SCTP vs. TCP: Comparing Packets Loss Rate of Transport Protocols in Best-Effort Networks

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Abstract :

With its new features such as multi-homing, multi-streaming, and enhanced security, the Stream Control Transmission Protocol (SCTP) has become a promising candidate to join UDP and TCP as a general-purpose transport layer protocol. However, the challenge faced by SCTP in the best-effort network, where the network does not provide any Quality of Service (QoS) for the upper layer. The objective of this research are to measure the performance of both TCP and SCTP over a wired network in term of packet loss rate in a network environment that has SCTP with UDP traffic, and then compare SCTP and TCP performance results. All experiments have been done through the network simulation NS2. The result indicates that the SCTP traffic had a highest packet loss rate than TCP traffic under similar conditions.

Keywords: SCTP, TCP, packet loss rate, NS2

1. INTRODUCTION

Until now, there have been two general purpose transport protocols widely used for applications over IP networks: UDP and TCP. Each provides a set of services that cater to certain classes of applications. However, the services provided by TCP and UDP are disjoint, and together do not satisfy ideally the needs of all network applications. The Stream Control Transmission Protocol (SCTP), designed to bridge the gap between UDP and TCP, addresses shortcomings of both[1, 2].

SCTP was originally developed by the IETF SIGTRAN Working Group to carry telephony signalling messages over IP networks for telecommunications and e-commerce systems. With continued work, SCTP evolved into a general purpose transport protocol that includes advanced delivery options[1, 3].

One commonality between TCP and SCTP is that they both try to prevent that the network they operate in becomes congested. Network congestion is a phenomenon that was discovered in the late 80's when the Internet began to grow large. Basically, the problem of congestion occurs when parts of the network infrastructure become over-utilized by large amounts of traffic and, therefore, are unable to deliver any data to its users. In both TCP and SCTP the mechanisms that try to prevent congestion are coupled with the mechanisms that are used to ensure that no data is lost in the communication process, which called reliability mechanisms. This is done because packet loss historically has been seen as a sign of network congestion, and it is therefore considered natural that these mechanisms work together[4].

The paper is organized as follows: Section 2 provides a brief introduction of a transport layer protocols while in section 3, SCTP has been discussed. Section 4 includes the simulation setup, scenarios and the simulation results. Finally, the conclusion summarized in section 5

2. TRANSPORTLAYER PROTOCOLS

In the TCP/IP model, the transport layer provides services to the application layer, accepts the data from the application layer and adds its header, then forwards to the lower layers for further processing. The transport layer provides efficient, reliable, cost effective services such as: Multiplexing/Demultiplexing, reliable data transfer, flow control and buffering, congestion control. The parameters that affect the quality of service of the transport layer are connection establishment delay, connection establishment failure probability, throughput, transit delay, residual error ratio, priority, resilience [5].

Many protocols have been proposed to meet the requirements of the transport layer, the most known are User Datagram Protocol (UDP) , Transmission Control Protocol (TCP) , and Stream Control Transmission Protocol (SCTP) .

UDP is one of the protocols that are widely used in the internet. The services provided by UDP are unordered delivery of packets, connectionless service, full duplex connection and message boundaries preserving. TCP has been proposed to overcome drawbacks of UDP and now has been widely used in internet for data transfer and for many applications. SCTP, which uses features of both TCP and UDP, was defined in the aim of removing drawbacks of TCP[5].

3. SCTP PROTOCOL

The Stream Control Transmission Protocol (SCTP)[6-8] is a Transport Layer protocol, serving in a similar role as the popular protocols: TCP and UDP. Indeed, it provides some of the same service features of both, ensuring reliable, insequence transport of messages with congestion control like TCP, and preserving data message boundaries similarly to UDP. However, differently to TCP and UDP, SCTP offers such advantages as multi-homing and multi-streaming capabilities.

SCTP was developed by the IETF Signalling Transport (SIGTRAN) working group with the goal of overcoming limitations of TCP. The main difference to TCP is the multihoming mechanism and the concept of several streams within a connection. Where in TCP a stream is referred to as a sequence of bytes, an SCTP stream represents a sequence of messages and these may be very short or long. Table 1 compares SCTP's services and features with those of TCP and UDP.

Table 1: Comparison of SCTP, TCP, and UDP [9]

Protocol Feature	SCTP	TCP	UDP
State required at each endpoint	yes	yes	no
Reliable data transfer	yes	yes	no
Congestion control and avoidance	yes	yes	no
Message boundary conservation	yes	no	yes
Path MTU discovery and message	yes	yes	no
Fragmentation Message bundling	yes	yes	no
Multi-homed hosts support	yes	no	no
Multi-stream support	yes	no	no
Unordered data delivery	yes	no	yes
Security cookie against SYN flood attack	yes	no	no
Built-in heartbeat (reachability check)	yes	no	no

3.1 Multi-homing mechanism

Using the multi-homing mechanism, an association between two endpoints can use multiple IP addresses (interfaces) rather than a single address (as in TCP)[10]. Even though using multiple IP address only one address is used for data exchange[11]. The other IP connections are used when initial connection breaks or when there is a heavy loss. Multi-homing mechanism is very helpful in case of networks with frequent failures.

3.2 Multi-streaming mechanism

The term multi-streaming refers to the capability of SCTP to transmit several independent streams of chunks in parallel (refer to figure 1), within a connection or association[12]. In a TCP connection, only one single data stream is allowed: all of the information must be passed through that one stream. SCTP allows multiple simultaneous data streams within a connection or association. Thanks to the SCTP multi-streaming

mechanism, only one stream is affected if there is a loss[13].

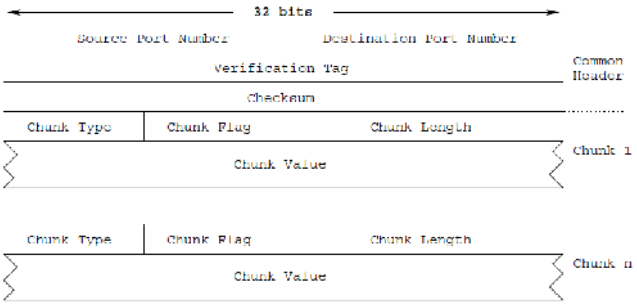

Figure 1: Packet format of SCTP

4. SIMULATION SETUP AND RESULTS

To conducting the experiments, a simple simulation has been designed using NS-2 Network simulator. NS-2 is a free software simulator that is very suitable for the protocol simulation. NS-2 is a discrete-time simulator whose development began in 1989 with the development of REAL Network Simulator. Probably one of the main reasons for its success is the fact that the distribution has General Public License (GPL) condition that drives the free development of the same. For this work , the latest version a network simulation was NS2-35 installed on Ubuntu 11.10 machine with core i3 (2.2 GHz) of processor.

4.1 Simulation Scenarios

A set of experiments was carried out using five constant bit rate CBR, SCTP and TCP traffic sources. In this setup, the node started transmission with predefined delays to simulate the real environment. Two simulation scenarios have been considered for measure a packet loss rate for both of SCTP and TCP separately.

4.1.1 Measure a packet loss rate of SCTP scenario

The network consist of ten source node, ten destination node along with two routers. The five source SCTP node would generating FTP over SCTP traffic destined for the SCTP node on the other portion of the network, while the CBR over UDP generated by the rest of the nodes would be used to simulate the actual networking environment. Figure 2 shows the network set-up to measure a packet loss rate of SCTP.

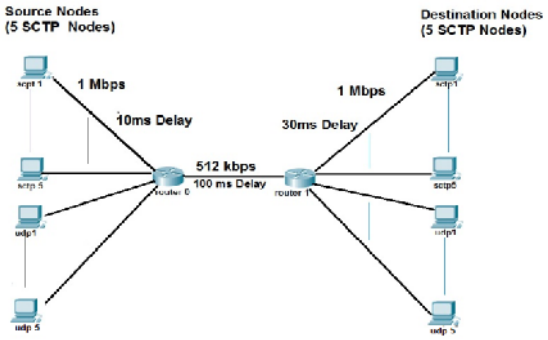

Figure 2: Simulation Topology with SCTP Nodes

4.1.2 Measure a packet loss rate of TCP scenario

Similar scenario was applied to measure a packet loss rate of TCP except for replace TCP nodes in place of SCTP nodes. Here, also FTP CBR traffic would be carried over UDP, respectively. Figure 3 shows the simulation network set-up.

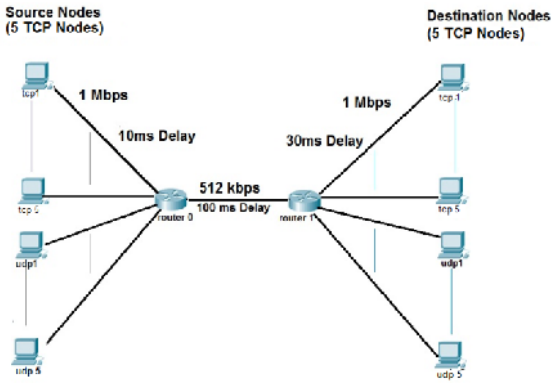

Figure 3: Simulation Topology with TCP Nodes

In both experiments, the source nodes generated FTP and CBR packets of 1000 kB at the rate of 1000 packets per second. The node router0 and router1 are the buffer-limited drop tail (FIFO) routers so that packets arrive at the routers are dropped when the buffer is full. (router0-router1) is the bottleneck link. The simulation process time is 1000 ms, and the nodes generated packets with a delayed start ordered to simulate a controlled real environment as shown in table 2.

Table 2: Nodes Traffic Start Transmitting Time

Protocol	Nodes	Start Transmitting Time
SCTP/TCP	Node 1	10 ms
	Node 2, Node 3	15 ms
	Node 4, Node 5	20 ms
UDP	Node 1, Node 3	30 ms
	Node 2	35 ms
	Node 4	40 ms
	Node 5	50 ms

4.2 Packet Loss Metric

Packet loss is an error that can occur in data network when the data in the network is congested. This means that packets of the data are not able to be transmitted as well as in a normal case, or the fail to reach the destination. Therefore, this can cause significant problems in systems that use Voice Over Internet Protocol (VOIP) as an application. In addition, packet loss is an important parameter affecting the performance of the networks[3]. Packet loss is expressed as a percentage of the number of packets lost to the total number of packets sent [7].

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The Packet loss can be caused by many factors such as: the signal degrading, hardware problems and networks that are in demand more than necessary (that creates a state of congestion) which males the network susceptible to packet loss in sending or receiving. Packet loss is one of the important performance metrics usage for measuring performance[3].

4.3 Simulation Results

This paper focus on comparing SCTP and TCP performance based on packet loss metric. Table 3

display a summary of simulation results related to SCTP traffic packet loss rate per nodes. Packet loss rate range was between (2.451 %) at node 2 and (2.775 %) at node 4. The variance between these values was (0.324) with average (2.543 %).

	Sent	Dropped	(%)
Node 1	13406	75	0.559
Node 2	11856	85	0.717
Node 3	9373	94	1.003
Node 4	11826	94	0.795
Node 5	8518	113	1.327

Table 3 : Summary of Packet Loss Rate per Node for SCTP

Table 4 displays a summary of the simulation results for TCP traffic packet loss rate per node. The packet loss rate range was between (0.559 %) at node 1 and (1.327 %) at node 5. The variance between these two values was (0.768) with an average of (0.88 %).

Table 4 : Summary of Packet Loss Rate per Node for TCP

Node Number	Number of Packets Sent	Number of Packets Dropped	Packet Loss Rate (%)
Node 1	8288	205	2.473
Node 2	7385	181	2.451
Node 3	7758	192	2.475
Node 4	7278	202	2.775
Node 5	7839	199	2.539

loss rate of SCTP was three times more than the average packet loss rate of TCP. However, the SCTP packet loss rate was affected by the network environment, where it showed a variation in packet loss ranging between increasing and decreasing.

The results of both experiments have been compared against each other in order to evaluate the rate of packet loss of both protocols. Figure 4 shows this comparison. It is obvious that SCTP traffic has registered a higher packet loss rate than TCP traffic under similar conditions. The average of the packet

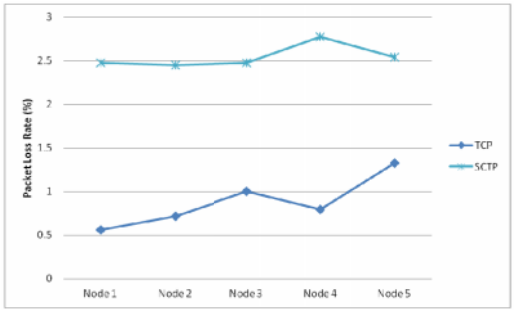

Figure 4: The Packet Loss Rate of SCTP and TCP Stream

5. CONCLUSIONS

This study focused on the SCTP protocol by measuring its performance in the best-effort network, where the network has other protocols in term of packet loss rate. All experiments have been done by simulate two network scenarios through the network simulation NS2. The first experiment was

carried out by exchanging the data between the source and destination of SCTP nodes through routers where the second network carried out TCP nodes when the network had a

different protocol, UDP. The network consist of ten source node, ten destination nodes, along with two routers. The first five source node would generate FTP over SCTP traffic destined for the SCTP nodes (SCTP nodes replaced by TCP nodes in second experiment), while the CBR over UDP generated by the rest nodes would be used to simulate the actual networking environment.

The result shows that the average packet loss rate on SCTP traffic was (2.543 %) where the rate on TCP traffic was (0.88 %). This indicates that the packet loss rate of SCTP was three times more than TCP. On the other hand, the packet loss rate of TCP increases as the number of packets in the network was increased with variance of loss rate per node equal to (0.768). However, SCTP shows more stability with (0.324) as a variance of loss rate per nodes under similar conditions the TCP has.

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مقارنه نسبه فقدان الحزم لبرتوكولات النقل TCP وSCTP

في شبكات Best-Effort

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الملخص:

اصبح برتوكول SCTP مرشحا واعدا للانضمام الى بروتوكولات النقل الشهيرة TCP و UDP كبرتوكول نقل للأغراض العامه بالاعتماد على المميزات الجديدة لبرتوكول النقل SCTP مثل تعدد الروابط و تعدد الارسال وأمنيته المحسنه. مع ذلك فان هناك تحدي يواجهه برتوكول SCTP في شبكات best-effort , حيث لا توفر أي (QoS) للطبقة العليا. الهدف من هذا البحث هو مقارنه اداء كل من بروتوكولات النقل TCP وSCTP باستخدام شبكة اتصال سلكية بالاعتماد على مقياس النسبه المئوية لفقدان الحزم (Packet Loss Rate) .

تم اجراء جميع التجارب باستخدام محاكي الشبكات NS2. و قد اظهرت النتائج ان بروتوكول SCTP له اعلى نسبه لفقدان الحزم من بروتوكول TCP في ظروف مماثله.