INTERNATIONAL JOURNAL OF RESEARCH IN COMPUTER APPLICATIONS AND ROBOTICS
ISSN 2320-7345

AN ACTIVE DISTRIBUTED CONGESTION CONTROL POLICY OVER RELIABLE HIGH SPEED NETWORK

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Abstract:- Now, Network are great friend of people , many of us are right now connected directly or indirectly with it, as a result producing heavy traffic, large data that does network complicated, peoples faced it frequently now are days. Congestion is key problem of communication technology that may be wired or wireless, since the network services increasing day to day ,it produce large volume of data that comes from many request due to high availability of electronic portable devices network are growing towards the objective of working in paperless data transmission environment. At present network are maintaining large number of clients that is the challenging task for every network management policy to give congestion free network.

The Proposed ADCC model are addressing the solution for congestion error , this research may also help to all the network manager to support smooth functioning of client request in high congested traffic area so that network can doing well at any step of traffic availability, research also proposed mathematical construct for managing buffer in order to remove complexity at the position of centralized node that performing decision to select correct address in order to fulfill each upcoming thousands of request from different nodes at same time. In proposed solution, Author design and proposed an algorithm and simulate them on GNS3, for performance analysis, where they take care the underline specification as low buffer size problem, bandwidth utilization, and data loss problems. Proposed algorithm comfortable with real time application environment in the network, which gets demand for higher bandwidth communication architecture, where traffic should be managed with proper fault tolerance, proposed distributed session object allows the network to works in distributed synchronized fashion to support run time real application environment.

Keyword: - Congestion Control, ADCC Model, GNS3.

I. Introduction

Proposed research provides an intelligent way to achieve reliable communication in highly congested network where research proposed synchronized network design that support real time application with reduced complexity, Protocol performed function of flow management along with the responsibility of error control services, to get easy in implementation this scheme researcher user timer function at centralized node to manage multiple interface to transmit and deliver data to remaining nodes, another useful function is buffer optimization at run time environment support communication network to get increases throughput and response time for every connection. This scheme is suitable not only for efficiency but also useful to get quality of communication in network. Algorithm has been tested over simulator (GNS3) , implemented algorithm functioned over reliable connection as result network show quality in communication by getting successful transmission of every data packet at every node for specific time duration , result conclude the experiment by producing result that define statistical value of data packet loss, response time , throughput in highly congested network , proposed experiment showing low complexity, highly reliable, efficient networking mechanism that has been performing well at highly traffic network which is mostly required by network that reduces many pitfall from present network in order to achieve highly reliable network services with great extent .

II. Related Work

Heavy load in the network causes Congestion error many time to reduce this number there are so many techniques has been proposed by traditional as well as present researchers to get know about the congestion before congestion happing, it good for reducing congestion problem many times, author discussing conceptual meaning of congestion, in this section author of this paper discussed questions like How one can know about congestion is going on? How to detect congestion on channel and the way by which network can have the capability to avoid detected congestion. In order to detect congestion error before happing congestion , there is only way is to get information through data loss and getting information regarding changes at throughput performance level [2], following are the techniques is used to get detect congestion in network.

(i) Slow Star and Search Technique

In this approach one observed that as load is getting higher throughput get higher in underlined load and levels off as network comes to its capacity. The Throughput cure represents as an indication of congestion problem. When connection are added the curve get level off and when connection are out the level becomes more linear, Here cure follows the rule by proposed algorithm that get alert for the congested network situation. To represent curve state, one can use following

$$T(CW_n) = \frac{(T(CW_n - T(CW_{n-1}))}{(CW_n - CW_{n-1})} \qquad (1)$$

Here Equation (1) represents CW as Congestion Window used window size parameter and T represent throughput Scenario. In order to calculate round trip time of multiple connections can be calculate by following equation (2).

$$TTG(W) = \frac{TG(W_{tt})}{TG(W_1)} \qquad (2)$$

Here TTG represents total throughput of connection, W_{tt} shows total connection throughput. The TTG get ranges over 1 to 0 values as traffic varies, as per the analytical study on simulator in a low traffic network Total throughput would be close to 1, as result when network reaches its capacity total throughput reaches to 0.

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(ii) DAUL Technique

Proposed a new scheme to correct the oscillation problem find in slow start algorithm, proposed algorithm address the solution by implementing round trip time computation to detect congestion as well as reduces loss of data that occurred in case when bottleneck detected previous algorithm.

The RTT is consisting of two parameter for propagation delay and queuing delay. Here RTT is equal to propagation delay.

$$Actual\,RoundTripTime_{min} = X_n \tag{3}$$

Here Highest Round trip Time would be measure as the total of Propagation delay and delay due to bottleneck Processes, that can also be represent as the actual scenario of proposed scheme where one can be able to analyze throughput ratio with delayed ratio in the form of entities like.

$$T\Delta_{max} = \frac{TP}{TRT} \tag{4}$$

In this case another arithmetic calculation can be processed for the working of calculated through put it get more optimal solution in case of higher level of traffic network , where network managed queue for the purpose of managing multiple connection at run time with the implementation of following construct.

$$RoundTripTime = \frac{Total\,Process}{ProcessingRate} \tag{5}$$

This algorithm identifies the congestion error with the help of detecting the information regarding the loss of data due the buffer overflow error happenings, Techniques provide solution by proposed idea, in which it estimates Round Trip Time minimum value and maximum value over threshold value, as result avoid bottleneck problem in high volume of traffic network, That can be done as.

$$Actual\,RoundTripTime = (1 - j)_{min} + RTT_{max} \tag{6}$$

Where j>=1 for every j_i,$j_{i.}$ If J<=1 then it get away from overflow error.

(iii) Gateway based Congestion Control Techniques

The traditional congestion control mechanism is not sufficient for the detection of congestion control problem because when one talk about the congestion error techniques , all the previously discussed algorithm and processes are good to manage congestion error, for the point of view of end to end connection [1], in such type of connection, congestion is not detecting as a buffer overflow error rather then it can be detected by the happing of data loss error, increased total round trip time or changes in throughput [10].

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$$CON_{max>j} = \frac{TRT_{j....j-i}}{TTH_{ji}} \quad (7)$$

Where CON found congestion rate, TRT is the Total Round Trip Time. In this way gateway performing the monitoring for congestion error to indicate congestion reasons therefore gateway is much able to take action regarding the congestion maintenance, it seems meaning full not only for detecting but also for avoiding the event of happening congestion [5].

(iv) Random Early Detection Technique

Drop packet can be a solution proposed by gateway in previous technique, where drop packet is done by the system to get resolve congestion problem. it done automatically when queue is full. In the same way Tail Drop algorithm is work, where every new arrived packet has been dropped out from the memory where approach get queue is full. One more approach has been proposed named early drop tail technique in which drop packet has been done randomly from run time connection, drawback of this approach is , throughput is getting low [4]. To carry out the average of queue for packet dropping algorithm uses following equation (8).

$$DetectionAVG = (1 - F_q)avg + W_q * MaxQueueSize \quad (8)$$

III. Performance Evaluation of Traditional Congestion Control Protocol with Proposed Active Congestion Control Policy

The proposed "Active Distributed congestion Control Policy" has been designed to traffic stability along with the parallel communication process to achieve fairness and efficient allocation of network resources to proper utilization of network bandwidth, in the previous section we have discussed the traditional congestion control protocol and strategies in detailed for reliable network services , now this section aims is to evaluating and comparing the traditional protocols with newly proposed scheme , which has been tested over the simulation platform named network simulator 3 with live internet capturing, approach are investigating the Packets loss Ratio, Delay Time Ratio and throughput Measurement with fairness testing. The result describes that the proposed policy is more flexible and fair in the way of throughput, high performance, proposed algorithm providing control congestion environment with reduced data loss, it does efficient networking with fair share of available bandwidth utilization. Author observes Vegas TCP is performing fair result but not utilizing the available bandwidth of channel when the situation has been raised during the implementation phase with Reno. Performance of TCP affected due to the loss of data or due to the Congestion in channel. The proposed implementation resolving both of the issues at the same time all the transmission is going through this approach having reached safely without much delay and no packet has been marked over there that reflect the quality of services factor of proposed research to proof its efficiency than other traditional one.

(i) Evaluation on Congestion Window with Time measurement

In this section author perform the experimenting analysis between multiple congestion control protocols with newly proposed research algorithm, the buffer optimization plays an important role to get optimum result in all the case of going network scenario, in figure 1 author already describes that the pictorial view of the running processes to proceed the sequences to perform different transmission on same channel capacity. In this experimental analysis author perform evaluation study on graphical network simulator 3 for the purpose of simulating Transmission loss based, High Speed TCP Versions of Protocols, midrange TCP, Low range TCP that has been consider for the experiment of difference scenario of network for performing delay based congestion control mechanism.

The overall performance of research is primarily focused on the growth and decrements in congestion window at the event of congestion. This session is using different network scenario to experience the performance of protocols in different way with variation in the flow, experiment has been done under some consideration as the data type transmission will be FTP at both end, and figure 2 describes the working efficiency of algorithm by measuring performance over congestion window with congestion elapsed time.

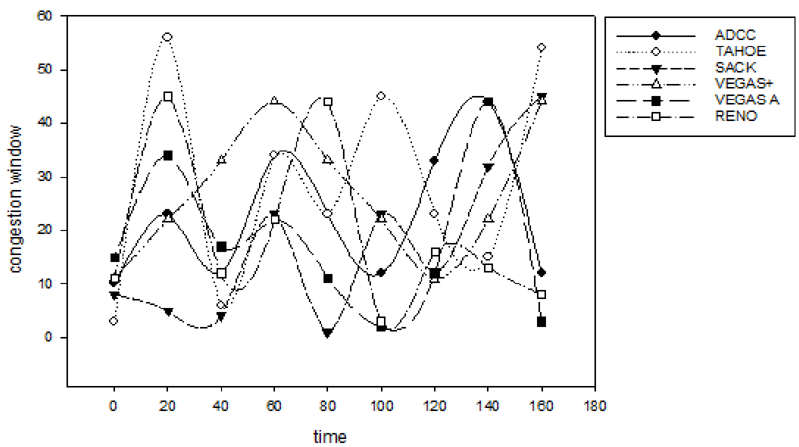

Figure 1 Congestion Window Vs time Measurement

(ii) Evaluation between Throughput Vs Flow

Channelization are one of the important functioning that is need to be consider whenever someone talk about the quality of services during the transmission process in the way of providing most superior flow over the available bandwidth with the parallel management of throughput performance [3,7] . In below figure 2. Author are analyzing the performance variation with different traffic has been used to capture the flow control measurement over different TCP communication level. Here author considering the network performance in some different situation as below.

Scenario I

Network considering the TCP connection level as high Speed TCP connection, comparing with traditional TCP to measure the disputes causing during the transmission, the overall story of the figure 2. is to focus on the perception of measuring congestion collapse events and its effect over the fellow of network.

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Scenario II

This author performing the evaluation over the estimated round trip time estimated rate with scalable TCP support flow with controlled flow over the channel, the ratio of throughput and flow control showing the disappear behave of TCP that not getting drops the packets but gathering the multiple packets delay that does performance down in the estimated time duration, it happened some times, TCP Reno depicted in figure 3. that unfair assignment of channel capacity.

Scenario III

In this evaluation concept TCP flow has been managing the congestion control over cubic TCP reliable services, this scenario reflects serious errors regarding the detection of congestion events that result maximum retransmission rate for the dropped packets transmission ,the higher frequency of happing such events frequently , in this case the congestion window growth is very slow down for the increment function, rate given by this scenario is very down compare to previous two scenario I,II as in figure 4.

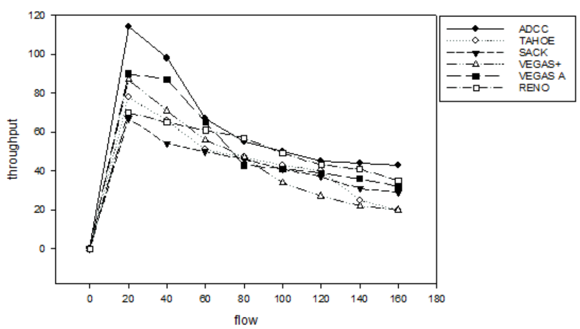

Figure 2 High Speed TCP Flow Vs Throughput

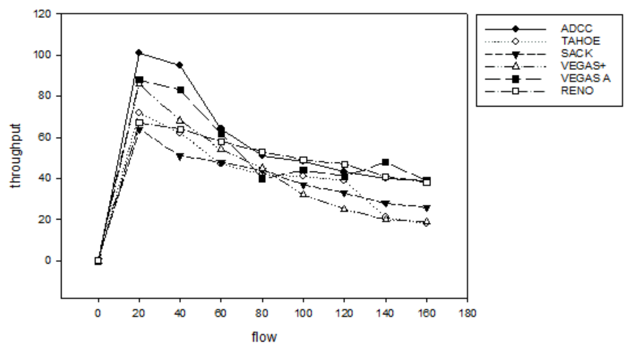

Figure 3 Scalable TCP flow Vs Throughput

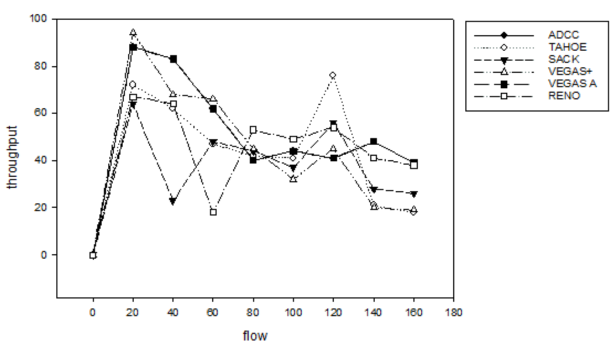

Figure 4 Cubic TCP Flow Vs Throughput

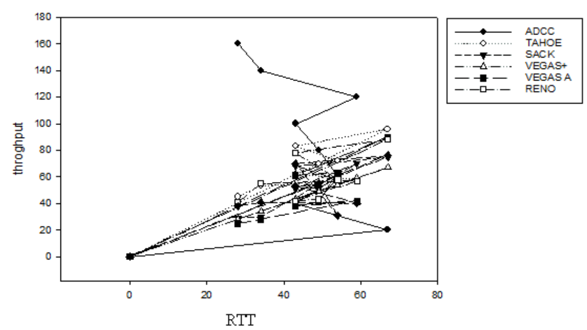

Figure 5 Throughput Vs RTT Estimation

Scenario IV

This time the network situation arises as per the incoming and outgoing overflow in beginning, in-between and at the end node where all the nodes are working together to find the optimum node with current congestion window state , flow control is important task here the proposed algorithm provides estimated RTT as per the required throughput level for performing the transmission that does control flow transmission and avoiding congestion state from the beginning of transmission [8,9] , figure 5.described the throughput and Estimated RTT values at different pattern of

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data transmission as discussed in figure 5. has been evaluated over the current congestion size that has been growing and sinking as long as the load over the network has been detected.

(iii) Average Queue Delay Vs RTT Estimation

The proposed algorithm is not considering the differential time gap between transmission due to the efficient working of the implemented network, the intermediate node are getting the decision for the next node selection from first initialized node as n_1t_1………n_nt_n to get the overall performance of entire network, one need to observe every discovered node for the active mode position to produce the required result in optimum, in the below figure 6. Proposed algorithm define the ratios between average delay and estimated round trip time vector as V_1……………………..V_n. on the available channel rate.

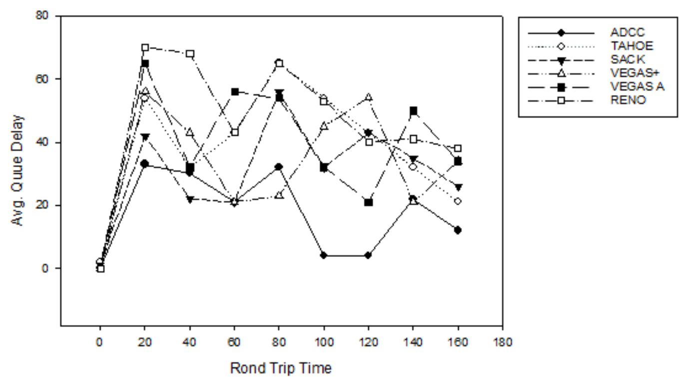

Figure 6 Avg. Queue Delay Vs Round trip Time Measurement

Figure 6.describes the functioning aspect of the implemented active distributed object at networking scenario that makes network more intelligent compare to the general working of previous one, to avoid average queue delay it managing the performance efficiency with RTT measurement factor. In figure 6. The Proposed protocol performing better compare to the other as they are degrading the throughput average at many times due to the congestion occurrence it happened especially with RENO, AIMD and Vegas A when the traffic load gets higher during the transmission of interlayer or outer-layer operations.

(iv) Fairness Vs Number of nodes

Below figure 7. analyzing the performance specially for the measurement of effective utilization of network bandwidth that has to be use efficiently on every pattern of ongoing network to maximize the performance result author proposed the scheme with fair load distributing approach that allow network to work effectively even in heavy traffic environment [6] , in figure 7. one can analysis that only the proposed algorithm are maintaining its performance consistently among all with different types of traffic pattern , to possible this buffer computation performing major role in between the process at intermediate node level . Active Congestion Control algorithm performing optimality operation and then use this optimum value to grow the congestion window accordingly that protect network performance at higher level.

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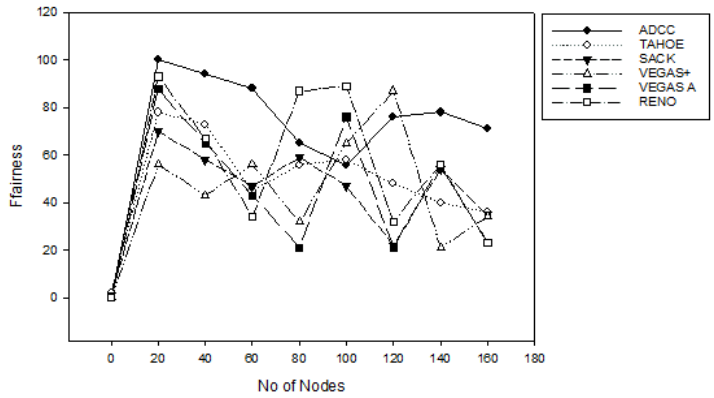

Figure 7 Fairness Vs No.of Nodes

In figure 7. Only the proposed ADCC algorithm only managing its performance throughput and other all are suffering from congestion events with different traffic patterns.

IV. Proposed algorithms

Following stages are playing an important role for the purpose of buffer optimization, here important is to ensure that how our approach is accurate and complete for the buffering operations.

1. At the very beginning all the Buffer_nodes of the network has been initialized with its initial value.
2. Assuming a parameter named f that is always follow f>0 with a barrier parameter as α>0.
3. The default initial rate value is pR(0),s(0), σR(0).
4. Initialize Loop1: Do until
   (Maximum value|pR(T+1)-pR(t)|<) and (maximum value | σR(t + 1) − σR(t) |<)
5. Initialize next loop 2: Do until
   (Maximum value | ‘pR(k+1)-‘pR(k)<) and (maximum value | ´σR(k + 1) − ´σR(k) |<).
6. Now calculate a new variable as H, to obtain negative and positive value of above instruction. If it is positive then goto step 5 else continue.
7. Now calculate d as a variable and go to step 6.
8. Ptem=’pr(k)+dpr;
9. σtem=’ σp(k)+dσr;
10. atemp=’s(k)+ds
11. Now compute the new value of φ(k + 1) by having new vale of computed buffer rate by substituting it from ptemp, σtemp,Stemp as from metrix problem.
12. If found (φ(k + 1) − φ(k) ≥ ´) is positive value then decrease R and go to step 4.
13. Calculate pR(k+1)=ptemp; σR(k+1)= σtemp; s(k+1)=stemp
14. End of loop.
15. Decreases the value of parameter μ.
16. End of first loop.

V. Conclusion and Future Scope

Every research required something special and much beneficial propose and design that cooperates for the future enhancement and implementation therefore first of all author of thesis is concentrating in the fundamental aspect of understanding the problem scenario therefore they referred the underlining problem scenario in the way as section 1 in which they focus on the fundamental aspect of proposed work , they parallel takes the literatures and already proposed solution with common problem factors to make them more accurate with proposed technique.

Traditionally defined TCP instruction set is not sufficient to perform high quality reliable communication network proposed TCP algorithm is efficient enough to perform the operation at node to node in order to works in highly reliable network environment that provides quality of networking services with the help of proposed instruction set one can have the connection oriented transmission services so that it co-operates today's real time application.

Future of networking technology will be more advance in each and every direction of communication services scenario where network has to manage more advance application and tools to overcome from the problem of congestion over large data set that is coming from reliable and unreliable sources, as the technology is growing the production of data is also getting higher , traffic increased with multiuser communication scenario leads to manage it with performance efficiency and quality , data loss issue are also getting increases as the users and application are increasing in wired or wireless network , it can be a big challenge in future networking scenario for the network protocol designers and developer that how to manage large set of application users in dynamic communication environment, the proposed solution and technique is designed for an specific type of networking infrastructure because there is the defined protocol supports the reliability issues , it cooperates with the quality service factors by the way of channel feature and referred parameter aspects , author just take the week factors into focus to improve reliability services only , but this technique showing improved performance only over high speed reliable network as per the proposed tilted and research requirements.

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