International Journal of Trend in Research and Development, Volume 2(1), ISSN: 2394-9333
www.ijtrd.com

Reducing Data Loss and Improving Reliability in Network by Active Distributed Congestion Control Policy

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Abstract

Invention of wireless portable devices plays critical role in communication where the production of data is getting higher causes traffic higher, these traffic may be in the form of digital or analog signals, when some device are interacting with other in communication network has to be well managed and designed according to the volume of upcoming and ongoing data services so that network can be safer from failure and data loss that can be the biggest drawback of network technology, now are days internet are known as such a dependable unit for the communication therefore everyone are having very much efficiency in communication.

Researchers can say that “Congestion, in terms of networking is known as a kind of thing where one media carries large volume of data beyond its capacity, causes congestion”, In such of environment as the traffic gets higher sender and receiver node suffer a kind of problem known as “congestion”, as result network performance degrades, quality in communication getting deteriorate, resulting delay and network suffered from loss of data packets. In such type of congested network one can analyze slow response time with reduced throughput performance, to overcome from the problem one need to focus on errors occurred due to congestion, need to target at bandwidth inefficiency and network capacity to manage traffic.

In this case data frame has been lost during the transmission due to congested network is getting countered by flow control network management protocol scheme known as retransmission policy and many others, which manage network traffic by reducing data load, this policy works in two sate mode under the management of parallel data traffic load where one mode manage initial load and other manage network throughput.

Keywords- Congestion, Performance, Flow Control Data Loss, Throughput.

I. INTRODUCTION

Presently network are facing so many problems regarding the “Quality of Services factors”, Transmission of data with QoS factor is very much important whenever one talk about the Reliable network services infrastructure, due the high volume of data transmission from limited resources causes many side effects in the form of Data loss [1], delayed transmission and communication process and low throughput scenario affect network working, Such type of issues are getting increasing as long as the user and data size are getting higher from network to network , peoples always required reliable and timely services , they want to meet quality in communication that can be achieve only when one design an effective mechanism that perform well compare to the traditional one , because congestion is one of the major factor that should be consider at primary level so that performance can be manageable at every stage of communication [2], in this way the algorithm and scheme gives opportunities to have quality of communication with desired level , in the coming section author of the research discussed the "Analytical study of congestion control policies " that has been proposed few years ago to make our network more effective and efficient [3], congestion error always has been considered as an serious error in network because it is the base problem behind the unreliability and less quality of services network system. Congestion degrades performance on each and every direct or indirect function of communication system.

II. Analysis of Congestion Control Policies

In this section author analysis and discussion the recently proposed and presented congestion control policies that have been proposed between 2000 to 2014. The object of this section is to provide analytical representation of recent research in order to resolve congestion error and to improve performance of network.

AIMD Based Congestion Control Policy

Traditional TCP congestion control protocol is the best solution for end to end flow control solution at transport layer for wired and wireless communication [4,5], however this protocol is not much effective for delay , bandwidth -intense and Loss-tolerant multimedia application like multimedia application that use stream delivery services and real time video conferencing [4,5], because TCP has trades transmission delay for quality of network services even information also loss during the transmission due to network congestion and errors, in this case TCP support retransmission services to make transmission successful but it causes long delay, traditional TCP also having two reasons one due to the ignorance of delay deadlines of multimedia information transmission packets which means packets is not decodable due to the long delay system will not entertain such type of packets [6,7].

Second is the TCP throughput is fluctuated over time, significantly increases the end to end delay and reflect poor performance for stream delivery data transmission. In order to address solution for this problem one proposed AIMD based congestion control policy to determine network performance for multimedia application [3]. Although UDP protocol is more suitable for multimedia communication system compare to TCP but video over TCP is tremendously popular due to the coordination with HTTP. HTTP support advantages of web caching. TCP is famous for real time application environment because TCP is comfortable to deal with firewall security environment scenario [8].

In the proposed model author design a media aware POMDP based congestion control technique called as learning TCP that provides solution for traditional TCP protocol and improve network performance when dealing width multimedia communication, the proposed protocol is working with additional parameters like distortion impact and delay deadlines when adapting its congestion window size , traditional TCP is only consist congestion window with packet loss rate and round trip time .The proposed media aware solution focus on the congestion window updation mechanism on sender end without changing anything at receiver end or at any intermediate node [16].

Here one need to focus on the important thing that multimedia quality obtained by destination node is affected by congestion error due to the bottleneck error activity links, that has been deal by the sender node with the acknowledgement, to capture congestion error in dynamic environment with the observation of performance optimization parameter that reflect the resulting scenario has been infected by the multimedia quality measurement , the proposed solution gives idea to compute the media aware congestion control issue with POMDP framework [17], the computing structure describes the way that how the congestion are getting raised on network and how the network performed variation over time during the stream data transmission. New technique are able to optimize the congestion window , in this calculation research shows the multimedia transmission can be describes and measure as the total distortion reduction of the received multimedia packets [4]. In practice the network working scenario will as comes under into the condition where source node needs to understand the network state and environment during the communication in order to works in congestion control policy, so that proposed technique also uses a POMDP based model to detect and identified the congestion error, in AIMD based solution provides a way to get network congestion state to be cauterized only through transmission acknowledgements [15]. It is very important thing because technical difference has been offered and observable between MDPs and POMPDs, describes that AIMD based congestion control is more effective than model based congestion control technique. In the given table shows that the comparison scenario between existing and proposed techniques. In the next section authors are discussing the mechanism to overcome from the problem of getting data and reliability loss due to congestion error , ni most of the cases it happens that when network are getting suffered due to heavy load and large data transmission on the signal channel gets the same issue as what author motivate to do this research project.to overcome from the situation and the issues from the network author proposed efficient TCP mechanism that enhance the capacity of tradition version of TCP with more efficient networking reliable services that makes proposed approach more effective, the proposed concept ont only reduces the pitfalls from previous networking system but also getting quality of communication services that causes network more efficient and more reliable that define the importance of modern networking technology .

III. Improving Transmission Control Protocol Efficiency by Proposed Algorithm

In the previous secession, one discussed the history of TCP variation form long time of invention, what one observed that there are sod many limitations and drawback has been found in the working of traditional TCP approach , changes upgrades the algorithm is it fact when network come under the position of high load network scenario it happens, that reflect significant changes to be required to reduce the error regarding flow and error control [13], in case ,if network it getting heavier due to traffic, then performance has been decreases and the quality of service parameter getting survival due to bad network implementation policy , that require to changes in the traditional algorithm, one discussed, working and drawback of TCP Tahoe [14], perform well for some specific designed network but the implemented part of TCP is not sufficient as much as required for different scenario , one analyze that it designed on the principal of slow start algorithm , performing retransmission and congestion avoidance with a limited network communication area that causes high maintenance and overheads, basically Tahoe , developed new one on the configuration of previously proposed RFC 793 specification, in which one analyze that Tahoe gives three concept to improve traditional algorithm, in which first one is the RTO

(Retransmission Timeout Estimation) based policy , if over estimation in retransmission has been detected then TCP performance has been degrades , in the next approach it comes with RTT_VAR (Round Trip Time Variations) ,in this , it found that the estimated measurement of the specify parameter like round trip time estimation for flow control in parallel communication network [12], progressively it gives idea to enhance the detection of data loss in which hit perform the computation and comparison between round trip time and retransmission time estimation, here RTO>RTT are used to detect and avoid the congestion situation, at the end it gives idea for avoid congestion by slow start congestion avoidance technique. Apart from all the benefit Tahoe causes significant drawback like straining the network along with high amplitude parameter on each periodic stage it affect the rate, round trip time and causes packets loss on different variation of changed amplitude. DAUL[11 proposed a new invention to improve the drawback of last TCP , in this, it improve the TCP functioning by defining the TCP DUAL algorithm with the implementation of Queuing Delay based estimation for getting notification for the congestion state of dynamically working network. Another important proposal named “TCP Reno” that has been evaluated, by the author of the thesis to get the comparison and practical performance scenario between DAUL and Reno. Dual technique is quite low compare to the Reno, author found in DUAL, network suffered from some technical drawbacks like, in that if delay parameter found then it is not always indicating the true message to the network regarding the loss or congestion affects that leads unfair share of network resources, another drawback is what happens? When network environment is not idea one like one DUAL is dealing with another DUAL on the same network. Another mechanism to improve TCP working has been discussed in the previous section named as “New TCP Reno”, it is the extension of previously discussed TCP Reno version [10]. In TCP Reno an important drawback is fast recovery algorithm is not suitable when multiple packets has been lost by single congestion effect, insignificantly degrades the performance in high traffic network environment.

IV. Mathematical Construct for improving High Speed Network

The basic idea behind the implemented of mathematical construct is “Congestion Avoidance” phase of the proposed algorithm for getting reliability over that , this has been tested over different category of network that provides different type of window size for controlling congestion errors and flow controlling issues. More importantly, author wants to address the solution on the very common issue regarding congestion error in high speed network is state of size of congestion window has been change rapidly due to this factor , in most common case increasing is going on rapidly and reducing operation with congestion window more slowly, to utilize full network bandwidth with congestion window maintenance at both end , one need to improve the Round Trip Time minimum and Maximum rate computation parameter should be estimated through following mathematical construct, with the help of this approach in heavy loaded network congestion window size has been parallel managed with the buffering optimization construct discuss in coming section .

As long as the process completed successfully and network perform additional operation for maintaining the congestion window as the arrival of every packets ACK response, high speed network protocol performed increments over initial congestion window value from its current state value.

$$Cw \leftarrow Cw + \frac{\alpha(w)}{Cw} \quad \textit{(1)}$$

When network detect congestion event is going to raised proposed protocol decreased the value of congestion window as like equation (2).

$$Cw \leftarrow (1 - \beta(Cw))Cw \quad \textbf{(2)}$$

$$\alpha(Cw) = \frac{2Cw^2 p(Cw)\beta(Cw)}{2 - \beta(Cw)} \quad \textbf{(3)}$$

$$\beta(Cw) = \frac{\log Cw - \log Tlow}{\log Thigh - \log Tlow}(\beta high - 0.5) + 0.5 \quad \textbf{(4)}$$

Equation (3,4) are resolving the common congestion factor by introducing parameterized solution rising due to the flowing difference value and over flowing of allocated buffer status. α (C_w) getting optimized variable value, $\beta(C_w)$ consisting the estimated value of round trip time and $p(C_w)$ consisting throughput thresh rate..

V. Proposed Active Distributed Congestion Control Policy

According to the traditional policy of data transmission states that XR2→R2Z will be the best and least way to transmission but it happens when R2 is available for X, if R2 has been engaged with some one else then it does heavy congestion at intermediate router R2 . To handle this situation proposed algorithm perform buffer optimization and comparison to avoid congestion state at long extent, after computation network found the best ways with intermediate node to assign "Active Distributed Session" for all the active network node that belongs to this session has been the participating node to perform transmission with quality of services, as in figure1, network collect active node that has been chosen after optimization process and assign active transmission session for timer T1.

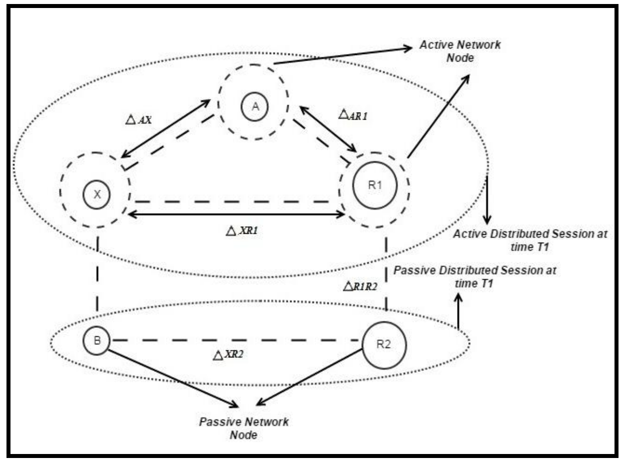

Figure 1. Proposed Active Distributed Congestion Control Policy

VI. Conclusion

At the end of the section author concluding the discussion regarding the algorithm and techniques which has been already proposed and implemented for controlling the congestion at node level or intermediate node level, reliability factors in high speed network is completely based on the implementation of offsite technology , TCP is one of the important protocol , author need to analyses the services proposed by traditional TCP algorithm in that authors gets the idea to achieve reliability factors with congestion control and avoidance scheme together in our proposed policy, proposed research performing the reliability in network state even at heavy traffic load with the improving and updating services of TCP [9] so that here author describing the performance measurement comparison between services on behalf of the features provided by traditional and newly proposed algorithm as mentioned on table that reflects the importance of proposed technology named as "Proposed TCP" showing in beneficial featuring as improving reliable network services with quality of service factor with losing data packets.

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