International Journal for Innovative Engineering and Management Research

PEER REVIEWED OPEN ACCESS INTERNATIONAL JOURNAL

www.ijiemr.org

IJIEMR Transactions, online available on 26th Dec 2022. Link :http://www.ijiemr.org/downloads.php?vol=Volume-11&issue=Issue 12

10.48047/IJIEMR/V11/ISSUE 12/136

TITLE: A STUDY OF RATE-BASED FRAMEWORK FOR CONGESTION CONTROL

Volume 11, ISSUE 12, Pages: 1027-1032

Paper Authors **ASHWIN KUMAR KOTI, DR. APARANA SACHIN PANDEY**

International Journal for Innovative Engineering and Management Research
PEER REVIEWED OPEN ACCESS INTERNATIONAL JOURNAL
www.ijiemr.org

A STUDY OF RATE-BASED FRAMEWORK FOR CONGESTION CONTROL

CANDIDATE NAME- ASHWIN KUMAR KOTI
DESIGNATION- RESEARCH SCHOLAR SUNRISE UNIVERSITY ALWAR
GUIDE NAME- DR. APARANA SACHIN PANDEY
DESIGNATION- SUNRISE UNIVERSITY ALWAR

ABSTRACT

As has been shown in standard protocols and router methods are inadequate when it comes to minimizing congestion, notwithstanding the gains achieved in congestion management. It's possible that rate-based congestion management strategies might help with this. In this study, I briefly explain the problems with conventional congestion management strategies and then outline some potential solutions. In light of these solutions, I offer a framework for congestion management that is built on a ‘sustainable rate' foundation, without relying on traditional paradigms like sliding windows and round-trip timers. Most of the discussed methods for reducing traffic congestion are forms of service reduction, in which transmission rates from various sources are lowered. A different approach is to pinpoint the presence and severity of network congestion; from there, it's up to each source to figure out the best way to react to the observed congestion.

KEYWORDS: Rate-Based Framework, Congestion Control, protocols and router methods

INTRODUCTION

Since end-to-end congestion management techniques must infer network congestion from circumstantial indicators like packet loss and round-trip delay increases, their effectiveness is limited. These assumptions are often incorrect, since packet corruption is only one of many causes of packet loss. It seems that techniques that relay congestion data from routers to traffic generators perform better than end-to-end mechanisms. Unfortunately, the'sliding window' method is often misused. Error recovery, end-toend flow management, and a rudimentary form of congestion control are all handled via sliding windows in protocols like TCP. Therefore, the answer to these three issues is not orthogonal. Bursty source broadcasts may result from rapidly varying window sizes: This may lead to temporary and permanent bottlenecks at routers and destinations. It is problematic because most end-to-end techniques rely on a round-trip timer to detect packet loss. The timeout is often either set too low, resulting in unnecessary packet retransmission, or too high, producing poor throughput. Because of its unreliability, end-to-end assessments of congestion based on packet loss are often inaccurate. The implementation of a round-trip timer also requires significant computer resources. Transport protocols should provide the entire orthogonality of error recovery, flow management, and congestion control to prevent these drawbacks. Congestion management with sliding windows is a bad idea. Packets should be admitted to the network as evenly as feasible, and both flow control and congestion management should make use of true rate control. Short-term router congestion is reduced and traffic bursts are smoothed out as a result.

International Journal for Innovative Engineering and Management Research
PEER REVIEWED OPEN ACCESS INTERNATIONAL JOURNAL
www.ijiemr.org

WEAKNESSES AND SOLUTIONS

1. Mechanisms for End-to-End Control

A real rate control protocol should learn the desired rate from the Network Layer's congestion management method. Selective Acknowledgments and Selective Retransmissions are recommended for error detection. The only packets that need to be resent are the ones that have been lost. A transport protocol should be given the freedom to use as much in-transit traffic as it needs for error recovery, rather than being subject to a sliding window. Only its data transfer capacity, memory, and ready application data may restrict this. Once a source has sent all the traffic it can and is waiting for acknowledgments, only then will a round-trip timer be needed. Good throughput may be ensured by maintaining a rate-controlled transmission regime until the protocol runs out of data. The estimate of the timer's value will not be computationally costly since its value is not mission-critical. In addition, it will be utilized seldom and provide little overhead costs.

2. Methods for Identifying Congestion

Any congestion scheme's end goal must be to decrease network load to an acceptable level. Minimalist methods like Source Quench reveal nothing to the source but the fact that there is congestion in the network. Dec Bit and other protocols are superior to Source Quench because they notify sources when congestion is detected. Again, just the fact that congestion exists is sent back to the original sources. Despite the merit of trying to solve the issues of end-to-end congestion management, schemes like RED can only signal congestion by deleting packets. If at all possible, packets should be reused. If rate control is used by transport protocols for congestion management, the optimal flow rate for each source should be calculated by the congestion scheme. Adjusting the rates of all sources back to their optimal levels could ease traffic congestion.

3. Methods for Routers

Any router's job is to forward traffic while using as little power as possible, as seen in the knee of the power diagram in Figure 2 (page 4). When reaction time is minimal and throughput is high, power is maximized. When network delays are small, such as when router buffers are not overloaded, response times are small. When packets are not dropped and connection utilization is high, throughput is good. When a received packet may be resent instantly without being held in a buffer, power consumption is minimized. Therefore, a router's buffer occupancy should never be more than one packet. Congestion techniques like packet reordering schemes and packet dropping schemes, which rely on large buffer occupancies, are only viable in rare cases when congestion recovery is needed. A well-designed congestion management solution will maintain full network capacity with minimal buffer use.

4. Admission Control for Data Packets

When packets are admitted in bursts, either from a traffic source or an intermediate router, it becomes more challenging to maintain low buffer occupancies in the routers and leads to temporary packet delays. Traffic sources and routers should use smooth packet acceptance strategies like Leaky Bucket to alleviate temporary congestion. Router packet reordering methods that ease traffic admission may

International Journal for Innovative Engineering and Management Research
PEER REVIEWED OPEN ACCESS INTERNATIONAL JOURNAL
www.ijiemr.org

also be used for congestion recovery if necessary

DESIGN CONSIDERATIONS FOR A RATE-BASED FRAMEWORK

The overarching goal of the aforementioned solutions is to collect comprehensive data on network congestion and utilize it to fine-tune transmission rates that maximize the whole network's power. A rate-based model presents itself as the natural starting point for such a congestion management system. These design considerations helped me create a system for rate-based congestion management. The problems I've highlighted with current congestion management methods are remedied by these. Router congestion data should be used for transport congestion management. Congestion may be detected by a source using bit-setting methods, but this does not reveal the severity of the problem. Congestion data should be sent by routers to transport protocols in more than one way. To avoid producing network congestion, it is ideal for routers to provide transport protocols at a pace that is sustainable for each traffic flow.

Dropping packets is highly undesirable

When packets are lost, the router is running in congestion recovery mode, which is an inefficient state of operation. Congestion recovery should be prioritized in any effective congestion management strategy. If routers are reporting congestion conditions to their sources, it is even more critical that packets be avoided wherever feasible. A queue size of more than one in a router is problematic. The average queue length of a router should be 1 packet, as shown by Jain. End-to-end delays and round-trip durations increase when the queue length increases beyond 1. The length of the line also has a role in the variability of wait times. When queue times increase, it means the router is overloaded with more incoming data than it can process and retransmit. The goal of any congestion management strategy should be to maintain average queue lengths at 1. It is not ideal to use round-trip clocks. As we've seen, calculating travel time between two points is a complicated and tricky endeavour. If feasible, round-trip times should be left out of the proposed congestion management architecture.

Sliding windows should not be used to regulate airflow. It's not easy to develop reliable and efficient methods for updating windows. In addition to contributing to congestion and disrupting time-sensitive network traffic, window updates may cause spikes in overall network activity. In high-speed networks with significant latencies, insufficient window widths may potentially cause underutilization. Therefore, methods of flow control based on a sliding window should be avoided. Unnecessary retransmission of packets should be avoided. In the case of packet retransmission protocols like Go-Back-N, previously received data is resent. Data is only retransmitted if it is known to have been lost, as is the case with more modern retransmission systems like Selective Retransmission. Retransmitting data only when absolutely necessary helps keep the network from being overburdened. It is preferable to keep error handling and flow management separate. Congestion and flow management in a transport protocol should never lead to packet loss. There are several causes of packet loss besides congestion. Packet loss should only be

International Journal for Innovative Engineering and Management Research
PEER REVIEWED OPEN ACCESS INTERNATIONAL JOURNAL
www.ijiemr.org

used to trigger retransmission in a transport protocol. The packet ingress and egress should be seamless. Short-term congestion isn't the sole effect of peak traffic times like window updates; end-to-end and round-trip variability also rise. Routers should make an effort to maintain the uniform spacing between packets that transport protocols allow into the network. Here, using Leaky Bucket and discouraging packet reordering are optimal strategies. If the queue size in a router is just one packet, rearranging the packets in the queue is almost impossible.

OVERVIEW OF THE FRAMEWORK

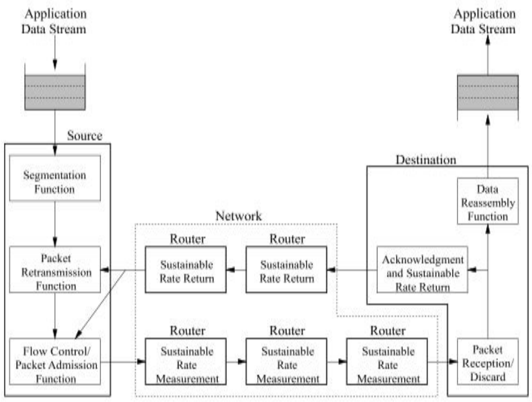

Figure 1 Overview of the Rate-Based Congestion Control Framework

All of these needs are accommodated in the framework I propose. The system, in essence, makes use of data on optimal traffic flow rates to distribute congestion management methods across the Transport and Network layers. The major parts of the framework are shown in Figure 4. Routers have the features seen in Figure 1 on page 3, plus the Sustainable Rate Measurement feature.

Rate-based congestion management architecture consists of the following key components:

1. The Network Layer provides the Transport Layer with rate information that is used by the Transport Layer to implement a rate-based flow control mechanism. Flow regulates and Packet Retransmission are two separate technologies that regulate error and flow independently.

2. Each traffic flow (a single data stream from an application) has its own unique source that allows packets into the network at regular intervals. The Packet Admission function does this by limiting the impact of temporary bottlenecks caused by spikes in traffic. It's important for routers to keep packets at about the same intervals of time when possible.

3. The Sustainable Rate Measurement function determines the maximum throughput that can be maintained by the network while still supporting all current traffic flows. The routers in the Network Layer execute a congestion management technique. This reliable measurement of throughput is sent to the receiving machine and back to the sending machine in the form of acknowledgement packets. The framework's central notion is to quantify the sustainable rate, and the method I propose to do this is called RBCC. In Chapter 6, we get into the specifics.

4. The remainder of the Network Layer congestion management system, including congestion avoidance and congestion recovery methods, is implemented by routers. In both the non-crowded and congested operation modes, routers equitably divide up the available resources among the various traffic streams. In the congestion cliff operational zone, routers begin dropping packets. The Packet Dropping, Packet Queuing, and Packet Selection processes carry out these tasks.

NETWORK LAYER RATE MEASUREMENT

Looking at the data flows that make intercomponent communication possible is necessary before addressing the various components of the framework. Each traffic flow in the framework will have its sustainable traffic rate throughout the network measured by the network routers and reported back to its origin. Several fields in the packets' Network headers, both originating and destined, are useful for this purpose.

When a source sends out a packet, it sets its desired bit rate in the Desired Rate field, its actual bit rate in the Rate field, and its identification in the Bottle Node field. If the source specifies a maximum Desired Rate, it means it is aware of the maximum data rate at which it can transmit. If the source requests a rate of, it will use the whole bandwidth at its disposal. A router's Sustainable Rate Measurement function may reduce the value of the Rate field while a packet travels through the router if it has assigned a lower rate to the traffic flow than the current value of the Rate field. If the Rate is changed, the router's identity is recorded in the Bottle Node field. The Bottle Node field indicates the node that set this lowest rate, and the Rate field contains the lowest sustainable rate assigned by the routers along the packet's journey when the packet arrives at its destination.

SOURCE FLOW CONTROL AND PACKET ADMISSION FUNCTIONS

The source should use a packet admission method to make optimal use of the Return Rate values from the network components and to ensure that packets are admitted to the network in a uniform fashion. Here's a twist on Leaky Bucket [Turner 86] that I've come up with. The bucket has an indefinite buffer and allows packets into the network such that the bit rate of the flow is equal to the Return Rate at the moment of packet admission. Figure 6 depicts the equation used to determine the inter-packet delay.

CONCLUSION

Only experimental verification of the suggested framework's effectiveness as a congestion control method exists at this time. The framework's validity may be established by rigorous mathematical study of the distributed algorithm it implements. The architecture relies heavily on the TRUMP transport protocol and the RBCC sustainable rate measurement function. Unfortunately, TRUMP lacked sufficient detail for proper integration into a functional protocol stack. The TRUMP protocol stack needs to have its specification finalized and an implementation developed. Similarly, RBCC has to be included into functional protocols. While this thesis demonstrates RBCC's use in routers, it is also necessary at traffic sources themselves because of the need of multiplexing several application data flows into a single or shared network interface. If RBCC were implemented at the source, application data flows may use the available bandwidth on the network interfaces more efficiently. Both routers and sources would need to be updated to use RBCC. Research into the proposed congestion management framework in realistic network settings would be made possible with a combined TRUMP/RBCC implementation. This would validate the findings of the network simulations presented in this thesis and provide investigation into how Link Layers and hardware affect the overall architecture. The computational burden of the framework on routers might be studied using a real-world implementation of RBCC, providing motivation to make the framework more efficient. TRUMP employs a sluggish retransmission method to prevent the establishment of any round-trip timer. When network capacity is high and round-trip delays are substantial, this might lead to an overload of the network. Lazy retransmission should be contrasted to other methods of requesting acknowledgements from the destination, such as actively polling the destination.

REFERENCES

[Mishra et al 96] P. P. Mishra, H. Kanakia, and S. K. Tripathi. "On Hop-by-Hop RateBased Congestion Control". IEEE/ACM Transactions on Networking, Vol. 4, No. 2, pp 224–239. April 1996.

D. Morrison. "PATRICIA – Practical Algorithm to Retrieve Information Coded In Alfanumeric". Journal of the ACM, Vol. 15, No. 4, pp 514–534. October 1968.

J. Nagle. "Congestion Control in IP/TCP Internetworks". RFC 896. January 1984.

J. Nagle. "On Packet Switches with Infinite Storage". IEEE Transactions on Communications, Vol. 35, No. 4, pp 435–438. April 1987.

Newman. "Traffic Management for ATM Local Area Networks". IEEE Communications Magazine, Vol. 32, No. 8, pp 44–50. August 1994.

K. K. Ramakrishnan and R. Jain. "A Binary Feedback Scheme for Congestion Avoidance in Computer Networks". ACM Transactions on Computer Systems, Vol. 8, No. 2, pp 158–181. May 1990.