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IMPROVED CONGESTION CONTROL FOR PACKET SWITCHED NETWORKS

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ABSTRACT

This study on Improved Congestion Control for Packet Switched Networks is motivated by inefficient utilization of resources, ultimately leading to network collapse. Suffice it to say that congestion has raised a lot of dust in packet switched network. Congestion shows lack of balance between various networking equipment. It is also a global issue. Congestion is seen as a state of being over crowded, overloaded or blocked that is too full of traffic. Congestion control refers to the mechanisms and techniques to control the congestion and will reduce below the capacity. The aim therefore is to design a software that will reduce network congestion within a packet switched network environment. This work disuses the various congestion issues and challenges with emphasis on packet switched network and develops a software that will reduce network congestion within a packet switched network environment. It addresses the problem of burst traffic, transmission delay and insufficient memory to store arriving packets. And the software was developed using Java programming language. The instruments for data collection are interviews, focus group discussions, observational methods and document analysis, focus group discussions, observational methods and document analysis.

KEYWORDS: Traffic Congestion, Networks, OOADM

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1. INTRODUCTION

The internet is a global infrastructure for information exchange that has revolutionized the social, economic, and political aspects of our lives. One of the most crucial building blocks of the internet is a mechanism for resource sharing and controlling congestion on the internet.

Congestion has been seen as one of the basic important issue in packet switched network. Sapna et al. (2012) asserted that congestion in a network may occur if the load on the network (the number of packets sent to the network) is greater than the capacity of the network (the number of packets a network can handle). Mathematically, congestion can be defined as if ∑Total number of packet sent + ∑Total number packet network can handle. Then it is in the state of congestion.

When the number of packets sent into the network is within the carrying capacity, they all are delivered, except a few that has to be rejected due to transmission error. And then the number delivered is proportional to the number of packet sent.

In the same vein, as traffic increases too far, the routers are no longer able to cope and they begin to lose packets. This tends to make matter worse. At very high traffic, performance collapse completely and almost no packet is delivered.

Kushwaha (2013) highlighted that congestion can be defined as a network state in which the total demand for resources, e.g. bandwidth, among the competing users, exceeds the available capacity leading to packet or information loss and results there will be a simultaneous increase in queuing delay, packet loss and number of packet retransmission. In other words, congestion refers to a loss of network performance when a network is heavily loaded. Without proper congestion control mechanisms there is the possibility of inefficient utilization of resources, ultimately leading to network collapse (Chandra, 2010).

Sabato (2014) disclosed that the main reason of congestion is more number of packets into the network than it can handle. Hence congestion control is an effort to adapt the performance of a network to changes in the traffic load without adversely affecting user's perceived utilities. Congestion control requires quick remedial measures both at the end host needs to decreases their data sending rates (or congestion windows) and routers need to drop packets until the congestion state is relieved.

Congestion control refers to the mechanism and techniques to control the congestion and keep the load below the capacity. This is done in order to avoid the telecommunication network reaching what is regarded as congestive collapse.

Research Paper

Yang & Reddy (1995) have divided congestion control mechanisms into two broad categories: congestion avoidance (open-loop congestion control) and congestion recovery (closed –loop congestion control). The strategy of congestion avoidance is preventive in nature; it is aimed to keep the operation of a network at or near the point of maximum power, so that congestion will never occur. Whereas, the goal of congestion recovery is to restore the operation of a network to its normal state after congestion has occurred. Without a congestion recovery scheme, a network may crash entirely whenever congestion occurs. Therefore, even if a network adopts a strategy of congestion avoidance, congestion recovery schemes would still be required to retain throughput in the case of abrupt changes in a network that may cause congestion. Congestion control is a (typically distributed) algorithm to share network resources among competing traffic sources. A network with a large band-width-delay product is commonly known as a high-speed network or long a fat network (shortened to LFN and often pronounced "elephant"). As defined in RFC1072 (Braden and Jacobson, 1988), a network is considered an LFN if its bandwidth-delay product is significantly larger than 105 bits (12,500bytes). In data communications, bandwidth-delay product refers to the product of a data link's capacity (bit/s)s and its end-to-end delay(s). The result, an amount of data measured in bits (or bytes), is equivalent to the maximum amount of data on the network circuit at any given time, i.e. data that has been transmitted but not yet received. Some important examples of systems where the band-width-delay product is large are high-capacity packet satellite channels.

However, packet switching is a method of grouping data that is transmitted over a digital network into packets. In order to transfer the file fast and efficient manner over the network and minimize the transmission latency, the data is broken into small pieces of variable length called packet. At the destination, all these small parts (packets) has to be reassembled, belonging to the same file. Packets are made of a header and a payload. Data in the header are used by networking hardware to direct the packet to its destination where the payload is extracted and used by application software. In every node of a packet switching network, queues (or buffers) are maintained to receive and transmit packets (store/forward network). Due to busty nature of the network traffic there may be situations where there is overflow of the queues. As a result there will be retransmission of several packets, which further increases the network traffic. This finally leads to congestion. Packet switching is more reliable as destination can detect the missing packet. It is cost effective and comparatively cheaper to implement. Telecommunication is the transmission of signs, signals, messages, words, writings, images and sounds or information of any nature by wire, radio, optical or other electromagnetic systems. Telecommunication occurs when the exchange of information between communication participants includes the use of technology. It is transmitted either electrically over physical media such as cables or via electromagnetic radiation. However, transmission of signals cannot take place if there is no network. Network is therefore a set of devices (often referred to as nodes) connected by communication links. A node can be a computer, printer or any other device capable of sending and / or receiving data generated by other nodes on the network. Most networks use distributed processing, in which a task is divided among multiple computers. Instead of one single large machine being responsible for all aspects of a process, separate computer (usually a personal computer or workstation) handle a subset.

Furthermore, network congestion can result when there is a rise in the transmission of a thereby leading to a decrease in throughput (Throughput is the percentage utilization of the network capacity). Also it can occur as a result of sending more data than the network elements can accommodate thus causing the buffers on the network elements to be filled and possibly having overflow.

Network protocol that use aggressive retransmissions to compensate for packet loss due to congestion can increase congestion, even after the initial load has been reduced to a level that would not normally have induced network congestion. Such networks exhibit two stable states under the same level of load. The stable state with low throughput is known as congestive collapse.

Networks use congestion control and congestion avoidance techniques to try to avoid collapse. These include: exponential backoff in protocols such as CSMA/CA in 802.11 and the similar CSMA/CD in the original Ethernet, window reduction in TCP, and fair queuing fair in devices such as routers and network switches. Other techniques that address congestion include priority scheme which transmit some packets with higher priority ahead of others and the explicit allocation of network resources to specific flows through the use of admission control.

Finally, congestion control is the controlling of traffic entry into a telecommunication networks in order to avoid congestive collapse. A system is said to be congested if it is being offered more traffic than its rated capacity due to too many active subscribers. System maintenance and repair actions can lead to system congestion but whatever be the cause of the overload, it will manifest as depletion of resources that are critical to the operation of the system.

Motivation

The study was motivated because of inefficient utilization of resources ultimately leads to network collapse.

Statement of the Problem

Kloth (2008), said that Internet can be considered as a Queue of packets, where transmitting nodes are constantly adding packets and some of them (receiving nodes) are removing packets from the queue. Some of the problems encountered include:

1. Insufficient memory to store arriving packets: Sometimes, all of the sudden a large chunk of data arrives simultaneously from several input sources and this data needs to be forwarded through a common output link, then a long queue of packets will be created at output link.
2. Slow processor: It has been observed that if the processor is slow in performing various tasks like processing queues,

processing routing table etc. There will be existence of large queue.
3. Burst traffic: This is one of the major causes of congestion.
4. Transmission delay: It has been observed with dismay that transmission delay is more because of rerouting.

Aim of the Paper

The aim of the paper is to improve on software that will reduce network congestion within a packet switched network environment.

Scope of the Study

This paper will cover the following:
1. A detailed look into packet switched network
2. Critical analysis of congestion issues in a packet switched environment and
3. Possible congestion control techniques. This has to do with the solution of network congestion within a packet switched network.

Significance of the Study

The significance of paper includes the following:
1. It will help to improve network performance in any business organization
2. It will help to eliminate data loss tendencies in a packet switched networked environment.
3. It will aid telecommunication companies in Nigeria to achieve a high performance level and optimal profit.

2. LITERATURE REVIEW

Mawhinney et al., (2004) designed a congestion management technique that smoothen packets transfer in a network. The system prevents congestion from reaching the point where data is lost or packets are dropped. The host session is divided into mission critical and non-mission critical sessions. The session types are prioritized in a way that during periods of congestion the mission critical session remains unaffected while the non-mission critical sessions are controlled. The mission critical sessions basically enjoy a reserved specific size of bandwidth up to point of congestion.

The design is quite similar to the Congestion Controlling using Network Border protocol designed by Sharadcandra et al., (2017), except that, for the former, the rate controlling is implemented in the host end and not in the network.

Kloth (2008), developed techniques for congestion control in IP network such as fiber channel network. Methods are provided to detect congestion in a host. As a controller sends packets through a link, the time elapsed between the sending and the receiving is measured. If the time elapsed is large, the path to destination is presumed to be congested and the port is blocked from receiving subsequent data. In the fibre channel, a data sequence at a buffer controller is received with a source matching one of many ports coupled to the buffer controller and destination accessible via a link also coupled to the buffer controller. The link is shared by traffic from numerous ports attached to the buffer to reach different destinations. The received packets are forwarded to the link and a transmission acknowledgement is received.

According to Kloth (2008), depending on the time the acknowledgment is provided, the port associated with the received packets is blocked. In another example, a congestion controlling setup in a fibre channel network is presented. The setup involves a buffer controller and various input ports. The various input ports are designed to receive packets having destinations accessible via a shared resource. The buffer controller is designed to receive packets from of the various inputs, send the packet through the link and receive acknowledgment from the link.

Floyd et al., (2000) proposed an "equation-based congestion control for unicast applications". According to them, (TCP) has been doing well in managing most "best-effort traffic" in the internet today. Nevertheless, a congestion control technique that is compatible with TCP and avoids reacting to packet drop by slowing down rate of transmission by half will be a good way of handling best-effort unicast streaming multimedia traffic. With this mechanism, the source regulates rate of sending depending on measurement of loss event. Meanwhile, loss event is the situation where some packets are dropped within a one round-trip time. Yang et al., (2011) opined that in shared networks (e.g. internet), the hosts should react to congestion by adjusting their rate of transmission thereby preventing a collapse and maximizing the available network resources. According to Yang et al., (2011), the internet today is robust as a result of TCP's host-based congestion control techniques. Much as TCP congestion control performs well for large data transmission applications, it would not work well for other newer applications which will find TCP behavior of halving transmission rate in response to congestion as harsh. As such, since internet traffic is mainly TCP based, it becomes pertinent that emerging congestion control techniques still be TCP compatible. Yang et al., (2010) then examined the fairness, responsiveness and assertiveness of TCP, General Additive Increase and Multiplicative Decrease (GAIMD) and some other two typical TCP-Friendly congestion control protocols.

These protocols are analyzed and simulated and their responses to network changes closely observed. Yang considered the integral instabilities in a static network environment and studied "protocol responsiveness and aggressiveness by evaluating their responses to a step increase of network congestion and a step increase of available bandwidth".

According to Rejaie et al., (1999), the stability and robustness of today's internet is mostly a function of end-to-end congestion control schemes and internet traffic today is mostly TCP and will remain so for quite a while. It is important, therefore, to have new applications to be "TCP friendly". Presented a rate adaptation protocol (RAP) which was TCP friendly and employs the TCP additive-increase, multiplicative –decrease AIMD algorithm. RAP was well suited for unicast real-time playback streams and the key goal was to separate network congestion control from application-level reliability while maintaining fairness and TCP compatibility.

Evaluating RAP through thorough simulation show that bandwidth is usually evenly shared between RAP traffic and TCP traffic. Rajaie, et al., (1999) state that "unfairness to TCP traffic is directly determined by how TCP diverges from the AIMD algorithm". Though RAP performs similarly to TCP in some situations, however, a simple rate control scheme is devised to widen the scope. Sharadcandra et al., (2017), presented a congestion controlling system that uses the network border patrol (NBP). NBP is a congestion control mechanism implemented on the network layer that controls congestion collapse by "a combination of per flow rate monitoring at an out-router and per flow rate control at an in-router", using feedback exchange instructions from the feedback controller in the out-router.

The out-router sends backward feedback to the in-router to inform about the rate flow's packets are entering the network. According to Sharadcandra et al., (2017), there are two unique types of routers introduced in the network called edge routers. Edge router may be viewed as in-router or as out-router depending on the active flow direction. An edge router, for example, acting on a flow entering into a network will be termed an in-router, while an edge router acting on a flow leaving a network will be termed an out-router. NBP uses feedback exchange between routers to manage problematic packets attempting to enter the network.

Subraman (2010), have presented a brief survey of major congestion control approaches and categorization characteristics, and elaborates the TCP-friendliness concept and then a state-of-the-art for the congestion control mechanisms designed for network. Kushwah & Gupta (2013) they pointed out the major pros and cons of the various congestion control approaches and evaluated their characteristics.

Zhang et al., (2007) proved that single-link congestion control methods with a stable radial Jacobian remain stable under arbitrary feedback delay (including heterogeneous directional delays) and that the stability condition of such methods does not involve any of the delays. They extended this result to generic networks with fixed consistent bottleneck assignments and max-min network feedback. They investigated the properties of internet congestion controls under non-negligible directional feedback directional feedback delays. They focused on the class of control methods with radial Jacobians and showed that all such systems are stable under heterogeneous delays. To construct a practical congestion control system with a radial (symmetric in particular) Jacobian, they made two changes to the classic discrete Kelly control and created a max-min version they called MKC. Combining the latter with a negative packet-loss feedback, they developed a new controller EMKC and showed in theory and simulations that it offers smooth sending rate and fast convergence to efficiency.

Qiaoyan et al., (2009) proposed an expert-control based multicast congestion control mechanism for wireless networks, termed ECBMCC. In this mechanisms, multicast receivers sent their feedback information to the expert controller rather than the sender, and the expert controller made sure the state of TCP connection by inferring according to the feedback information. Multicast congestion control is one of the key factors which restrict the development of multicast application, especially in the wireless environment. Here they proposed an expert-control-based multicast congestion control mechanism for wireless network - ECBMCC, and analyze the performance of ECBMCC by simulation. Results showed that ECBMCC adapts to wireless environment well. And ECBMCC works normally in wireless environment with high BER. Moreover, ECBMCC achieved excellent performance in TCP-friendliness on low BER wireless channel.

3. METHODOLOGY

Research methodology can be defined as a scientific method of enquiry involving formal process of verifying knowledge. It entails the plan and systematic collection, analysis and presentation of data. It compasses sources of data collected, method, analysis and interpretation of data collected. Research methodology therefore contains information on how data was collected, while design referred to planned structure and strategy of investigation though in order to get responses to research question and control variance. It can be said to be general program of study. It defines how the objects of the study will be accomplished and how problems encountered will be treated.

However, the researcher will employ two methodologies – The Structured Systems Analysis and Design Methodology (SSADM) which is systems approach to the analysis and design of information systems and Object –Oriented Analysis Design Methodology which comprises of the procedure of identifying software engineering requirements and developing software specifications in terms of a software system's object model and implementation of the conceptual model produced during object oriented analysis. The main reason why I decided to adopt SSADM is that it provides a set of techniques and graphical tools. They allow the researcher to develop a new kind of system specifications that are easily understandable to the user. In SSADM the researcher uses graphic symbols, Data Flow Diagrams (DFDs) and Data Dictionaries (DDs) to represent the system. Furthermore, the SSADM which was adopted in this research work has the following components which are contextually presented below:

1. Problem identification
2. Feasibility study
3. Analysis
4. Design
5. Implementation
6. Post Implementation Maintenance

1. **Problem Identification:** The problem identified in this case is congestion control in a packet switched network. This implies that each time there is need for communication flow, there is a kind a jam in a network due to huge traffic which in turn storms the efficiency of communication. The problem as identified calls for a means of reducing traffic to quicken communication exchange very efficiently.
2. **Feasibility Study:** This involves the identification of

various techniques available to solve the problem of congestion.

3. **Analysis:** This involves analyzing the present situation to identify the new situation in the relative to present realities, to enable reconfiguration in the light of the new findings. During the preliminary investigation some findings were made which now has been attributed, say fibre cut, further analysis revealed a different ball game which request a new approach to tackle.
4. **Design:** The ultimate design will now be achieve in line with the present analysis. Having identified causes of congestion, the design will take care of congestion obstacle and present complete control of communication flow.
5. **Implementation:** This will be design blue print to carry out implementation recommendations.
6. **Post Implementation Maintenance:** This involves proper examination of implementation activity to ensure that the design has been religiously followed and where there are mistakes, it is correctly put right.

4. CONCLUSION

This paper presents the study on Design of Improved Congestion Control for Packet Switched Networks. As the congestion control is the most important factor of any packet switching network, the whole performance and accuracy of network is directly related to it, the congestion control becomes more important. The researcher briefly surveys the methodology and techniques that will be used in designing of Improved Congestion Control for Packet Switched Network. It is expected that at the end of the paper that the software designed will help reduce network congestion by ensuring that the source host will be able to send data at a uniform rate and also have sufficient memory to store arriving packets.

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