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A Novel Rate Control Metric for Packet Congestion Control in Distributed Networks

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Abstract

The existing internet architecture becomes insufficient in meeting the demands of exponentially growing dynamic customers worldwide. This scenario is in need of higher channel capacity and efficient congestion management techniques. It is observed that the existing methods and approaches are becoming inefficient in managing current trend of communication and congestion. In this paper, we propose a method based on a novel metric known as "stochastic rate controlling factor." This metric is devised to measure the performance of the rate control components and in turn control the congestion effectively. The proposed system is validated and demonstrated through a system simulation approach.

Keywords: Rate Control, Congestion Control, Internet Architecture, Bandwidth

Introduction

In Information Technology (IT), the technological advancement is a continuous process and Internet is the manifestation of this. Since ARPANET [1], the growth of internet has seen multifold in speed and number of hosts. As per RFC 1958, the principle of constant change is perhaps the only principle of the Internet that should survive indefinitely. Evolving design principles in the architecture of the future internet is most discussed issue in the area of distributed networks. The continuous growth and incredible success of internet (distributed network) services encouraged researchers to devise novel safety and security services, which may not be supported by the current Internet architecture. As per the AKARI project "New Generation Network Architecture AKARI Conceptual Design" [2], there will be heterogeneity of nodes such as computers, terminals, mobile devises, sensors etc and the dimension as large as of 100 billion in size. Various services have been conceptualized in the domain of emergency Services, Transportation, healthcare etc in pervasive or ubiquitous way. At the same time, because of the advancement in video technologies a larger amount of multimedia content will also be flowing across the networks.

Based on the experimental observation made by Medeiros [3], it was found that there is massive volume of data being shared among 32 millions of user, especially from the Internet based video. This fact highlights that resources available in Internet continuously increases its size with the dynamic demands of the massive volume of online users. Handling such data is quite a challenging task as it genuinely leads to worst traffic congestion. Hence, it is essential that congestion protocol should be design considering such real-time challenges for the purpose of enhancing the availability, reliability, and interoperability of Internet-based services. Another author Mahonen et al. [4] have also discussed about the growing traffic in Internet and suggested that such congestion related issue could be possibly mitigated by over-dimensioning the network. Similar issues were also discussed by Jacobson et al. [5] for the purpose of enhancing the channel capacity on the backbone network to cater up the advances Internet-based services.

From the recent cumulative survey conducted by Pan et al. [6], it is observed that most of the current test beds for future Internet architecture research in different countries are the results of previous research projects that are not related to future Internet architectures. Hence, one contradictory question arises based on the efficiency of the architecture as a restricting factor towards the design and development of future Internet architecture. For an instance, as discussed in [7] “the end-to-end arguments are insufficiently compelling to outweigh other criteria for functions such as routing and congestion control”. On the other hand, the evolution of the Internet architecture is driven by incremental and reactive additions [8], rather than by major and proactive modifications. Furthermore, the researchers have exhibited that efficient performance or functionality define necessary but not sufficient conditions for change in the Internet architecture (and/or its components); hence, it demands to demonstrate limits of the current architecture [9]. Thus, scientists and researchers from companies and research institutes world-wide are working towards understanding these architectural limits so as to progressively determine the principles that will drive this massive network to meet the requirements in the present scenario.

We propose a novel metric known as Rate Control Metric (RCM), formulated through stochastic principles. The performance of this metric in view of effective congestion control in the complicated Internet work system is analyzed through simulation.

This paper is organized as follows. Initially the related work carried out in the area of congestion control in Distributed networks is discussed and then the motivations for the design of RCM, the formulation of RCM are discussed. In the next section we briefly describe the significance of RCM. The later sections elucidate the implementation details and results respectively. Finally the threat to validity followed by conclusion is presented.

Related Work

Congestion management in Network

The work carried out by Seferoglu et al. [10] has presented a framework for mitigating congestion issues raised from TCP network. The study integrated TCP Friendly Rate Control standards with standard Forward Error Correction. The evaluation strategy highlighted that standard FEC techniques depends on the statistics of data loss and can produce potential network overhead in transmitted redundant parity packets. The outcome of the study was analyzed using the performance parameter of delay and loss characteristics in various simulation scenarios for determining the significant duration of maximized congestion condition. Mao et al. [11] have developed hybrid traffic Active Queue Management (AQM) router with classifier and scheduler that confirms the link capacities. The author has confirmed some stability conditions for the AQM policy to stabilize the TCP and UDP queues in routers. Shiang and Schaar [12] have proposed a content-aware congestion management for multimedia system streaming over TCP/IP networks and achieved more than 3dB improvement in terms of PSNR over TCP congestion control approaches. Rahman et al. [13] have introduced Datagram Congestion Control Protocol (DCCP) and found it is s appropriate for multimedia applications. Zhou et al. [14] have presented a congestion window adaptation formula for MPTCP (Multipath Transport Control Protocol). It adjusts the congestion window for every TCP sub-flow dynamically and therefore mitigating the variable end-to-end path delay.

Congestion Control in Wireless Sensor Network

Waghole and Deshpande [15] provided a solution on congestion issue by reducing average End-to-End Delay through deploying Movable Mobile Sink in uniform Random Wireless Sensor Network. Yedavalli [16] has analyzed the queue dynamics within the wireless sensor network using fluid models and exponential back-off based service rate models to control congestion. Reddy [17] have designed protocol for a queuing model for generating heterogeneous traffic among every sensor node in line with high priority to the sink. Priya and Terence [18] have projected a protocol namely EECP (Energy efficient and Congestion-aware Protocol) that helps to realize correct event detection, energy potency and fewer traffic. They claim that this protocol will offer an effective MA (Mobile Agent)-based clustering algorithm (MACA) to realize the energy potency and congestion resolution. Gupta et al [19] suggested a protocol that deals with the minimum energy consumption and speedy transmission between multi-hop clustering in wireless sensor network. The proposed protocol minimizes the congestion at the base station and improve throughput by using round Robin programming in inter-cluster communication scenario. Chakravarthi and Gomathy [20] proposed a cost effective protocol in wireless sensor network to discover and manage congestion at the MAC layer. The level of congestion is measured using a metric called Depth of Congestion (DC). Based on the measured value the node effectively adapts its transmission data rate to control congestion.

Congestion management in Mobile Adhoc Network

Bullibabu and Ramesh [21] have proposed a multi-rate multicast congestion management policy which supports mobile ad-hoc networks. The projected theme overcomes the disadvantages of existing schemes that resist them from being applied to MANET situations (e.g., being affected adversely by link access delays caused by access competition and by high link error rates; having excessive management traffic overhead). Rathod and Patel [22] projected a network cryptography and congestion aware routing mechanism in MANET by performing elaborated analysis of existing coding and congestion aware routing protocol. Sheeja and Pujeri [23] proposed to develop a scheme called as an Effective Congestion avoidance that consists of congestion observation, effective routing establishment and congestion-less routing. Bawa and Banerjee [24] propose a load equalization approach in AOMDV (Adhoc On-Demand Multipath Distance Vector Routing) protocol that uses queues for congestion monitoring. Reddy [25] has introduced a new cross layer and path restoration procedure in painter that derives two algorithms for Path discovery and congestion management correspondingly. Rajeswari and Venkataramani [26] evaluate the performance of four queuing disciplines (FIFO, PQ, RED and WFQ) that is enforced within the adaptive Energy efficient and a Reliable Gossip Routing (AEERG) protocol in mobile ad-hoc networks. Rao et al. [27] have proposed an energy efficient and reliable congestion management protocol for multicasting in mobile ad-hoc networks (MANETs).

The demand of the present day scenario is the congestion free and high performance network. In view of this we propose effective rate control metric which will address congestion control effectively.

Motivation

Problems Identified in the current system

It is noted that the existing internet architecture lacks effective methods to ensure quality and reliability when it comes to dynamic traffic management. It encounters the following quality and performance problems.

- The network becomes unmanageable during critical applications due to its less cognitive in nature. Majority of the services are static and lacks in flexibility and customization.
- As majority of the services are designed on the basis of best effort the system suffers with standardization problems
- The existing network services are not designed to meet the challenges associated with future internet user demands in terms of quality and services. Hence, the current network services are not governed by the rules and policies which will support the Future demand.

In addition to the above design issues, the existing internet architecture has several detrimental features of bandwidth, traffic congestion, application performance as well as security. Hence, the usage of reservation as well as differentiated services could be an added advantage to overcome the dynamic requests. Hence it becomes necessary to design an effective internet architecture that supports large scale dynamic end user requirements.

This could be achieved by incorporating self-adaptation and self-learning capability in the future internet architecture. Therefore we propose a Rate control Metric (RCM) based on stochastic principle to understand the traffic behavior in a better way. This metric provides rate control in communication between nodes/ routers by registering the capacity of the link at that point of communication to mitigate the congestion that exists in the network.

Significance of Rate Control in Communication

Controlling the rate is one of the crucial exercises for the purpose of minimizing the response time of user as well as catering up the dynamic traffic demands. The currently practiced techniques of TCP based connection are focused for specialized application that deploys long-live traffic flows. By controlling the rate of the traffic ensures that the traffic flow is accomplished to the minimum resulting to the discernible enhancement for the multiple online users in distributed network.

Therefore the rate controlling parameter in distributed network can be developed to incorporate following characteristics.

- Exponentially faster access to massive files as compared to conventional TCP based connection.
- Should be deployed on advance networking systems like optical network and should adopt high-bandwidth delay product.
- Should ensure stability in the network and free from much dependencies on round trip time, flows, and link capacities.
- Should be compatible for incorporating traffic-based policies for ensuring the adherence to congestion control schemes.
- Flexibility to network operators for incorporating privileges to certain traffic flow.

Hence, it is anticipated the proposed rate control metric should adhere to the above mentioned characteristics to ensure better congestion control strategies. The parameter for controlling rate should ensure the mechanism for evaluating the rate control factor at the end host in congestion layer that has surfaced up owing to IP/TCP based network. The parameter for controlling rate should also ensure performing comparison to the currently existing transmission rate of link that has data packet stamped at the destination end. The minimized value of the transmission rate is updated in the data packet and forwarded to the next link. The proposed techniques ensure fairness in bottleneck link and can cater up the dynamic traffic needs.

Formulation of RCM

The formulation of RCM is carried out, considering the design challenges and the execution model of rate controlling factor in future internet architecture.

The proposed model considers design of rate controlling factor in simulation mode where a networking device like router controls a unit rate R_{unit} (t) for every communication channel. The router time “stamps” R_{unit} (t) on every transmitted data packet and forwarded to the sender node to indicate slowest rate along the routes. It supports the sender to organize itself for a low dense data transmission to avoid congestion. Figure 1 depicts the design of RCM.

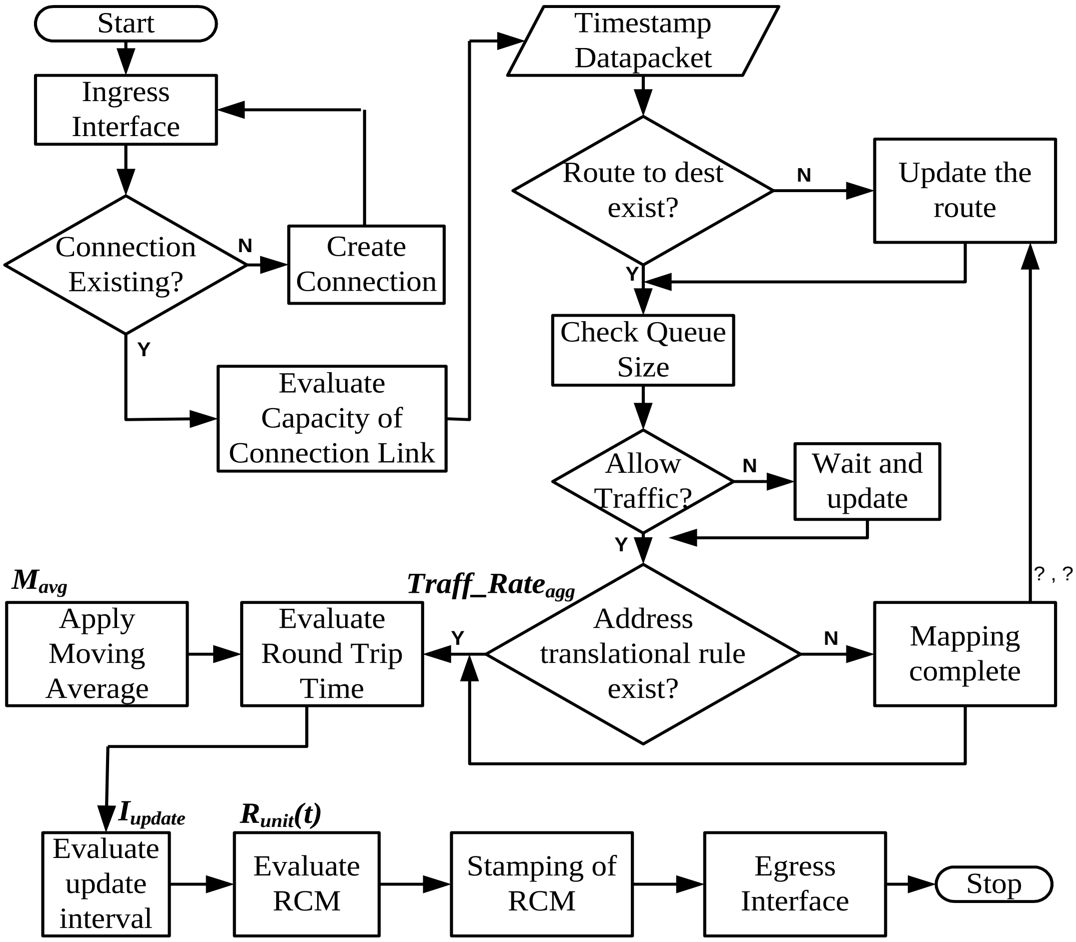

Figure 1: Flow Chart of the Proposed Mechanism

Fig.1 shows the flowchart of the proposed system. When a packet is received at the ingress interface of the router it will be checked for communication link existence, then it is being evaluated. Under feasible condition the data packets are time stamped and routed to the destination. In view of formulating RCM the RTT is considered as a function of Moving Average (M_{avg}), Aggregated traffic rate ($Traf\_Rate_{Agg}$) and Link Capacity(L_{cap}). Based on RTT and I_{update} we could arrive R_{unit} (t) which is the proposed metric RCM.

The router regularly updates R_{unit} (t) around once per round trip time. Intuitively, in order to simulate processor sharing, the router will also be designed to offer the equivalent rate to every flow, attempt to fill the outgoing communication path with established traffic, and keep the queue occupancy near to zero. Considering the deterministic traffic, the prime target of the proposed study is to understand the behavior of the future internet architecture. Considering L_{cap} as capacity of the communication link, $Traf\_Rate_{Agg}$ as estimated aggregate input traffic rate during the last update interval, the empirical remnant buffer of the channel can be computed as,

Remnant Buffer = Lcap-Traf_RateAgg (1)

The remnant buffer computed from above equation may vary according to the type of channel. As the proposed system is investigated using simulation-based study, a stability parameter is considered for better accuracy (ψ) Therefore, eq (1) becomes

$$\text{Remnant Buffer}=\psi\,(L_{cap} - T_{raf\_}Rate_{Agg}) \tag{2}$$

The proposed system however bears the bottleneck rate of the highly congestion communication link, which is overwritten by the router as it passes through the network. The receiver communicates the bottleneck rate to the sender using I_{update} parameter where it updates the interval regularly. The round trip time (RTT) carries the sender's data and is used for updating RTT. Hence, eq (2) can be modified as

$$\text{Remnant Buffer} = R = \frac{I_{update}}{M_{avg}}\,(\psi\,(L_{cap} - Traf\_Rate_{Agg})) \tag{3}$$

The above equation supports in estimating the remnant buffer used for Rate Control Metric (RCM) formulation. The attributes ψ and ρ are frequently used in the empirical evaluation of congestion control mechanism [34]. For enhancing the performance of the new metric RCM, We have introduced a novel performance attribute (NA) in our equation (4). The NA is arrived based on Queue size and M_{avg}.

$$\text{New attribute for end host} = NA = p.\frac{Queue_{size}}{M_{avg}} \tag{4}$$

The proposed new metric RCM is devised considering the current estimated link capacity. It is defined as depicted in the equation 5.

$$\gamma = 1 + \frac{R - NP}{L_{cap}} \tag{5}$$

Now, we have arrived at the proposed new metric RCM ($R_{unit}(t)$). It is defined as the product of $Rate_{last}$ (last rate which is equivalent to (t-I_{update})) and the current estimated link capacity (γ).

$$\text{RCM} = R_{unit}(t) = (t - I_{update}).\gamma \tag{6}$$

It is observed that, RCM has effective control over the stability, independence of the link capacities, the flow frequency and the network round trip delay.

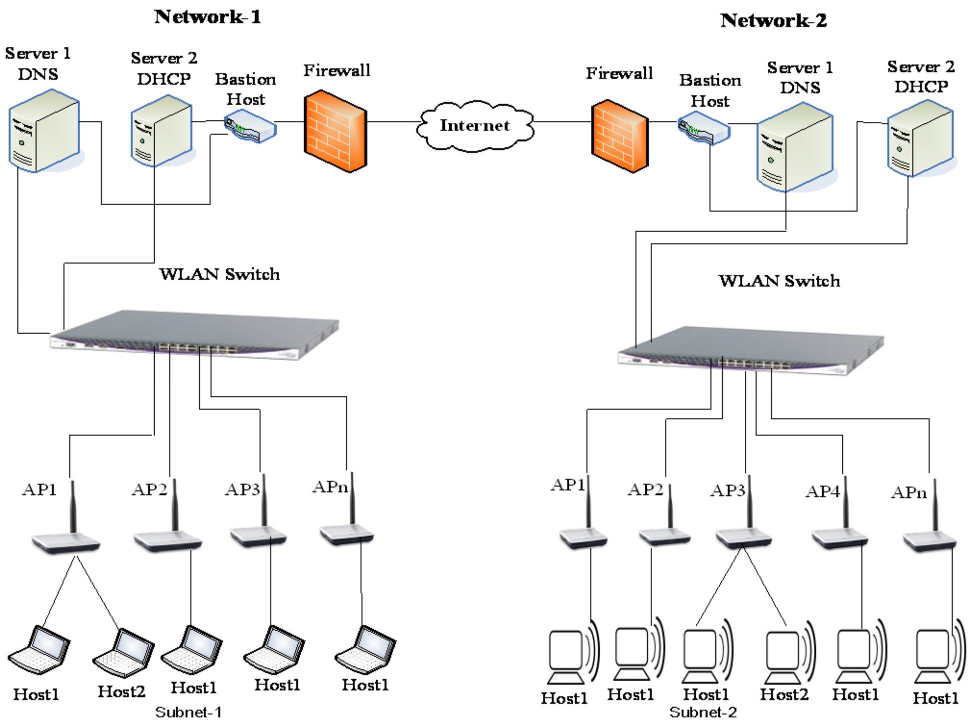

Figure 2: Distributed network

Consider, h1 be the sample period, C be the Link Capacity, T_p be the network delay, N_0 be the number of senders, and p_0 be the desired dropping probability for establishing the stability over the selected network. The window at equilibrium can be formulated as $W_0=\sqrt{}(1/p_0)$ and Round Trip Time (RTT) in equilibrium (or steady stage) can be represented as $R_0=(N_0.W_0)/C$. Similarly, queue length at equilibrium can be indicated as $q_0=C.(R_0-T_p)$, while the maximum and minimum queue length is represented as $Q_{max}=q_0+100$ and $Q_{min}=q_0-100$. Hence eigenvalue of the system matrix of the distributed networking system is represented as

$$b_0 = Max\left(E_0\left(\alpha + \beta.\exp\left(\sqrt{-1}.Freq.R_0\right)\right)\right) \quad (7)$$

In the above equation (2), E_o is the function for performing eigenvalue of system matrix, α is system matrix evaluated from number of senders N_o and Round Trip Time R_o, while β is another system matrix considering the delay due to congestion.

$$\beta = \{(-\frac{N_o}{R_o^{\ 2}}.C),2.\frac{C.R_o - q_o + q_{\min}}{C.(q_{\max} - q_{\min})}.R_o^2\} \quad (8)$$

The above equation (7) gives the value of β which is used in equation (8). The reason for finding the eigenvalue of the system matrix is to establish a probability of stability in future internet architecture. Hence, if $b_o>0$ than the distributed network can be considered as unstable. It is required to design b_o as it is highly non-deterministic in nature owing to the dynamic traffic state in future internet architecture. Hence, in order to design a stochastic model, a distributed network system is considered in its high traffic state as b_o is a non-deterministic state.

Significance of RCM

The literature review has shown that although there were multiple attempts for the purpose of controlling congestion on various types of traffic, but very few have presented a reliable congestion control technique using rate control mechanism. The proposed rate control metric (RCM) is supposed to be incorporated in router, which will have the capability of computing fairness in rate and time-stamped it in the headers of the control message. The proposed system maintains mean RTT for outbound traffic on each interface, which performs computation of rate in every control interval. The system captures the parameters like $Traf\_Rate_{Agg}$ and mean RTT, which are encapsulated in each control interval for the purpose of rate evaluation. The computation of RCM is quite faster for any types of router (g-based or n-based router), which makes the system quite cost effective and reliable in terms of controlling congestion in distributed system.

Implementation

The proposed protocol is implemented using Matlab and Java. Matlab is used for performing mathematical simulation study, while java is used for performing network test bed using TCP/IP socket in wireless environment. Network protocol analyzer (WireShark) is used for shriveling the packet flow. The proposed system is inherently fair as all flows at a bottleneck receive the same rate. The flow completion time is better than that of conventional internet system and close to what flows would have achieved if they were ideally processor shared. This is because the proposed model allows flows to jump-start to the correct rate .Even short-lived flows that perform badly under TCP (because they never leave slow-start) will finish quickly with proposed method. The proposed method allows flows to adapt quickly to dynamic network conditions i.e, quickly grabs spare capacity when available and backs off by the right amount when there is congestion, so flows don't waste RTTs in figuring out their transmission rate. There is no per-flow state or per-flow queuing. The per-packet computations at proposed protocol router are simple. The algorithm to implement RCM is as follows

Input: data from sample multimedia file
Output: Estimation of data flow rate Start

1. Input sample packet
2. Design congestion Header (14 byte)
   Con_{Header}= {$bottleneck_{rate}$(x), $reversepath_x$, RTT}
3. Design end host function
4. Read inbound packets $\rightarrow$update RTT
5. Timestamp rate in outbound packets.
6. Estimate

7. Initiate packet processing on Arrival of packet
8. Inbound_Bytes + = Data_size_Bytes
9. If (Current_data_RTT < Max_RTT)
10. $\sum RTT_{Tx}$ += Current_data_RTT
11. $\sum$ Data_with_RTT + = 1
12. Perform processing on outbound data
13. If (Data_BW_Request > Estimate_Ctrl_rate)
14. Data_BW_Request = Estimate_Ctrl_rate
15. Perform evaluation (throughout, delay, end-to-end bw)
    Stop

Result Analysis and Discussion

For the purpose of performance evaluation, the proposed metric is compared with that of Cicco et al. [33] work. Here the real time experiments were performed to evaluate the Google congestion control (GCC) and the results are published. The authors have found that the algorithm is suitable when the GCC flows access the bottleneck in isolation and when bandwidth is shared by multiple flows of data its effectiveness is reduced. This problem is addressed in our work. The proposed system is evaluated with respect to the network parameters Viz., throughput, channel capacity and delay.

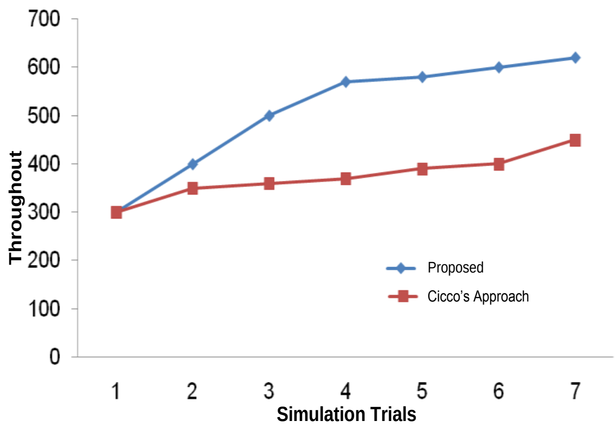

Figure 3: Performance Analyses for Throughput

Figure 3 highlights the evaluated results for the proposed system when compared with the work done in [33]. The main issue addressed in this work is sharing the bottleneck with loss-based flows and a mechanism using a multimedia application for evaluating the results. In our system we have considered a small amount of per packet processing performed in the data path and hence it has better throughput.

Figure 4 exhibits the performance analysis of the cumulative bandwidth utilization where it can be seen that proposed system can actually optimize the bandwidth to a greater extent as compared with the recent work done (Cicco et al. [33]). The prime reason behind this is that the previous work uses Google Congestion Control (GCC) algorithm which runs over the UDP and it encapsulates the audio/video frames in RTP packets. It also employs Forward Error Correction (FEC) and re-transmissions to counteract packet losses. However, in the proposed protocol, upon packet arrival the router must update counts for the corresponding output port of the running RTT sum, the number of arriving bytes, and the number of packets carrying a valid RTT. On packet departure the router overwrites the bottleneck rate carried in the packet if needed so as to optimize the bandwidth utilized.

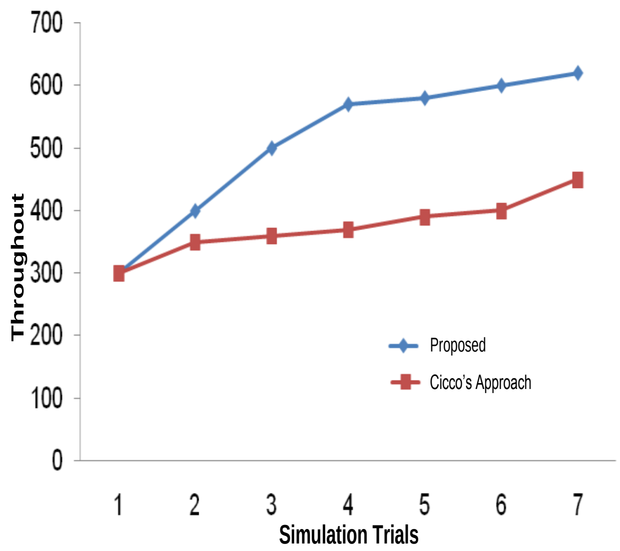

Figure 4: Performance Analyses for Bandwidth Utilization

Figure 5 highlights the performance analysis with respect to delay of the proposed system as well as the recent work done .Here the video flows controlled by the GCC get starved when sharing the bottleneck with a TCP flow, if the bottleneck capacity is less than or equal to 1000 kbps; when two GCC video flows share the bottleneck, the algorithm behavior appears unpredictable and exhibit poor fairness. However, the proposed system additionally computes the fair-share rate using RCM.

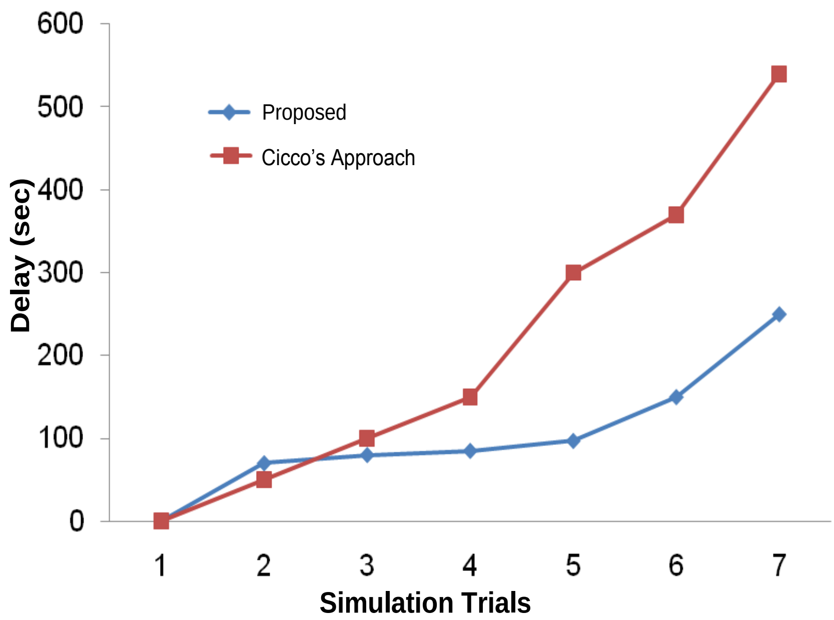

Figure 5: Performance Analyses for Delay

The rate computation requires the router to maintain estimation of average round-trip of the outgoing traffic on each interface. This is carried out based on RTT information stamped in every packet. The rate of flow is calculated once in every control interval (RTT). The aggregate incoming traffic and average RTT are gathered during each control interval and the actual rate of flow is computed with these data. When a packet arrives with the time stamping (proposed method) the router adds RTT value of packet header to the running sum (the time stamping before departure). Because of this technique it is evident that the delay in the communication is highly reduced as shown in figure 5.

Table 1 show the statistical outcome of the work where it can be seen that total time of simulation is very less with less time complexity. The error estimation is also found to be reduced from 0.6 to 0.3, and randomness is reduced from 220 to 0.2 with only 0.14 overhead of simulation stating the model to have less space complexity.

Table 1: Statistical data of simulation

Total Simulation Time	0.410963 seconds
Congestion Mitigation (hypothetical)	2.3979
Congestion Mitigation(simulation)	10.9011
Error (simulation)	0.3052
Error (Hypothetical)	0.6458
Randomness (hypothetical)	220
Randomness (simulation)	0.2061
Anticipated Packet Delivery Ratio	211.0320
Overhead (simulation)	0.1412

Threat to Validity

Internal validity

The development of Software Defined Radios and dynamic approaches to spectrum efficiency shows that cross-layer interaction will span the entire range of layered architecture. However, layering has been tremendously useful networking paradigm because it limits the interaction with the internals of the protocol at one layer with that in another. As a consequence, protocols can be designed and implementations are taken place in isolation, leading to maintainable software. Different protocols of the same layer and their implementations can be "plugged in and out" without affecting the functionality of protocols at other layers. A cross-layer algorithm by its nature destroys this useful characteristic, because each layer must make some of its internals visible and accessible to other layers. Further, it is rather brittle, because changing the protocol at a given layer or even just the implementation of the same protocol may break cross-layer interactions. Thus the explosions of cross layer methods have raised the fear of a regress to monolithic software, unmanageable and un-maintainable.

External Validity

The complexity of router design will be increased because of any additional features to the router. Because of this complexity there may be delay in router processing in addition to that power consumption may increase.

Conclusion

The proposed metric RCM supports congestion free traffic in future internet architecture. This metric enables a regulated flow at a particular device (bottle neck device) when there is a cumulative flow of data because of increased demand on the network. This metric is expected to reduce the time taken for the rate of flow of data to much less than the time taken for flow completion in existing TCP and XCP. This is because the rate controlling factor will permit flows to jump-start to the precise rate as even connection set-up data packets are stamped with the fair-share rate. The short-lived flows will finish quickly with the proposed RCM. This factor will allow the flow to adapt quickly to dynamic network conditions (i.e.) quickly grabs spare channel capacity when it is available and backs off by the right amount during congestion period. The advantage of introducing RCM in the communication network is the improvement in quality of transmission as discussed in the previous sections. This is achieved because there is no requirement to figure out the transmission rate and to record the state of flow at each queue.

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