

Latency in the Http Protocol and its Measurement and Analysis in the Channel Network

Vikas Tripathi¹, Durgaprasad Gangodkar², Devesh Pratap Singh³, Dibyahash Bordoloi⁴

¹Department of Computer Science & Engineering, Graphic Era Deemed to be University, Dehradun, Uttarakhand India, 248002

²Department of Computer Science & Engineering, Graphic Era Deemed to be University, Dehradun, Uttarakhand India, 248002

³Department of Computer Science & Engineering, Graphic Era Deemed to be University, Dehradun, Uttarakhand India, 248002

⁴Department of Computer Science & Engineering, Graphic Era Hill University, Dehradun, Uttarakhand India, 248002

ABSTRACT

The increased need for bandwidth may be traced back to the emergence of internet protocol (IP) services in multimedia applications on the internet, which in turn impacted QoS efficiency. The internet's great service, however, hasn't come without its drawbacks, most notably the congestion that has arisen as a direct consequence of the proliferation of online users. Congestion Controlling bandwidth has emerged from the increasing complexity of managing network services, which in turn has hampered the delivery of high-quality internet services. Ping plotter was consequently utilised to track the network's latency as part of this study. The administrator may make an informed judgement on how to attain high performance based on throughput after observing the network for a while without boosting bandwidth

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

INTRODUCTION

Before recently, the term "information technology" was not often used. Information can now be found all around the globe in a matter of seconds thanks to advancements in computer technology. This development, however, has resulted in significant internet latency as a result of congestion. Due to the negative impact of congestion resulting from high demand and lengthy lineups at the server, resulting in poor upload and download speeds on the internet network, congestion management is one way to optimise available bandwidth. According to Panos et al., congestion occurs when the constant demand for network resources is equal to or greater than the capacity of the network[1].

Several studies have been done in the past to manage congestion because of delays in internet network services. These include the use of algorithms to control congestion, the operating of parallel network infrastructure, and the use of an acceptance policy. An appropriate design is essential for

the long-term viability of a network after deployment, just as it is for any other kind of project. Successful networks are the product of careful planning by network designers and technicians who assess a company's needs, determine which technologies would best suit those needs, and then implement those technologies[2]. However, despite the aforementioned mechanisms, users have not been completely satisfied, which prompted this study. In this study, a highly effective tool is used to monitor the network server and calculate the round trip time of the target server at different times, allowing the network administrator to determine the throughput on the network and subsequently manage the bandwidth, as well as providing some contextual information regarding the root cause of the delay experienced by the target server. It's possible to talk about bandwidth in terms of network bandwidth, data bandwidth, or even just plain old digital bandwidth. Bandwidth is the range of frequencies or wavelengths inside a band, bandwidth is the maximum data rate that can be sent across a network connection or interface. The ping plotter pro programme is useful for keeping tabs on how quickly data is being sent to and from a server, and it also allows users to notify their ISPs through email if they see any signs of network congestion or slowdown.

REVIEW OF RELEVANT LITERATURE

In order to manage traffic congestion on the internet, created congestion avoidance algorithms. All web servers are now required to implement these algorithms. The widespread adoption of such controls at the time was crucial to their ultimate success. It's clear that certain customers are being treated unfairly when it comes to bandwidth allotment, which is a major drawback of the authors' method. Unfair users are taking advantage of the system and lowering the standard of service for everyone else. This method of management increases earnings while decreasing user frustration with slowdowns. Substantial problems with demand forecasting, connection times, and price breaks are discussed and analysed. However, resource allocation was not addressed in this study despite the fact that it gave key insights on provisioning and peering (pricing).

[3] said emphatically that the lack of management at most research centres and educational institutions in Africa and the developing world is the primary cause of network failures and, in some cases, the extinction of such networks.

[4] further shown that, contrary to popular belief, internet congestion occurs mostly in the outbound direction. In order to take advantage of better provisional and less congested classes,

[5] proposed running parallel network infrastructure for different classes of service at higher prices. Users require a cost-effective control mechanism to achieve good quality of service in order to achieve efficient network performance given that pricing will be the primary control tool for achieving quality of service in a congested network in a best-effort network.

Since it has been reported that most of the bandwidth is used for video services, causing delay on the network,

[6] proposed that in order 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. However, the quality assessment in this

study was performed offline. Objective measuring methods that mimic the human visual system have advanced in recent years, but they still need a lot of processing power, have a significant learning curve, and cause delays due to decoding and buffering.

[7] of Midlands State University, Zimbabwe, found that five institutions in that country were taking steps to control their bandwidth and decrease internet congestion in order to provide efficient internet network service. From what we can see, most educational institutions did not have a formal Acceptable Use Policy (AUP) in place to help them cut down on their internet bandwidth. Here's how we'll go about doing the observation:

The location of the packets had to be selected, two separate sites were picked to guarantee that the packets passed across different networks. Google's servers and those at universities are the sites in question [8]. The packet size was set to 56bytes. The packet size of 56 bytes was selected only for the purpose of verifying the destination server's connection and realising a more compact set of objectives. In the actual world, however, this package size is rather little. However, users upload packets with bigger size, without loss of generality we chose 56bytes for simplicity reason [9].

The path of the packet was tracked, and the names and numbers of each intermediate node were recorded. Cross-references between IP addresses and DNS names were recorded.

The ping plotter screen showed not only the lowest and maximum RRT timings, but also the average RRT time.

Each hop's duration was recorded, and a chart depicting this duration's relationship to the following jump was also created. Congestion and bottleneck indicators were shown in a timeline figure for both sample sets and their respective target hops. Target server round-trip times as a function of observation time have been shown graphically [10].

administer a university's internal network. These items are notoriously hard to come by and prohibitively pricey. This research thesis will implement a method that may be utilised to improve the aforementioned work irrespective of the network configuration or devices already in place. The software's computed round trip durations at various periods will be used to assess throughput for the research thesis. Nothing in the aforementioned analysis has considered the channel's throughput.

METHODOLOGY

Using ping plotter pro, we analysed the performance of the google.com and Unilagspgs.edu.ng servers as part of our main study. By transmitting data packets from a distant system, we were able to test whether or not the two servers were experiencing internet congestion. Every sixty seconds, the network's activity was monitored. Where T is the throughput and RRT is the round trip time, this technique was selected to measure the round trip times necessary to compute the throughput of the channel via which messages may be carried and to notice congestion for each point of the network. But $Dt = RRT$ (4) XVIII.

RESULTS

The purpose of this research, which used a 3.5G network from SWIFT 4G Network with a bandwidth of 3Mbps, was to track the latency or delay at a certain hop or server over time using ping plotter software. Swift network is an ISP that offers very fast broadband connections. Provides cutting-edge broadband options. Swift Networks operates in Nigeria's licenced and interference-free 3.5GHz spectrum thanks to its unique wireless spectrum licence from the Nigerian Communications Commission (NCC). It was determined that the network's signal strength was sufficient for the study project at hand. Ping plotter packet size was set to 56 bytes, and this app keeps tabs on the target hop, which is always the last hop in the trace graph, by showing trace and timeline graphs for it. To do the trace, we chose two servers and recorded their hops.

Here's what our example set looked like from the perspective of a university server: Trace it four times:

60 Seconds Between Traces. To be sampled, we need 4 In this regard, it's important to remember that, The Outcomes of the University Server's Analysis

For the sake of this study, just the current round trip timings will be utilised; the maximum and lowest, as well as the average, will be saved for future research. The round-trip timings to the target server are 735ms, 793ms, 730ms, and 636ms, respectively. Throughput (in kilobytes per second) for a round-trip time of 735 milliseconds is 0.6 kbps.

The other two RT values, 730ms and 636ms, result in throughputs of 0.61kbps and 0.71kbps, respectively. In Excel, we'll plot a graphical representation of the correlation between throughput and round-trip delay.

Results and Discussion

According to the findings shown in figures 5 and 10, it is clear that the channel's throughput decreases with increasing round-trip times. The throughput drops down noticeably with higher RRT, but with lower RRT, it soars to excellent levels on both target servers. The round-trip time will change regardless of the message size. However, it is important to note that if the RRT of the destination server can be decreased by working on the hops (via the ISP) with high RRT (with packet loss), then the throughput will increase and the bandwidth will be used more efficiently.

Ping plotter data for the university server engine server showed that hop number 3 consistently had 100% packet loss, indicating that packets were being transmitted to the hop but not being responded to. It's possible that this is the consequence of a malfunctioning piece of hardware. Also, sample sets 1 and 2 experienced 100% packet loss from hop 14, which increased round trip timings, but for the other sample sets, hop 14 replied. This might have been because of a cable connection between hops 13 and 14. Hop 14 is trembling or has unstable parts. Even though congestion at hops 9, 11, and 12 caused packet loss in the second, third, and fourth sample sets, the remarkable thing is how the current response time was reduced in the fifth and final sample set, which means that in brief when the packet losses is not much then the latency or delay in the network will be reduced. Once the ISP or network administrator has this information, they may quickly determine which hop is causing the issue and fix it. Ping plotter results for a search engine server showed that hop 1 wasn't responding

in the first sample display, which is typical in a tracing utility graph, but hop 3 was experiencing 100% packet loss in all four sample displays. This means that hop 3 was never responding to the client. As a consequence, we may assume that the third hop will be a major bottleneck. Because this hop had the same results as the university server, it should be properly diagnosed, replaced, or reported to the ISP. Through reporting to the ISP and doing some debugging on the user's end, we may determine the root cause of the congestion and take steps to alleviate it. However, if additional users are connected over the same channel, the bandwidth will be shared, resulting in some congestion, even if the packet size is raised up to the limit set by the ISP.

CONCLUSION

Ping Plotter Pro, a robust piece of software, has helped this study accomplish its aims. In this research, we calculated the throughput values of two servers—a search engine server and a university server—and then evaluated the findings to better understand internet congestion and its root causes. Based on the findings, it is clear that congestion and traffic delays originated at several routers. An issue known as "over blocking" exists. If all of the subscribers tried to download files at once, resources would be overstretched and congestion would occur at the routers all the way to the target server, causing the target hop to be overstretched. This is what happens when ISPs rent less bandwidth than their subscribers require in the case of large upload or download of data. The fact that every single one of the outgoing packets took the exact same path each time is a fascinating and significant discovery made by the researchers.

As a result, the route to a destination seldom changes, even if there were delays at some routers along the way. This came as a shock since it is common knowledge that routers utilise neighbour and router status information to construct routing tables that circumvent slow routers. If the hops (routers) are properly analysed, either via replacement or troubleshooting, service quality may be restored. When bandwidth is completely at capacity, it is clear that the routers or ISP services are to blame for any delays or congestion in the network, and not the software.

REFERENCES

1. Rath, M., Rout, U. P., Pujari, N., Nanda, S. K., & Panda, S. P. (2017). Congestion control mechanism for real time traffic in mobile adhoc networks. In *Computer communication, networking and internet security* (pp. 149-156). Springer, Singapore.
2. Al-Kashoash, H. A., Kharrufa, H., Al-Nidawi, Y., & Kemp, A. H. (2019). Congestion control in wireless sensor and 6LoWPAN networks: toward the Internet of Things. *Wireless Networks*, 25(8), 4493-4522.
3. Nuha, H. H., & Prabowo, S. (2018, May). Tcp congestion window analysis of twitter with exponential model. In *2018 6th International Conference on Information and Communication Technology (ICoICT)* (pp. 61-65). IEEE.
4. Claypool, M., Chung, J. W., & Li, F. (2018, June). BBR' an implementation of bottleneck bandwidth and round-trip time congestion control for ns-3. In *Proceedings of the 10th Workshop on ns-3* (pp. 1-8).
5. Yan, F. Y., Ma, J., Hill, G. D., Raghavan, D., Wahby, R. S., Levis, P., & Winstein, K. (2018). Pantheon: the training ground for Internet congestion-control research. In *2018 USENIX Annual Technical Conference (USENIX ATC 18)* (pp. 731-743).

6. Al-Kashoash, H. A., Amer, H. M., Mihaylova, L., & Kemp, A. H. (2017). Optimization-based hybrid congestion alleviation for 6LoWPAN networks. *IEEE Internet of Things Journal*, 4(6), 2070-2081.
7. Goyal, P., Alizadeh, M., & Balakrishnan, H. (2017, November). Rethinking congestion control for cellular networks. In *Proceedings of the 16th ACM Workshop on Hot Topics in Networks* (pp. 29-35).
8. Mishra, A., Sun, X., Jain, A., Pande, S., Joshi, R., & Leong, B. (2019). The great internet tcp congestion control census. *Proceedings of the ACM on Measurement and Analysis of Computing Systems*, 3(3), 1-24.
9. Li, W., Zhou, F., Meleis, W., & Chowdhury, K. (2017, May). Dynamic generalization kanerva coding in reinforcement learning for tcp congestion control design. In *Proceedings of the 16th Conference on Autonomous Agents and MultiAgent Systems* (pp. 1598-1600).
10. Pramanik, A., Luhach, A. K., Batra, I., & Singh, U. (2017, March). A systematic survey on congestion mechanisms of CoAP based Internet of Things. In *International Conference on Advanced Informatics for Computing Research* (pp. 306-317). Springer, Singapore.