Simulation of TCP, UDP and SCTP with constant traffic for VOIP services

Pallavi Gangurde*, Sanjay Waware*, Dr. Nisha Sarwade**
*(M.Tech.(Electronics and Telecommunication)
Department of Electrical Engineering, VJTI, Mumbai, Maharashtra, INDIA
** (Associate Professor, Department of Electrical Engineering, VJTI, Mumbai , Maharashtra, INDIA.)

ABSTRACT

In recent years, Voice over IP (VoIP) has gained a lot of popularity. Signaling being an important part of VoIP has been addressed by the (IETF) SIGTRAN working group to meet Quality of Service as given by Public Switched Telephone Network (PSTN), so that both PSTN and VoIP can co-exist and work together in a seamless manner.

SIP (Session Initiation Protocol) for VoIP signaling is a communication control protocol capable of running on different transport layers, e.g., TCP, UDP or SCTP. Today's SIP application is mostly operating over the unreliable transport protocol UDP. In lossy environment such as wireless networks and congested Internet networks, SIP messages can be lost or delivered out of sequence. The SIP application then has to retransmit the lost messages and re-order the received packets. This additional processing overhead can degrade the performance of the SIP application. Therefore to solve this problem, the researchers are looking for a more appropriate transport layer for SIP. SCTP, a transport protocol providing acknowledged, error-free, non-duplicated transfer of messages, has been proposed to be an alternative to UDP and TCP. The multi-streaming and multi-homing features of SCTP are especially attractive for applications that have stringent performance and high reliability requirements and an example is the SIP proxy server.

In this research, we have analyzed the performance offered by SCTP for SIP message delivery in the perspective of historic research work as well as determined call setup time using UDP and SCTP by simulating SIP traffic in Network Simulator- 2 (ns-2). We also evaluate TCP, UDP and SCTP traffic with constant bit rate of traffic through ns-2.

Keywords - Simulation, TCP - SCTP - UDP Comparison, VoIP

1 INTRODUCTION

VOICE over Internet Protocol (VoIP) is a technology that allows users to make telephone calls using a broadband Internet connection instead of an analog phone line. VoIP holds great promise for lowering the cost of telecommunications and increasing the flexibility for both businesses and individuals. VoIP leverages existing IP-based packet-switched networks to replace the circuit-switched networks used for voice communications since the invention of the telephone. Voice over IP (VoIP) applications are gaining an ever increasing popularity in the Internet community, favored by the massive deployment of wireless access technologies. For instance, more than eighty million users have already subscribed to Skype [6], the most popular VoIP commercial application for personal use, roughly 10% of which are estimated to be simultaneously online at any time.

The paper has been organized as follows: Section 1 gives an overview of VoIP, Section 2 briefs on Background of the various protocols used for transmission in VoIP Networks. Section 3 gives a discussion on the Research carried on these protocols and the results found and we will conclude the paper in Section 4.

2 BACKGROUND

2.1 Session Initiation Protocol and Signaling

Session Initiation Protocol has gained a lot of popularity for carrying signaling information for multimedia applications. Session Initiation Protocol (SIP), developed by the Internet Engineering Task Force (IETF), is a control protocol which creates, modify and terminate session with one or more participants and this session can be an Internet call, multimedia conference session, or multimedia distribution. The IETF RFC 3261 defines this protocol [4]. SIP is a lightweight protocol because it requires very few messages, called methods, for managing a basic session. These methods are INVITE, BYE, ACK, REGISTER, OPTIONS, CANCEL and INFO. In the case of video, audio, or multimedia session, the session information will be used for setting up an RTP stream, running on UDP that in turn operates on IP. SIP is not dependent of transport layer, i.e., it can be used over UDP, TCP, or SCTP. In SIP case when using UDP, messages may be lost or received out of sequence. SIP, therefore, uses its own reliable mechanisms via retransmission timers, command sequence (CSeq) numbers, and positive acknowledgments. The SIP-T detailed are specified in RFC 3372 [5].

2.2 TCP, UDP and SCTP

SCTP was developed by IETF SIGTRAN Working Group to carry telephony signals on IP from networks like SS7 [7]. The main propose of design consideration was to beat the limitations of TCP and UDP as signaling carrier. Since its close similarities with TCP in clogging and flow control it has undergone a lot of studies and investigations in terms of performance evaluation and judgment with TCP. SIP is most fruitful signaling protocol now days. SIP can operate on UDP, TCP or SCTP. UDP provides unreliable and untrustworthy datagram service [1], and relies on the application layer for error control, detection of message repetition, duplication, and retransmission of lost messages.

On the other side, TCP gives error and flow control [2]. However, its strict byte order delivery creates performance issues. It also suffers from other downsides as mentioned in [8]. SCTP overcomes some of the limitations of TCP and SCTP also provides a reliable datagram transport mechanism. SCTP also provides features which required by a SIP system such as multi stream message passing for performance, cookie mechanism for security, and multi homing for fault tolerance and high availability [8].

The selection of protocols is influenced by the fact that SCTP, TCP and its all variants form one category of protocols (reliable, have flow and congestion control, connection oriented) whereas UDP is a protocol without connection orientation, without flow and congestion control. Thus UDP has minimum of overhead, but retransmissions have to be implemented in application layer which could be a major disadvantage. So this paper makes a comparison between UDP and SCTP.

3 RESEARCH

3.1 NS2 introduction

NS2 (Network Simulator, version 2) [3] is a set of object-oriented network simulator, developed by UC Berkeley. It can simulate the real network structure and characteristics in the network structure, there are router, link with the end point in the network's characteristics, have packet delay or packet drop. Put simply, NS2 is a OTcl script interpreter, it is a script written by the user (OTcl Script, describes the simulated network) to interpret, produce simulation results, thus analysis of the results, or through the NAM tool program to simulate the process of visual show, to simulate the situation to gain a better understanding.

3.2 Simulation Scenarios

In order to analyze the packet loss, delays and average throughput offered by transport protocols some scenarios are simulated. One is to study the effect of traffic on delays and second to observe the effect of various packet loss conditions on delays.

3.3 Transmission Protocols Simulations

3.3.1 TCP

The packet size is 1000 byte and channel capacity is 0.2 Mb. The tcl script runs over ns and we found a .trace file and a nam file then added some awk script and found required trace file. Then got the results and generate the graphs shown in fig 3.2.

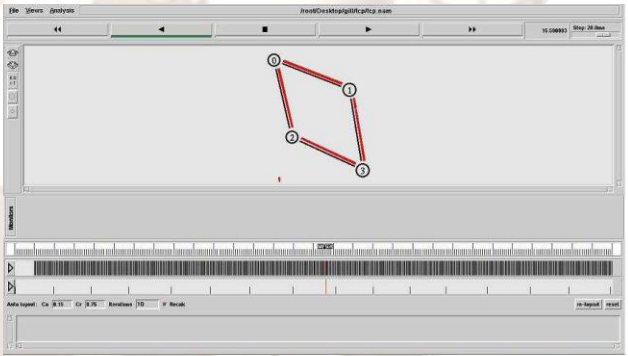

Figure 3.1: TCP.nam

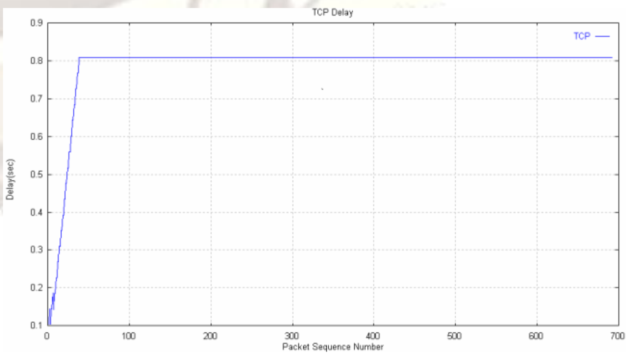

Figure 3.2: TCP delays

The time at which a message is en-queued at transport layer at node 0 is subtracted from the time when it is delivered to application at node 3 to get delay for a particular request.

Number of packet read: 1405
Number of packet sent: 712
Number of packet received: 693
Packet lost: 19
Average delay of packets: 0.787624
Variance of delay is: 0.010288
% Throughput: 97.331460

3.3.2 UDP

The packet size is 1000 byte and channel capacity is 0.2 Mb. The tcl script runs over ns and we found a .trace file and a nam file then added some awk script and found required trace file. Then got the results and generate the graphs shown in fig 3.4.

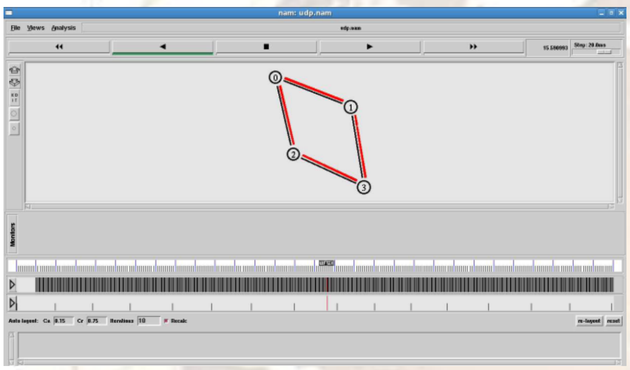

Figure 3.3: UDP.nam

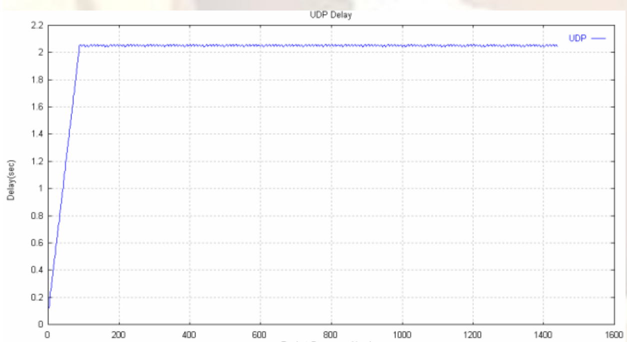

Figure 3.4: UDP delays

We can see here that the delay is increasing in term of time w.r.t TCP that 0.82 and in UDP it is upto 2.05.

Number of packet read: 2292
Number of packet sent: 1569
Number of packet received: 723
Packet lost: 846
Average delay of packet: 1.930832
Variance of delay is: 0.142850
% Throughput: 46.080503

3.3.3 SCTP

The packet size is 1000 byte and channel capacity is 0.2 Mb. The tcl script runs over ns and we found a .trace file and a nam file then added some awk script and found required trace file. Then got the results and generate the graphs shown in fig 3.6.

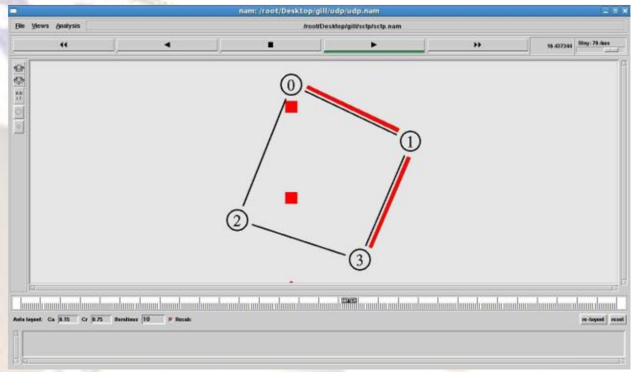

Figure 3.5: SCTP.nam

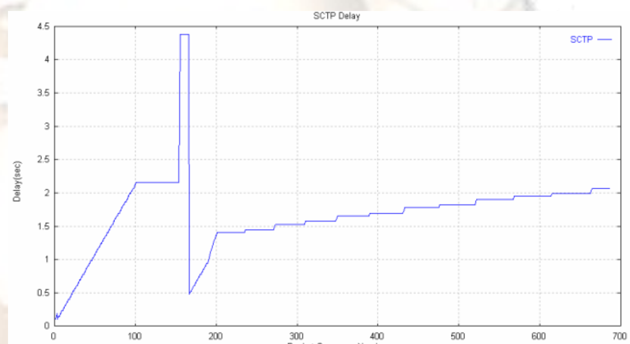

Figure 3.6: SCTP delays

Number of packet read: 1437
Number of packet sent: 750
Number of packet received: 687
Packet lost: 63
Average delay of entries: 1.680858
Variance of delay is: 0.315453
% Throughput: 91.6

We can see that delays are increasing up till packet no 100 that is 2.3 then constant to 150 then 150 to 160 it boosted up 4.4 then minimize to 0.5 then it is increasing slowly till last packet.

3.3.4 Comparison of TCP, UDP and SCTP

The graphs below show the comparison of delays of TCP, UDP and SCTP respectively. Also table 1 gives a brief comparison w.r.t. various parameters.

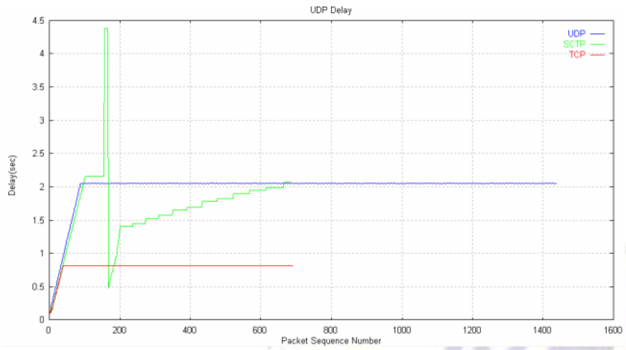

Figure 3.7: Comparison of TCP, UDP and SCTP

Table 1. Comparison of TCP, UDP and SCTP

Protocol	**Sent**	**Receive**	**Packet loss**	**Avg. Delay**	**Variance of delays**	**% Throughput**
TCP	712	693	19	0.787	0.01028	97.331
UDP	1599	723	846	1.931	0.1428	46.080
SCTP	750	687	63	1.681	0.3154	91.6

Depending upon the bandwidth, if we do more results, packet lost decrease. And SCTP looks in a better manner.

4 CONCLUSION

The above simulations were conducted to compare the performance of TCP, UDP and SCTP with traffic analysis. We kept the packet size of 1000 bytes and run the simulation with constant bit rates over all transport protocols and we had the channel capacity 0.2 Mb. As shown in figure 3.7, on comparing the results it can be seen that TCP is performing the best with least number of packet loss as compared to that of SCTP and that of UDP. SCTP is best effort because its multi homing and multi association but its packet delivery acknowledgement is time consuming and researches are required to be done in future so that SCTP would be more advantages over TCP.

Also more or less satisfactory performance is observed in competing traffic with UDP and SCTP, although UDP has an edge being free from all sorts of transport overheads. But in the case of packet loss, where SCTP suffers a bit of delay variations UDP suffers from the effect of application layer retransmission. With increasing effect of packet loss the performance of SCTP undergoes a severe degradation.

UDP on the other hand keeps a consistent behavior as the packet drop has no effect on its application. Same rate of packet loss in SCTP causes packets drops at transport layer and delays increase in a consistent manner. So it is very easily observable that SCTP has no comparison with UDP. Since internet traffic is burst in nature and it is difficult to predict traffic density and loss rates, in the same way it is not simple to give a clear verdict regarding choice of a transport protocol, but the kind of performance given by SCTP makes it clear that SCTP cannot be used with confidence foe a VoIP traffic, since even a loss rate of 0.2% degrades it performance significantly compared to UDP.

Serious considerations are required to be made in the future regarding protocol redesign that can be best suitable to carry VoIP signaling messages.

REFERENCES

[1] J Postel User Datagram Protocol IETF RFC 768, August 1980.
[2] J Postel Transmission Control Protocol IETF RFC 793, September 1981.
[3] KO, Chih-Heng、CHENG, Jung-Hsiang、HSIEH, His-Kun、HUANG, Wen-Hsiang “Computer Network experiment – implementation with simulation tool -NS2” XBOOK MARKETING CO., Ltd. 2005.
[4] ROSENBERG, J, et al: 'SIP Session Initiation Protocol'. KFC3261, IETF, June 2002.
[5] A Vemuri, J Peterson et al RFC 3372 SIP-T Context and Architectures September 2002.
[6] Skype, http://www.skype.com, continuously updated.
[7] IETF Signaling Transport Working Group: “sigtran”, www.ietf.org/html.charters/sigtran-charter.html
[8] R Stewart, Q Xie, K Morneault, C Sharp, H Schwarzbauer, T Taylor, I Rytina, M Kalla, L Zhang, and V Paxson Stream Control Transmission Protocol IETF RFC 2960, October 2000.