Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

NEW RE-ROUTING AND RETRANSMISSION TIMEOUT POLICY THROUGH REAL TIME WEB PERFORMANCE ANALYSIS

M.S.GANESH[1], DR. S.RAMKUMAR[2]

[1]Associate professor, Department of Computer applications
RM.D Engineering College, Kavaraipettai, Tamil Nadu, India
Email ID:ganeshsrnvs@gmail.com, Tel:+919444732412
\
[2]Professor, Department of Computer Science,
RM.D Engineering College, Kavaraipettai, Tamil Nadu, India
Email ID: set.dean@gmail.com

ABSTRACT

Most important performance metrics quantifies TCP retransmission timeouts (RTOs) is the Round Trip Time (RTTs), which create havoc on network and application performance by introducing huge amount of retransmission packet over the internet routing. This paper tries to locate various means of non-fair RTOs using real-time web-based internet domain server access procedure. The impact is identified through Wireshark tool, Tracert, windows Application Programming Interface (APIs) procedure and other parameters of internet connection such as bandwidth, time of access, traffic intensity time zone with respect to routing parameters. The experiment is to collect packet parameters on the internet on various factor and analyze the same for the impact. Through the study the need for the fine tuning on Retransmission along with Round trip time is identified .The study also provide the new re-routing strategy to improve the overall performance of routing and retransmissions suing simple communication agent architecture on TCP level.
Keywords: *Transmission control protocol, Retransmission Timeouts, Round Trip Time, Web domain, Routing, Wireshark*

1. INTRODUCTION

The Retransmission Timeout (RTO) algorithm of Transmission Control Protocol (TCP) plays an important role in reliable data transfer. A rigorous real time practical analysis of the RTO algorithm is important based on RTTs. Over the internet the percentage of retransmitted packet are being introduced due to non-fair values of RTO, calculated based on available RTTs. Considering the scenario of sender receive the acknowledgement (ACK) immediately after the timeout occurs which introduce the duplicate packet into the internet. To find the fairness in the RTO not totally on the RTTs and also on the following

- Characteristics of varying receiver advertise window with RTT.
- Traffic intensity time zone considered for the Router load and routing path for the delivery of packet over period of time.
- Link quality estimate defines the nature of link as factor of number of segment sent versus number of retransmission over the link,
- Based on the link quality the rerouting strategies can be derived to make path more reliable rather than shorter.

In this paper it's proposed to increase fairness in retransmission timeout to improve overall performance of routing and TCP throughput. This is done through identifying the areas of Routing and TCP parameter such as Round trip time, number of Retransmission packet and retransmission timeout mechanism suing proposed experimental setup discussed further

2. RELATED WORK

Nandita Dukkipati et.al.,[1] explore some of the weaknesses of the standard algorithm described in RFC3517 and the non-standard algorithms implemented in Linux. It is found these algorithms deviate from their intended behavior in the real world due to the combined effect of short flows, application stalls, burst losses,

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

acknowledgment (ACK) loss and reordering, and stretch ACKs. Linux suffers from excessive congestion window reductions while RFC3517 transmits large bursts under high losses, both of which harm the rest of the flow and increase Web latency.

P. Papadimitriou and V. Tsaousidis[2], finds the Real-time transport over wired/wireless networks is challenging, since wireless links exhibit distinct characteristics, such as limited bandwidth and high error rates, due to fading or interference Based on an analytical approach, as well as extensive simulations, it is shown that local recovery prevents wasteful end-to-end retransmissions and allows the transport protocol to utilize a higher fraction of the available bandwidth. However, it is found that uncover undesirable effects of local error control which degrade the performance of real-time delivery in several occasions.

Honda, Osamu, et al.[3] describes the factors that may affect the end-to-end TCP performance include: network parameters such as the link bandwidth, the propagation delay, MTU (Maximum Transmission Unit) and the router buffer size; network workload such as the number of TCP flows , the number of TCP tunnels, the traffic pattern of TCP flows and the background traffic; TCP configurations such as TCP version (e.g., Tahoe, Reno, New Reno and Vegas), existence of the SACK option, the socket buffer size and the initial value of RTO (Retransmission Time Out).

Liangping Ma; Barner, K.E.; Arce, G.R.,[4] finds the impact of retransmission timeout (RTO) algorithm of Transmission Control Protocol (TCP), which sets a dynamic upper bound on the next round-trip time (RTT) based on past RTTs, on reliable data transfer and congestion control of the Internet. It is found nevertheless, such an analysis has not been conducted to date. In this paper they present such an analysis from a statistical approach through an auto-regressive (AR) model for the RTT processes. Experimental results that indicate: 1) RTTs along a certain path in the Internet can be modeled by a shifted Gamma distribution and 2) the temporal correlation of RTTs decreases quickly with lag. This model is used to determine the average reaction time and premature timeout probability for the RTO algorithm. The theoretical analysis strengthens a number of observations reported in past experiment-oriented studies

Mark Allman et.al ,[5] states that deriving accurate estimates of the loss rate from TCP transfers has been largely unaddressed. In his paper first it is shown that using a simple count of the number of retransmissions yields inaccurate estimates of the loss rate in many cases. The mis-estimation stems from flaws in TCP's retransmission schemes that cause the protocol to spuriously retransmit data in a number of cases. Using new techniques for refining the retransmission count to deduce the loss rate estimate for both Reno and SACK variants of TCP. Finally, it they explore the benefit of reducing the number of needless retransmits is a reduction in the amount of shared network resources used to accomplish no useful work.

Loukili et.al[11] use the wirehsark as tool to estimate the performance of tcp retransmission time setting to understand in various operating system such as Linux, windows to verify the RTO of TCP SACK and congestion control.

3. APPROACH AND EXPERIMENTAL ETUP

Using ping, trace route and communication using wireshark and Visual basic APIs [6] various parameters of real time internet domain are collected with help of 100 web domain URL In various time intervals the applications communicate with servers of the URL which are not restricted by the firewall.**Fig-1** show the running instance of program in visual basic environment.

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

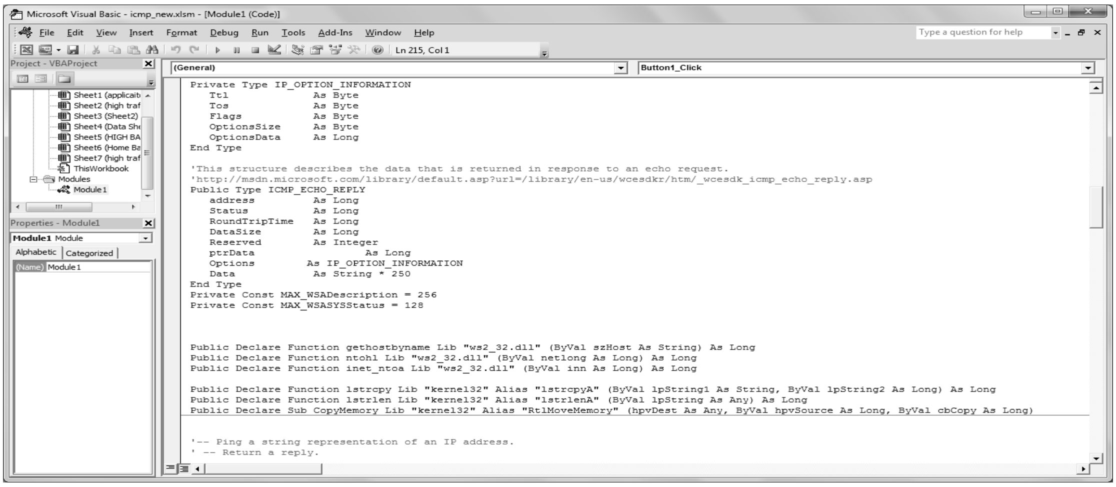


Fig-1 Visual Basic API Program To Communicate With Various Webs URL

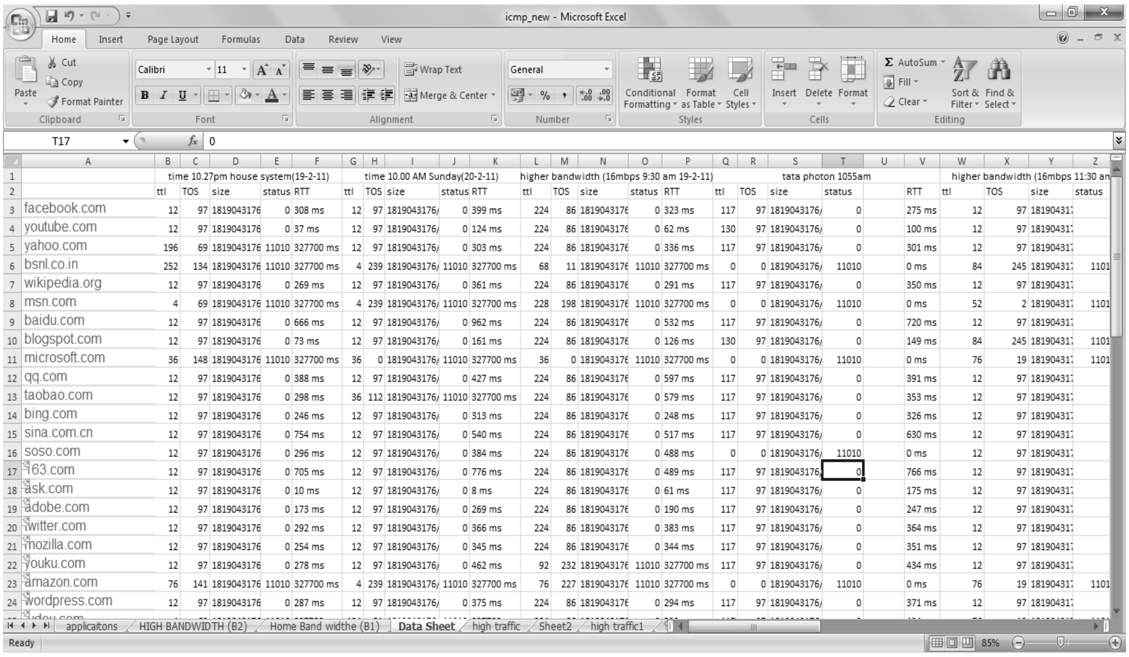


Fig-2 Information Collected Using Visual Basic Apis

Wireshark[7] is the world's foremost network protocol analyzer. It lets you capture and interactively browse the traffic running on a computer network. It is the de facto (and often de jure) standard across many industries and educational institutions.

With the help of the wireshark about 30000 packets are captured and analyzed for various parameters of network which are discussed in the result section of this paper. **Fig-3** shows the capturing with various point of traffic intensity with same bandwidth using same set of URL list.

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

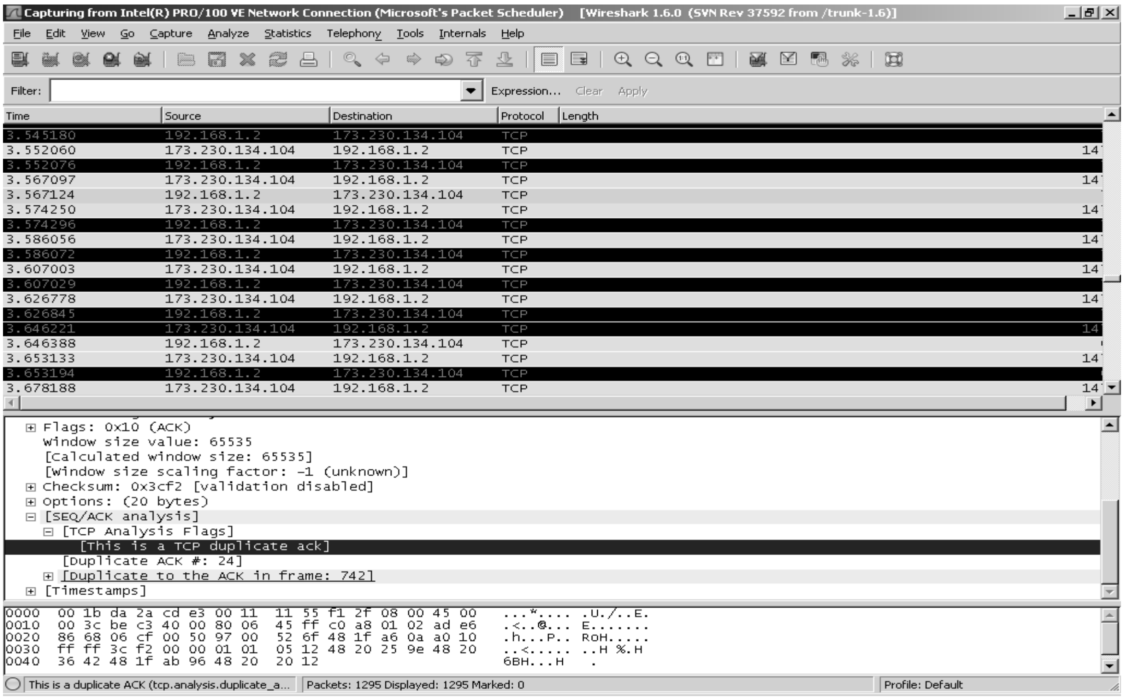


Fig -3 Wireshark Capture Packet With Retransmission Packet Indicated In Black

4. USING DATA COLLECTED FROM API PROGRAM WITH RESULTS AND DISCUSSIONS

With help of Visual basic API the change in the RTT are noted in various time interval which depicts the traffic intensity from the host and destination that is the RTT changes in rapidly as the time with higher or lower bandwidth. It is also found that the traffic intensity time after 6pm to 10pm the RTT are in the higher side with single bandwidth capacity.

The Fig -4 shows the relation between the RTT values on various time intensity time zone as high intensity and low intensity. It is found that for one web url www.facebook.com communications provide divergent RTT values various from 10ms to 632 ms where 1000 ms states the website is unreachable. The same thing is found for various web url given the graph.

The Fig-5 shows the RTT value variation along with comparison using bandwidth and the traffic intensity time. It is noted that here also the RTT values are not linear with bandwidth or traffic but non-linear with parameters. The variation is not predictable with parameters, as the high bandwidth having low RTT values than the other. it holds good for the traffic intensity also. The nature is due to retransmission of packet in the particular time for a connection which may due to congestion or load in particular route or reduced delivery ratio.

The Fig-6 shows the impact of bandwidth alone on the RTT values for set URL web communications. It is found that the RTT values of non-predictable and divergent in nature values various from 491ms to 991ms, which are in comparison higher than with respect to adding traffic intensity. The result is compared with variant payload details suggested by Choudhury, Sayantan, et al [8] in their paper and Ramasubramanian,et.al[9].

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

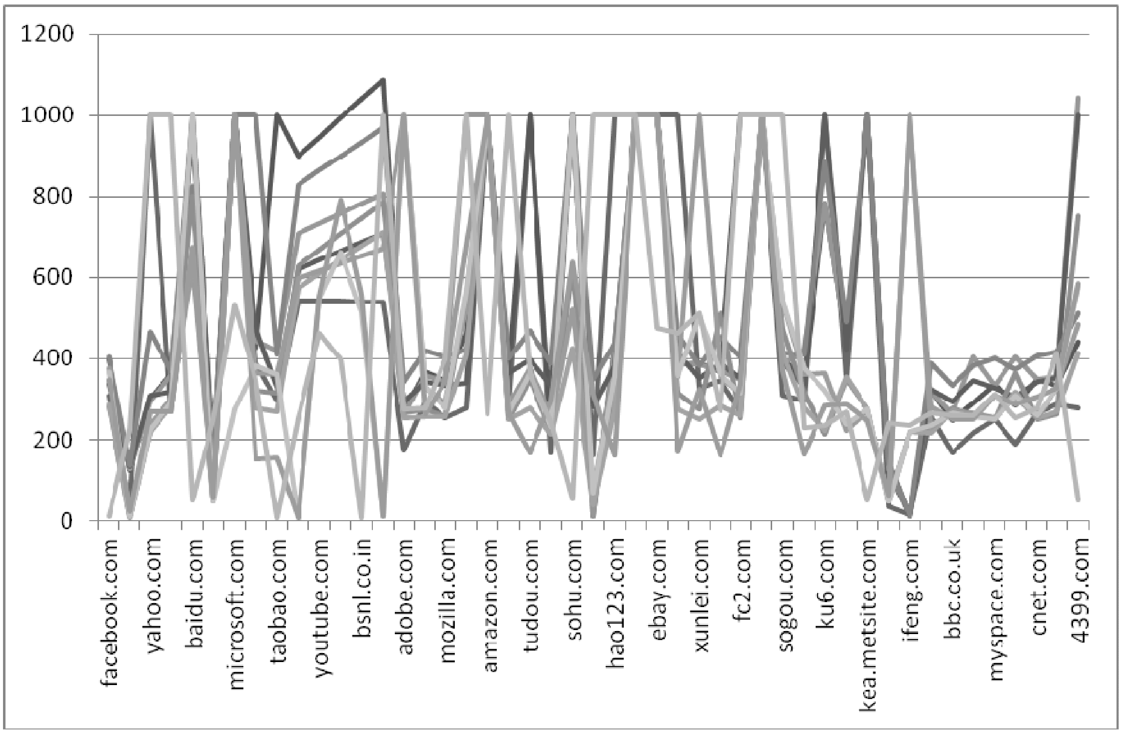


Fig-4 show the RTT vs web URL Collected on various traffic intensity time

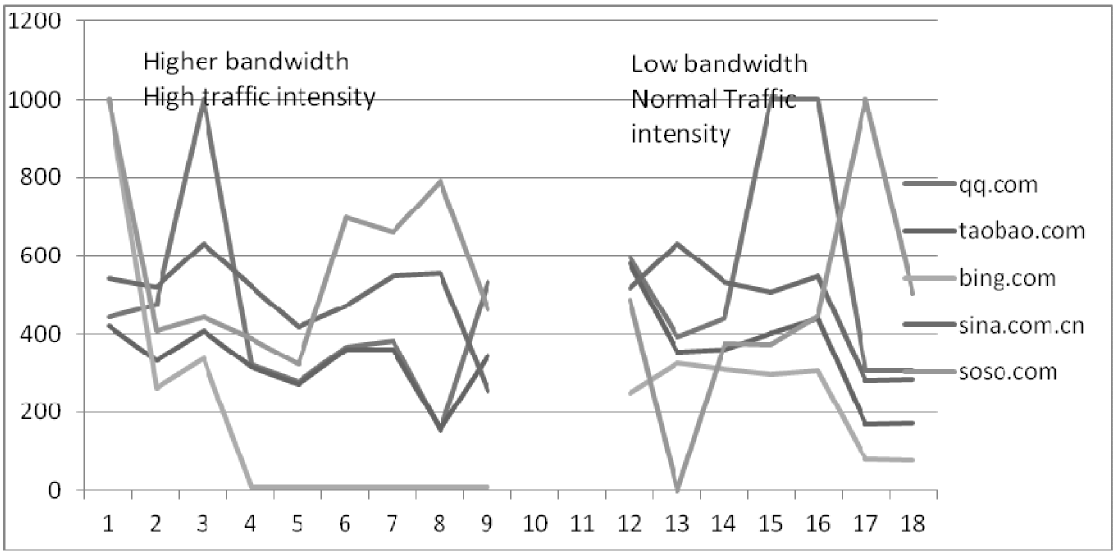


Fig-5 show the RTT variation with respect to bandwidth and traffic intensity for sample URL selected

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

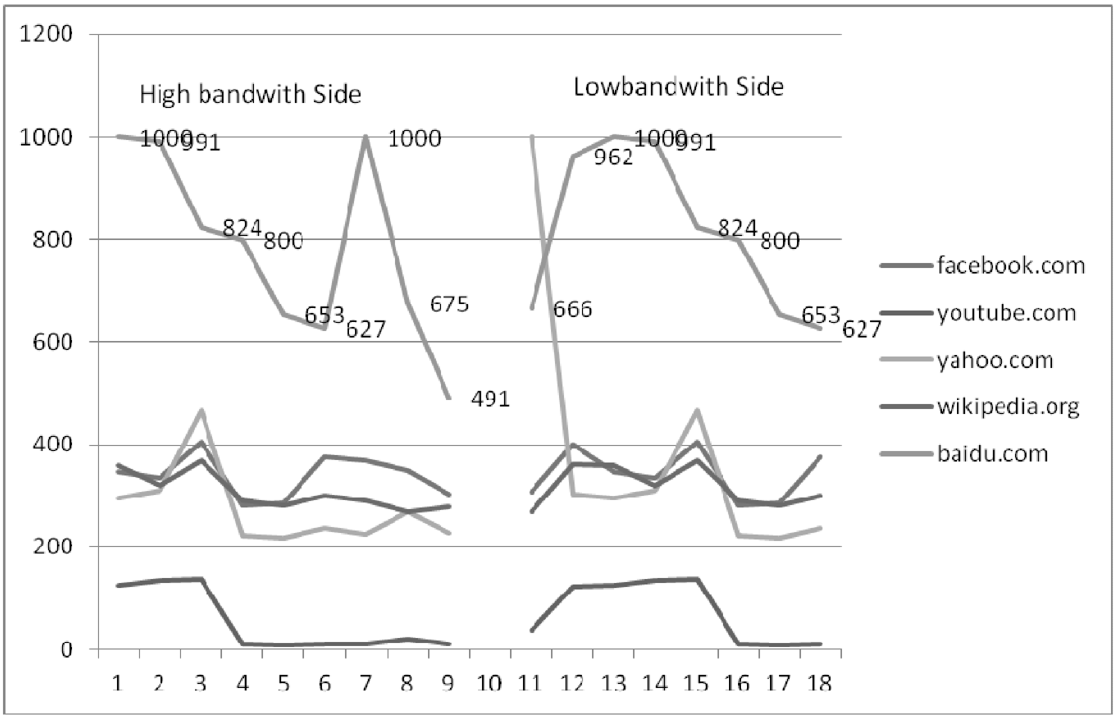


Fig-6 show the RTT variation with respect to bandwidth alone for sample URL selected

5. WITH TRACE ROUTE OF VARIOUS WEB URL RESULTS AND DISCUSSIONS

It is also noted that the route taken by the host to destination plays vital role in the reducing the RTT and in turn the link reliability. From the Trace route collected data some of the servers are loaded even though the direction path is not matched with route. It is suggested that the rerouting policy can be implemented to make path are less loaded during the traffic intensity time.

Table 1 shows data collected from selected 100 web URLs Trace route values are being analyzed using MSACESS database and queried to find the load on the routers along the path. It is found that Japan server are being routed through MUMBAI-LONDON-USA ROUTE instead a Korean web server is being directed through SINGAPORE this anomaly increase the RTT values to 271ms instead later with 71ms.

It is found from the Tracert Experiment the routing from various points of network to URL domains are routed independent of the actual location of router or server. It is suggested that the routing can made dynamic based on the parameter collected such as the reliability, load on the router path, and maximum or minimum bandwidth availability along the path of the data transfer. For example table given below it is found that to reach an Korea server the path diverted in the direction opposite to actual path that is Mumbai-London-Dallas-Korea instead we reroute towards the Singapore server which can reach destination with less RTT values in turn reduce the RTO and retransmission of missing packet. The impact of RTT and RTO in web latency is studied and provided by Flach, Tobias et.al [10] which encourages the study towards the need of fairness in RTO.

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

Table 1 : Sample Trace Route Path Of Sample Web Urls With Unmatched Path Selection

Hop no.	RTT1		RTT2		RTT3		Intermidate ip path
1	<1 m	s	<1 m	s	<1 m	s	10.0.0.10
2	1	ms	1	ms	1	ms	Static-97.109.93.111.tataidc.co.in [111.93.109.97]
3	4	ms	4	ms	4	ms	Static-189.108.93.111.tataidc.co.in [111.93.108.189]
4	5	ms	5	ms	5	ms	121.241.157.149.static-chennai.vsnl.net.in [121.241.157.149]
5	29	ms	38	ms	29	ms	172.31.16.173
6	30	ms	76	ms	32	ms	ix-0-100.tcore2.MLV-Mumbai.as6453.net [180.87.39.25]
7	155	ms	153	ms	154	ms	if-6-2.tcore1.L78-London.as6453.net [80.231.130.5]
8	*		152	ms	158	ms	Vlan704.icore1.LDN-London.as6453.net [80.231.130.10]
9	149	ms	170	ms	149	ms	Vlan533.icore1.LDN-London.as6453.net [195.219.83.102]
10	153	ms	163	ms	152	ms	vl-3604-ve-228.csw2.London1.Level3.net [4.69.166.157]
11	153	ms	154	ms	154	ms	ae-57-222.ebr2.London1.Level3.net [4.69.153.133]
12	251	ms	252	ms	251	ms	ae-42-42.ebr1.NewYork1.Level3.net [4.69.137.70]
13	261	ms	266	ms	261	ms	ae-81-81.csw3.NewYork1.Level3.net [4.69.134.74]
14	259	ms	255	ms	256	ms	ae-82-82.ebr2.NewYork1.Level3.net [4.69.148.41]
15	248	ms	248	ms	248	ms	ae-3-3.ebr2.Dallas1.Level3.net [4.69.137.121]
16	255	ms	249	ms	249	ms	ae-92-92.csw4.Dallas1.Level3.net [4.69.151.165]
17	*		251	ms	251	ms	ae-4-90.edge2.Dallas3.Level3.net [4.69.145.204]
18	252	ms	252	ms	252	ms	RACKSPACE-M.edge2.Dallas3.Level3.net [4.59.36.50]
19	271	ms	271	ms	271	ms	corea.dfw1.rackspace.net [74.205.108.34]
20	252	ms	252	ms	252	ms	core3.dfw1.rackspace.net [74.205.108.7]
21	253	ms	252	ms	252	ms	jt-w1.japantoday.com [108.166.65.155]
trace rt foro www.korea.net.gccdn.net[119.31.253.205]							
1	<1 m	s	<1 m	s	<1 m	s	10.0.0.10
2	1	ms	1	ms	1	ms	Static-97.109.93.111.tataidc.co.in [111.93.109.97]
3	4	ms	4	ms	4	ms	Static-189.108.93.111.tataidc.co.in [111.93.108.189]
4	5	ms	5	ms	5	ms	121.241.157.149.static-chennai.vsnl.net.in [121.241.157.149]
5	5	ms	5	ms	5	ms	ix-4-2.tcore1.CXR-Chennai.as6453.net [180.87.36.9]
6	38	ms	37	ms	46	ms	if-5-2.tcore1.SVW-Singapore.as6453.net [180.87.12.53]
7	38	ms	38	ms	38	ms	if-0-0-0-213.core2.IH4-Singapore.as6453.net [180.87.12.66]
8	37	ms	37	ms	37	ms	if-0-0-0-500.core1.IH4-Singapore.as6453.net [180.87.136.1]
9	41	ms	39	ms	42	ms	203.116.20.153
10	37	ms	37	ms	38	ms	an-ats-int10.starhub.net.sg [203.118.3.230]
11	50	ms	41	ms	48	ms	203.117.6.22

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

12	72	ms	71	ms	71	ms	175.41.4.230
13	38	ms	38	ms	38	ms	119.31.253.205
trace rt foro www.australia.gov.nu[67.215.65.132]							
1	1	ms	1	ms	<1	ms	10.0.0.10
2	1	ms	1	ms	1	ms	Static-97.109.93.111.tataidc.co.in [111.93.109.97]
3	4	ms	4	ms	4	ms	Static-189.108.93.111.tataidc.co.in [111.93.108.189]
4	5	ms	5	ms	5	ms	121.241.157.149.static-chennai.vsnl.net.in [121.241.157.149]
5	69	ms	70	ms	71	ms	Vlan1331.icore1.HK2-HongKong.as6453.net [116.0.67.121]
6	248	ms	247	ms	246	ms	opendns-RGE.hkix.net [202.40.160.189]
7	247	ms	248	ms	247	ms	hit-nxdomain.opendns.com [67.215.65.132]

6. ANALYSIS FROM WIRESHARK

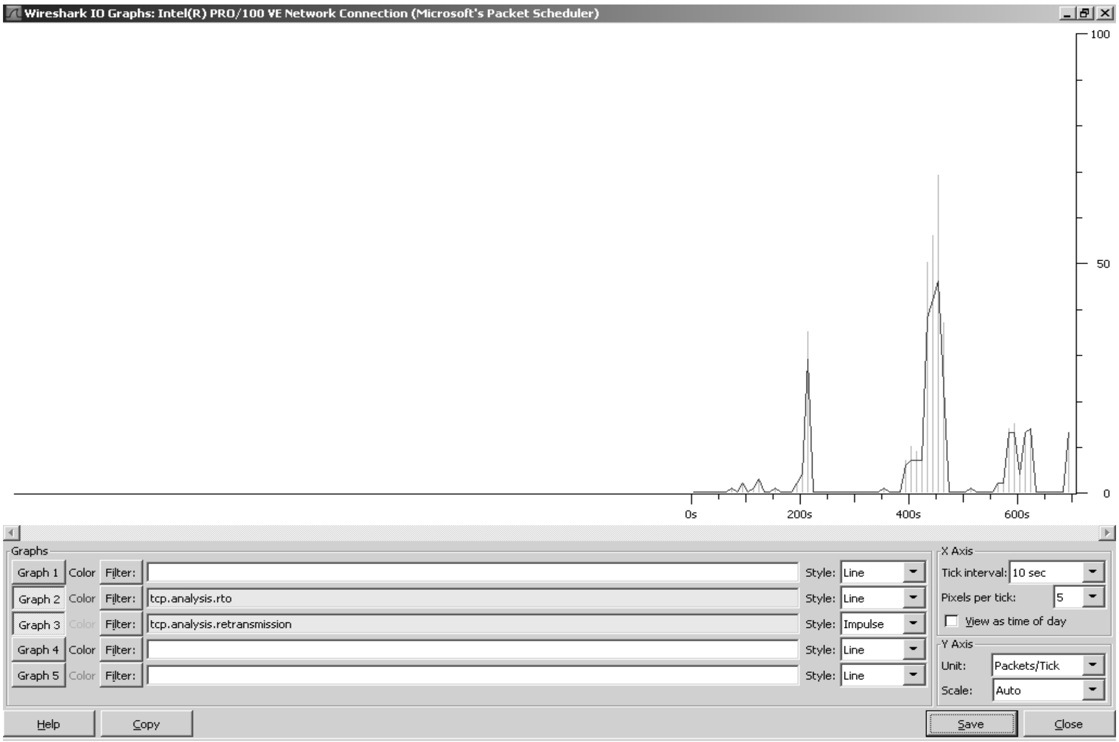


Fig-7 Shows The Wire Shark Analysis Of Retransmission Of Packet With RTO And RTT Values

The Fig-7 from wireshark with varied URL communication shows the relation among RTO, RTT and Out of order packet. It shows the impact of RTT on RTO and out order packet are similar as they tend to increase with respect to other. These results justify the need of the fine tuning of the RTT and RTO to have better throughput and reduce retransmission in unpredictable link failure environment.

The Fig-8 from wireshark with varied URL communication shows the nature of Time To Live (TTL) for the captured packet in wireshark tool. It is found that the TTL has strong mapping with previous graph parameter RTO and RTT which shows that the retransmitted packet roaming around the network may cause the congestion which result in increase in RTO.

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

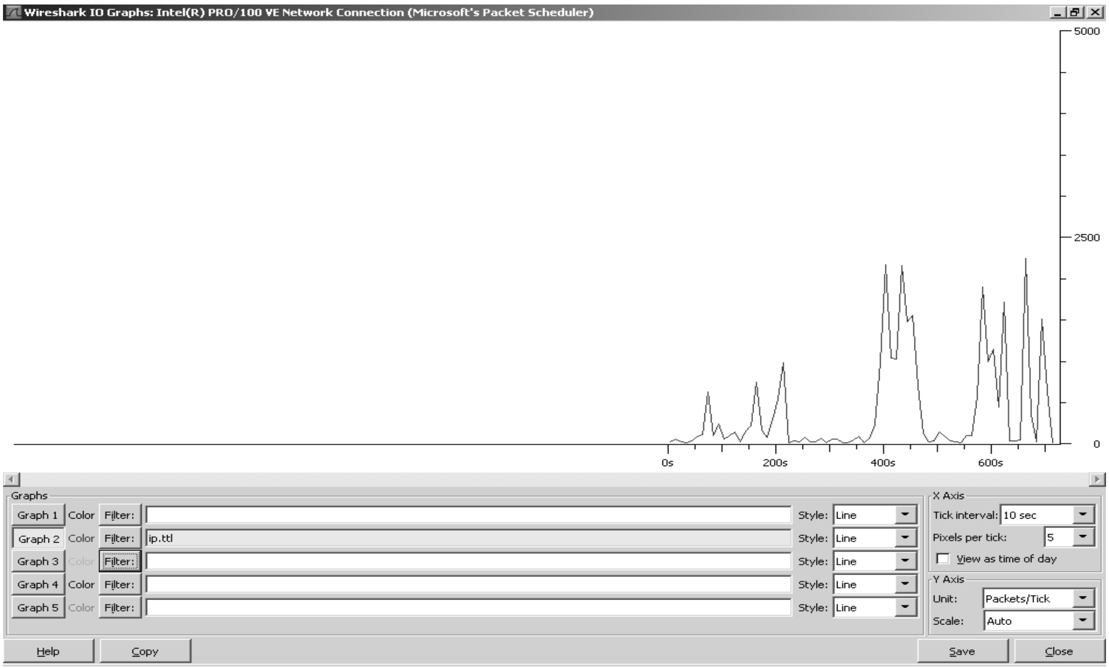


Fig – 8 shows the TTL values for the communication of URL using Wireshark

The following graph with data captured form wireshark show the intense relationship between retransmission timeout and RTT for period of time. The variation shows the two attributes, which may be influenced by the number of unacknowledged packet in the network to greater extent as around 450ms RTO and Number of retransmitted segment are high.

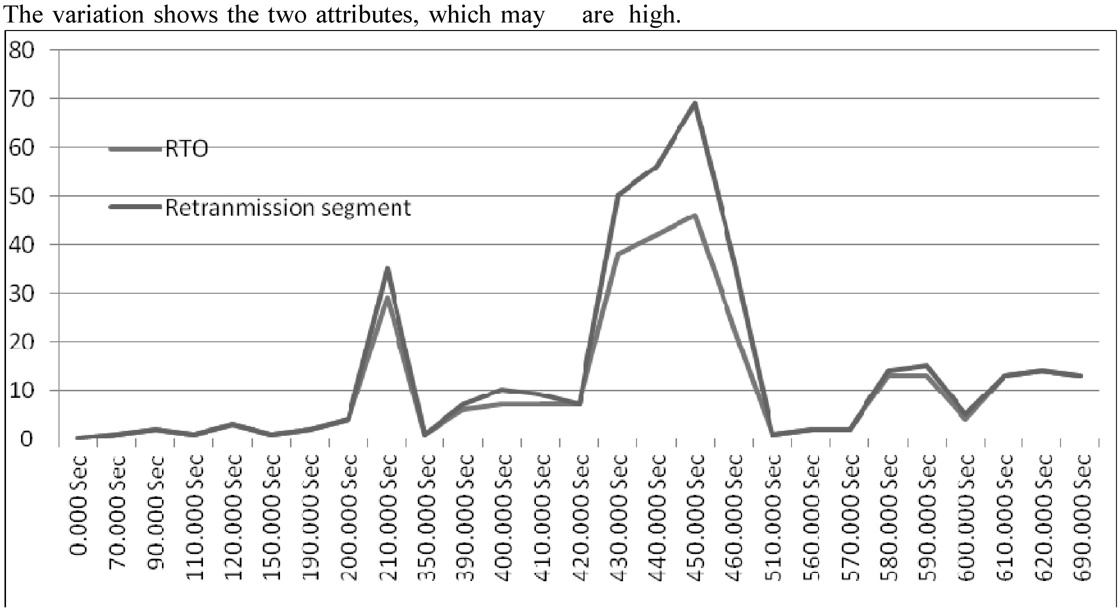


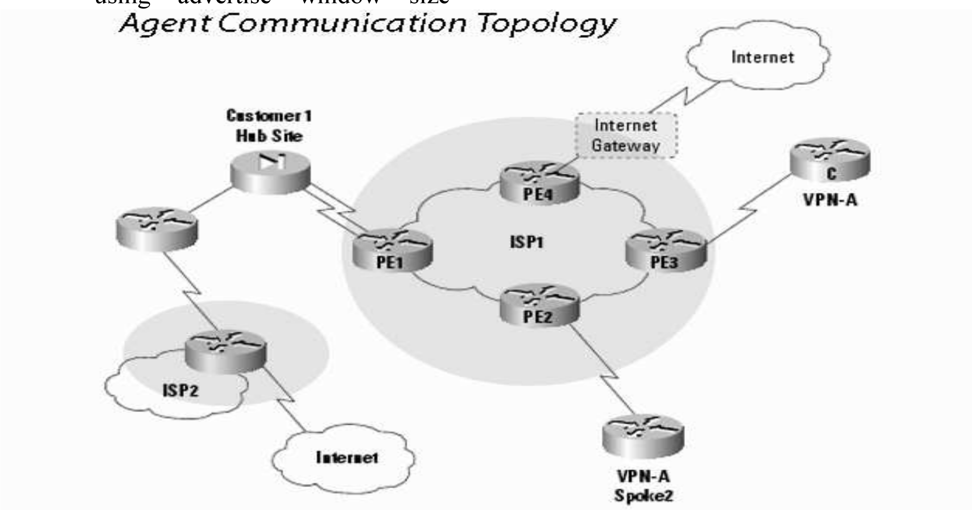
Fig-9 Relationship between Number Retransmission segment and Round trip time

Journal of Theoretical and Applied Information Technology
10th October 2014. Vol. 68 No.1

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

7. SUGGESTED FINE TUNING METHODS AND RE-ROUTING POLICY

1. Varying RTO formula based on the real time data provide by both IP and TCP sliding window algorithm such that the advertise window, rate of deliver of packet with RTT, Variance of the RTO are considered to make final decisions on the value of RTO for every transmission of segment by TCP.A simple algorithm based on the sliding window protocol parameter such as Advertize window size, rate of deliver of packet are calculated and fix the better Retransmission timeout Calculation. It is suggested following sample calculation will lead to better retransmission timeout which will reduce the number of retransmission packet.
   a. Base rate of delivery calculated using advertise window size ,sender window size and previous RTT variance
   b. Using intermediate RTT the Base RTT is calculated if variation is larger and this reflects the dynamic nature of network.
   c. An Expected rate of change of delivery also calculated base on the present scenario and previous values to fix new Retransmission timeout values. so that it will introduce dynamism in the RTO.
2. Agent based RTO fine tuning with intelligent inference of domain performance with various parameters to add predefined expectation value to RTT and RTO. An expert system which share the values of various parameters of network and appropriate decision in routing and timeout are fixed.



3. Swarm intelligent based re-routing of packet to increase throughout with less time delay. It is proposed that a re-routing policy based on the best match route that takes less web latency to reach servers based on time intensity factor. That is routes are selected rather than calculated based on the intelligent inference made by the agent on various intensity time zone and bandwidth availability etc., on long range training of set of parameters.

8. CONCLUSION AND FUTURE WORK

In this paper it is ascertained that the number of retransmission packet affect the throughput of the real time network. Along with various other parameters which actually support in calculating the web latency delays such as traffic intensity, RTO and sliding window parameters is also analyzed. The future work is to implement and test the methodology using appropriate simulator in various network architecture like wired, wireless and ad-hoc where routing place the vital role in throughput.

REFERENCE

[1] Nandita Dukkipati , Matt Mathis , Yuchung Cheng , Monia Ghobadi, Proportional rate reduction for TCP, Proceedings of the 2011 ACM SIGCOMM conference on Internet measurement conference, November 02-04,

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

2011, Berlin, Germany doi>10.1145/2068816.2068832

[2] P. Papadimitriou and V. Tsaoussidis, "Performance Evaluation of Real-Time Transport with Link-Layer Retransmissions in Wired/Wireless Networks", Journal of Mobile Multimedia, 2(4), December 2006, pp. 327-343.

[3] Honda, Osamu, et al. "Understanding TCP over TCP: effects of TCP tunnelling on end-to-end throughput and latency." *Optics East 2005*. International Society for Optics and Photonics, 2005 http://dx.doi.org/10.1117/12.630496

[4] Liangping Ma; Barner, K.E.; Arce, G.R., "Statistical analysis of TCP's retransmission timeout algorithm," *Networking, IEEE/ACM Transactions on* , vol.14, no.2, pp.383,396, April 2006,doi: 10.1109/TNET.2006.872577

[5] Mark Allman , Wesley M. Eddy , Shawn Ostermann, Estimating loss rates with TCP, ACM SIGMETRICS Performance Evaluation Review, v.31 n.3, p.12-24, December 2003 [doi>10.1145/974036.974038].

[6] Steven Roman, 1999, Win32 API Programming with Visual Basic O'Reilly Media

[7] Network Protocol Analyzer. http://www.wireshark.org – Wireshark

[8] Choudhury, Sayantan, et al. "Effect of payload length variation and retransmissions on multimedia in 802.11 a WLANs." *Proceedings of the 2006 international conference on Wireless communications and mobile computing,* ACM,2006,doi>10.1145/1143549.1143625

[9] Ramasubramanian, Venugopalan, Dahlia Malkhi, Fabian Kuhn, Mahesh Balakrishnan, Archit Gupta, and Aditya Akella. "On the treeness of internet latency and bandwidth." In *ACM SIGMETRICS Performance Evaluation Review*, vol. 37, no. 1, pp. 61-72. ACM, 2009.

[10] Flach, Tobias, Nandita Dukkipati, Andreas Terzis, Barath Raghavan, Neal Cardwell, Yuchung Cheng, Ankur Jain, Shuai Hao, Ethan Katz-Bassett, and Ramesh Govindan. "Reducing web latency: the virtue of gentle aggression." In *Proceedings of the ACM SIGCOMM 2013 conference on SIGCOMM*, pp. 159-170. ACM, 2013

[11] Loukili, Alae, Alexander L. Wijesinha, Ramesh K. Karne, and Anthony K. Tsetse. "TCP's Retransmission Timer and the Minimum RTO." In *Computer Communications and Networks (ICCCN), 2012 21st International Conference on*, pp. 1-5. IEEE, 2012.