Route Diversity: A Future For Transmission Protocols?

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Abstract—This contribution is attempting to show how a route diversity can improve the traffic behavior of a connection between two network entities. The most used transmission protocols TCP and UDP control the quasi-totality of the Internet traffic. However they are not capable to carry out all the traffic generated by the Internet as they lead to congestion and packet loss out of the quality of service required specially for critical connections that need to be treated with a higher priority. In this paper, we focus on multipath routing and multipath transport protocols that seem to be a relevant solution to avoid the drawbacks generated by the use of a single path by each connection packet. Pushing connection packets on multiple paths during all the data transaction should bring a better QoS. We mention a new protocol design for the next generation Internet. This protocol has been developed in order to avoid or reduce the congestions appearance and the high variability of the Internet traffic. We suggest a TCP protocol based on Multiple Paths MPTCP for a data transmission between two network entities in a well-defined environment. A simulator has been developed in order to evaluate the gain provided by the route diversity.

I. INTRODUCTION

The Internet supports multiple connections that start randomly in the space-time domain. Each user thereby manages its connections in respect with the network configuration in order to maintain the required quality of service (QoS). The most used transmission protocols TCP and UDP control the quasi-totality of the Internet traffic. Connections start from an emission network node with a route discovery that provides a fixed path where all data packets will be propagated towards the destination. However, the route can not stay the “shortest path” between the nodes involved in the connection during all its duration, specially if it lasts for a long time. Then for large file transfers, the connection route state can change radically from its original one. In these cases it can be benefit to switch to another path that offers a better QoS. Moreover an issue appears quickly as the same ressources are shared simultaneously by several traffic sources spread over the entire topology. We assume that each router applies the basic FIFO queuing strategy. Then a connection path can move to a congested state as a high transverse traffic crossing one of its router can vastly affect its behavior. A point of congestion can occur locally on a network router if it can not treat all packets that must cross it at their arrival rate. As a consequence, packet loss decreases drastically the QoS required by the defined connections. In fact all connection sources whose route crosses the congested node are notified to adapt their emission throughput in order to avoid the traffic jam maintenance. Then the ressource allocated for these connections are misused during the idle time needed for the congestion resorbance.

II. RELATED WORK

Traffic traces recorded over the Internet show an irregular and bursty behavior often modeled by self-similar processes. We have shown in Ref. [1] that a fixed routing protocol is responsible of the creation of congestion nodes. Afterwards they modulate traffic sources of connections that cross it in respect with the TCP protocol. Congestions statistically become more and more probable as the connection hop length grows. This represents a large issue during the design of a network and its optimization. Reducing the presence of congestions seems to be the major way to keep a desired performance over the network. N. Larrieu and P. Owezarski showed the assets of their Measurement Based approach of Congestion Control (MBCC) in Ref. [2]. Their approach relies on a real time analysis of traffic characteristics and QoS evolutions. They designed mechanisms that able to adapt their reactions accordingly. MBCC is able to cope with the large variability in traffic throughputs. Its conception aims to optimize the use of ressources and to improve the connections QoS. Their protocol uses the new TCP Friendly Rate Control (TFRC) algorithm which has been designed in order to provide a service suited for stream oriented applications requiring smooth throughput. TFRC tries as much as possible to avoid brutal throughput variations that occur with TCP because of the loss recovery. They concluded that MBCC improves the network robustness and it provides high quality services. If ressources are available, traffic sources are notified and they are allowed to send more traffic without generating congestions or packet loss. Traffic sources become pro-active and they adapt their throughput in function of available ressources. The prime purpose rests on the generation of smoother and more regular traffics. However, if no ressources are available, the gain of MBCC stays insignificant. In our approach, we consider that congestions rise from the use of a fixed routing protocol that provides a single path for each connection. Our solution lies on the routing complexity of each connection. Multipath routing and multipath transport protocols are well studied with many existing works. For instance, in Ref. [3] S. Lee and M. Gerla...
suggested an on-demand routing schemes called Split Multipath Routing (SMR) that establishes and utilizes multiple routes of maximally disjoint paths. After a route discovery process, the traffic is switched to two routes for each session. In Ref. [4], I. Díaz et al. proposed a multipath routing in combination with a Multiple Description Coding (MDC) in order to improve the performance of multimedia connections in a well connected network. The multimedia traffic is splitted up into several substreams named descriptions that are sent along different paths from the source to the destination. In Ref. [5], T. Nguyen and S. Cheung suggested the MultiTCP system, a receiver-driven, TCP-based system for multimedia streaming over the Internet. In our work, we have presented a new transmission control protocol designed for the next generation Internet and based on TCP. The MPTCP protocol based on multiple paths has been developed in order to avoid or reduce the congestions appearance and the high variability of the Internet traffic. Several non-negligible assets rise from the use of MPTCP. For a connection between two network entities, the defined traffic does not remain located on the same path but it is stretched over many routes so network resources are better used. The congestion appearance is reduced as a routes manager modifies the probability to take a path in respect with its load. This dynamic route selection implicates that, contrary to the case where the TCP protocol is used, the traffic remains more regular for the reception node involved in the connection. The routes manager supports the choice of the best paths for each connection packet in order to increase the global quality of service.

III. Main Results

Our goal is to understand how different aspects of networks as physical topology and routes on that topology, along with packet arrival models lead to congestion and other observed network characteristics as its fractal and periodic behavior. We have already implemented a simulator on the top of ns-2 in order to lay our traffic model based on log-normal distributions. We revealed in Ref. [1] topology and routing effects on TCP data transmissions between two routers linked by a fixed connection path in a well-defined network environment.

What are the reasons that lead to congestions? Users behavior or network infrastructures? We can not separate them in our model as the traffic behavior is a direct result of their mutual interactions. How does a connection attribute a route between two network entities? Without hard changes on the topology, as the disjunction or the breakdown of a router, a fixed path is allocated for all connection packets. Afterwards the source is only allowed to manage its sending rate. It is assumed that the network configuration will not change radically during the data transmission. However with the high development of multimedia applications, we more and more share large files, even surpassing many gigabits. The connection duration can consequently be spread out from many ms to many hours in function of the path bottleneck throughput. Thus the route performed at the connection start and followed by each data packet could not be optimal during all the transfer duration.

We now investigate an alternative routing method that avoids drawbacks rising from the use of a fixed path per connection. We reject the dynamic routing because it generates much more traffic that explodes with the network size. Each node must periodically informs the network of its level of congestion. We suggest to allocate a set of multiple disjoint routes per connection if they are available. This new approach allows to avoid or partially reduce the problem described above in a congested network where a fixed routing manages the data transmissions. Multiple paths enter in contest for one connection. The route followed by each data packet sent on the network will be selected in order to optimize the connection QoS. For instance in our survey we aim to minimize the connection duration.

IV. Simulator

We develop a software on the top of Scilab in order to test and validate new transmission protocols as the one describe above. Scilab is an open source platform for numerical computation [6]. TCP and UDP have already been implemented. The results are in adequacy with the ones produced by other network simulators as ns-2. Our simulator have been divided into two modules. The first one is entirely dedicated to the generation of topologies that respect the Internet ones. According to the available knowledge of the Internet topology [7], [8], hierarchical topologies can be created in respect with the required network size. We aim to efficiently generate random but “realistic” topologies by duplicating a set of representative graph metrics. We avoid to reproduce the node degree distribution. First the degree of a node is the quantity of its direct neighbors. Second the node degree distribution is the probability that a randomly selected node is k-degree. For instance, Fig. 1 shows a random topology composed by 230 nodes.

![Fig. 1. Random graph created by our topology generator. 230 nodes are linked together in respect with the node degree distribution found on the Internet.](image)

The second module allows to study the traffic dynamic on a topology in the discrete-time domain. Thus the appearance of congestion can be observed in real time. Our
target is to evaluate the connection behavior when congestion phenomena occur in function of its transmission protocol. The other network connections are managed by each source node. Each network node can play this role as the session start time could be selected randomly. The node behavior is defined by a distribution of delay between connections that follows Poisson. The connection data size distribution is taken uniform, but a new release allows to select the exponential one. For the connections different from the survey one, 90% of the generated traffic is governed by TCP, the remainder uses UDP. For instance, in Fig. 2 we have generated a random topology composed by 100 nodes. A TCP connection between the nodes \( N_d \) and \( N_f \) is studied.

![Fig. 2. Topology used for simulations. 100 nodes are linked together. The node \( N_d \) sends packets towards the node \( N_f \). Each network node manages its connections.](image)

We collect 2 important parameters that permit to characterize the network effects on the defined connection: the inter-arrival time process at the destination node (see Fig. 3) and the travel time process of connection packets that reflects the time needed by each connection packet to move from the emission node towards the destination node (see Fig. 4).

![Fig. 3. Inter Arrival Time process of connection packets at the destination node \( N_f \) provided by a simulation. The sequence presents a high variability.](image)

We have constructed a new protocol based on TCP spread on multiple disjoint paths. Several distinct routes between 2 network entities, named “diamonds” [9] can be discovered by the flood approach, the n-modified Dijkstra’s algorithm or the modified Bellmann-Ford algorithm. Each method is already implemented in our software.

V. CONCLUSION

In conclusion, our simulator permits to study and visualize the topology and routing effects on a defined connection. We aim to select the best strategy providing a better route diversity in order to increase the connection QoS. A large wave of simulations have been launched in order to find and validate many contestant strategies. We will analyse the results as soon as they will be available. We expect a considerable gain because the route diversity can be applied quickly in a well-connected network as the Internet.

REFERENCES


