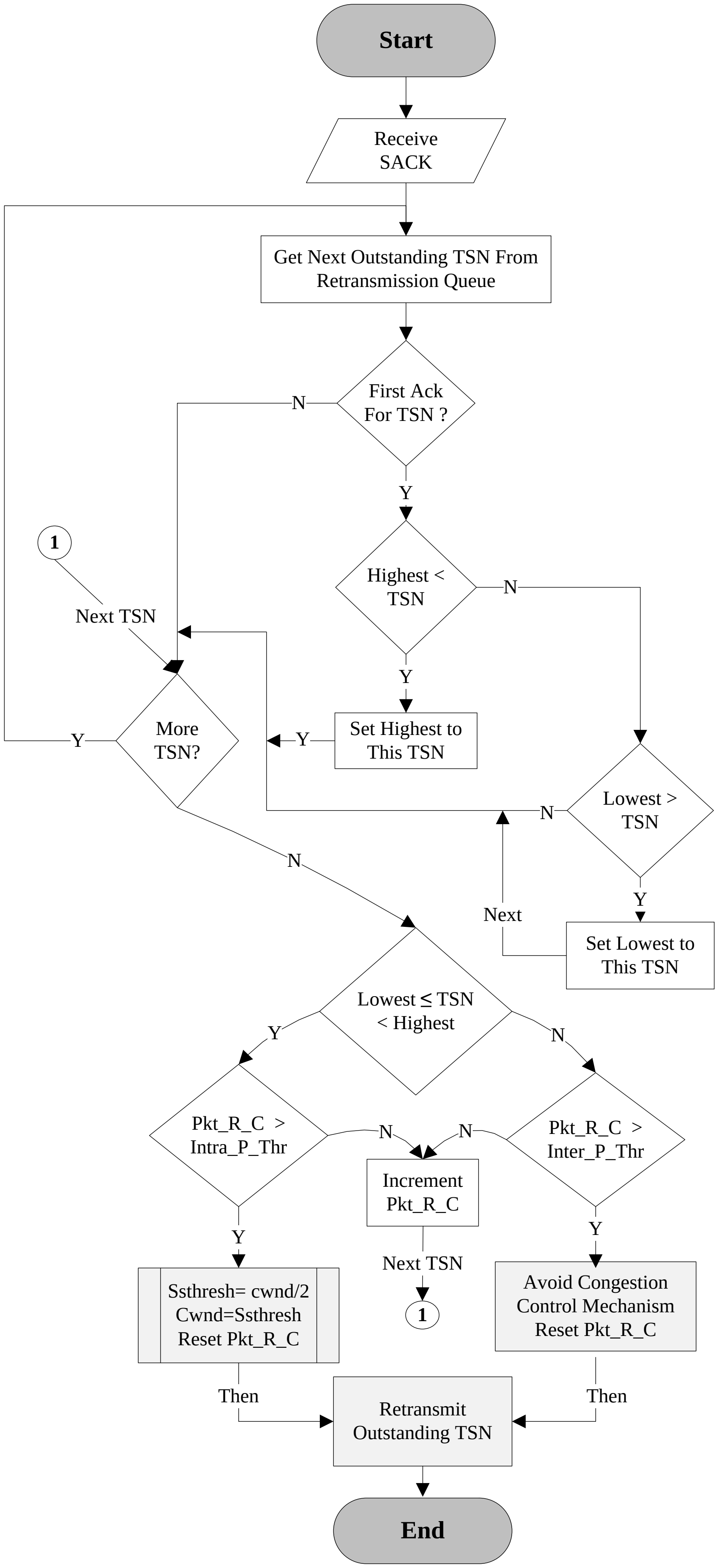

Figure 3.3: IMPD and MCC of SMT-MFR-II scheme

due to multiple path effects, not due to congestion in the network. Therefore, there is no need to reduce the cwnd in this scenario. This strategy helps in avoiding the Rbuf blocking problem as well as by maintaining normal cwnd growth.

On the other hand, excessive retransmission of duplicate packet on fast link diminishes the SMT benefit of bandwidth aggregation by resending slow link traffic on the fast link. This chaotic performance degradation can be solved by adaptive multihomed congestion control mechanism as mentioned in the next section.

3.5 SMT-Adaptive Modified Fast Retransmit

The traditional single path congestion control mechanism uses static fast retransmit threshold to retransmit the missing packets with the assumption that delayed packet after a specific time interval (3 Duplicate Acknowledgements) is considered to be a lost packet. In case of multipath transmission, the inter-path missing packets are due to disparities in bandwidth and delay features of multiple paths. According to the 3rdinference of IMPD algorithm (in section 3.3.2), there is a need for adaptive multihomed congestion control (AMCC) mechanism to provide an arrival opportunity to the inter-path missing packet. AMCC mechanism is designed to efficiently utilize the available Rbuf space by increasing the fast retransmit threshold value to an extent just before the Rbuf blocking occurs. This provides enough waiting time for reception of inter-path missing packets. The AMCC mechanism decreases the amount of fast retransmit events. AMCC has reduced the duplicate data transmission and hence, improved the aggregated throughput.

AMCC mechanism uses a tradeoff between the fast retransmit threshold value and available Rbuf space to provide extra time for inter-path missing packets. This tradeoff is decided on the basis of a number of simulation scenarios, where normalized Rbuf space is considered with respect to different values of fast retransmit threshold. In these simulation scenarios, the multihomed sender (MHS) transmits data simultaneously to the multihomed receiver (MHR) using two multiple paths, named path A and B as shown in Figure 3.4.

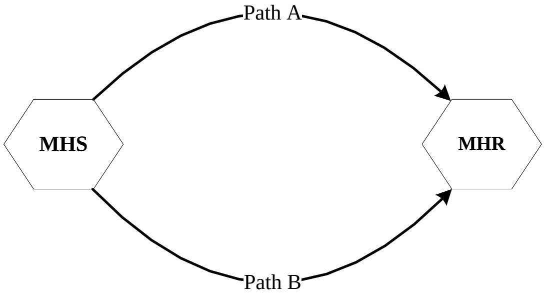

Figure 3.4: Multipath transmission between multihomed sender (MHS) and receiver (MHR)

Table 3.1: General configuration parameters of five simulation scenarios

Parameters	**Values**
Traffic source	File transfer protocol (FTP)
Stream	(single stream) 1
Transport protocol	SMT-SCTP
Packet size	1500 Bytes
Receiver Buffer (Rbuf)	65536 Bytes (65 KBytes)
Probabilistic Packet Losses in Path A	0.01
Fast retransmit Thresholds ($F_{RT}(m\|n)$)	$F_{RT}(3\|4), F_{RT}(4\|5), F_{RT}(5\|6)$
Rbuf Size Wise Pivot Points	10, 20, 30,.,., 100 (%)

These simulation scenarios have the bandwidth or delay based disparity and 0.01 probabilities of packet losses in path A. The general configurations of parameters used in these scenarios are mentioned in table 3.1.

In table 3.2, there are five scenarios that use different values of bandwidth and delay for multiple companion paths. The first one is the simplest scenario, where the bandwidth and delay of both paths are kept same. In 2nd and 3rd scenario, the delay of multiple paths is kept same while the bandwidth is configured differently for each path in order to evaluate the bandwidth based disparity issue. In these scenarios, the bandwidth of one path is increased by multiple times of bandwidth of a second companion path, to thoroughly evaluate the bandwidth based disparity effect on multipath transmission.

Table 3.2: General configuration parameters of five simulation scenarios

Scenario	**Path**	**Bandwidth (Mb/Sec)**	**Delay (ms)**
1st	A	1.0	45
	B	1.0	45
2nd	A	0.2	45
	B	1.0	45
3rd	A	0.5	45
	B	1.0	45
4th	A	1.0	45
	B	1.0	90
5th	A	1.0	45
	B	1.0	135

In last two simulation scenarios, (i.e., 4^{th} and 5^{th}), the bandwidth is kept same while the delay of one path is increased multiple time of delay of the second path. This has helped us during the analysis of the issues due to delay based disparities in multipath transmission.

In these simulation scenarios, the total Rbuf space is classified into multiple pivot points, i.e., 10, 20,.,., 100 in terms of percentage. Each pivot point in a scenario indicates the available Rbuf space. Various fast retransmit threshold (F_{RT}(m|n) values are used for each pivot point where m and n are the values of fast retransmit threshold (F_{RT}), used before and after that pivot point. This means that the inter-path missing packet will be retransmitted on receiving m duplicates Acks, if the available Rbuf space is less than pivot point and n duplicate Acks if the available Rbuf space is greater than or equal to the pivot point.

At least 20 percent Rbuf space is essential for the SMT-AMFR scheme to buffer at least more than 3 inter-path missing packets to generate 3 duplicate Acks. Figure 3.5 provides better overview to understand the relation between the available Rbuf spaces and fast retransmit threshold policies. This enables us

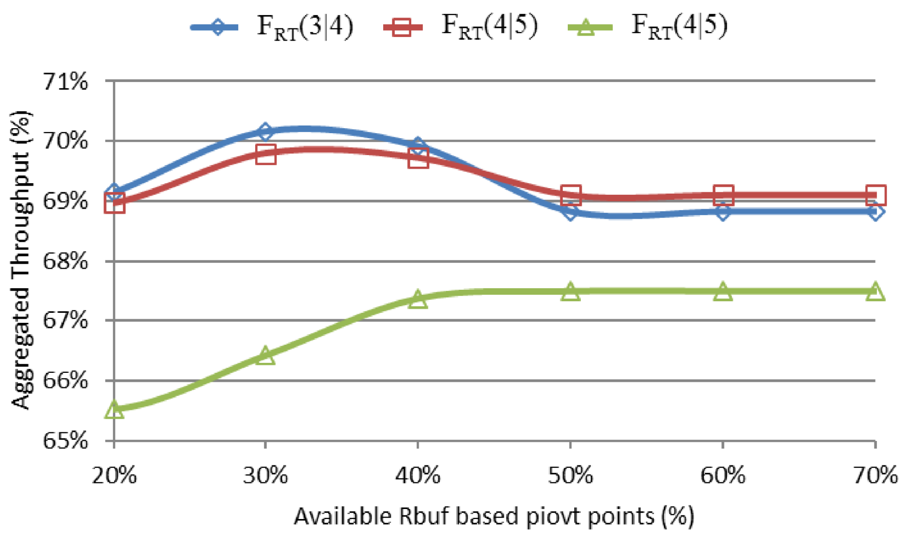

FIGURE 3.5: Average aggregated throughput of multiple paths (A & B), with bandwidth and delay based disparities

to derive the following analysis, which assists us in presenting the SMT-AMFR scheme of the multihomed congestion control mechanism.

1. $F_{RT}(3|4)$ helps in high aggregated throughput, when the multihomed receiver has available Rbuf size less than 45%.
2. $F_{RT}(4|5)$ is more effective when the multihomed receiver has available Rbuf size greater than 80%.
3. $F_{RT}(5|6)$ should be avoided as its aggregated throughput is less than both $F_RT(3|4)$ and $F_RT(4|5)$.

There is a need for an adaptive fast retransmit strategy, where the fast retransmit threshold jumps from one policy to another with respect to available Rbuf space. In light of these recommendations, Rbuf space is categorized into risk zones of inter- path missing packets arrival i.e., critical, substantial and moderate zone, marked by two pivot points i.e., border of critical zone (B_C) and the border of the substantial zone (B_S), as shown in Figure.3.6.

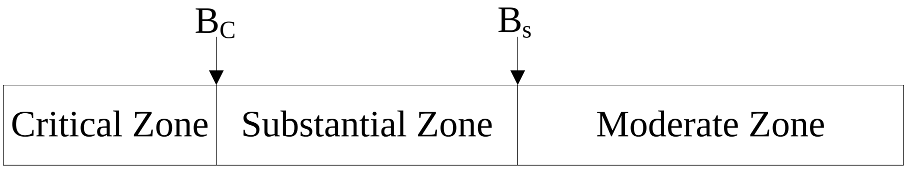

Figure 3.6: Rbuf space categorization with respect to performance risk

The initial position of these pivot points is selected as by default values, based on previous results of simulation scenarios. These pivot points move forward or backward to increase or decrease the specific Rbuf zone with the intention to maximize the aggregated throughput. The selection of specific fast retransmit threshold is based upon the following SMT-AMFR algorithm, as shown in Algorithm 2. To further elaborate on the functionality of SMT-AMFR scheme, Figure. 3.7 represents the state transition diagram of the SMT-AMFR scheme. The shifting of the a_rwnd from one zone to another depends on its size. At the start of the multipath transmission, the whole Rbuf size is advertised to the sender as a_rwnd. At this stage, if the a_rwnd is greater than B_S pivot point then it is in the moderate zone where the fast retransmit threshold value is set to 5 for the missing packet. In the moderate zone, the receiver has enough available Rbuf space to accommodate the incoming OOS packet and provide an arrival opportunity for delayed inter-path missing packets. During the moderate zone stage, if a_rwnd is reduced and its size occurred between B_C and B_S pivot point ($B_C <$ a_rwnd $\leq B_S$), a_rwnd jumps to substantial zone. In the substantial zone, the fast retransmit threshold value is set to 4. The inter-path missing packets whose fast retransmit counter is equal or greater than 4 are retransmitted.

Advertised receiver window (a_rwnd) in a substantial zone can move gradually back to moderate zone if the a_rwnd remains in substantial for a specific number of cycles mentioned by occurrence threshold (O_T). The advertised receiver buffer (a_rwnd) in substantial zone moves to critical zone, if its value is less than a B_C pivot point. The fast retransmit threshold value is configured to 3 in critical zone and by default, values for B_S and B_C pivot points are initialized again. In the case of drastic degradation of a_rwnd, each zone i.e., moderate and substantial zones have the ability to shift to the critical zone. In this way, a_rwnd

Algorithm 2 SMT-Adaptive Modified Fast Retransmit (AMFR) Scheme

Input: (a_rwnd, B_C, B_S, O_T).
Output: Fast Retransmit Threshold (F_{RT}) value that efficiently manages the waiting time for retransmission of the missing packet at specific destination D.
Let available receiver buffer size advertised to the receiver is $Rbuf_A$ and occurrence threshold value be O_T .

if $a\_rwnd < B_C$ **then**
  $F_{RT} = 3$; {Critical zone}
  Initialized B_C and B_S to by-default values
else
  if $a\_rwnd \geq B_C$ && $a\_rwnd < B_S$ **then**
    $F_{RT} = 4$ {Substantial zone}
    Moderate_count ++
    if $Moderate\_count \geq O_T$ **then**
      $B_S = a\_rwnd$ - 0.001
      $Moderate\_count = 0$
    end if
  end if
else
  if $a\_rwnd \geq B_S$ **then**
    $F_{RT} = 5$ {Moderate zone}
  end if
end if

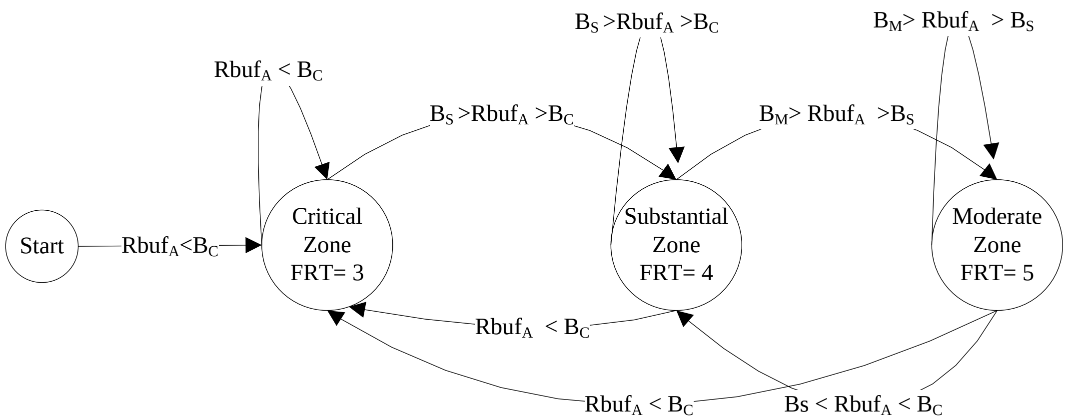

FIGURE 3.7: State transition diagram (STD) of adaptive modified fast retransmit (SMT-AMFR)

moves between the risk zones in order to provide an arrival opportunity to the inter-path missing packets and avoid Rbuf blocking at the same time.

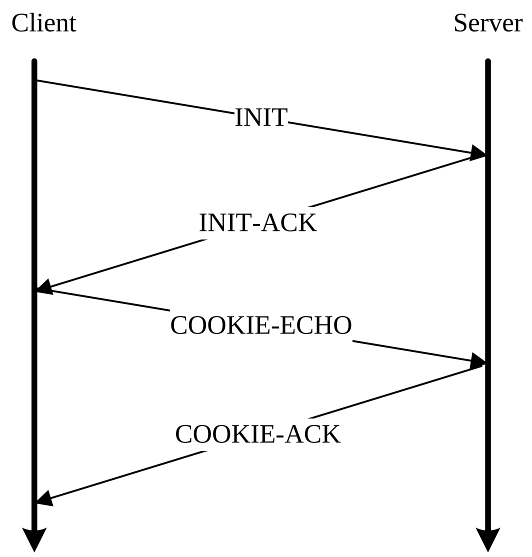

FIGURE 3.8: SCTP Four-way handshake for connection establishment

3.6 Proposed SMT Architecture

The SMT schemes (SMT-MFR and SMT-AMFR) have utilized the association based connection features of SCTP to exchange the list of IP addresses between sender and receiver. SCTP has four-way handshake method to establish the association between the client (sender) and server (receiver), as shown in Figure 3.8. The detailed description of the four-way handshake method is given below:

1. The server executes the socket, bind and listen command to prepare itself for the incoming association request from the clients side.

2. The client creates an association by sending a connect message command to the server. This enables the client SCTP to send INIT message to the server, which contains the list of internet protocol (IP) addresses at the client side, initial sequence number and other association-related information.

3. After receiving the initiation (INIT) message from the client side, the server stores the list of available IP addresses on the client side. The server acknowledges the INIT message of the client with its own initiation acknowledgment (INIT-ACK) message. This server INIT-ACK has the list of IP addresses

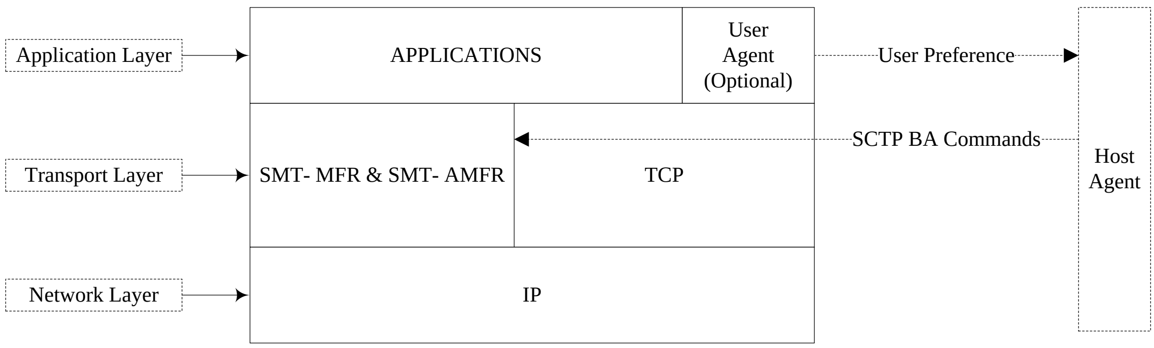

Figure 3.9: Proposed SMT architecture

available on the server side, initial sequence numbers and other association-related information. In this way, the client has detailed information about the list of available IP addresses on the server side.

4. In the next phase, the client sends COOKIE-ECHO message to the server, which is acknowledged by the server with the COOKIE-ACK message. COOKIE-ECHO and COOKIE-ACK have client and server related information and may also contain user data.

Once the association is established between the sender and receiver, the heartbeat packets used to check the present path status and further explore the additional IP address available for multipath transmission (Stewart, 2007). SCTP can handover packets to IP packets or can be tunneled using UDP packets for ease of deployment (Tuexen and Stewart, 2007).

Figure 3.9 shows the proposed architecture to provide the SMT schemes at the transport layer. This architecture is a combination of the application layer, transport layer, network layer and an optional cross layer component named host agent. The application layer must have the support of SCTP and have an optional user agent. If the application has a lack of support for user agent then all the available paths are used for bandwidth aggregation. The user agent will be used for the willful bandwidth aggregation service and configuration of user preferences about applications and links preferences.

In case of the presence of more than two multiple paths, the user agent provides an interface to the user for selecting the desired number of multiple paths

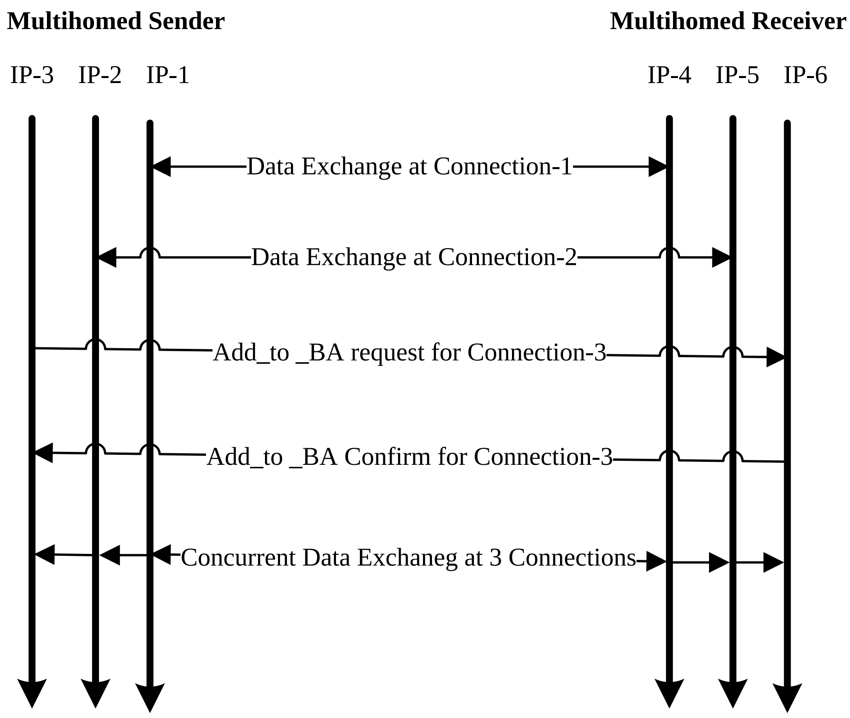

FIGURE 3.10: Bandwidth aggregation exchange between multihomed sender and receiver

for transmission. The availability of multiple paths related information to the user agent is provided by the cross-layer decision module called the host agent. The host agent has access to the association module of SCTP. The host agent receives user preferences about multiple companion paths from user agent and communicates these bandwidth aggregation related instructions to the association module using add to bandwidth aggregation (Add_to_BA) commands as shown in Figure 3.10.

Multiple connections of an application can be established under each association as shown in Figure 3.11. In this way, single application flow is distributed over multiple connections in light of user preferences.

3.7 Summary

This chapter 3 presents the proposed SMT schemes (SMT-MFR and SMT-AMFR). The SMT-MFR scheme differentiates the missing packet into intra and inter-path

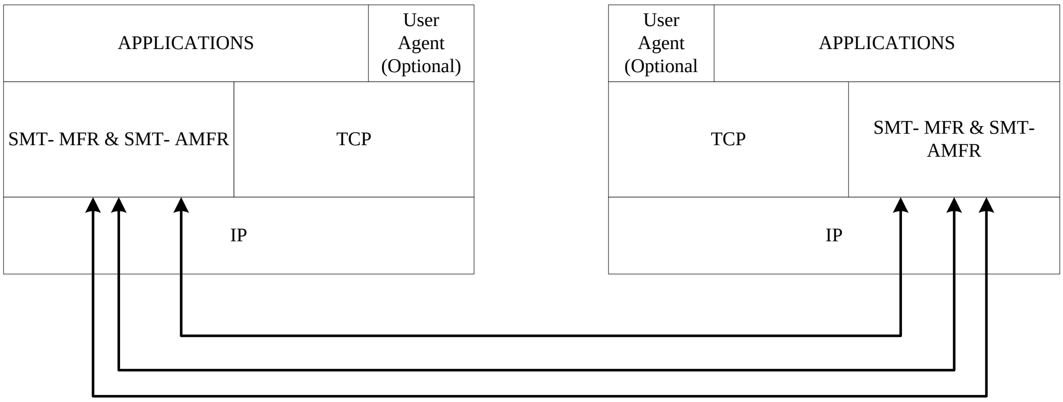

FIGURE 3.11: The connections diversity in SMT-MFR and SMT- AMFR

and manages the multihomed congestion window accordingly, which has improved the aggregated throughput. The SMT-AMFR further optimizes the SMT-MFR by efficiently utilizing the available Rbuf by partitioning it into three zones, i.e., critical, substantial and moderate zones. The fast-retransmit threshold value is not static in this case and kept changing concerning its zone. This chapter ends with proposed SMT architecture which briefly mentioned the SMT schemes (SMT-MFR and SMT-AMFR) at the transport layer and their interaction with host agent. The exchange of connection setup and bandwidth aggregation messages between multihomed sender and receiver are discussed.

Chapter 4

RESULTS AND DISCUSSION

This chapter elaborates the obtained results of the SMT schemes i.e., SMT-MFR and SMT-AMFR and their discussion. Before proceeding towards the results and discussion of SMT schemes, section 4.1 presents the implementation detail of the SMT schemes in network simulator -2 (NS-2) (Network Simulator, 2009). Section 4.2 provides the topological detail of the simulation scenarios and their configuration used for the performance analysis of the proposed SMT schemes. The definition and mathematical formulation of performance analysis parameters used in theses simulation scenario are mentioned section 4.3. Finally, the performance analysis of the SMT-MFR and SMT-AMFR using these performance analysis parameters is given in section 4.4 and 4.5 respectively. This chapter ends with concluding remarks that wrap up the merits and demerits of the SMT-MFR and SMT-AMFR schemes and a gap of research work to be tackled in future.

4.1 SMT Schemes Implemented in NS-2

The SMT (SMT-MFR and SMT-AMFR) schemes are implemented in network simulator-2 (NS-2), which is a discrete event network simulation tool. NS-2 is publicly available for personal, educational and industrial uses to perform research related activities and development.

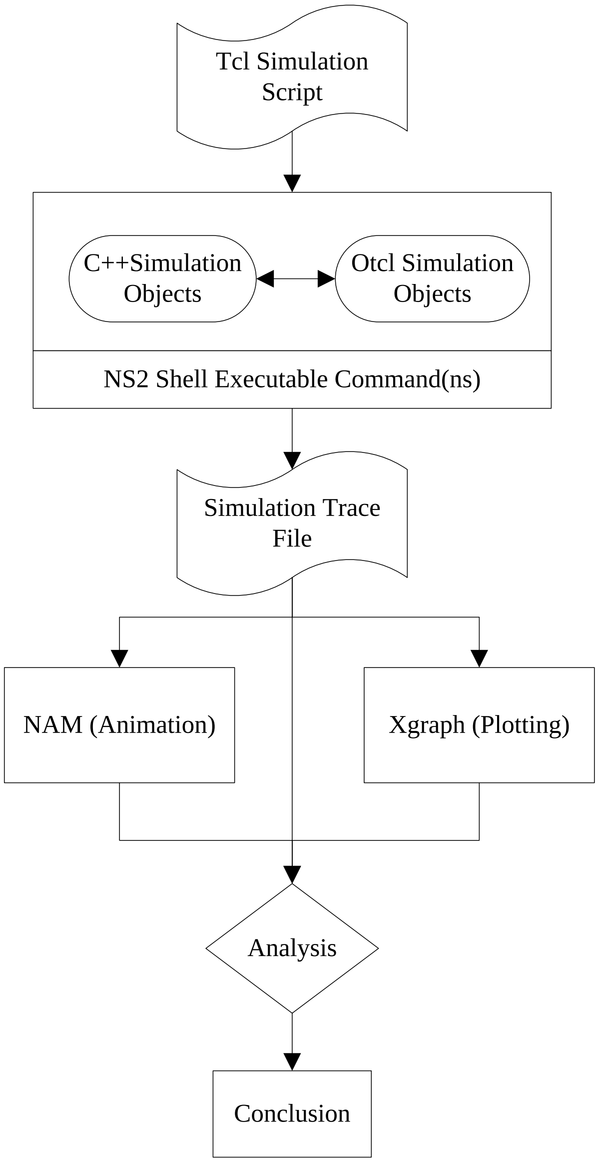

FIGURE 4.1: Basic operational data flow diagram of network simulator -2 (NS-2)

NS-2 uses the split-programming concept where OTCL is used at the front end interface and C++ language runs at the back end, as shown in Figure 4.1. C++ is used to run the main objects of simulation, while their parameters are written in OTCL. OTCL provides a user-friendly interface where the user can easily configure realistic network scenarios by writing script files. These OTCL script files are processed by NS-2 to generate network animator (NAM) and trace files. Network animator (NAM) and xgraph tool are used for the analysis of trace files in order to conclude results. SMT schemes are implemented by extending the SCTP module of NS-2 hierarchy, as shown in Figure 4.2.

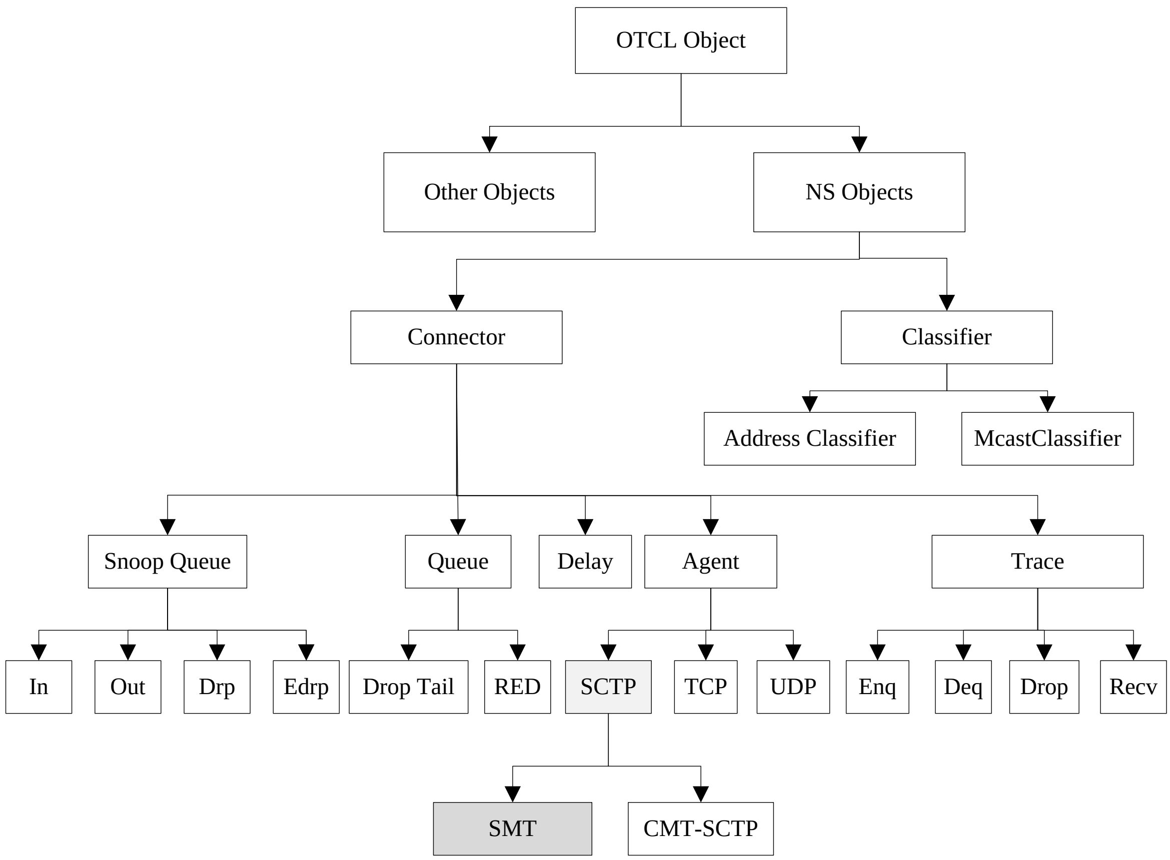

FIGURE 4.2: SMT schemes implementation in NS-2 class hierarchy

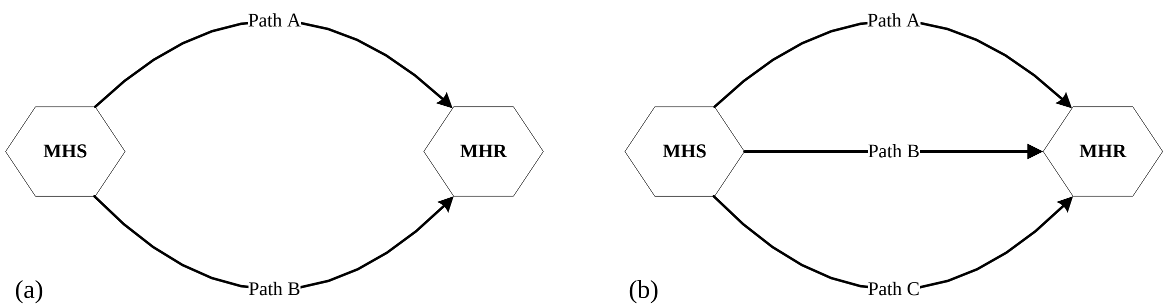

FIGURE 4.3: Simple simulation scenarios (a) with two paths (b) with three paths

4.2 Simulation Scenarios

Six simulation scenarios are implemented using a multihomed system model as discussed in chapter 2. The simple multihomed scenario with two paths (A & B) and three paths (A, B & C) are shown in Figure 4.3 (a & b). Each path is configured with specific bandwidth, delay and path feature.

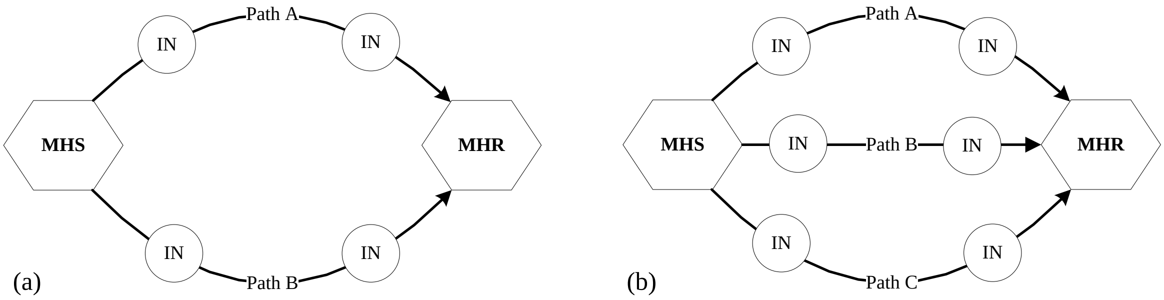

FIGURE 4.4: Simulation scenarios having intermediate nodes (a) with 2 paths (b) with 3 paths

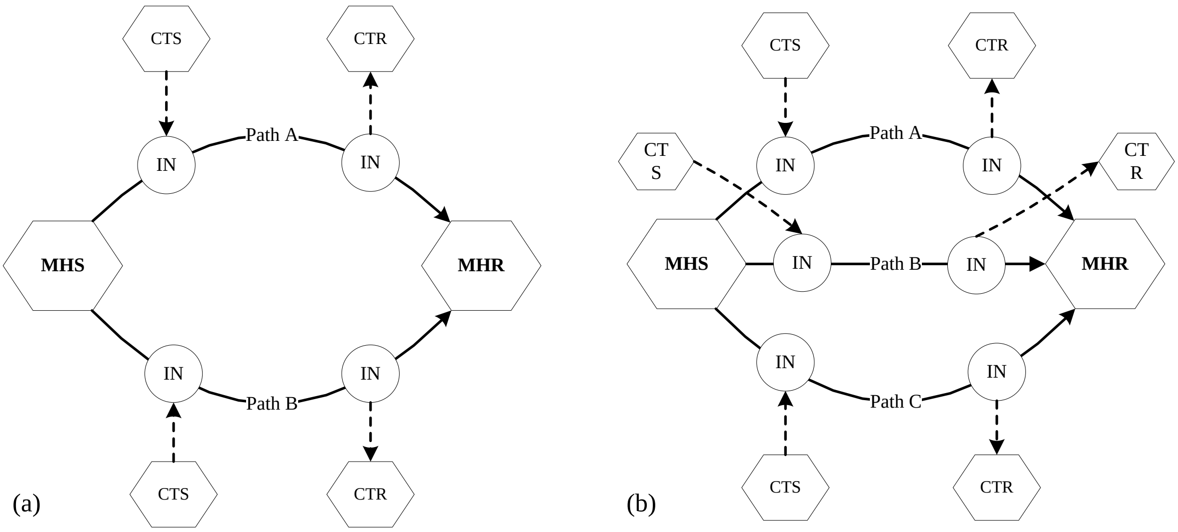

FIGURE 4.5: Simulation scenarios having cross traffic (a) with 2 paths (b) with 3 paths

The scenarios with three paths are used to find out the effect of increasing the number of paths from 2 to 3. The relatively complex scenarios are also designed to find out the effect of intermediate nodes and cross traffic for performance evaluation of SMT schemes, as shown in Figure 4.4 and 4.5.

The cross traffic is initiated before the data is transmitted on multiple paths. Doing so, the cross traffic pre-occupies bandwidth resources on the path and hence, creates a comparatively complex realistic simulation scenario.

The generalized configuration of simulation parameters is mentioned in table 4.1. Same parameters are configured, which are being used normally by other research studies in this area (Lyengar et al., 2006; Cao et al., 2015; Wallace et al., 2010). The file transfer protocol (FTP) is used for data transmission using

Table 4.1: Generalized configuration of simulation scenarios

Parameters	**Values**
Traffic source	File transfer protocol (FTP)
Stream	(single stream) 1
Transport protocol	SMT-MFR
Packet size	1500 Bytes
Receiver window (rwnd)	65536 Bytes
Cross traffic	Constant bit rate (CBR)

single stream, which is split among multiple connections in order to support the single sequence number concept. The SMT schemes are used at the transport layer with a packet size of 1500 bytes. The receiver buffer size of 65536 bytes is reserved as by default Rbuf space for each connection that can accommodate theoretically the aggregated flow of 11.65 Mbps $\left(\frac{65536*8}{\frac{45}{1000}}\right)$ between the sender and the receiver. The fixed propagation delay is 45 milliseconds between the sender and the receiver. The UDP is used as cross traffic at the transport layer, whereas constant bit rate (CBR) multimedia streaming is used at the application layer. This research study does not focus on a multipath scheduler that is why, the CBR cross traffic is selected due to the fact that CBR generates multimedia streaming with the uniform rate throughout the simulation time. This helps us to analyze the performance of SMT schemes in a coherent way.

4.3 Performance Analysis Parameters

In this study, normalized format of performance analysis parameters is used such as, congestion window (cwnd), receiver window (rwnd), throughput, aggregated throughput, aggregated bandwidth utilization and end to end delay (E2e) for SMT schemes (SMT-MFR, SMT-AMFR). The brief descriptions of data normalization and these parameters are given below.

4.3.1 Congestion Window (cwnd)

Congestion window (cwnd) is defined as the sender-side limitation on the amount of data that a sender can transmit into the network without receiving an ACK. Both a_rwnd and cwnd are used to regulate the data flow using flow control and congestion control mechanism.

$$Sending\ window(swnd) = \min(a\_rwnd, cwnd) \tag{4.1}$$

4.3.2 Advertised Receiver Window (a_rwnd)

a_rwnd is a control variable which is used to state most recently calculated receiver advertise window. This is the receiver side limit on the amount of outstanding segments used for end to end flow control. In simultaneous multipath transmission, the receiver announces maximum a_rwnd with the reception of insequence data until OOS segment arrives, which causes a decrease in a_rwnd space.

4.3.3 Throughput

Throughput refers to the quantity of error-free data received at the receiver side per unit of time. The paths throughput is limited by a_rwnd and the round trip time (RTT).

$$Throughput \leq \frac{a\_rwnd}{RTT} \tag{4.2}$$

Even if there is no segment loss, the throughput cannot be increased more than (a_rwnd/RTT) at any time.

4.3.4 Aggregated Throughput

Throughput can be defined as the data that has actually received per unit time. In this study, simultaneous data transmission on multiple paths, aggregated throughput (T_{Agg} %) in percentage is used to have precise knowledge of achieved throughput (in bits/Sec) with respect to available path capacity (in bits/Sec). Mathematically, aggregated throughput (T_{Agg} %) is given by the following equation:

$$Aggregated\,Throughput(T_{Agg}\%) = \frac{T_{Agg}}{C_{Agg}} \times 100 \tag{4.3}$$

Where, T_{Agg}, C_{Agg} represent the aggregated throughput and aggregated capacity of the sum of n multiple paths (P_n) respectively.

In case of using more than one path for data transmission, the capacity of multiple paths is combined using aggregated capacity (C_{Agg}) parameters. Mathematically, aggregated capacity is represented by the following equation:

$$Aggregated\,Capacity(C_{Agg}) = \sum_{i=1}^{n} C_i \tag{4.4}$$

Where, C_i represents the capacity of the i^{th} path. The aggregated throughput (T_{Agg}) is the summation of average throughput of multiple paths, as mentioned in the following equation:

$$Aggregated\,Throughput(T_{Agg}) = \sum_{i=1}^{n} T_{Aver_i} \tag{4.5}$$

Where, T_{Aver_i} represents the average throughput of i^{th} path. The averaged throughput is defined as the total size of data received during simulation time and calculated by the following formula:

$$Average\,Throughput(T_{Aver}) = \frac{\sum_{i=1}^{n} Pkt\_count_i \times Packet\_size}{Total\_time(T_N)} \tag{4.6}$$

Whereas, the total data received at receiver side is calculated by multiplying total number of packets received (Pkt_count_i) with their packet size.

4.3.5 Aggregated Bandwidth Utilization

Aggregated bandwidth utilization is defined as the achieved aggregated throughput related to the total bandwidth of multiple paths. It is measured in percentage.

$$Aggregated\,Bandwidth\,Utilization(\%) = \frac{\sum_{i=1}^{n} Throughput_i}{\sum_{i=1}^{n} Path\ Bandwidtht_i} \times 100 \quad (4.7)$$

4.3.6 End to End (E2e) Delay

The E2e delay can be defined as the time, consumed by the segment during transmission across a network from source to final destination. E2e delay can be represented as:

$$E2e = d_{Trans} + d_{Prop} + d_{Proc} + d_{Queu} \quad (4.8)$$

Whereas,
d_{Trans} = Transmission delay,
d_{Prop} = Propagation delay,
d_{Proc} = Processing delay and
d_{Queu} = Queuing delay.

In this study, the transition and processing delay are considered negligible (Wallace, 2012). The propagation and the queuing delay are two main contributing factors towards E2e delay. With efficient congestion control mechanism and queuing management techniques in multipath communication, the unnecessary delay can be avoided to minimize the arrival of OOS segment and Rbuf blocking problem.

4.3.7 Data Normalization

Parameters such as throughput, advertised receiver window (a_rwnd) and cwnd are normalized to bring them into proportion with one another. Normalization adjusts the appropriate scaling of the coefficients associated with these different parameters for disparity. The normalization has helped us in the analysis of meaningful relative activity between these parameters (Bolstad, 2012). Normalization scales the values of data between 0 and 1, as shown in the following equation:

$$Z_i = \frac{x_i - x_{\min}}{x_{\max} - x_{\min}} \tag{4.9}$$

4.4 SMT-Modified Fast Retransmit Results

The multihomed congestion control (MCC) mechanism in SMT-MFR creates a virtual queue at sender side for non-acknowledged packets for each destination. The virtual queue keeps the record of highest and lowest sequence numbers of the packets transmitted to a destination. If the sequence number of the missing packet is within the range of the lowest and the highest sequence number of the same path, where it is transmitted, then this packet is considered to be intra-path missing packet, otherwise, it is an inter-path missing packet. The MCC mechanism maintains a fast retransmit threshold value, static for intra-path and dynamic for inter-path missing packet. The fast retransmit event is triggered by the arrival of a number of SACK packets as mentioned by respective fast retransmit threshold. In case of the intra-path missing packet, the MCC mechanism reduces the cwnd to its half to decrease the congestion in the respective path, while being same in case of inter-path missing packet.

The previous multipath transmission schemes (i.e., NMT-SCTP and SCTP-CMT) do not differentiate the missing packet into intra and inter-path. SMT-MFR scheme is compared with NMT-SCTP and SCTP-CMT using simple simulation scenario, as mentioned in Figure 4.3(a). In this simple scenario, the multihomed

Table 4.2: Parameter configuration for simulation scenarios

Parameters	**Values**
Path A bandwidth & delay	0.2 Mbps & 45 milliseconds
Path B bandwidth & delay	1 Mbps & 45 milliseconds
Simulation scenario	Simple scenario with 2 multiple paths

sender sends a data file (file transfer protocol (FTP)) using a single stream, which is split between the two paths. The multihomed receiver has a standard Rbuf size of 65532 bytes, as shown in table 4.1.

Both paths have bandwidth disparity with same delay feature to create the effect of OOS packet arrival at the receiver side as mentioned in table 4.2. The path A is a slow link that creates the OOS packet effect on path B that is why the behavior of path A is found to be normal. Hence, the performance of path B is affected by OOS packet and is worthy of discussion here.

SMT-MFR has overcome the shortcoming of SCTP-CMT by maintaining persistent cwnd and higher aggregated throughput, as shown in Figure 4.6. The sending capability of a transport layer is defined by the cwnd and a_rwnd, as mentioned in equation 4.1. During data transmission, the sending window has to send the least data allowed by the cwnd and a_rwnd. Figure 4.6 shows that the cwnd remains high while the a_rwnd decreases or increases, which repeats this pattern throughout the multipath transmission. Here, a_rwnd is the effective parameter that has an upper limit on the sending window for data transmission.

There is a need to study the behavior of SMT-MFR in handling the OOS and missing packets in order to understand the pattern of increase and decrease in a_rwnd size. The complete Rbuf space is advertised as a_rwnd to the sender during the association establishment phase. a_rwnd is maximum at the early stage of data transmission, as shown in Figure 4.6. The data transmission of a single stream of data over multiple paths with a disparity in bandwidth causes the arrival of the OOS packets at the receiver side. The transport layer is responsible

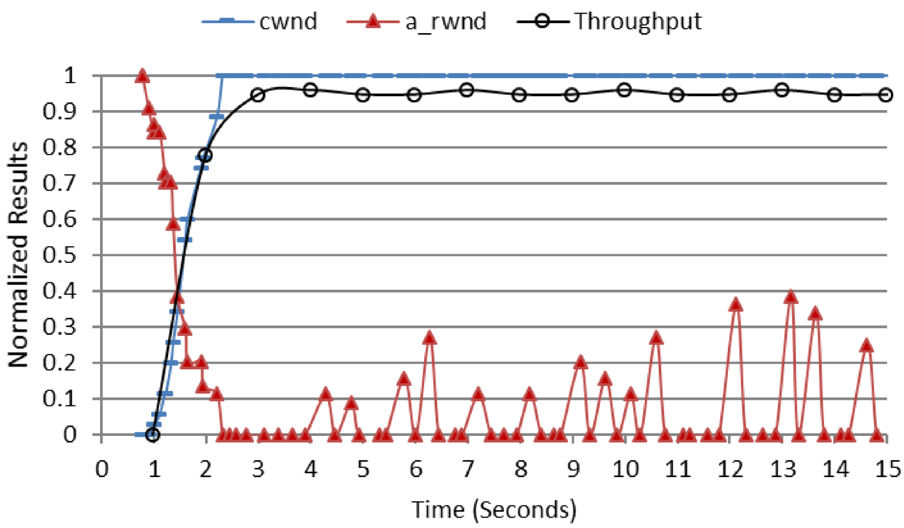

FIGURE 4.6: SMT-MFR path Bs cwnd, a_rwnd and throughput

for transmitting insequent data to the application layer, while the OOS packets are buffered in Rbuf. This decreases a_rwnd, which is conveyed to the SMT-MFR at the sender side through the SACK. The SMT-MFR inferences the missing packets information from the gap block variables mentioned in SACK. The SMT-MFR differentiates the missing packets into intra and inter-path. The SMT-MFR fast retransmits the missing packet to the multihomed receiver using the multihomed congestion control (MCC) mechanism. On getting these retransmitted missing packets, the receiver gets the OOS packet into insequent data packets, which are transmitted to the application layer. This makes free space in Rbuf due to which, larger a_rwnd is advertised to the receiver. Here, the packet gets OOS and inter-path missing packet notifications are due to multipath effect. There is no congestion in the multiple paths and no packet is dropped due to congestion. That is why the SMT-MFR does not decrease cwnd during fast retransmission of inter-path missing packet and cwnd remains high, as shown in Figure 4.6. This ability of differentiating the missing packets into inter-path and intra-path avoids the CC mechanism for inter-path missing packet arrival yielding high throughput, which is maintained throughout the simulation time. The fast retransmission of intra-path missing packet helps in avoiding the Rbuf blocking issue at the receiver side which is the main cause of throughput degradation in SCTP-CMT.

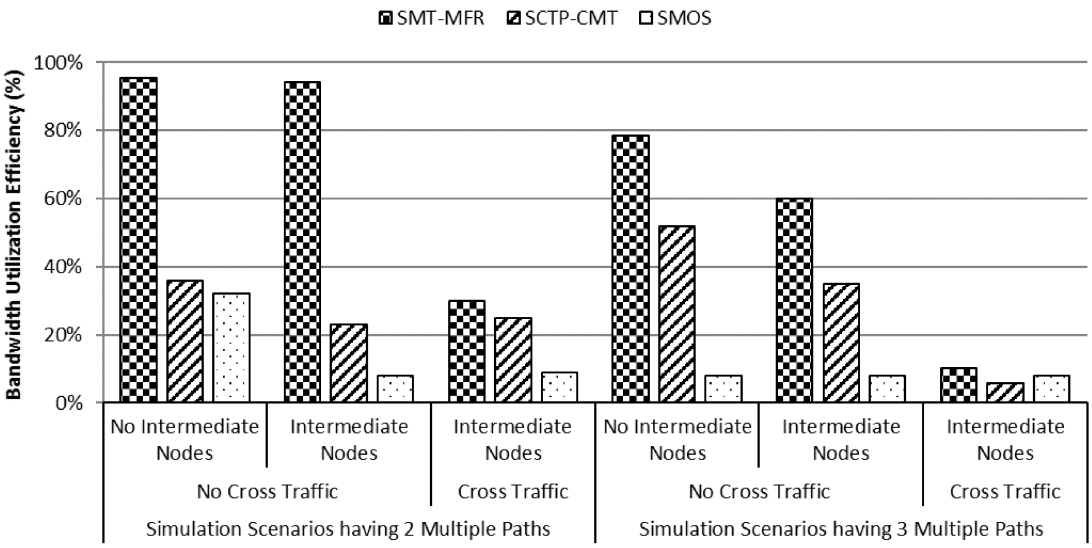

Figure 4.7: Aggregate bandwidth utilization of SMT-MFR, SCTP-CMT & SMOS

Extensive simulations are carried out to thoroughly analyze the SMT-MFR performance gain along with SCTP-CMT in various topological configurations, as shown in Figure 4.5. The SMT-MFR has improved aggregated throughput ranging from 164% to 72.4% (in a worst case scenario) as shown in Figure 4.7. The throughput of SMT-MFR is degraded in three paths scenario as compared to two paths scenario. However, it is better than CMT and SMOS [37] in all the cases. SMT-MFR does not deal with scheduling of packet on multiple paths. The transmission of data on more than two paths can be handled in a better way by a suitable scheduler, which will be the focus of future research work.

A weaker side of SMT-MFR is the fast retransmission of the inter-path missing packet by waiting for 3 duplicates acknowledgments and without considering the availability of free Rbuf space. In the worst case scenario, most of the inter-path missing packets of the slow link are transmitted over a fast link. This problem is solved by SMT-adaptive modified fast retransmit (AMFR) scheme. The SMT-AMFR utilizes the dynamic adaptive threshold points for fast retransmission of missing packets with respect to available Rbuf space.

4.5 SMT-AMFR Results

SMT-MFR successfully avoids the Rbuf blocking in multipath transmission by inter-path missing packet differentiation (IMPD) and multihomed congestion control (MCC) mechanism as mentioned in section 4.3. The SMT-MFR performance is degraded by introducing other path features such as, intermediate nodes and cross traffic, as shown in Figure 4.7. The intermediate nodes add more end-to-end delay to the path, while cross traffic causes random packet losses. SMT-AMFR is proposed to efficiently utilize the Rbuf space while retransmitting the inter-path missing packet, as discussed.

SMT-AMFR is simulated using diverse types of scenarios that have a disparity in path features such as, bandwidth and delay. Each scheme (SCTP-CMT, SMT-MFR & SMT-AMFR) is evaluated against various receiver buffer sizes (ranging from 32 to 512 kilobytes) to find out the effect of limited Rbuf space on the efficiency of multipath schemes. These experiments are limited to two path simulation scenarios to minimize the effect of a number of parallel paths (more than two) with the intention that the suitable number of path selection is handled by an appropriate scheduler in future research work. The general parameter configurations in the scenarios are according to table 4.1, unless explicitly specified in a particular simulation scenario. The bandwidth and delay of a path A kept on changing in each scenario, while path B has the same bandwidth and delay (of 1Mbps and 45 milliseconds respectively). The simulation scenarios configuration like this helps us to generate the OOS packet and missing packet effect at path B. Performance efficiency of a scheme on the path B is most likely to be affected due to multipath features.

4.5.1 Bandwidth-Based Disparity

The proposed SMT-AMFR mechanism is evaluated along with SMT-MFR and CMT-SCTP in bandwidth based disparity scenarios. In bandwidth based disparity scenarios, the bandwidth of Path B is fixed (1 Mb/Sec), while path A bandwidth

Table 4.3: Bandwidth based disparity scenario parameters

Parameters	**Values**
Path A Bandwidth & Delay	0.1, 0.2, .,.,., 1.0 Mbps & 45 milliseconds
Path B Bandwidth & Delay	1 Mbps & 45 milliseconds
Receiver window (rwnd)	32/64/128/256/512 Kilo Bytes
Simulation scenario	Simple scenario with 2 multiple paths

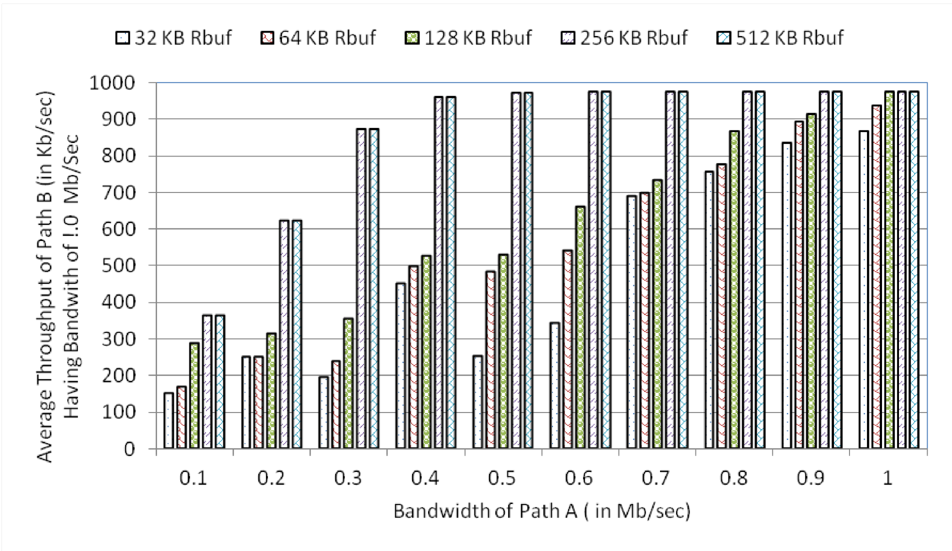

Figure 4.8: SCTP-CMT: average throughput of path B in bandwidth based disparity scenario

varies from 0.1 to 1 Mb/Sec with 0.1 increments in each subsequent scenario. In addition to this, the delay for both paths is kept same, as mentioned in table 4.3.

The performance of SCTP-CMT in terms of average throughput is gradually degraded by the incremental increase in bandwidth disparity and decrease in size of total Rbuf as shown in Figure 4.8.

The increase in bandwidth disparity motivates more generation of OOS packets and missing packet notifications. This situation becomes worse with a decrease in usage of total Rbuf space. Using very low Rbuf space like, 32KB, the SCTP-CMT average throughput oscillates in an abnormal way, due to low space for holding enough OOS packets, until fast retransmission of the missing packets is triggered. This causes degradation in an average throughput of SCTP-CMT.

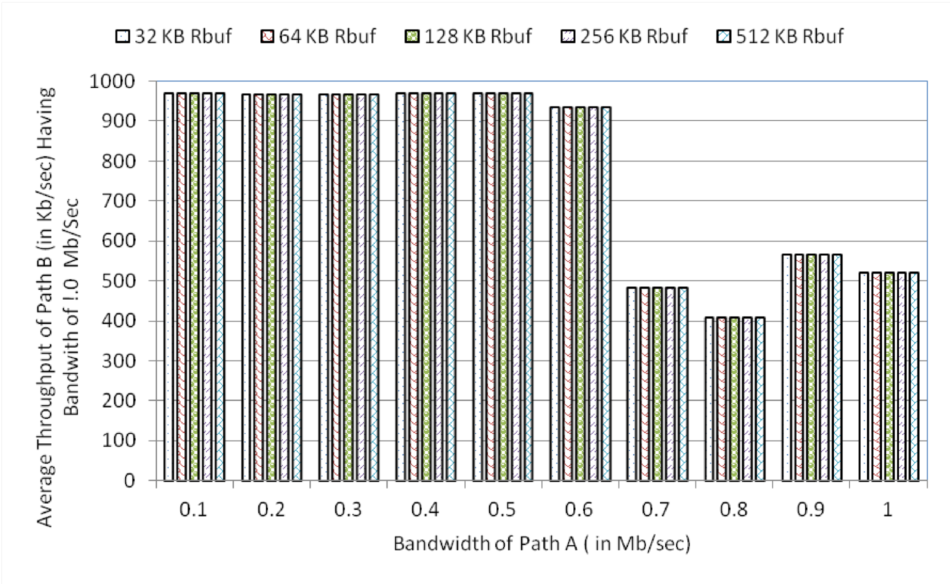

FIGURE 4.9: SMT-MFR: average throughput of path B in bandwidth based disparity scenario

On the other hand, SMT-MFR outperforms SCTP-CMT by maintaining higher average throughput until the disparity increases with the increase in bandwidth of path A beyond 0.6 Mb/Sec, as shown in Figure 4.9.

The average throughput of SMT-MFR remains high up to 0.6 Mb/Sec. In these scenarios, the missing packet notifications are mostly received at path B. By increasing the bandwidth of path, A from 0.7 Mb/Sec to 1Mb/Sec, the average throughput of SMT-MFR decreases. Here, the missing packet notifications are received approximately on both paths (A & B), due to less difference in their bandwidth. Both paths fast retransmit the missing packet on an alternate path. This further increases the queue of outstanding packets on each path for retransmission. The average throughput of SMT-MFR does not affect considerably by an increase in Rbuf size (ranges from 32KB to 512 KB). This reveals that SMT-MFR is quite resistant to variation in Rbuf size.

This complex situation of missing packet retransmission on both paths leads to a decrease in the throughput. This shows the aggressive nature of SMT-MFR in handling the missing packet, which can be minimized by considering the available Rbuf space before retransmission. For example, if a_rwnd is large enough, that

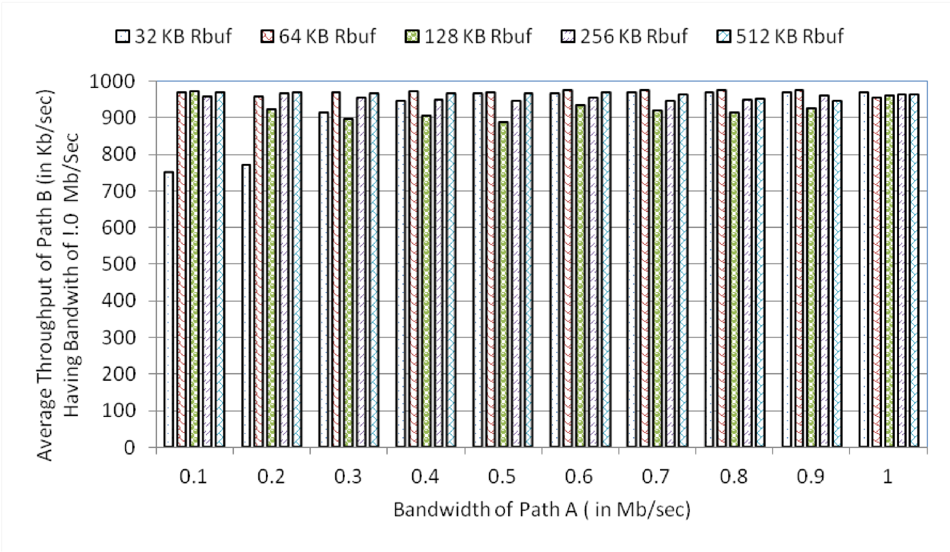

Figure 4.10: SMT-AMFR: average throughput of path B in bandwidth based disparity scenario

can hold the OOS packets by providing an arrival opportunity for delayed missing packets as practiced in SMT-AMFR.

SMT-AMFR has outperformed the SCTP-CMT and SMT-MFR, as shown in Figure 4.10. SMT-AMFR performance is not considerably affected by the disparity in bandwidth and Rbuf size. The SMT-AMFR comparative performance gain is due to the efficient utilization of available Rbuf space with respect to OOS packet arrival. SMT-AMFR holds the OOS packets for a while and provides an opportunity to the missing packets for arrival by providing some extra time until Rbuf blocking occurs. SMT-AMFR fast retransmits the missing packet just before the occurrences of Rbuf blocking. In this way, the SMT-AMFR efficiently utilizes Rbuf space to maximize the average throughput. Similarly, the dynamic setting of fast retransmit threshold values also helps in achieving higher average throughput.

The SMT-AMFR has an inherited feature of SMT-MFR by differentiation of inter-path missing packets, and its fast retransmission. This helps SMT-MFR in maintaining higher average throughput, and remains unaffected by variation in Rbuf size (range from 32KB to 512KB). In short, the SMT-AMFR comparatively outperforms SCTP-CMT and SMT-MFR by yielding higher average throughput, as shown in Figure 4.11.

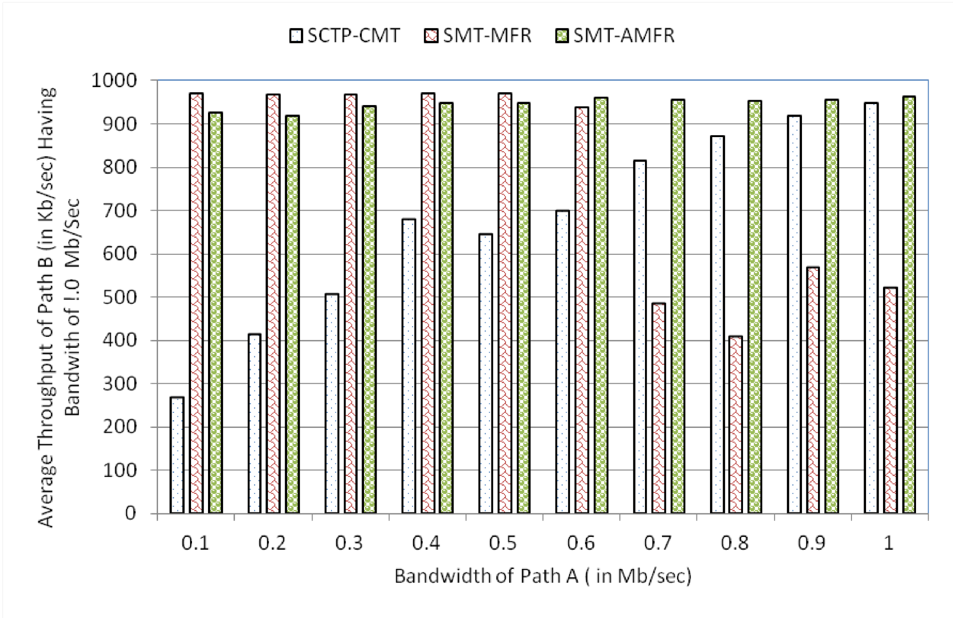

FIGURE 4.11: Average throughput of SCTP-CMT, SMT-MFR and SMT-AMFR with respect to varied Rbuf sizes (32, 64, 128, 256, 512 KB) in bandwidth based disparity scenario

TABLE 4.4: Delay based disparity scenario configuration

Parameters	**Values**
Path A Bandwidth & Delay	1 Mbps & 20, 30, 40, 50 milliseconds
Path B Bandwidth & Delay	1 Mbps & 45 milliseconds
Receiver window (rwnd)	32/64/128/246/512 Kilo Bytes
Simulation scenario	Simple scenario with 2 paths

4.5.2 Delay-Based Disparity

Delay based disparity scenarios are designed to find out the effect of variation in delay feature of a path on the performance efficiency of other paths. For this purpose, delay of path B remains same (i.e., 45 milliseconds) in all scenarios, while the delay of path A varies (from 20 to 45 milliseconds) in each scenario. Both multiple paths (A & B) have the same bandwidth (1Mbps) for data transmission, as mentioned in table 4.4. In addition to this, various simulation scenarios are configured having different Rbuf size (range from 32 KB to 512 KB).

Figure 4.12 shows that average throughput of SCTP-CMT is affected by variation in delay and Rbuf size usage in parallel transmission in multiple paths.

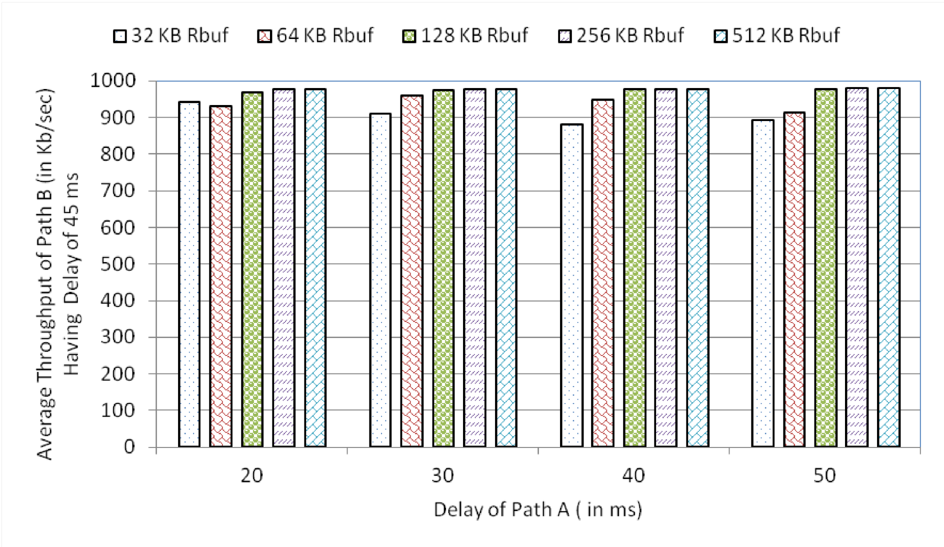

FIGURE 4.12: SCTP-CMT: average throughput of path B in Delay based disparity scenario

Although, the average throughput of SCTP-CMT is higher in delay based disparity scenario, as compared to the bandwidth disparity scenario. Since both paths have the same bandwidth, the chances of inter-path missing packets are rare. The average throughput of SCTP-CMT with low Rbuf space (64KB) decreases with increase in delay variation due to the inefficiency of SCTP-CMT in the presence of low Rbuf space.

On the other hand, the performance of SMT-MFR enhances with increases in delay disparity of multiple paths, as shown in Figure 4.13. SCTP updates the cwnd with every round trip time (RTT). The size of cwnd of one path with lower RTT is larger than 2nd path with higher RTT. The missing packet belongs to a path, which has larger RTT (Path B) retransmitted on the path with smaller RTT (Path A where delay is 20, 30). In scenarios where both paths have less disparity in delay (i.e., when path A delay is 40 and 50), the missing packet notifications are received approximately on both paths (A & B). Both paths fast retransmit the missing packet on an alternate path. This further increases the queue of outstanding packets on each path for retransmission, which causes a reduction in average throughput. The average throughput of SMT-MFR does not affect considerably by an increase in Rbuf size (ranges from 32 KB to 512 KB). This reveals that SMT-MFR is quite resistant to variation in Rbuf size.

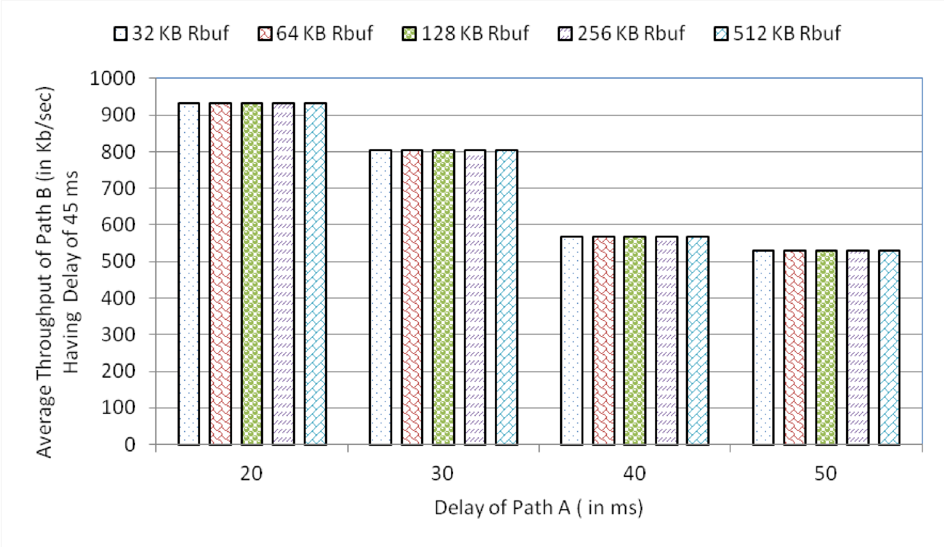

FIGURE 4.13: SMT-MFR: average throughput of path B in Delay based disparity scenario

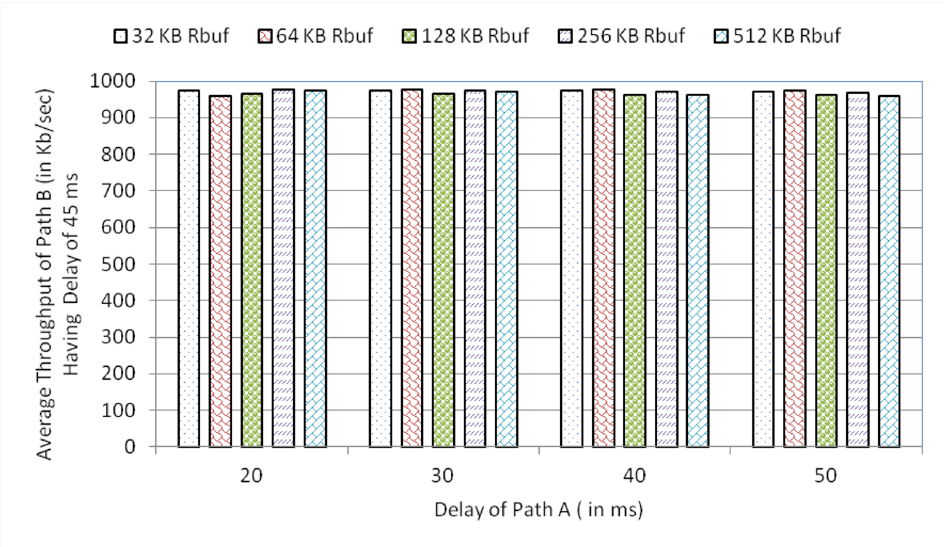

FIGURE 4.14: SMT-AMFR: average throughput of path B in Delay based disparity scenario

The SMT-AMFR shows good performance in almost all delay based disparity scenarios and remains unaffected by usage of Rbuf sizes, as shown in Figure 4.14. The advantage of performance of SMT-AMFR with respect to SMT-MFR is due to efficient utilization of Rbuf space and dynamic fast threshold values.

The Figure 4.15 shows the average result of multipath schemes with respect to Rbuf size. The SMT-AMFR outperforms SMT-MFR and SCTP-CMT in delay based disparity scenario.

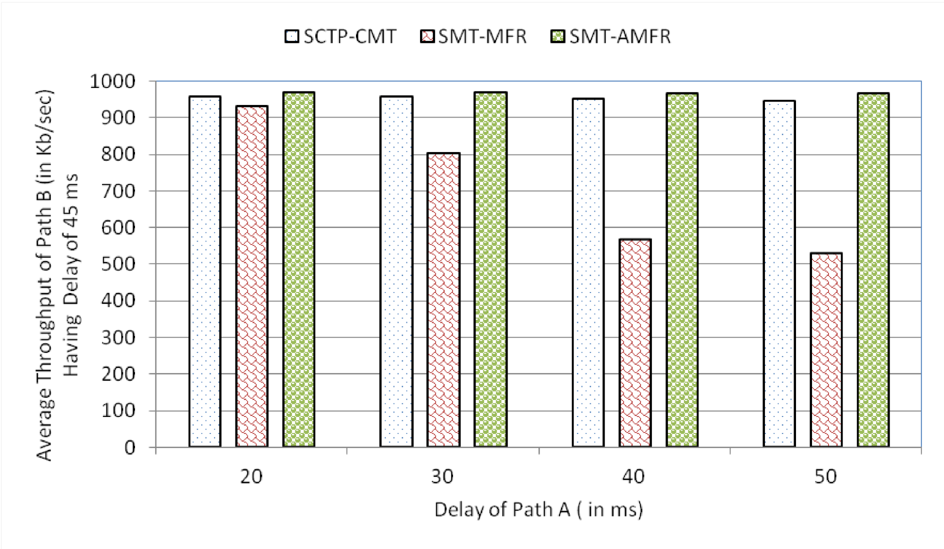

FIGURE 4.15: Average throughput of SCTP-CMT, SMT-MFR and SMT-AMFR with respect to varied Rbuf sizes (32, 64, 128, 256, 512 KB) in Delay based disparity scenario

TABLE 4.5: Percentage difference of average throughput SMT-AMFR-to-SCTP-CMT, SMT-MFR-to-SCTP-CMT, and SMT-AMFR-to-SMT-MFR

Simulation Scenarios	**SMT-MFR TO SCTP-CMT**	**SMT-AMFR TO SCTP-CMT**	**SMT-AMFR TO SMT-MFR**
Bandwidth-based disparity	13.4%	31.4%	19.8%
Delay-based Disparity	-14.6%	2.5%	16.9%

Hence, from the analysis of graphical results mentioned above in this chapter, it is revealed that SMT-MFR has successfully handled the differentiation of the missing packets, frequent cwnd collapses and throughput degradation issues found in NMT-SCTP and SCTP-CMT schemes. However, SMT-MFR adopts an aggressive mechanism by having static fast retransmit threshold. SMT-AMFR inherits the SMT-MFR features with additional advantages by using dynamic fast retransmit threshold values, which provide an arrival opportunity to the inter-path missing packets by waiting and holding incoming OOS packets until just before the occurrence of Rbuf blocking. This has helped SMT-AMFR to have a comparatively high throughput gain with respect to SMT-MFR and SCTP-CMT, as shown in table 4.5.

4.6 Summary

In this chapter, various simulation multihomed scenarios are presented to compare and evaluate the performance of SMT schemes (SMT-MFR & SMT-AFMR) with SCTP-CMT, which acts as benchmark multipath transmission scheme. SMT schemes implementation in the NS-2 hierarchy is mentioned here. The performance analysis parameters used for evaluation of SMT schemes are mentioned here theoretically and mathematically.

Chapter 5

ANALYTICAL PERFORMANCE EVALUATION OF SMT SCHEMES

The mathematical model is considered as a substitute to simulation and is often preferred to use it as a time saving device to analyze the performance of networking protocols. The mathematical model provides deep insights into the man-made systems. In computer networks, the mathematical models are used to create the theoretical framework of the protocols in order to analyze their performance constrains such as, throughput, delay and packet losses.

5.1 Review of Transport Layer Modelling

The renewal theory and Markov model are used in the past to reflect the working behavior of transport layer protocol. The renewal theory is presented by [47] to create a simple model for the throughput estimation of transport layer protocol, where long-lived connections with independent periodic packet losses are considered. This renewal theory has used the fast recovery and linear congestion window growth only. The author avoided the time out mechanism in modeling the renewal theory with the assumption that cwnd will be decreased only with the arrival of packet loss notification. Latter one, [48] has modified the renewal theory by including the time out events along with triple duplicate losses. This model is again revisited by [49] [50] and named it PFKT-model in order to handle the inaccuracies found in its previous version. The authors incorporated the slow start phase after time out events to improve the accuracy of the transport layer model.

On the other hand, the Markov chains based fixed point method has evolved during this time to model transport layer protocols [51] [52]. This model starts with an arbitrary arrival rate of packets over the M/M/1/K receiver buffer where the probability based independent packet loss is used. These parameters are feeding back into the Markov chain, which decides the next arrival rate for the M/M/1/K receiver buffer. The same Markov model is used by [53] to provide a theoretical framework for multihoming features of transport layer protocol. The Markov chain model is modified by considering the number of packet losses in a previous round. This causes an increase in the number of state spaces, which is handled by using performance estimation interim of round.

[40] have used renewal theory and Deterministic Markov chain (DTMC) model to evaluate the throughput of the concurrent multipath transfer in SCTP. In this model, the author has considered the limited receiver buffer size along with a probability based independent packet losses. The authors have concluded that the DTMC Markov model is more accurate as compared to renewal theory. That is why, the DTMC is used and modified in order to model the SMT-Schemes.

The parameters used in the previous DTMC model of SCTP protocols are congestion window, packet loss, slow start threshold, time out and number of transmitted packets. Here, DTMC model has ignored the receiver buffer size in the throughput estimation of SCTP with the assumption that the packet transmission rate of all the flows will never exceed the shared receiver buffer size. The receiver buffer size plays an important role in deciding the sending window according to the flow control mechanism in a transport layer protocol. That is why, one additional parameter of receiver buffer size is considered in DTMC model for throughput estimation of SMT Schemes.

5.2 Modeling SMT Schemes

The mathematical modeling of SMT is considered as the multihomed network topology, where a sender transmits data packets using multiple paths to a receiver. Each path is treated as independent path and has its own path related features, such as bandwidth, packet loss ratio and end to end delay. The whole process of data transmission between sender and receiver is divided into discrete units called rounds [54] [53]. Each round starts with the transmission of the data packet, and ends with the reception of a SACK. The duration of time between the start and end of a round is measured as a single round trip time (RTT) or one Retransmit timeout (RTO). The status of each round is found to be in one of the four discrete states, i.e., slow start (SS), congestion avoidance (CA), fast retransmit (FR) and time out (TO) states.

The DTMC is used for SCTP-CMT to predict the next state of the round based on the packet loss probability of a network in a round [55]. A process is considered to have the Markov property, when its future state depends only on its current state and not on the historical behavior. In DTMCs, the process (round) transits from one state " i " to a target state " i' " with 1 probability [56]. In DTMCs, the steady-transition probability matrix (Q) is generated, which is used to find the next steady-state probability distribution (π) of a round. The state of a round is defined by four parameters, i.e., congestion window (ω), the slow start threshold (τ), packet dropped (γ) and advertised receiver window (υ). The packet is the unit of measurement for all these parameters in a state. The system transition takes place at the end of a round from i to i' using the following equations:

$$State\; of\; a\; round(i) = (\omega, \tau, \gamma, \upsilon) \tag{5.1}$$

$$State\; of\; next\; round(i') = (\omega', \tau', \gamma', \upsilon') \tag{5.2}$$

Whereas, the transition probability $Q(i, i')$ and steady-state probability distribution (π) is defined by the following equation:

$$\pi = Q\pi \tag{5.3}$$

For ease of study, the mathematical model is sorted out into three submodels i.e., network model, sender side model and receiver side model.

5.3 Network Model

The packet gets lost during data transmission due to path error or congestion in the network or due to Rbuf blocking issue in multipath transmission. In this model, the probability based packet loss is used with respect to send data packets, as mentioned below.

5.3.1 Probability of Missing Packets

During a round, the packet may be lost. There are two packet loss models, i.e., independent and correlated. In independent loss model, each packet has the same probability of being lost within a round. The probability of a packet loss is independent of its sent pattern. The independent packet loss model is used in a path, where the packets are dropped due to error or lossy nature. The Bernoulli probability formula is used to find out the probability of packet loss (γ) during the transmission of (ε) packets considering the independent loss model.

$$P(\gamma, \varepsilon) = \binom{\varepsilon}{\gamma} P^{\gamma} (1 - P)^{\varepsilon - \gamma} \tag{5.4}$$

Table 5.1: Probability of packet loss with respect to number of sent packets[1-10]

Number of Packets	Probability of packet Loss
0	0.2281
1	0.0317
2	0.0026
3	0.0001
...	...
9	0.0
10	0.0

Whereas, P is the probability of packet loss of a path. The independent loss model is used when the Rbuf is empty or not full. The correlated loss model is utilized in case of congestion in the network or the Rbuf is full. In the correlated loss model, the probability of first packet loss is P, while all other packets after first packet loss will be lost with probability of 1. This is the scenario where the Rbuf is full and all incoming packets are dropped due Rbuf overflow or Rbuf blocking. In correlated loss model, the probability of losing "γ" packets during the transmission of "ε" packets is given below:

$$P(\gamma,\varepsilon) = \begin{cases} (1-P)^{\varepsilon} & \text{if } \gamma = 0; \\ P^{\gamma}(1-P)^{\varepsilon-\gamma} & \text{if } 0 < \gamma \leq \varepsilon; \\ 0 & \text{otherwise.} \end{cases} \tag{5.5}$$

The Bernoulli equation helps us to find the probability of packet losses when numbers of packets are sent. For example, if there is a need to find the packet loss probability of 10 packets sent on a path with 0.03 probability (i.e., binopdf (1:10,10,0.03), then the following results are obtained, as mention in table 5.1 (by using Mat Lab):

Using the above information, an algorithm is proposed as mentioned in Figure 5.1 which is used to find the number of packets dropped in the current round. In order to randomly decide the number of packets dropped in each round, random number is used. A random value rnd is generated and the loss probability of each packet is compared with this random value.

Algorithm 3 Finding the number of packets dropped randomly

Input: (Sent_Packets (N), Packet Loss Probability (P)).
Output: Dropped Packets(i)
Let N be the number of packets sent in a round with "P" packet loss probability.
The Bernoulli equation is used to calculate the probability of packet loss "p" of each packet in a round, i.e. packet loss probability of i^{th} packet is p^i
Following steps helped us to find the number of packets dropped in current round
Step 1: Generate a random number "rnd"
Step 2: To determine the number of packets dropped in a round.
for i=0; i $\leq$ N; i++ **do**
  if $p^k \geq rnd$ **then**
    return i
  else
    p^k += p^i
  end if
end for
Whereas i is the number of packets dropped in current round and p^k is the sum of packet loss probability of k packets and $k \leq N$.

If the probability of the packet is greater or equal to the random number, then the amount of the packets shown by its index number is considered as dropped packets in the current round. On the other hand, if the probability of the packets is less than random value, then this probability is added to the sum of the probability of previous packets.

5.4 Sender Side Modeling

Sender manages congestion window (cwnd), send window and round trip time (RTT) in order to decide the amount of data transmitted in a round. The cwnd, send window and RTT are explained below with mathematical equations.

5.4.1 Congestion Window (cwnd)

The cwnd manages the congestion in a path with the help of slow start threshold (SSThreshold) and fast retransmit threshold (FRThreshold). The SSThreshold is used to categorize the cwnd into slow start and congestion avoidance phase, as shown in equation 4.

$$State\ of\ cwnd = \begin{cases} Slow\ Start & if\ cwnd < SSThreshold; \\ Congestion\ Avoidance & if\ cwnd \geq SSThreshold. \end{cases} \tag{5.6}$$

The initial value of SSThreshold is arbitrarily high and equal to the receiver window (rwnd). In slow start phase, the value of cwnd is initialized as per following rules[14]:

$$Initialcwnd = \begin{cases} \min(4 \times MTU, \max(2 \times MTU, 4380B) & At\ start\ of\ Transmission; \\ 1 \times MTU & After\ Retransmit\ Time\ out. \end{cases} \tag{5.7}$$

5.4.2 Slow Start Phase

The basic assumption for the growth of cwnd in slow start phase is given below:

(a) All the packets in the previous cwnd are transmitted.

(b) The recent SACK has increased the cumulative Ack(Cum_Ack) value. i.e.,

$$New\ Ack = Cum\_Ack_{n+1} - Cum\_Ack_n \tag{5.8}$$

(c) Congestion window (cwnd) is not in fast recovery phase followed by fast Retransmit.

If the above three conditions are verified, then increase in cwnd takes place with each Ack arrival, and is measured by using the following equation[14]:

$$cwnd_{n+1} = cwnd_n + \min(NewAck, MTU) \tag{5.9}$$

Whereas, the maximum transmission unit (MTU) is the size of data packet, which is transmitted in the network without fragmentation. In view of these features of cwnds growth, the state parameters of a round in slow start phase of SMT schemes are mentioned below:

$$SS = (\omega, \tau, \gamma, \upsilon);\ \omega = 2^i, i \,\epsilon\, Z \tag{5.10}$$

Whereas,

$$\{2 \leq \omega \leq SS_{Thresholds}, \tau \,\epsilon\, SS_{Thresholds},\ 0 \leq \gamma \leq \omega, 2 \leq \upsilon \leq \upsilon_{\max}\} \tag{5.11}$$

The $\upsilon_{\max}$ is the initial maximum Rbuf and $SS_{Thresholds}$ is the set of slow start thresholds and is given by:

$$SS_{Thresholds} = 2^i \colon 0 \leq i \leq \left[\log_2(\frac{\varepsilon_{\max}}{2})\right], i \,\epsilon\, Z \tag{5.12}$$

Whereas, $\varepsilon_{\max}$ is the maximum sending rate. If there is no packet loss in the current round, then the system will transit from $(\omega, \tau, 0, \upsilon)$ to $(2\omega, \tau, 0, \upsilon)$ in one RTT with probability $P(0, \omega)$. In case of packet lost i.e., $(0 \leq \gamma \leq \omega)$, the system will transit to FR in next round having $(\frac{\omega}{2}, \frac{\omega}{2}, \gamma, \upsilon')$ with probability $P(\gamma, \omega)$. The system will move to an exponential back-off (EB) state with $P(\omega, \omega)$ (i.e., $P(1)$), if all packets are lost in the previous round as mentioned in following equation:

$$SS = (\omega, \tau, \gamma, \upsilon) = \begin{cases} SS(2\omega, \tau, 0, \upsilon) & \text{if } \gamma = 0; \\ FR(\frac{\omega}{2}, \frac{\omega}{2}, \gamma, \upsilon') & \text{if } 0 < \gamma < \omega; \\ EB(1, \frac{\omega}{2}, \gamma, \upsilon') & if \gamma = \omega. \end{cases} \tag{5.13}$$

The a_rwnd is reduced (i.e., $\upsilon' < \upsilon$) due to packet loss in previous round as mentioned in algorithm 5.2.

5.4.3 Congestion Avoidance Phase

The congestion avoidance phase starts at the moment, when cwnd is greater than or equal to the SSThreshold. During the congestion avoidance phase, the cwnd is increased by 1 MTU with reception of Acks for all the packets sent in the previous round, i.e.,

$$cwnd_{n+1} = cwnd_n + MTU \tag{5.14}$$

The set of the states for the congestion avoidance (CA) phase is modeled as mentioned in the following equation:

$$CA = (\omega', \tau, \gamma, \upsilon);\ \omega' = \omega + 1 \tag{5.15}$$

Whereas,

$$\{SS_{Threshold} \leq \omega \leq \omega_{\max}, \tau \,\epsilon\, SS_{Thresholds},\ 0 \leq \gamma \leq \omega, 2 \leq \upsilon \leq \upsilon_{\max}\} \tag{5.16}$$

The maximum size of congestion window is $\omega_{\max}$. A round in CA state may remain in CA state or it may transit to FR or EB state; depending upon the probability of packet loss, as mentioned in the following equation:

$$CA = (\omega', \tau, \gamma, \upsilon) = \begin{cases} CA(\omega + 1, \tau, 0, \upsilon) & \text{if } \gamma = 0; \\ FR(\frac{\omega}{2}, \frac{\omega}{2}, \gamma, \upsilon') & \text{if } 0 < \gamma < \omega; \\ EB(1, \frac{\omega}{2}, \gamma, \upsilon') & if\ \gamma = \omega. \end{cases} \tag{5.17}$$

During the CA phase, the amount of sent data (ε) in a round with respect to packet losses (γ) is given by the following equation:

$$\varepsilon' = \begin{cases} \omega - \gamma & \text{if } \gamma \geq 0; \\ \omega + 1 & \text{if } \gamma = 0 \,\&\, \omega < \omega_{\max}; \\ \omega & Otherwise. \end{cases} \tag{5.18}$$

5.4.4 Fast Retransmit Phase

The packet loss notification is received at sender side as selective acknowledgment (SACK). The congestion control mechanism maintains a separate fast retransmit counter for each missing packet. The fast retransmit counter of a missing packet is incremented with the arrival of each SACK. The congestion control mechanism retransmits the missing packet whose fast retransmit counter is equal to fast retransmit threshold. In other words, the missing packet is considered lost with reception of three SACKs and is fast retransmitted with a reduction in the cwnd and SSThreshold value, as shown in the following equation:

$$cwnd_{n+1} = \frac{cwnd_n}{2} \tag{5.19}$$

$$SSThreshold_k = cwnd_{n+1} \tag{5.20}$$

In the case of SMT-MFR and SMT-AMFR, the fast retransmit event is triggered with respect to intra and inter-path missing packet. Probability of packet loss is used in this model. This probability of missing packets is subdivided into intra and inter-path missing packets according to a ratio as mentioned in table 5.2. Following algorithm is proposed to model the fast retransmit event for intra and inter path missing packets in SCTP-CMT and SMT schemes (SMT-MFR & SMT-AMFR).

Static fast retransmit threshold value is used in SCTP-CMT and SMT-MFR. In case of SMT-AMFR, the dynamic threshold value is used in such a way that the Rbuf space is divided into three zones i.e., critical, substantial and moderate zones. The fast retransmit threshold is kept different for each zone, i.e., 3 for the critical zone, 4 for the moderate zone and 5 for the substantial zone. The selection of specific fast retransmit threshold in a round with respect to a_rwnd takes place according to algorithm mentioned in Algorithm 2.

Algorithm 4 SCTP-CMT, SMT-MFR & SMT-AMFRs fast retransmit model for intra and inter-path missing packets

Input: ($Inter\_PMP$, $Intra\_PMP$, $cwnd_{p-1}$)
Output: ($cwnd_p$)
Let the ratio of intra and inter-path missing packet be Inter_PMP and Intra_PMP respectively i.e.
Inter_PMP + Intra_PMP = 1
Congestion window of previous round is $cwnd_{p-1}$.
A uniform random number (Frt_random_i) is generated between 0 and 1.

- FOR SCTP-CMT

if $Frt\_random_i < Inter_PMP$ **then**
  Ignore the $cwnd_{p-1}$ reduction
  Ignoring fast retransmission of missing packets
else
  if $Frt_random_i \geq Inter\_PMP$ **then**
    $cwnd\_p = \frac{cwnd_{p-1}}{2}$
    $SSThreshold\_k = cwnd\_p$
    Fast retransmit the missing packets
  end if
end if

- FOR SMT-MFR & SMT-AMFR

if $Frt\_random_i < Inter\_PMP$ **then**
  Ignore the $cwnd_{p-1}$ reduction
  Fast retransmsision of missing packets
else
  if $Frt\_random_i \geq Inter\_PMP$ **then**
    $cwnd_p = \frac{cwnd_{p-1}}{2}$
    $SSThreshold\_k = cwnd_p$
    Fast retransmit the missing packets
  end if
end if

5.4.5 Exponential Back-off (EB)

The feedback system of acknowledgment is used to regulate the transmission of data on the transport layer. The exponential back-off phase starts when RTO gets expired as an acknowledgement of transmitted data is not received within the specified time interval called retransmit timeout (RTO). In this situation, the RTO becomes double for next transmitted data and the cwnd is reduced according to the following equation:

$$cwnd_{n+1} = \max\left(\frac{cwnd_n}{2}, 4 \times MTU\right) \tag{5.21}$$

$$RTO = \min\left(2 \times RTO, RTO_{\max}\right) \tag{5.22}$$

The twofold increase in RTO is repeated until the RTO reaches at some maximum value ($RTO_{\max}$). RTO occurs during the slow start phase and the congestion avoidance phase. In case of multihoming, RTO is calculated independently for each path. Smoothed round-trip time (SRTT) and round-trip time variation (RTTVAR) are used to calculate the RTO for each destination. On reception of first acknowledgment of data transmission, the RTO is calculated using SRTT and RTTVAR as mentioned below:

$$SRTT = RTT \tag{5.23}$$

$$RTTVAR = \frac{RTT}{2} \tag{5.24}$$

$$RTO = SRTT + 4 \times RTTVAR \tag{5.25}$$

At the reception of next acknowledgment, the RTO, SRTT and RTTVAR are calculated by using following equations.

$$RTTVAR = (1 - \beta)RTTVAR + \beta|SRTT - RTT| \tag{5.26}$$

$$SRTT = (1 - \alpha)SRTT + (\alpha \times RTT) \tag{5.27}$$

Whereas,

$$\alpha = \frac{1}{8} \ and \ \beta = \frac{1}{4} \tag{5.28}$$

$$RTO = SRTT + 4 \times RTTVAR \tag{5.29}$$

In light of these basic features, the set of EB states is mentioned in the following equation:

$$EB = (\omega', \tau, \gamma, \upsilon) \tag{5.30}$$

Whereas,

$$\{\omega' = 1, \tau \,\epsilon\, SS_{Thresholds}, \gamma = \omega, 2 \leq \upsilon \leq \upsilon_{\max}\} \tag{5.31}$$

The EB phase continues until an Ack is received for delivery of a packet. The transition probability of EB state into slow start phase is $P(0, 1)$.

5.4.6 Round Trip Time (RTT)

RTT is the duration of time between the sent data and the acknowledgment of its reception. This measurement of RTT is done once per round. If B is the bandwidth of the network with a fixed propagation delay of d and cwnd is the size of the congestion window in a round, then RTT for the current round is calculated by using the following equations:

$$RTT = \begin{cases} d + \frac{1}{B} & \text{if } B \times d \geq cwnd; \\ \frac{cwnd}{B} & Otherwise. \end{cases} \tag{5.32}$$

5.4.7 Send Window

Sending window is the amount of data that a sender can send to the receiver without violating the congestion and flow control mechanism. This size of send window is the minimum of:

(a) the amount of data that a receiver can receive (a_rwnd) i.e., flow control and

(b) the amount of data that a network can handle according to its capacity (cwnd) i.e., congestion control. Mathematically, it can be shown as:

$$Sending\,Window = \min(a\_rwnd, cwnd) \tag{5.33}$$

In any situation, the send window for a destination cannot exceed its outstanding data transmission limits imposed by the bandwidth of the path.

5.5 Receiver Side Model

The SMT is based on sender side modification. The receiver responds according to the standard procedure, which is sending the available Rbuf space as advertised receiver window (a_rwnd) and missing packet information using SACK, as briefly explained in the next subsection.

5.5.1 Advertised Receiver Window (a_rwnd)

Receiver window is a mandatory variable (in each Acks or SACK), used by the sender (to regulate the sending window in order) to have the information about the recent available receiver buffer space at the receiver side.

Algorithm 5 Finding a_rwnd and newly buffered packets in the current round

Input: (a_rwnd_{n-1}, $Dropped\_Pkt_k$, $Receiving\_Pkts_m$).
Output: (a_rwnd_n, $Buffered\_Pkt_r$)
Let a_rwnd_{n-1} be the a_rwnd of the previous round (n-1). Let k be the number of packets dropped randomly during receiving (m) packets.
Missing Packets(k) = $randperm$ ($Dropped\_Pkt_k$), ($Receiving\_Pkts_m$)
Following steps are used to find the a_rwnd_n for the current n^{th} round.

- Find the smallest missing packet, i.e. first missing packet in the newly buffered packets of current round

$First\_Missing\_Pkt = min(MissingPackets)$

- The number of packets to store in Rbuf in current round is given by

$Buffered\_Pkt_r = (Receiving\_Pkts_m - First\_Missing\_Pkt) + 1$

- The a_rwnd_n for current n^{th} round is calculated by the following equation.

$a\_rwnd_n = a\_rwnd_{n-1} - Buffered\_Pkt_r$
if $a\_rwnd < 0$ **then**
  $a\_rwnd = 0$
end if

This information is advertised by the receiver in each SACK packet as a mandatory advertised receiver window (a_rwnd) to inform the sender about the amount of data that can be buffered or received. Initially, the a_rwnd is set to maximum Rbuf space during the association establishment phase. During each round, the size of a_rwnd decreases with the arrival of OOS packets at the receiver side. An algorithm is proposed (mentioned in Figure 5.3) to calculate the a_rwnd for each round at the receiver side:

5.6 Throughput

In this model, the throughput is expressed as the numbers of packets which are received at per unit time. The average throughput (η) is calculated using the following formula.

$$\eta = \frac{E[\varepsilon] - E[\gamma]}{E[\delta]} \tag{5.34}$$

Whereas,

$E[\varepsilon]$ = is the total number of packets sent,

$E[\gamma]$ = is the total number of packets dropped and

$E[\delta]$ = is the total time of mathematical model.

If $\varepsilon(i)$ is the expected number of packets sent in a round and $\pi(i)$ is the probability of a round being in a state(i), then the expected total number of packets sent is given by:

$$E[\varepsilon] = \sum_{i\epsilon\{SS,CA\}} \pi(i)\varepsilon(i) + \sum_{i\epsilon EB} \pi(i) \tag{5.35}$$

According to equation 5.4 and 5.5, the expected number of packets dropped is given by the following equation:

$$E[\gamma] = \sum_{i\epsilon\{SS,CA\}} \pi(i) \sum_{j=1}^{\varepsilon(i)} P(j, \varepsilon(j)) + \sum_{i\epsilon\varepsilon} \pi(i)P(1,1) \tag{5.36}$$

The total time of the mathematical model is the summation of the time duration between the rounds and is expressed as:

$$E[\delta] = \sum_{i\epsilon\ A} \sum_{i'\epsilon\ A} \pi(i)\ D(i,i') \tag{5.37}$$

Where "A" is the set of the states, i.e., A= SS,CA,EB and D is defined as the matrix of the state transition between the rounds.

5.7 Results and Discussion

Here, the accuracy of the mathematical model is compared to simulated results of SCTP-CMT, SMT-MFR, and SMT-AMFR. In multihomed network topology, each path behaves like an independent path, which has its own features such as bandwidth, delay, and probability of packet loss. That is why, there is a need for a single-homed network topology in order to evaluate the accuracy of the simulation. The usage of single-homed network topology in mathematical modeling helps us to focus on the core functionality of the SMT-MFR and SMT-AMFR in handling intra and inter-path missing packets. SMT-MFR and SMT- AMFR are designed to handle the bandwidth degradation due to missing packets. Therefore, the scenario based upon the probability of missing packets is designed. The parameters configuration for mathematical model and NS-2 simulation are mentioned in table 5.2. The Deterministic Markov Chain model is implemented using MATLAB R2015a.

The probability of missing packets ranging from 0.01 to 0.09 is used. These packets may be missing due to congestion in the network or due to multipath effect. To simulate and model this behavior, the probability of missing packet is further subdivided into intra and inter-path missing packets. During parallel multipath transmission, the chances of missing packets due to multipath effect are more as compared to the congestion in a network, that is why, intra and inter-path missing packets are configured 90% and 10% respectively of the probability of missing packets in each scenario.

Table 5.2: Percentage difference of average throughput SMT-AMFR-to-SCTP-CMT, SMT-MFR-to-SCTP-CMT, and SMT-AMFR-to-SMT-MFR

Parameters	**Values**
Bandwidth	1 Mbps
Path delay	45 milliseconds
Packet size	1500 Bytes
Receiver buffer (Rbuf)	65536 Bytes (64 KBytes)
Probability of missing packets	0.01 to 0.09
Inter-path missing Packets	90% of probability of missing packets
Intra-path missing Packets	10% of probability of missing packets

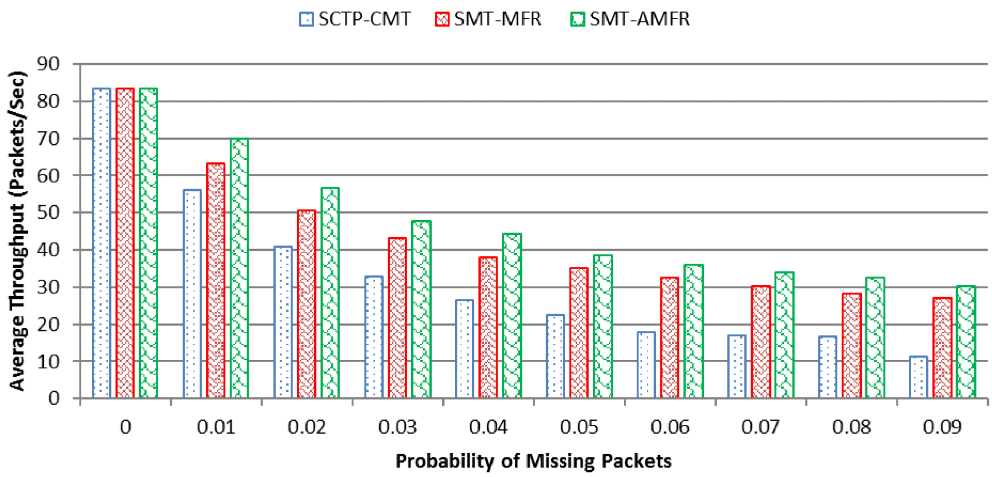

Figure 5.1: Mathematical model: Average throughput of SCTP-CMT, SMT-MFR, and SMT-AMFR

Figure 5.1 shows the average throughput of SCTP-CMT, SMT-MFR and SMT-AMFR with respect to the increasing probability of missing packets in the mathematical model. This mathematical model shows that the average throughput of all the three multipath protocols i.e., SCTP-CMT, SMT-MFR and SMT-AMFR remains same in absence of missing packet. The gradual decrease in average throughput of these multipath transmission protocols takes place due to the increase in probability of missing packets in subsequent scenarios. Here, the SMT-AMFR outperformed SMT-MFR and SCTP-CMT.

Table 5.3: General configuration parameters of five simulation scenarios

Protocol To Protocol	Percentage Difference
SMT-MFR TO SCTP-CMT	8.54%
SMT-AMFR TO SCTP-CMT	19.65%
SMT-AMFR TO SMT-MFR	11.15%

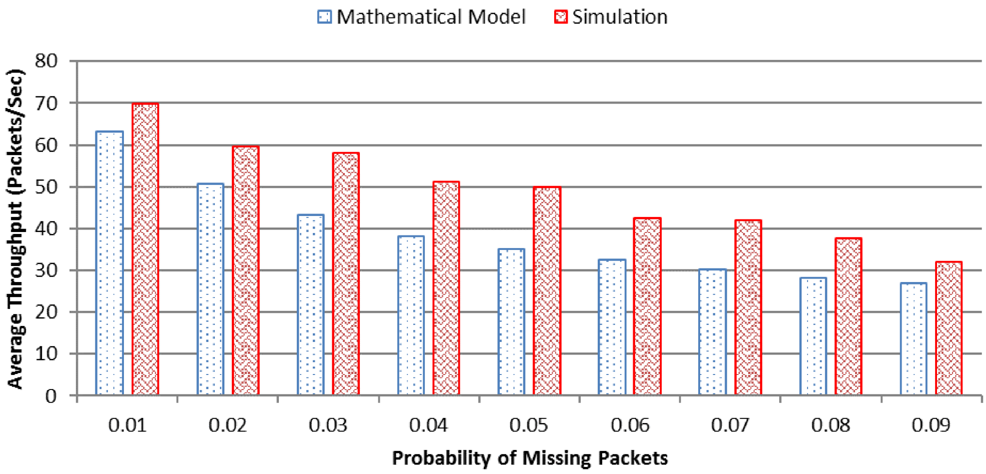

Figure 5.2: Comparison of analytical and simulation results of average throughput of SMT-MFR

The percentage difference of average throughput of SMT schemes and SCTP-CMT are mentioned in table 5.3. SMT-MFR has average throughput gain of 8.54% as compared to SCTP-CMT. The SMT-AMFR outperformed SCTP-CMT and SMT-MFR with an average throughput gain of 19.65% and 11.15% respectively.

Figure 5.2 and 5.3 show mathematical model performance comparison of SMT schemes (SMT-MFR & SMT-AMFR) with their simulation results. There is a little difference between the results of mathematical model and simulation. In these scenarios, the packets are randomly dropped during transmission using a probability based random number. There is a uniform random module in network simulator-2, which produced a uniform random number.

The mathematical model is implemented in Mat Lab, where there are 101 patterns of random numbers called a seed. The result of a simulation varies with the changes in the seed value from 0 to 101. This may be the reason for the

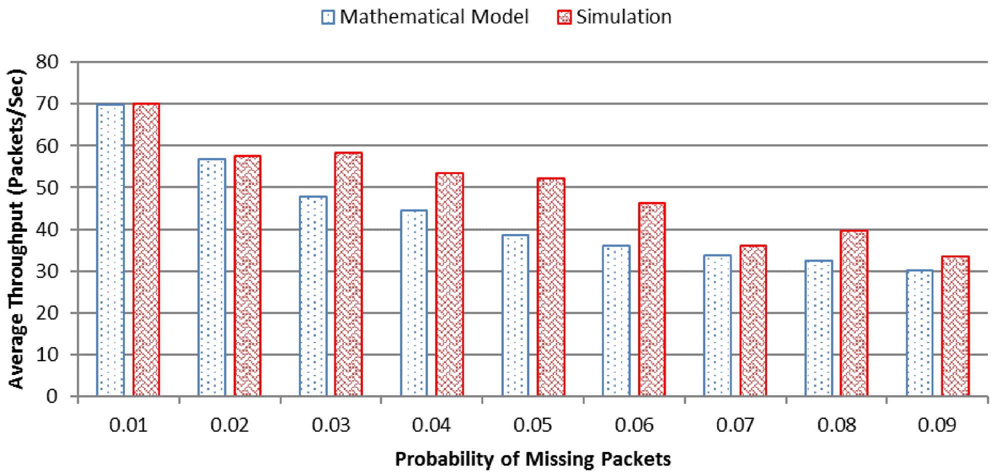

Figure 5.3: Comparison of analytical and simulation results of average throughput of SMT-AMFR

difference between the result of mathematical model and simulation scenarios. In short, the mathematical model supports the proposed SMT schemes (SMT-MFR and SMT-AMFR), which are implemented in network simulator-2.

5.8 Summary

This chapter presents the analytical models of SMT-MFR, SMT-AMFR and SCTP-CMT schemes. This chapter starts with analysis of existing transport layer mathematical models. Deterministic-Time Markov Chain (DTMC) model is used to mathematically model the SMT schemes (SMT-MFR and SMT-AMFR schemes) and SCTP-CMT. Result and discussion section revealed the comparative analysis of mathematical and simulation results of SMT schemes in same scenarios.

Chapter 6

CONCLUSION AND FUTURE WORK

6.1 Summary

This thesis has critically analyzed the transmission of data concurrently on multiple paths and have identified the root causes of its throughput degradation. The research hypothesis mentioned in this thesis, states that:

The degradation of aggregate throughput in simultaneous multipath transmission in SCTP can be improved by managing the congestion window based on the differentiation in the intra and inter-path missing packets and making fast re-transmit threshold value adaptive with respect to available Rbuf space.

The research hypothesis is proven by using various theoretical, mathematical and simulation-based investigations, as mentioned in the previous chapters. Chapter 1 is based on the critical analysis of the transmission of data concurrently on multiple paths and identification of the root causes of its throughput degradation. This helped us to formulate the hypothesis. Chapter 2 presents the background of this research study and details of the contemporary state-of-the-art multipath transmission schemes. The present and past research studies are discussed and it has been concluded that the OOS packet arrival, non-differentiation of missing packets into intra and inter-path and traditional use of single homed congestion window mechanism for multipath transmission play a significant role in throughput degradation. Chapter 3 presents the proposed SMT-MFR scheme that differentiates the missing packet into intra and inter-path and manages the multihomed congestion window accordingly, which has improved the aggregated throughput. In order to further optimize the solution, the SMT-AMFR is proposed. SMT-AMFR is used to handle the aggressiveness of SMT-MFR by efficiently utilizing the available Rbuf by partitioning it into three zones i.e., critical, substantial and moderate zones. In chapter 4, various simulation multihomed scenarios are used to compare and evaluate the performance of SMT schemes (SMT-MFR & SMT-AFMR) with SCTP-CMT which acts as a benchmark multipath transmission scheme. The performance analysis parameters used for evolution of SMT schemes are mentioned here theoretically and mathematically. Chapter 5 presents the analytical models of SMT-MFR, SMT-AMFR and SCTP-CMT schemes. Deterministic-Time Markov Chain (DTMC) model is used to mathematically verify simulation results of SMT-Schemes.

6.2 Critical Analysis and Conclusion

The existing transport layer schemes for concurrent multipath transmission, neither differentiates the missing packet into intra and inter-path nor consider the available Rbuf space in the fixation of fast retransmit threshold value. This results in OOS packet arrivals, which causes Rbuf blocking and degrade the aggregated throughput. This study introduces the simultaneous multipath transmission schemes with the name of modified fast retransmit (SMT-MFR) and adaptive fast retransmit (SMT-AMFR). The simulation and analytical results of chapter 4 and 5 validate the research question and hypothesis that aggregated throughput can be successfully improved by intra and inter-path missing packet differentiation and managing congestion window accordingly. In order to further improve the aggregated throughput, adaptive fast retransmit threshold is essential for efficient utilization of available Rbuf space. The main conclusions of this research are the following.

(a) In multipath transmission protocols, the management of the congestion window and fixation of fast retransmit threshold value do not take into consideration intra and inter-path missing packets in selective acknowledgments. This addresses the question 1 and partially question 3 which asserts that non-differentiation of missing packets into intra and inter-path and static fast retransmit threshold value independent of available Rbuf space are the root cause of the aggregated throughput degradation in multipath transmission.

(b) There are two reasons for the degradation of throughput, first is non differentiation in intra and inter-path missing packets and the second is OOS packets. OOS packets may cause a reduction in Rbuf free space (causing Rbuf blocking), which reduces the sending capability of the sender and hence, degrades the throughput. This validates the research question 2 which states that OOS packet arrival during multipath transmission plays a significant role in aggregate throughput degradation.

(c) The multihomed congestion control (MCC) creates a virtual queue at sender side for non-acknowledged packets for each destination. The virtual queue keeps the record of highest and lowest sequence numbers of the packets transmitted to a destination. If the sequence number of the missing packet is among the lowest and the highest sequence number of the same path where it is transmitted, then this packet is considered to be intra-path missing packet, otherwise, it is an inter-path missing packet. In this way, the missing packet in SACK is classified. *This addresses the research question 4 which deals with the differentiation of missing packet into intra and inter-path.*

(d) The MCC mechanism maintains a fast retransmit threshold value, static for intra-path, while dynamic for inter-path missing packet. The fast retransmit event is triggered by the arrival of a number of SACK packets, as mentioned by respective fast retransmit threshold. In case of the inter-path missing packet, the MCC mechanism reduces the cwnd to its half, in order to decrease the congestion in the respective path, while it remains the same in case of the inter-path missing packet. The simulation and analytical results in chapter 4 and 5 revealed that SMT-MFR has overcome Rbuf blocking with improvement in aggregated throughput ranging from 164% to 72.4% (normal to worst scenario, respectively).This addresses the research question 5 which states that multihomed congestion control mechanism based upon the cause of missing packet generation (i.e., due to inter-path or due to intra-path) is essential to solving the aggregated throughput degradation in multipath transmission.

(e) Traditional congestion window management (CWM) quickly retransmits the missing packet using static fast retransmit threshold value, independent of available Rbuf space. In case of inter-path missing packets, the re-transmission of the missing packets may be delayed, when there is a large Rbuf available to accommodate the incoming OOS packets . If the fast re-transmit threshold value is made according to the available Rbuf size (adaptive to Rbuf size), that is, if the available buffer size is small, there is a need to retransmit the missing packet early and if the buffer size is large, the re-transmission needs to be delayed (by increasing the fast retransmit thresholds value), so that the delayed packets may reach the destination. This way the chances of Rbuf blocking are reduced and the throughput is increased. The simulation and analytical results in chapter 4 and 5 revealed that SMT-AMFR has outperformed SMT-MFR and SCTP-CMT.This addresses the research question 3 which states that adaptive fast retransmit threshold with respect to available Rbuf space in multipath transmission is essential to enhance the aggregated throughput.

These conclusions endorse the hypothesis mentioned in chapter 1. These conclusions also answer the research questions asserted in chapter 1. The advantage of SMT schemes is that they need sender side modification only. The SMT-MFR is found to be very successful in avoiding the throughput degradation of the fast link due to slow link. SMT-AMFR maintains a high throughput gain in presence of minimum Rbuf space. This enables SMT-AMFR to be extremely useful in smartphones with limited minimum Rbuf space. This research study is unique in the sense that it tries to handle the OOS packet arrival. Other multipath transmission schemes focus on the scheduler to ensure the sequence packet delivery. The OOS packet arrival is an inevitable issue due to the heterogeneity of multipath path features. The beauty of the SMT schemes is that they complement the other multipath transmission schemes by handling the OOS packet arrival along with their own multipath scheduler and other features.

6.3 Future Work

This research study can be extended for the efficient multipath scheduler, seamless connectivity, and appropriate number multiple path selections. The multipath scheduler is required to handle the changing multipath features efficiently, such as, bandwidth, end to end delay and packet dropped ratio. This scheduler should be adapted for dynamic multiple path features by monitoring network conditions and proactively control data transmission on multiple paths. The scheduler adoption of SMT schemes (SMT-MFR & SMT-AMFR) to dynamic bandwidth and delay features of the Internet and their optimization is next in the queue. The scheduler should be optimized to handle the scalability issues using more than two multiple paths. In addition to this, the seamless connectivity problem will become more severe in simultaneous multipath transmission. There is also a need to solve the issue arising due to path failure or hand over management to another suitable access network. There is a need to analyze an appropriate number of parallel path selections for multipath selection with a motivation to achieve a higher aggregated throughput.This thesis focused on flow control modification for SMT schemes using wired networks. The congestion control mechanism can be modified to further improve the SMT schemes. In future, the SMT schemes may be thoroughly analyzed using wireless network.

6.4 Potential Limitation of SMT Schemes

The main limitations of SMT schemes are given below.

(a) The performance of SMT schemes degrades with an increase in the number of multiple paths. The situation becomes worse in the presence of cross traffic in multiple paths.

(b) The SMT-MFR schemes are based on redundant fast retransmission of an inter-path missing packet of the slow path on the alternate fast path in order to avoid the Rbuf blocking problem. This aggressive nature of SMT-MFR is handled by SMT-AMFR up to some extent. Still, this aggressive retransmission of missing packets may lead to wastage of useful network bandwidth.

(c) The SMT schemes are implemented using SCTP protocol, which is not widely deployed standard protocol on the Internet. We have selected the SCTP to solve the concurrent multipath transmission issues due to having mature multihoming in SCTP. Moreover, the header of SCTP packet can be configured to behave like TCP header by adjusting header size, using a single sequence number (TSN) for data transmission and selective acknowledgment scheme just like TCP SACK. In this way, we can utilize SCTP to address the concurrent multipath transmission issues face by the TCP, until it becomes mature enough in the near future to utilize more than one path at a time.

(d) The SMT schemes are not implemented and evaluated using Linux in a real-world environment. The SMT schemes have extended the SCTP-CMT based extension of SCTP. At the time of this research study, the SCTP-CMT was not implemented in the Linux kernel stack (Retrieved November 29, 2016, from https://www.nntb.no/projects). We have modified the SCTP-CMT version of SCTP available in network simulator -2 (version 2.34) in order to implement the SMT schemes as mentioned in chapter 3. The simulation result of SMT schemes is verified by using analytical model i.e., Deterministic Time Markov Chain (DTMC) model as mentioned in chapter 5.

Bibliography

[1] A. L. Caro, J. R. Iyengar, P. D. Amer, S. Ladha, G. J. Heinz, and K. C. Shah, "Sctp: a proposed standard for robust internet data transport," *Computer*, vol. 36, no. 11, pp. 56–63, 2003.

[2] T. J. Hacker, B. D. Athey, and B. Noble, "The end-to-end performance effects of parallel tcp sockets on a lossy wide-area network," in *Parallel and Distributed Processing Symposium., Proceedings International, IPDPS 2002, Abstracts and CD-ROM.* IEEE, 2001, pp. 10–pp.

[3] L. Ong, "An introduction to the stream control transmission protocol (sctp)," 2002.

[4] Y. Hasegawa, I. Yamaguchi, T. Hama, H. Shimonishi, and T. Murase, "Improved data distribution for multipath tcp communication," in *Global Telecommunications Conference, 2005. GLOBECOM'05. IEEE*, vol. 1. IEEE, 2005, pp. 5–pp.

[5] Y. Dong, D. Wang, N. Pissinou, and J. Wang, "Multi-path load balancing in transport layer," in *Next Generation Internet Networks, 3rd EuroNGI Conference on.* IEEE, 2007, pp. 135–142.

[6] D. Sarkar, "A concurrent multipath tcp and its markov model," in *Communications, 2006. ICC'06. IEEE International Conference on*, vol. 2. IEEE, 2006, pp. 615–620.

[7] M. Zhang, J. Lai, A. Krishnamurthy, L. L. Peterson, and R. Y. Wang, "A transport layer approach for improving end-to-end performance and robustness using redundant paths." in *USENIX Annual Technical Conference, General Track*, 2004, pp. 99–112.

[8] K. Rojviboonchai and H. Aida, "An evaluation of multi-path transmission control protocol (m/tcp) with robust acknowledgement schemes," *IEICE transactions on communications*, vol. 87, no. 9, pp. 2699–2707, 2004.

[9] D. S. Phatak and T. Goff, "A novel mechanism for data streaming across multiple ip links for improving throughput and reliability in mobile environments," in *INFOCOM 2002. Twenty-First Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE*, vol. 2. IEEE, 2002, pp. 773–781.

[10] A. Gurtov, *Host identity protocol (HIP): towards the secure mobile internet.* John Wiley & Sons, 2008, vol. 21.

[11] B. M. Sousa, K. Pentikousis, and M. Curado, "Multihoming management for future networks," *Mobile Networks and Applications*, vol. 16, no. 4, pp. 505–517, 2011.

[12] D. Wischik, M. Handley, and M. B. Braun, "The resource pooling principle," *ACM SIGCOMM Computer Communication Review*, vol. 38, no. 5, pp. 47–52, 2008.

[13] J. Postel, "User datagram protocol," Tech. Rep., 1980.

[14] R. Stewart, "Stream control transmission protocol," 2007.

[15] J. R. Iyengar, "End-to-end concurrent multipath transfer using transport layer multihoming," DELAWARE UNIV NEWARK DEPT OF COMPUTER AND INFORMATION SCIENCES, Tech. Rep., 2006.

[16] L. Magalhaes and R. Kravets, "Transport level mechanisms for bandwidth aggregation on mobile hosts," in *Network Protocols, 2001. Ninth International Conference on.* IEEE, 2001, pp. 165–171.

[17] A. Jungmaier and E. P. Rathgeb, "A novel method for sctp load sharing," in *International Conference on Research in Networking.* Springer, 2005, pp. 1453–1456.

[18] H. Adiseshu, G. Parulkar, and G. Varghese, "A reliable and scalable striping protocol," *ACM SIGCOMM Computer Communication Review*, vol. 26, no. 4, pp. 131–141, 1996.

[19] J. R. Iyengar, P. D. Amer, and R. Stewart, "Concurrent multipath transfer using sctp multihoming over independent end-to-end paths," *IEEE/ACM Transactions on networking*, vol. 14, no. 5, pp. 951–964, 2006.

[20] S. Khan, S. Z. Ahmed, and M. A. Qadir, "Throughput enhancement of simultaneous multipath communication using modified fast retransmit scheme," in *Computer Networks and Information Technology (ICCNIT), 2011 International Conference on.* IEEE, 2011, pp. 9–12.

[21] G. De Marco, M. Longo, F. Postiglione, S. Loreto, and A. Senatore, "On some open issues in load sharing in sctp," *CCCT 2003*, 2003.

[22] A. Argyriou and V. Madisetti, "Bandwidth aggregation with sctp," in *Global Telecommunications Conference, 2003. GLOBECOM'03. IEEE*, vol. 7. IEEE, 2003, pp. 3716–3721.

[23] C. Casetti and W. Gaiotto, "Westwood sctp: load balancing over multipaths using bandwidth-aware source scheduling," in *Vehicular Technology Conference, 2004. VTC2004-Fall. 2004 IEEE 60th*, vol. 4. IEEE, 2004, pp. 3025–3029.

[24] A. Abd, T. Saadawi, M. Lee *et al.*, "Ls-sctp: a bandwidth aggregation technique for stream control transmission protocol," *Computer Communications*, vol. 27, no. 10, pp. 1012–1024, 2004.

[25] J. Liao, J. Wang, and X. Zhu, "cmpsctp: An extension of sctp to support concurrent multi-path transfer," in *Communications, 2008. ICC'08. IEEE International Conference on.* IEEE, 2008, pp. 5762–5766.

[26] J. R. Iyengar, P. D. Amer, and R. Stewart, "Retransmission policies for concurrent multipath transfer using sctp multihoming," in *Networks, 2004.(ICON 2004). Proceedings. 12th IEEE International Conference on*, vol. 2. IEEE, 2004, pp. 713–719.

[27] ——, "Receive buffer blocking in concurrent multipath transfer," in *Global Telecommunications Conference, 2005. GLOBECOM'05. IEEE*, vol. 1. IEEE, 2005, pp. 6–pp.

[28] J. Liu, H. Zou, J. Dou, and Y. Gao, "Reducing receive buffer blocking in concurrent multipath transfer," in *Circuits and Systems for Communications, 2008. ICCSC 2008. 4th IEEE International Conference on.* IEEE, 2008, pp. 367–371.

[29] J. R. Iyengar, P. D. Amer, and R. Stewart, "Performance implications of a bounded receive buffer in concurrent multipath transfer," *Computer Communications*, vol. 30, no. 4, pp. 818–829, 2007.

[30] M. Becke, T. Dreibholz, H. Adhari, and E. P. Rathgeb, "On the fairness of transport protocols in a multi-path environment," in *Communications (ICC), 2012 IEEE International Conference on.* IEEE, 2012, pp. 2666–2672.

[31] C. Raiciu, C. Paasch, S. Barre, A. Ford, M. Honda, F. Duchene, O. Bonaventure, and M. Handley, "How hard can it be? designing and implementing a deployable multipath tcp," in *Proceedings of the 9th USENIX conference on Networked Systems Design and Implementation.* USENIX Association, 2012, pp. 29–29.

[32] C.-M. Huang, M.-S. Lin, and L.-H. Chang, "The design of mobile concurrent multipath transfer in multihomed wireless mobile networks," *The Computer Journal*, vol. 53, no. 10, pp. 1704–1718, 2009.

[33] F. H. Mirani, N. Boukhatem, and M. A. Tran, "A data-scheduling mechanism for multi-homed mobile terminals with disparate link latencies," in *Vehicular Technology Conference Fall (VTC 2010-Fall), 2010 IEEE 72nd.* IEEE, 2010, pp. 1–5.

[34] A. Afanasyev, N. Tilley, P. Reiher, and L. Kleinrock, "Host-to-host congestion control for tcp," *IEEE Communications surveys & tutorials*, vol. 12, no. 3, pp. 304–342, 2010.

[35] Y. Yuan, Z. Zhang, J. Li, J. Shi, J. Zhou, G. Fang, and E. Dutkiewicz, "Extension of sctp for concurrent multi-path transfer with parallel subflows," in *Wireless Communications and Networking Conference (WCNC), 2010 IEEE.* IEEE, 2010, pp. 1–6.

[36] F. Song, H. Zhang, S. Zhang, F. M. Ramos, and J. Crowcroft, "Relative delay estimator for sctp-based concurrent multipath transfer," in *Global Telecommunications Conference (GLOBECOM 2010), 2010 IEEE.* IEEE, 2010, pp. 1–6.

[37] J. Liao, J. Wang, T. Li, X. Zhu, and P. Zhang, "Sender-based multipath out-of-order scheduling for high-definition videophone in multi-homed devices," *IEEE Transactions on Consumer Electronics*, vol. 56, no. 3, 2010.

[38] K. Lan and N. Sha, "A cmt congestion window updates mechanism based on tcp westwood," in *Mechatronic Science, Electric Engineering and Computer (MEC), 2011 International Conference on.* IEEE, 2011, pp. 2119–2122.

[39] S. Shailendra, R. Bhattacharjee, and S. K. Bose, "Optimized flow based modeling of multi-path multi-homed transmission," in *Communication Systems (ICCS), 2010 IEEE International Conference on.* IEEE, 2010, pp. 411–415.

[40] T. D. Wallace, K. A. Meerja, and A. Shami, "On-demand scheduling for concurrent multipath transfer using the stream control transmission protocol," *Journal of Network and Computer Applications*, vol. 47, pp. 11–22, 2015.

[41] Y. Cao, C. Xu, J. Guan, and H. Zhang, "Cmt-cqa: Cross-layer qos-aware adaptive concurrent multipath data transfer in heterogeneous networks," *IEEJ Transactions on Electrical and Electronic Engineering*, vol. 10, no. 1, pp. 75–84, 2015.

[42] B. Wang, W. Feng, S.-D. Zhang, and H.-K. Zhang, "Concurrent multipath transfer protocol used in ad hoc networks," *IET communications*, vol. 4, no. 7, pp. 884–893, 2010.

[43] I. A. Halepoto, F. C. Lau, and Z. Niu, "Management of buffer space for the concurrent multipath transfer over dissimilar paths," in *Digital Information, Networking, and Wireless Communications (DINWC), 2015 Third International Conference on.* IEEE, 2015, pp. 61–66.

[44] J. Wu, C. Yuen, M. Wang, and J. Chen, "Content-aware concurrent multipath transfer for high-definition video streaming over heterogeneous wireless networks," *IEEE Transactions on Parallel and Distributed Systems*, vol. 27, no. 3, pp. 710–723, 2016.

[45] L. P. Verma and M. Kumar, "An adaptive data chunk scheduling for concurrent multipath transfer," *Computer Standards & Interfaces*, vol. 52, pp. 97–104, 2017.

[46] S. Liu, W. Lei, W. Zhang, and Y. Guan, "Cmt-sr: A selective retransmission based concurrent multipath transmission mechanism for conversational video," *Computer Networks*, vol. 112, pp. 360–371, 2017.

[47] M. Mathis, J. Semke, J. Mahdavi, and T. Ott, "The macroscopic behavior of the tcp congestion avoidance algorithm," *ACM SIGCOMM Computer Communication Review*, vol. 27, no. 3, pp. 67–82, 1997.

[48] J. Padhye, V. Firoiu, D. F. Towsley, and J. F. Kurose, "Modeling tcp reno performance: a simple model and its empirical validation," *IEEE/ACM Transactions on Networking (ToN)*, vol. 8, no. 2, pp. 133–145, 2000.

[49] R. Dunaytsev, Y. Koucheryavy, and J. Harju, "The pftk-model revised," *Computer communications*, vol. 29, no. 13, pp. 2671–2679, 2006.

[50] Z. Chen, T. Bu, M. Ammar, and D. Towsley, "Comments on modeling tcp reno performance: a simple model and its empirical validation," *IEEE/ACM Transactions on Networking (ToN)*, vol. 14, no. 2, pp. 451–453, 2006.

[51] C. Casetti and M. Meo, "An analytical framework for the performance evaluation of tcp reno connections," *Computer Networks*, vol. 37, no. 5, pp. 669–682, 2001.

[52] M. Garetto, R. L. Cigno, M. Meo, and M. A. Marsan, "Closed queueing network models of interacting long-lived tcp flows," *IEEE/ACM Transactions on Networking*, vol. 12, no. 2, pp. 300–311, 2004.

[53] S. Fu and M. Atiquzzaman, "Performance modeling of sctp multihoming," in *Global Telecommunications Conference, 2005. GLOBECOM'05. IEEE*, vol. 2. IEEE, 2005, pp. 6–pp.

[54] C. Casetti and M. Meo, "A new approach to model the stationary behavior of tcp connections," in *INFOCOM 2000. Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE*, vol. 1. IEEE, 2000, pp. 367–375.

[55] T. D. Wallace and A. Shami, "Concurrent multipath transfer using sctp: Modelling and congestion window management," *IEEE Transactions on Mobile Computing*, vol. 13, no. 11, pp. 2510–2523, 2014.

[56] M. Kwiatkowska, G. Norman, and D. Parker, "Prism: Probabilistic symbolic model checker," *Computer Performance Evaluation/TOOLS*, vol. 2324, pp. 200–204, 2002.