

PERFORMANCE ANALYSIS OF STRATEGIES FOR CONTROLLING OVERLOAD AND IMPROVING SCALABILITY IN SIP BASED NETWORKS

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by

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To my parents, husband and son.

Certificate

This is to certify that the thesis entitled “**Performance analysis of strategies for controlling overload and improving scalability in SIP based networks**” being submitted by **Mrs. Garima Mishra** to the Bharti School of Telecommunication Technology and Management, Indian Institute of Technology Delhi, for the award of the degree of **Doctor of Philosophy** is the record of the bonafide research work carried out by her under our supervision. In our opinion, the thesis has reached the standards fulfilling the requirements of the regulations relating to the degree.

The results contained in this thesis have not been submitted either in part or in full to any other University or Institute for the award of any degree or diploma.

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Abstract

Recent collapses of Session Initiation Protocol (SIP) servers in the carrier networks indicate two potential problems of SIP: (1) the built-in SIP overload control mechanism cannot handle overload conditions effectively, and (2) the current SIP design does not easily scale up to large network sizes. In order to help carriers prevent widespread SIP network failures effectively, this thesis presents a systematic investigation of current state-of-the-art overload algorithms and SIP-based group communication frameworks adopted by SIPPING, XCON and MMUSIC working groups of IETF. To achieve this goal, first, we propose an analytical model for the re-designed SIP proxy server to control overload locally and develop an algorithm to control overload hop-by-hop. Then a packet aggregation and compression mechanism is provided which improves the capability of single conferencing servers to handle large number of concurrent users.

SIP has a retransmission mechanism to maintain the reliability for its real time transmission. But these real time transmissions cause overload in the server and creates redundant messages. A feedback response code 503 (Service Unavailable) is sent from a SIP server once it experiences overload. It is a built-in functionality of SIP which provides a limited overload control, thus making SIP servers susceptible to overload. This limited overload control mechanism oscillates signaling traffic and shifts it between their neighbouring servers and make overload condition worse (RFC 5390). In this thesis, local overload mechanism is implemented within a SIP proxy server without requiring cooperation between servers. The local overload control is achieved by redesigning the SIP proxy server as a multi-server cut-off priority queue model. We design SIP proxy server by separating signaling traffic in two different classes (**Invites** and **Non-Invites**) and creating a cut-off priority queueing model. In order to study

the impact of overloaded SIP proxy server on throughput, the entire end-to-end SIP signaling flow is modeled between a sending and a receiving SIP servers. We propose a complete end-to-end SIP signaling flow for session establishment between two end users using Generalized Stochastic Petri Nets (GSPN) technique.

Next, we study the **hop-by-hop overload control** mechanism by extending the previously developed SIP proxy model for timed-out packets which cause retransmissions. SIP retransmission mechanism with timeout is modeled as a queueing system with impatient customers. Using this model, the effect of unnecessary retransmissions is studied and delay distribution and loss probability (*Pb_loss*) is calculated. The proposed analytical model is verified with simulations which demonstrate that the inclusion of timeout gives better delay performance. Using *Pb_loss*, an algorithm is developed to control the overload in hop-by-hop transaction as described in RFC 6357 [1]. The simulation results for the proposed model with overload control and the standard SIP as per RFC 3261 [2] are compared. The results demonstrate that the server utilization factor of an overloaded server always remains less than one and hence avoids system collapse. Further, the redundant messages in the system are reduced by 30% as compared with the standard SIP network.

SIP signaling protocol is also used for providing group communication in IP-based next generation network. The conferencing framework proposed by Internet Engineering Task Force (IETF) working groups has limitation for large scale networks due to their control over entire conference by a single central server. In order to overcome this limitation, we propose multiple SIP packet aggregation and compression of common header part of each aggregated packet process. This mechanism increases the bandwidth utilization and processing capability of conference server. A queueing model is presented to describe the process of packet aggregation and compression at the aggregator node. Using the proposed queueing model, the steady state probability distribution of the queue length is derived and the average packet delay is obtained. We show the trade-off between the overhead gain and delay. Based on analytical results obtained, the maximum frame size is optimized for which the average delay of a packet is minimum. The packet aggregation and compression mechanism is implemented using NS2 simulator. The key session setup performance measures discussed in **ITU E.721** are obtained which particularly focus on the SIP protocol requirements. The proposed

queueing model is further studied under $\text{Min}(N, T)$ policy. We consider two control parameters queue length threshold N and timer T to suppress the increase in delay due to the long frame formation itself. These parameters are optimized such that we get the maximum bandwidth utilization and minimum delay.

We may also note, as a interesting observation based on the above results, that if multi-party games are played on the Next Generation Networks (NGN), it would be better to aggregate and compress packets at the edge of the network. A gaming API with this provision would perform better than one without it.

Table of Contents

Certificate	i
Acknowledgements	ii
Abstract	iv
List of Figures	xi
List of Tables	xv
List of Abbreviations	xvi
1 Introduction to the Session Initiation Protocol	1
1.1 Overview of SIP	1
1.2 SIP Retransmission Mechanism	5
1.3 SIP Overload Problem	7
1.4 Multi-party Conferencing	11
1.5 Motivation	18

1.6	Summary of Contributions	19
1.7	Organization of this Thesis	21
2	Local overload control in combination with Hop-by-Hop overload control in a SIP signaling network	24
2.1	Introduction to SIP Proxy Server	24
2.2	Motivation	25
2.3	Approach for Re-designing the SIP Proxy Server	28
2.4	Local Overload Control	29
2.5	Analysis of SIP Signaling System with Impatient Customers	41
2.6	Hop-by-hop Overload Control	48
2.7	Performance Evaluation of the Proposed Overload Control Model and Simulation Results	50
2.8	Summary	59
3	Increasing scalability of SIP based group communication using packet aggregation	60
3.1	Introduction to Multi-party Communication Mechanism in SIP	60
3.2	Packet Aggregation Overview	62
3.3	Multi-party Conferencing Architecture	63
3.4	Edge-proxy Model for the Proposed Packet Aggregation Mechanism . . .	67

3.5	Measuring Performances: Packet Loss Rate, Bandwidth Utilization and Average Waiting Time	71
3.6	Performance analysis of Group Communication with the Proposed Packet Aggregation Mechanism	72
3.7	Simulation Model	75
3.8	Optimization Problem for Queue Threshold and Vacation Time	76
3.9	Summary	79
4	Increasing scalability of SIP based group communication using packet aggregation and compression process	80
4.1	Introduction to the Need for Packet Aggregation and Compression in SIP	80
4.2	Queueing Model for the Proposed Packet Aggregation and Compression Mechanism	84
4.3	Implementation of Packet Aggregation and Compression using NS2 . . .	92
4.4	Key Session Setup Metrics and Simulation Results	100
4.5	Analysis with the Aggregation Parameters-Idle Duration T and Queue Threshold N	104
4.6	Optimizing queue threshold N	106
4.7	Performance Analysis of Proposed Packet Aggregation and Compression Mechanism	108
4.8	Summary	113
5	Conclusions	116

Bibliography	119
Appendix	127
A SIP Model implementation in SimEvents	127
B NS2 Simulation Example	130
Research Publications	135
Author Biography	136

List of Figures

1.1	IMS Layered Architecture	2
1.2	SIP signaling call flow for session setup	3
1.3	SIP Presence service call flow	5
1.4	SIP message retransmissions	6
1.5	SIP server configuration for local overload Control	10
1.6	SIP server configuration for hop-by-hop overload Control	10
1.7	SIP server configuration for end-to-end overload Control	11
1.8	Summary of contributions	22
2.1	Two versions of multi-server queue model	26
2.2	Simulation result comparison between the two multi-server queue models	27
2.3	SIP server configuration for local overload Control	29
2.4	State transition diagram for two types of traffic in a SIP server	31
2.5	SIP message call flow between a client and a server to initiate a session .	36

2.6	n -stage tandem network of queues representing sequence of message handshake to establish a session [3]	37
2.7	GSPN Model of SIP message handshake	40
2.8	State transition diagram for the service and idle time periods	45
2.9	Queueing model of the modified system with timeout	47
2.10	System model for hop-by-hop overload control	49
2.11	Effect on probability of delay by varying SIP signaling <Non-Invite> traffic	52
2.12	Effect on probability of delay by varying SIP signaling <Invite> traffic .	52
2.13	Effect of server reservation with cut-off priority on probability of delay .	53
2.14	Average goodput of the network on varying signaling traffic (λ)	53
2.15	Average end-to-end delay in case of overload in the system with and without OC algorithm	56
2.16	Effect of feedback signal on fraction of redundant messages in the system	56
2.17	Utilization of the server under overload condition	57
2.18	Effect of OC algorithm on goodput and throughput of SIP server	57
2.19	Goodput comparison with varying SIP signaling traffic for different overload control mechanisms	58
3.1	Packet Aggregation overview for sending three packets	64
3.2	Multi-party conference in an IMS domain	64
3.3	Signaling flow for the conferencing session establishment	66

3.4	State transition diagram for the $M^X/M^Y/1/B(E, MV)$ queueing model .	70
3.5	Blocking Probability on varying buffer size for different mean vacation times $(1/\theta)$	73
3.6	Blocking Probability on varying mean vacation time for different arrival rates (λ)	73
3.7	Effect of mean vacation time on mean queue length for different arrival rates (λ)	74
3.8	Effect of ρ on mean queue length for various values of average batch size	74
3.9	Simulation topology in NS2 for multiple UEs connected to the IMS using SIP	77
3.10	Effect of traffic load on mean waiting time for different vacation times $(1/\theta)$ in the proposed model	77
3.11	Effect of traffic load on mean waiting time for different vacation times $(1/\theta)$ in the Traditional model	78
3.12	Relationship among packet loss rate PLR , bandwidth utilization BU and average waiting time Wq	78
4.1	Packet aggregation and compression process and signaling flow to establish a conferencing session	83
4.2	Equivalent queueing model for packet aggregation and compression process	84
4.3	State transition diagram for the proposed queueing model for packet aggregation and compression	86
4.4	Basic simulation scenario	93

4.5	Session request delay for the given SIP requests with and without packet aggregation mechanism	101
4.6	Network capacity in terms of session establishment ratio for a given system with and without packet aggregation mechanism	102
4.7	Call setup failure rate or ineffective session attempts (ISA) with respect to the injected requests	104
4.8	Mean delay versus traffic load for different values of maximum frame length	109
4.9	Mean delay versus traffic load for different values of minimum frame length	110
4.10	Mean delay versus maximum frame size L for various values of μ_2	111
4.11	Trade-off between overhead $(1/\mu_2\bar{l})$ and mean delay	113
4.12	Effect of packet aggregation and compression on packet loss rate	113
4.13	Trade-off between average delay and throughput	114
4.14	Queue threshold value N versus average cost function for various Timer values T	114
4.15	Timer T versus average cost function for various values of N	115
A.1	End-to-end SIP signaling simulation model	128
A.2	The main components of SIP proxy modeled in simevents	129

List of Tables

1.1	Comparison of scalability in types of SIP-based conferences [4]	13
2.1	List of places	42
2.2	List of timed transitions	43
2.3	List of immediate transitions	43
2.4	Average delay of SIP messages without and with timeout	55
3.1	Parameters assumed for the wired link	75
4.1	Target values for post selection delay from ITU E.721	101
4.2	The probability of having n packets in a frame	112

List of Abbreviations

3GPP	Generation Partnership Project
AS	Application Server
CSCF	Call/Session Control Function
CTMC	Continuous Time Markov Chain
DA	Destination-based Aggregation
DES	Discrete Event Simulation
FIFO	First In First Out
GSPN	Generalized Stochastic Petri Nets
I-CSCF	Intermediate-Call/Session Control Function
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
LST	Laplace Stieltjes Transform
MFS	Maximum Frame Size
OC	Overload Control
P2P	Peer2Peer
P-CSCF	Proxy-Call/Session Control Function
PGF	Probability Generating Function
QBD	Quasi Birth Death
QoS	Quality of Service
RTP	Real-time Transport Protocol
RTT	Round Trip time
S-CSCF	Server-Call/Session Control Function
SDP	Session Description Protocol
SIP	Session Initiation Protocol

TCP	Transmission Control Protocol
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UE	User Equipment
UDP	User Datagram Protocol