Measurement and Evaluation of Congestion in The Internet Protocol of The Channel Network

G.A. Gbotoso, L.A.Akinyemi, O.A Ajayi, A.O Ademodi, O.O Green

Abstract: The introduction of internet protocol (IP) services in multimedia applications on the internet had affected the Quality of service (QoS) efficiency in one way or the other which has resulted in the requirement of more bandwidth. However, exceptional service enjoyed on the internet had come with its challenges, purely the congestion of the internet as a result of growth in number of users. Congestion Management in the network services becomes one of the main problem in achieving high performance and good quality of internet services which also has resulted to managing bandwidth. This research therefore, used the ping plotter software to monitor the network in order to observe the delay on the network. After studying the network for a period of time without increasing bandwidth, the network administrator can make the right decision on how to achieve high performance based on the throughput.

Keywords: Bandwidth, Congestion, Internet, Internet Protocol (IP), Throughput.

I. INTRODUCTION

Some decades back, the word information technology was not pronounced until recent times. The breakthrough in information technology has made the world a global village where information is being sourced at some seconds' click of a computer keyboard. This breakthrough however has led to a lot of delay in the internet network due to congestion. Congestion control is one of the technique to maximize available bandwidth due to adverse effect of congestion arising from excessive demand and long queues at the server leading to slow upload and download on the internet network. Congestion is the state of sustained network overload where the demand for network resources is close to or exceeds capacity, Panos et al.(2001). Due to delay observed in the internet network services, several works had been done in the past to control congestion such as the use of algorithms to control congestion; running of parallel network infrastructure; the use of acceptance policy by deploying the application of many tools encompassing different techniques.

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Like every other project, a network projects must have a proper design for the network to survive expansion after deployment. According to ANAND (2005), good networks do not happen by accident rather good networks are the result of hard work by network designers and technicians, who identify network requirements and select the best solutions to meet the needs of a business.Nevertheless with all these methods users have not been satisfied with the usage of all these mechanisms highlighted above, thus prompting this research work. In this work, the use of a very powerful tool that can monitor the network server and calculate the round trip time of the target server at different times to enable the network administrator to ascertain the throughput on the network in order to be able to manage the bandwidth and at the same time give basic background of the cause of the delay to the target server while monitoring the network is employed. Bandwidth may be characterized as network bandwidth, data bandwidth or digital bandwidth. Behrouz (2012) defined bandwidth as a range within a band of frequencies or wavelengths while Devajitet al. (2013) defined bandwidth in computer networking as, a reference to the data rate supported by a network connection or interface. This tool is a software called the ping plotter pro that can monitor the links to the server and also give the end user the ability to trigger the software to send mail to the internet service provider (ISP) when delay is observed in the links and at the same time to report to the ISP some causes of the congestion or delay in the network.

II. REVIEW OF RELEVANT LITERATURE

Jacobson and Karels (1987) developed congestion avoidance algorithms to control the congestion on the internet network in order to manage traffic congestion on the network. These algorithms became mandatory requirements for all internet hosts. These control mechanism owes its success to its popularity back then. The limitation of the authors technique is the obvious users unfairness in the allocation of bandwidth .Greedy users have more than their fair share and thereby degrade the quality of service of either users.

Errin W. F and Douglass S. Reeves (2004) describes a scalable connection management strategy for quality of service enabled networks. The management technique maximizes profit, while reducing blocking experienced by users. Important issues regarding demand estimation, connection duration, and pricing intervals, are addressed and analysed. While this research work provided important insight into provisioning and peering, it did not address resource allocation (pricing).

Sara gywnn (2013) stated categorically that most research centers and educational institutions in Africa and the developing world are not managed at all thereby causing network failures and sometimes the extinction of such networks.

Avister (2009) also proved that although most people assume that internet congestions is only on the link to the internet, but congestion is mainly in the incoming direction. Odlyzko (1999) proposed the running of parallel network infrastructure for different classes of service with higher prices in order to use better provisional and less congested classes. With this method, pricing will be the primary control tool for achieving quality of service in a congested network in a best effort network, in other words users need a cost effective control mechanism to achieve good quality of service to achieve efficient network.

Reininger (1998) proposed that to dynamically reduce internet congestion, for a better quality of service on the network channel, he construct utility functions from experimental data based on the 5-level mean-opinion-score (MOS) test for subjective video quality (because it was reported that most of the bandwidth is used for video services which causes delay on the network. However, this reported work is based on an off-line process for subjective quality testing. Even though there have been some recent developments in objective measurement techniques that model the human visual system, these schemes are computationally intensive and involve decoding and buffering delay.

Lockias Chitanana (2012) at Midlands State University, Zimbabwe carried out research on five universities in Zimbabwe on what they were doing to manage their bandwidth in order to reduce congestion on the internet so as to have effective internet network service. The results show that most of the universities did not have an official Acceptance Use Policy (AUP) to assist in reducing internet congestion. The author proposed the use of application of many tools encompassing a number of different techniques to manage network within a university. These products are often expensive and are rarely available.

This research thesis will deploy the use of a technique that can be used to further enhance the above work that has been highlighted above regardless of the existing network configuration and installed devices. The research thesis will involve the use of the round trip times calculated by the software at different times and use it to evaluate the throughput in each case. All the review above has not looked into the throughput of the channel.

III. METHODOLOGY

The primary research was carried out using the ping plotter pro to analyze the search engine server -google.com and university server-Unilagspgs.edu.ng. The two servers were tested for internet congestion by sending out packets of data from the remote system. The behaviour of the network was observed for every sixty seconds interval. This method was chosen to measure the round trip times required to calculate the throughput of the channel through which message can be sent and to observe congestion for each location of the

service provided at that particular time. The process of carrying out the observation is as follows:

The location of the packets had to be decided, two different locations were chosen to ensure that the packets travelled over different networks. The locations are search engine server (Google) and university server. The packet size was set to 56bytes. A packet size of 56bytes was chosen just to realize a miniaturise set of goals and to test the connectivity of the target server. Although this packet size is small in real world. However, users upload packets with larger size, without loss of generality we choose 56bytes for simplicity sake.

The packet was traced, noting the names and the number of the many hops through which it had travelled. The IP addresses and the Domain naming system (DNS) names that were crossed were noted.

The average of the return Round Trip (RRT) times was displayed, on the ping plotter screen, as well as the current minimum and maximum RRT time of the sample set.

The time taken at each hop was also recorded with a figure showing these times, in relation to the next hop.

A time line Figure was shown for the each sample set and for the target hops which exhibits signs of congestion or bottleneck. The time line figure has been set to show the graph of current round trip times related to the duration of observation for each sample set for the target server

The information received from the ping plotter was used to realize the throughput of the channel, the graphical relationship displayed using excel and using the following expression ;

$$ms = ps * 8 \text{ bit} \qquad (1)$$
$$ms = 2 * B * D_t \qquad (2)$$

Where ms is the message size and Dt is the delay time in seconds which is also equivalent to round trip time. B is the bandwidth in bits per seconds and ms is the message size.

$$T = \frac{ms}{RRT} \qquad (3)$$

Where T is the throughput and RRT is the round trip time. However,

$$D_t = RRT \qquad (4)$$

IV. RESULTS

This project was aimed at monitoring the latency or delay that occurred at a particular hop or server using ping plotter software to monitor the network congestion over a period of time using a 3.5G network provided by SWIFT 4G Network with a bandwidth of 3Mbps.Swift network is a high speed broadband internet service operator. They offer state-of-the-art broadband services. Swift networks holds an exclusive wireless spectrum license from the Nigerian communications Commission (NCC), it operate an end-to-end reliable, fibre-like connectivity services in the exclusively licensed and interference-free 3.5GHz spectrum. Based on this, the network has strong signal strength, reason it was used for this research work.

The ping plotter packet size was set to 56byte and this software monitors the hop in question which is always the last hop in the trace graph; by displaying trace and the timeline graphs for that hop. Two servers were selected for the trace and there hops captured in table 1.

Table 1.0 Number of Hops to Destinations

Destination Server	Number of hops
Unilagspgs.edu.ng	22
Google.com	11

The result displayed for university server were four (4) different samples at sixty (60) seconds trace interval and also 4 different samples were displayed for the search engine server at sixty (60) seconds trace interval as was selected for this research work.

A. Results Obtained

University Server Results Time of trace for the university server is between 2.07.03 a.m and 2.10.03 a.m

For the university server sample set was as follows
Number of times to trace = 4
Trace Interval = 60 seconds
Number of Samples to include = 4
It should be noted that ,

$$ART = \frac{M_aRT}{M_iRT} \quad (5)$$

Where,

ART = Average Response time

M_aRT = Maximum Response Time

M_iRT = Minimum Response Time

Warning speed was set between 201ms-500ms
Critical speed was set at 500ms upward
It should be noted that,

$$PL(\%) = \frac{NSL}{TNSD} \quad (6)$$

Where, PL(%) = Packet Loss in %

NSL = Number of Samples Lost

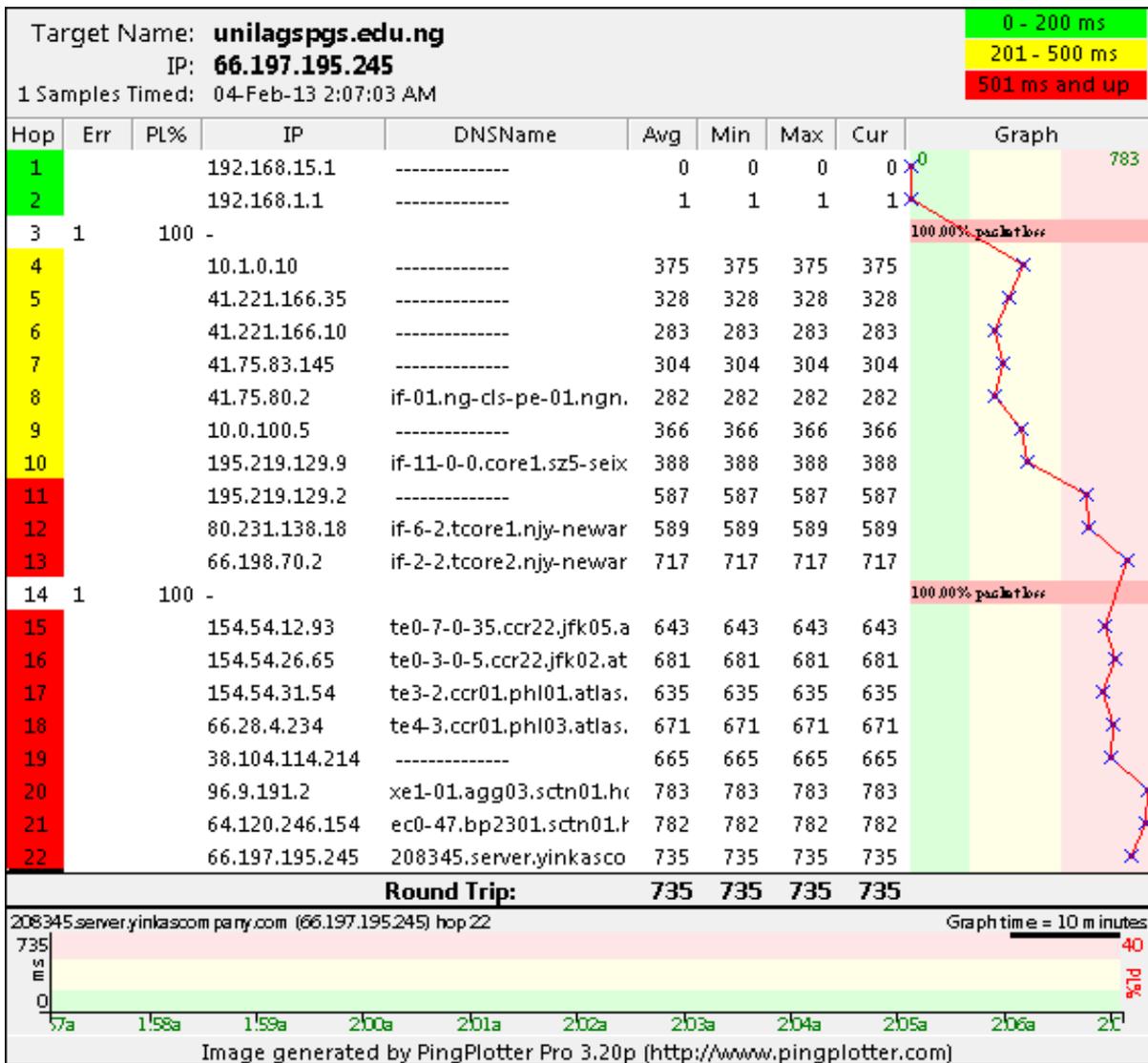TNSD = Total Number of Samples Displayed

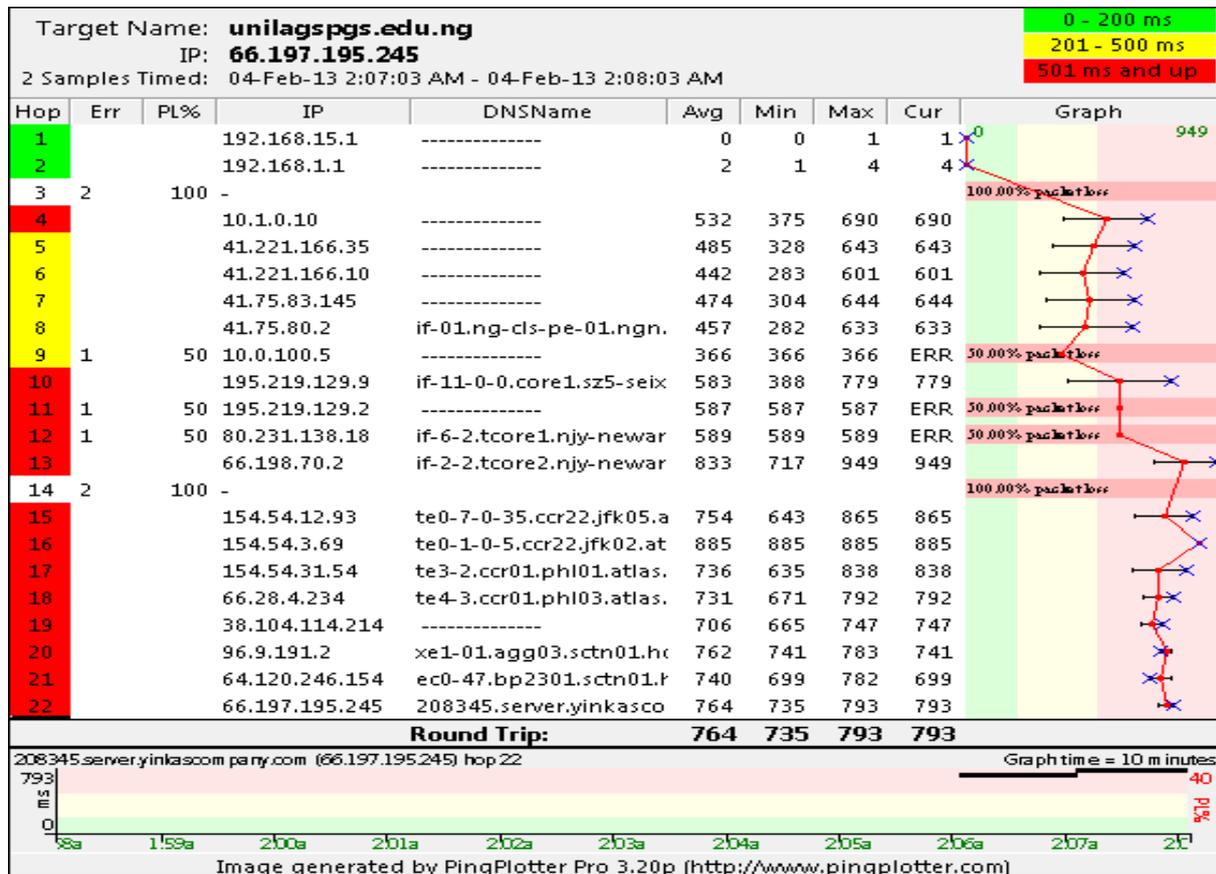**Fig 1 First Sample Display For unilagspgs.edu.ng**

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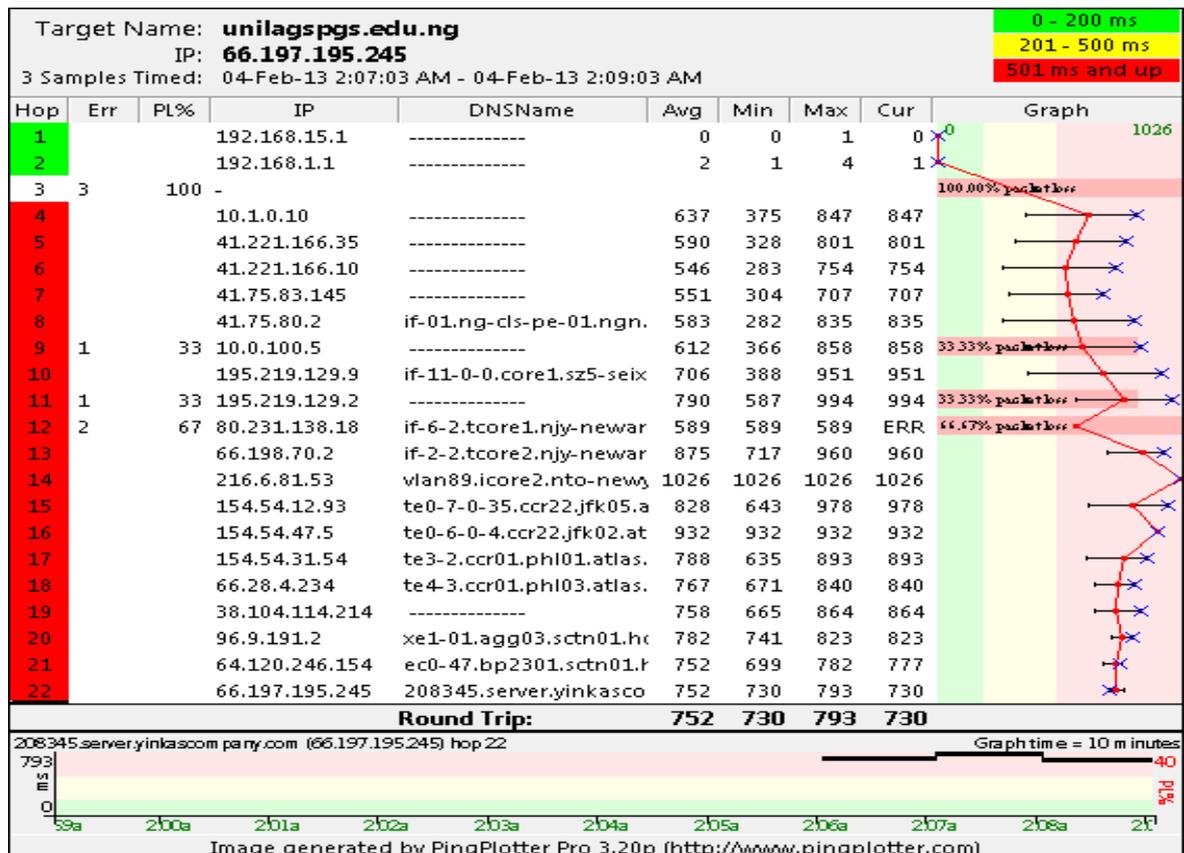**Fig 2 Second Sample Display For unilagspgs.edu.ng**

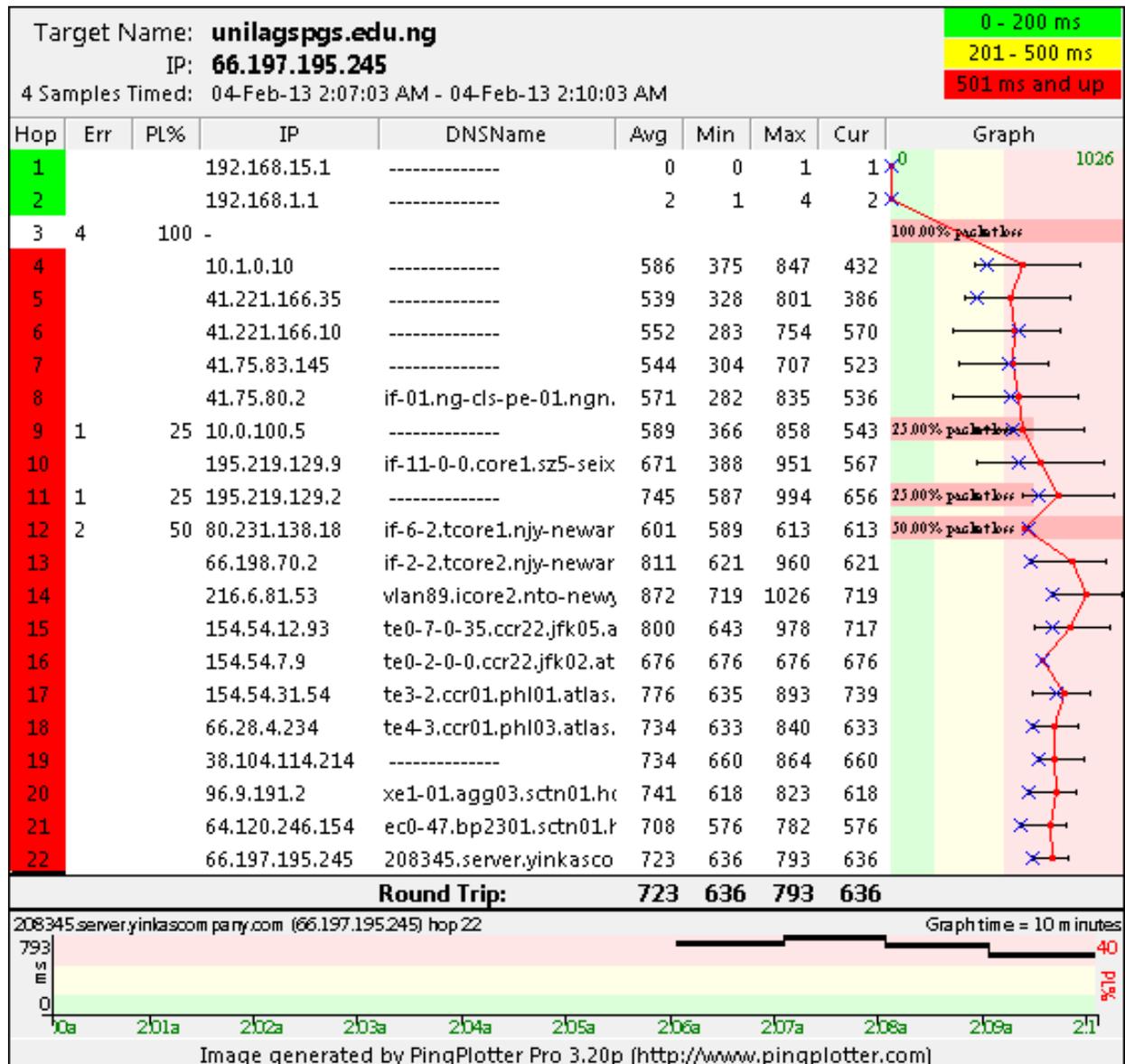**Fig 3 Third Sample Display For unilagspgs.edu.ng**

Fig 4 Fourth Sample Display For unilagspgs.edu.ng

B. Analysis of the Results Obtained for the University Server

In this work, the current round trip times not the maximum and minimum nor the average current times will be used as these are for further analysis in future.

The round trip times of the target server from the result displayed from figures 1 through 4 are 735ms, 793ms, 730ms and 636ms.

With message size = 56bytes = 56 ×8 =448bits, the throughput values can be realized from equation (3);

With RRT of 735ms
 Throughput = 0.6kbps
 RRT of 793ms
Throughput =0.56kbps

With the other round trip times of 730ms and 636ms, the throughput values are 0.61kbps and 0.71kbps.The graphical relationship between throughput and return round trip time will be displayed using excel.
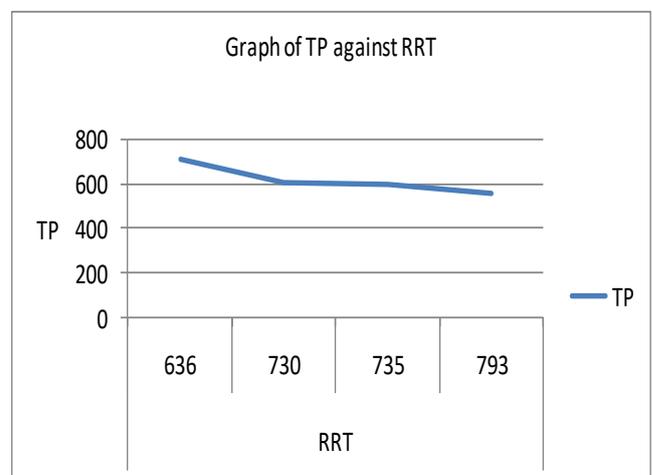

Fig 5 Graphical relationship between Throughput (in bps) and Return Round trip (in milliseconds) for unilagsps.edu.ng

Where TP means throughput.

C. Search Engine Server Results

For the search engine server:

Time of trace for the search engine server is between 2.07.03 a.m and 2.10.03

For the search engine server sample set was as follows:

Number of times to trace = 4

Trace Interval = 60 seconds

Number of Samples to include = 4

It should be noted that Average Response time = (Max response time +Min Response Time)/2

Warning speed was set between 201ms-500ms

Critical speed was set at 500ms upward

It should be noted that the equation for packet loss(%) in this case is the same as equation (6).
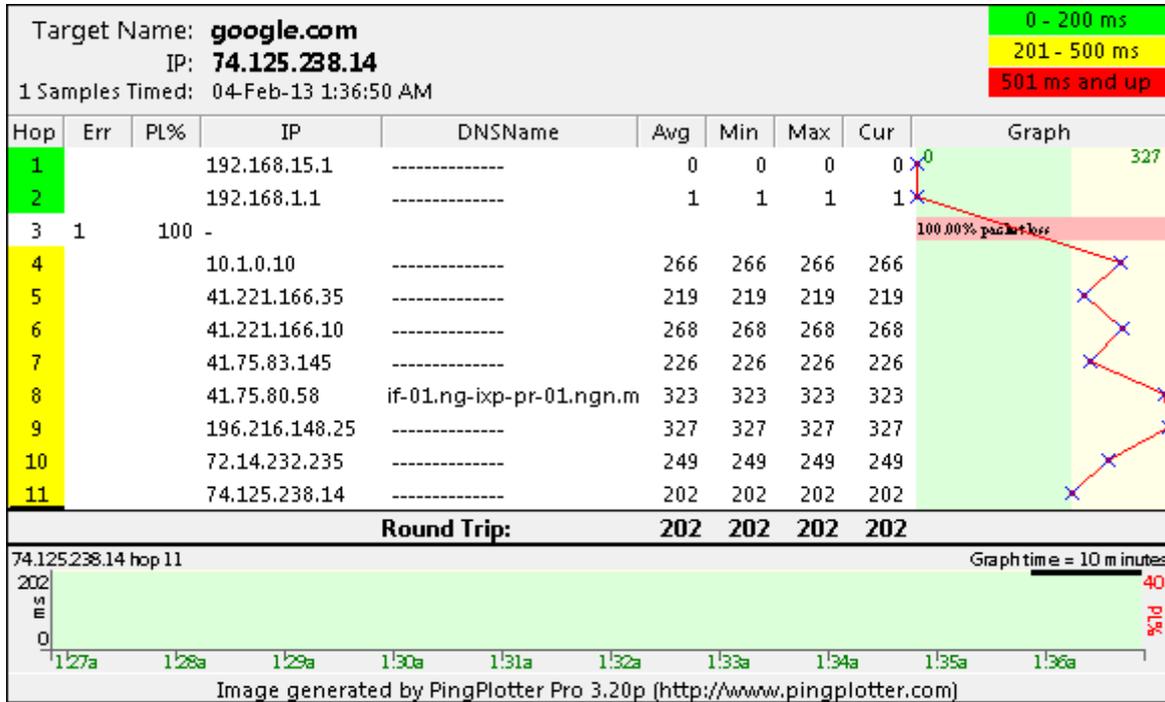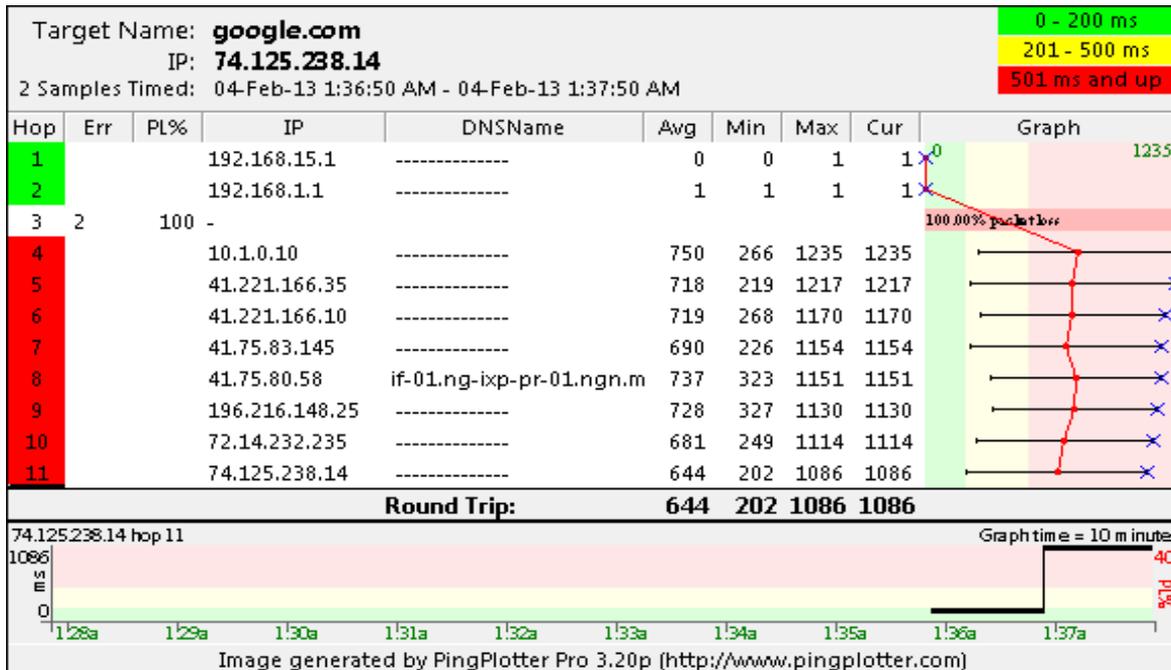

Fig 6 First Sample Display For google.com
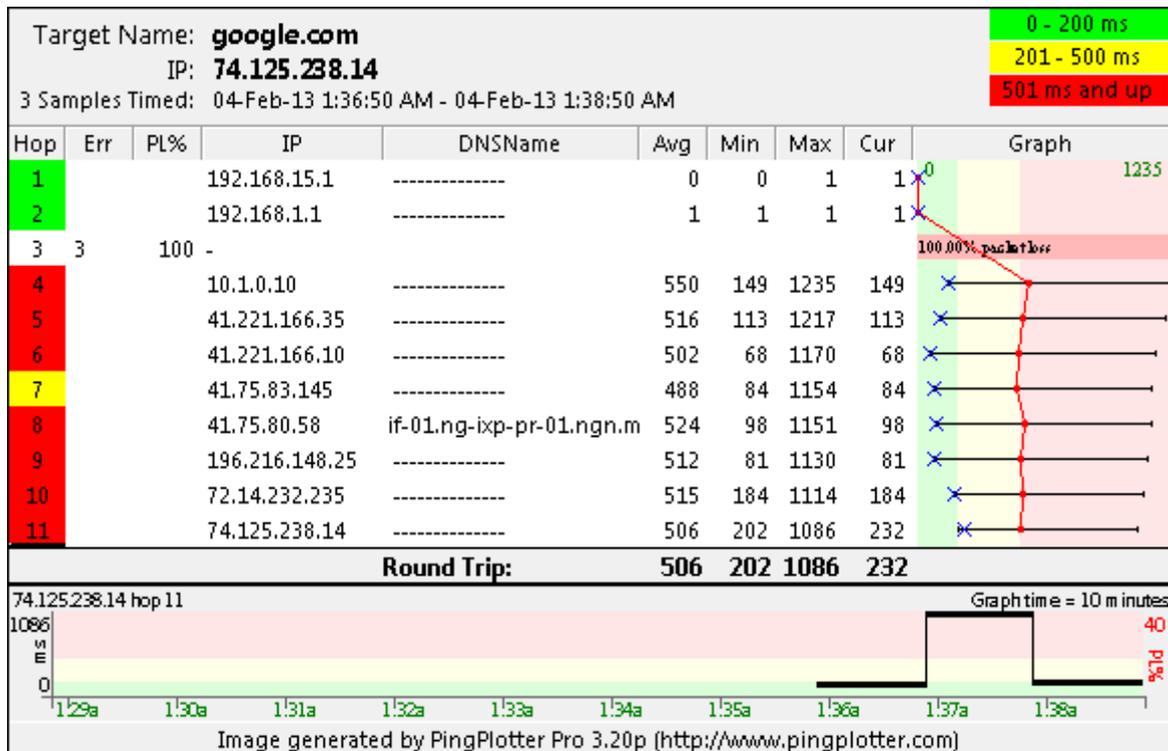

Fig 7 Second Sample Display For google.com

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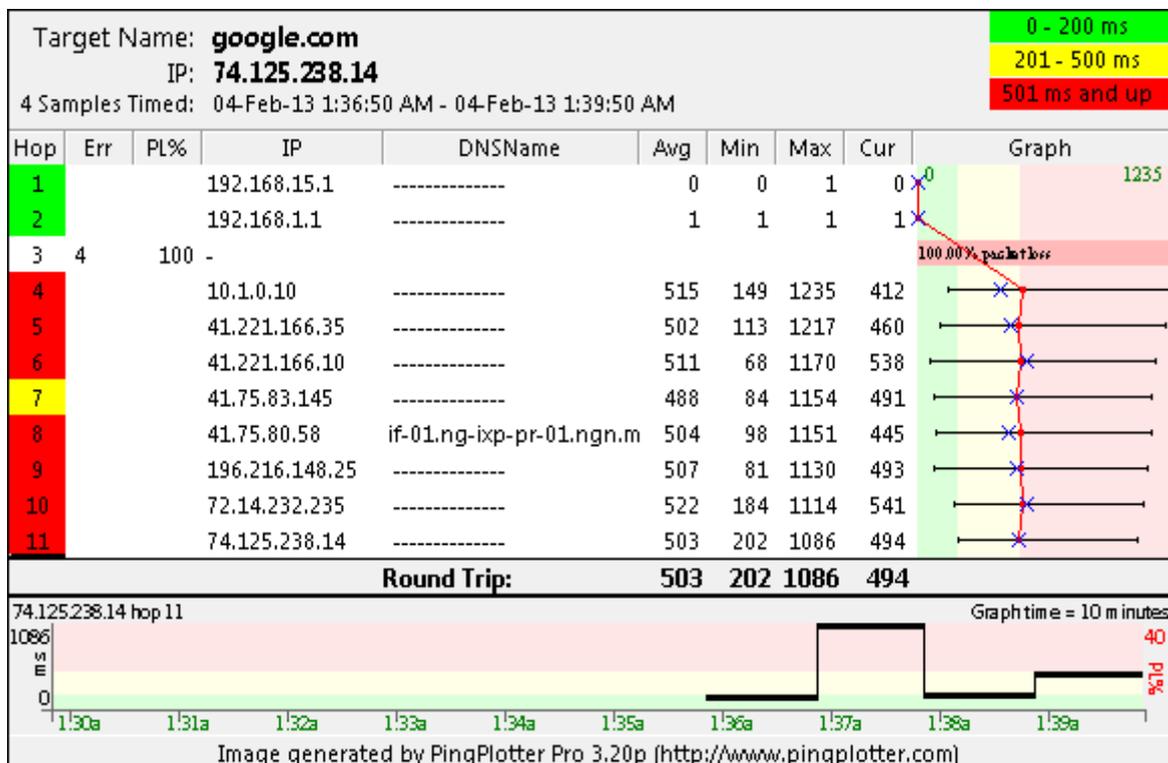**Fig 8 Third Sample Display For google.com**

Fig 9 Fourth Sample Display For google.com

D. Analysis of the Results Obtained for the Search Engine Server

We are only going to use the current round trip times not the maximum and minimum nor the average current times will be used as this are for further analysis in future.

The round trip times of the target server from the result displayed are 202ms, 1086ms, 232ms and 494ms.

With message size = 56bytes = 448bits form equation(1), the throughput values can be realized from equation (3);

With RRT = 202ms, throughput = 2217bps = 2.2kbps,
With RRT = 1086 ms, throughput = 413 bps= 0.413kbps
With RRT = 232ms, throughput = 1931bps =1.9kbps
With RRT = 494 ms, throughput = 907 bps =0.91kbps
The graphical relationship between throughput and return round trip time for the search engine server will be displayed using excel.
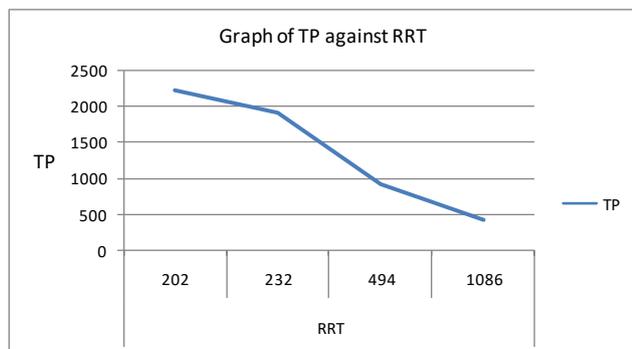
Fig.10 Graphical relationship between Throughput (in bps) and Return Round trip (in milliseconds) for google.com

V. DISCUSSION OF THE RESULTS

From above result of the graphical relationships in fig.5 and fig.10, it was observed that the higher the rounds trip times the lesser the throughput of the channel and vice versa. We can observe the deficiency in throughput with greater RRT but with lesser RRT the throughput value was very high which is a very good value and this is true for both target servers. Even if the message size is increased the round trip time will vary too. But it is pertinent that if the delay time is reduced by working on the hops(through the ISP) with high RRT(with packet loss) then the RRT of the target server can be reduced, which will result in high throughput and better utilization of the bandwidth.

It was observed from the ping plotter results that for the four set of samples displayed for the university server engine server- hop number 3 was always having 100% packet loss meaning that packet was sent to hop 3 but the hop is not responding. This could be as a result of hardware problem. Also hop number 14 had 100% packet loss at first and second sample sets, making the round trip times high at these sample sets , but later responded for the remaining sample sets this might due to cable link between hop 13 and 14 shaking or component instability in hop 14.Also hop numbers 9,11, and 12 were having some packet loss at the second, third and fourth sample sets this might due to congestion at this hops, but the amazing part is the way the current response time was reduced at the last sample set meaning that in brief when the packet losses is not much then the latency or delay in network will be reduced. This result can then be sent to the Internet Service Provider (ISP) or the network administrator so that they can easily diagnose which hop is giving the network problem and can be resolved. From the result obtained from the ping plotter for search engine server, it could be observed that from the first sample display hop number 1 was not responding which is normal in trace utility graph, but hop number 3 is having 100% packet loss meaning that hop number 3 is not responding to the client at all even throughout the sample displays, that is for both second, third and fourth displays. Then hop number 3 will be causing much traffic delay as expected from the result. It was also observed that this hop had the same result in the result of the university server meaning that this hop should be diagnosed properly or should be replaced, should be reported to the ISP provider. We could see that with this method we can reduce

congestion by diagnosing the source of the problem, which involves reporting to the ISP and also troubleshooting from the end-user point. It should be noted that if the packet size was increased the bandwidth will increase but cannot exceed the maximum allocated by the ISP, but with more packet size also there might be more delay times depending on the number of users on the network because if more users are connected through same channel then the bandwidth will be shared resulting in some congestion.

VI. CONCLUSION

This paper has been able to achieve the set out goals by using powerful software called the **Ping Plotter Pro**. The objective of this paper was to analyze congestion on the internet and cause of the problem using two servers by calculating the throughput values, which are the search engine server and the university server, the results were analyzed.

From the results, it is apparent that the delays in traffic and congestion were at the routers. There is a problem referred to as 'over blocking' .It occurs when ISPs rent less bandwidth than their subscribers required in the case of large upload or download of data, if they were all to download files at the same time, resources are overstretched and this also causes congestion at the routers even at the target server, causes the target hop to be over stretched. An interesting and important finding from the research was the fact that all the packets that were sent out followed that same route each time. Even if there had been delays at particular routers on the path to a destination, the route to the destination rarely varies. This was a surprise as it is known that dynamic routing exist between routers and that their neighbour is functioning, and that this builds routing tables from this information, avoiding routers which cause delays. Quality of service can be achieved if proper analysis is done on the hops (routers) either by replacing them or by troubleshooting them. It can be concluded that congestion or delays in the network is mainly from the routers or ISP services when bandwidth is totally saturated and not majorly from software point of view.

REFERENCES

1. Errin ,W. F., and Douglas, S. R., Bandwidth Provisioning and Pricing for Networks with Multiple Classes of Service, Computer Network Journal, Elsevier , Vol.46(1),41-52.Sept, 2004.
2. Lockias, C., Bandwidth management in Universities in Zimbabwe: Towards a responsible user base through effective policy implementationInternational. Journal of Education and Development using Information and Communication Technology (IJEDICT),Vol.8(2), 62-76.Aug,2012.
3. Sara, G., IAP Workshop. Promoting access and capacity building for scientific information resources, 30-32, Darker, Senegal, Jan,2013.
4. BehrouzA, F., Data communications and networking 5th Ed., Introduction Global Edition, McGraw Hill Education, Europe (2012).
5. Devajit, M., Majidul, A., Utpal J., and Bora ., A study of bandwidth management in computer networks. International Journal of Innovative Technology and Exploring Engineering, Vol. 2(2), 69 -73.Jan, 2013.
6. Panos, G., Jon C., Peter, K., and Saleem, B.(2001). Congestion Control Mechanisms and the Best Effort Service Model, IEEE Network. University College London.16-26, May, 2001.

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7. Avistar Communications Corp. (2009).Product approach document bandwidth management, pp. 2-10, 2009.
8. ANAND. Wireless Hotspots: Current challenges and future directions, mobile networks and applications,10, pp 265–274, 2005.
9. Odlyzko, A.M., Paris Metro Pricing for the Internet ,Proc. ACM Conf. Ecommerce, Vol.2(1), pp 4-10,1999.
10. Reininger. D., Dynamic Quality-of-Service Framework for Video in Broadband Networks, PhD thesis, Rutgers University, New Jersey, 1998.
11. Floyd, S. and Jacobson, V., Congestion control mechanism and best effort service model, IEEE/ACM Trans. Net., vol. 3(4), pp 314-329, 1995.