

Simulation Based performance evaluation and comparison of wired VoIP services over UDP and SCTP protocol

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Abstract

In current years, Voice over IP (VoIP) has increased a lot of acceptance. Waving being significant part of VoIP has been deployed by the (IETF) SIGTRAN working team to meet QOS as given by Public Switched Telephone Network (PSTN), since both PSTN and VoIP can cooperate and work composed in a seamless manner. VoIP messaging is a communication control protocol applicable of running on different transport layers protocols, e.g., Transmission control protocol (TCP), User Datagram protocol (UDP) or stream control transmission protocol (SCTP). Today's VoIP application is widely running over the unreliable transport protocol (UDP). In loss situation such as wireless networks and overfilled Internet networks, VoIP messages can be lost or delivered out of order. The VoIP application then has to resend the lost messages and re-arrange the accepted packets. This extra processing overhead can damage the performance of the VoIP service. Therefore to resolve this problematic, the scholars are looking for a more applicable transport layer for VoIP protocol. SCTP, a transport protocol giving acknowledged or recognized, error-free, not duplicated transmission of messages, has been intended to be an alternative to UDP and TCP. The multi-streaming and multi-homing advantages of SCTP are mainly attractive for applications that have strict performance and high reliability requirements. In this project, I have examined the performance offered by SCTP and UDP for VoIP message delivery in the viewpoint of old research works as well as determined call situation time using UDP and SCTP by simulating constant bit rate traffic in Network Simulator- 2. Typically the Transport Layer Protocols, UDP and TCP are used for transmission of data over the internet protocol based network like Internet. TCP gives a reliable way to transfer the data but its reliable mechanism results in rise in delay. UDP absence in providing any acknowledgment mechanism and it does not give any congestion control technique also. However the unreliable feature of UDP results in less delay in message transfer.

Keywords: Simulation, TCP, SCTP, UDP, Comparison, VoIP, performance

1. INTRODUCTION

Voice over Internet Protocol is a technology that permits users to make telephone calls through a broadband Internet connection in its place of an analog phone line [1]. VoIP holds great capacity for lowering the cost of telecommunications improve and growing the flexibility for both businesses applications and individuals. VoIP influences existing IP-based packet-switched networks to substitute the circuit-switched networks used for voice communications since the creation of the telephone [6]. VoIP applications are gaining an ever growing popularity in the Internet community, preferred by the huge deployment of wireless access technologies [7]. This paper has been prepared in following form: part 1 gives an indication of VoIP, part 2 briefs on Contextual of the different protocols used for transmission in VoIP Networks. Part 3 gives an argument on the Research approved on these protocols and Section 4 gives the results and discussion.

UDP offers an unreliable facility with no guaranteed distribution and no congestion control technique [5]. So there is lack of quality of service in UDP. But it is used for video streaming due to its less overhead. SCTP is a reliable and connection oriented transport layer protocol that offers ordered data delivery and also there is no error in that data [4]. Another important advantage provided by SCTP is Multi-homing. In multi-homing, an end point can have provision for multiple IP addresses or interfaces. SCTP also provides Multi-streaming advantage in which, the data to be sent can be breakdown into streams. Multi-homing and Multi-streaming can give good bandwidth utilization.

This project is for superior understanding of how VoIP operates on underlying layers of the network [5]. The wired VoIP topology is built through using network simulator-2. In case to make the simulation more accurate as in the real world, the background traffic is added. The simulation is preliminary start with UDP (User Datagram Protocol), and secondly goes with Stream Control Transmission Protocol (SCTP). To have good understanding on the difference between the VoIP on UDP and SCTP network, packet loss, throughput, delay and jitter are calculated and analyzed. To better imagine the measured output, the simulation results are plotted in to graph [11].

2. UDP, SCTP AND TCP APPLICATIONS

SCTP was established by IETF SIGTRAN Working Group to transport telephony signals on internet protocol from networks [2]. The basic advantage of design consideration was to solve the shortcomings of TCP and UDP as signaling carrier. Since its close likenesses with TCP in congestion and flow control it has practiced a lot of studies and surveys in

terms of performance evaluation and decision with TCP. VoIP is most productive signaling protocol today. VoIP can work with UDP, TCP or SCTP. UDP gives unreliable and untrustworthy datagram facility [1], and relies on the application layer for error control, identification of message repetition, duplicate, and retransmission of lost packets. On the other case, TCP provides error and flow control [1]. Though, its firm byte order delivery makes performance subjects. It also suffers from other weaknesses as mentioned in [2]. SCTP solves some of the problems of TCP and UDP. SCTP also offers a consistent datagram transport mechanism. SCTP also provides advantages which required by a VoIP system such as multi stream packets pass for performance, cookie mechanism for security issue, and multi homing for fault tolerance and high obtainability. The choice of protocols is prejudiced by the fact that SCTP, TCP and all alternatives form one group of protocols while UDP is a protocol lacking connection orientation, lacking flow and congestion control [3]. Thus UDP has min of overhead. Packet loss has an effect on all TCP, SCTP and UDP applications. But the only difference is that the TCP application provides a packet guarantee which means it has a packet retransmission technique for the packet which is lost during its transmission. This research mainly concerned with the UDP, and SCTP application. Which includes the following examples: Domain name system (DNS), voice over IP (VoIP), Trivial files Transfer Protocol (TFTP), and online multimedia games etc. Mainly my implementation is highly concerned with VoIP. A VoIP is a telephone set designed precisely for use in a voice over IP (VoIP) system by changing standard telephone audio into a digital format that can be transferred over the Internet, and by changing incoming digital phone signals from the Internet to normal telephone audio [8]. SCTP is similar like TCP in different ways. They are both support unicast connection-oriented protocols that offer reliable transport, in-sequence packet delivery and s congestion control. SCTP does offer some functions not found in TCP. SCTP is message-oriented whereas TCP is stream-oriented. [3] SCTP can handle or switch multiple simultaneous streams and multiplexed streams where TCP can contain only a single stream of data per connection. SCTP's stream-aware connection control is one of its most prominent features. SCTP also offers for multi-homing in that the end points can use multiple IP addresses for the connection. The SCTP connection endpoints can use IP addresses from different ISPs for network-level fault tolerance. If, during the connection, one of those ISPs were to fail, the connection would just use the IP address from the operational ISP for the connections.

3. SERVICES AND FEATURES PROVIDED BY SCTP AND UDP

Features and Services	SCTP	UDP
Connection Oriented	Yes	No
Un ordered Data Delivery	Yes	Yes
Ordered Data Delivery	Yes	No
Congestion Control	Yes	No
Flow Control	Yes	No
Multi-streaming	Yes	No
Multi-homing	Yes	No
Reliable data transfer	Yes	No

Table 1: Services and Features provided by SCTP and UDP

3.1 Connection orientation of SCTP

SCTP is connection oriented transport layer protocol. In SCTP, the connection is called association. When a process at host A wants to share a data from a Process at host B, then first of all an association is created between each other [3]. After that data is shared between them, and finally the association is terminated.

3.2 Ordered data delivery of SCTP

Data delivery out of order is allowed in SCTP. In case of a stream is affected in SCTP then only the affected stream is blocked temporarily while the other streams will be allowed to pass or deliver. SCTP does not guarantee wait for the messages to be ordered numerically at the end receiver, in fact, it process them on their order of arrival, no concerns what order they have. These messages are delivered reliably by SCTP. This

feature of SCTP of simply transient the data minimizes the overhead of message reordering on the server [2].

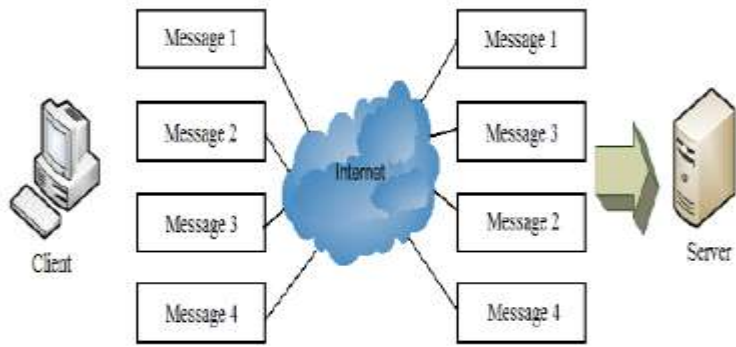


Fig 01: Unordered data delivery

3.3 Reliability in SCTP

SCTP provides a reliable transmission facility like TCP [5]. SCTP guarantees that the data is delivered to the network in sequence and without error. Reliable delivered in SCTP is also performed by sensing or detecting when the data is corrupted, duplicated or discarded. [2].

3.4 Multi-Streaming function in SCTP

SCTP permits multiple data streams in an association. All the streams in suggestion are linked to that specific association but they are independent. A stream number is given to each stream, which is prearranged in to the packets flowing through the association. If there is some packet loss within a stream then only that specific stream will be blocked (until the lost packets are re-transmitted) without affecting other streams in an association. This problem is called head-of-line blocking [4]

4. CONCEPTS AND TERMS OF SCTP AND THEIR RELATION

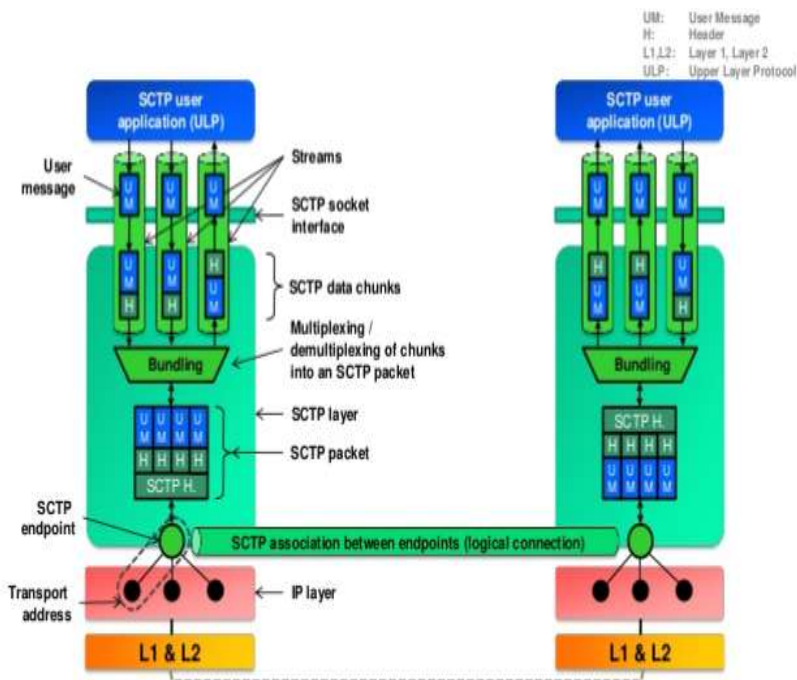


Fig 02: The SCTP model [12]

Association: The SCTP link is the logical relationship between two SCTP endpoints and the building packs many chunks user data chunks and SCTP control chunks in to an SCTP packet. Chunk: unit of information within an SCTP packet chunks contain either user data user data chunk or SCTP control (control chunk). Each chunk has its own header (chunk header).Endpoint: an SCTP endpoint is the addressable logical endpoint of an SCTP relationship. An SCTP endpoint represents exactly one SCTP port number, but may contain many transport addresses in case of multi-homing. A stream is a logical channel transporting in order application messages. Streams are unidirectional. If an application requests bidirectional stream, it must open one outgoing and one incoming unidirectional stream and treat them together as a bidirectional stream.

4. PURPOSE

Measuring and evaluating network traffic dynamics between end hosts has offered the basis for the existence of many different network protocols and systems. The main purpose of this project is evaluate and compare the loss of packets, latency or delay, jitter, and throughput of over the UDP and SCTP protocol over VoIP with scenario the network simulation-2. Finally plot the results from the simulation in to gnu-plot graph to visualize the result for more clarity.

5. SCOPE

The scope the project is to accurately measure and evaluate the performance of VoIP over the UDP and SCTP protocols with independent scenarios for each protocol which includes total loss of packets, jitter, latency, and throughput with both protocols on end-to-end communication of the transport layer.

6. LITERATURE REVIEW

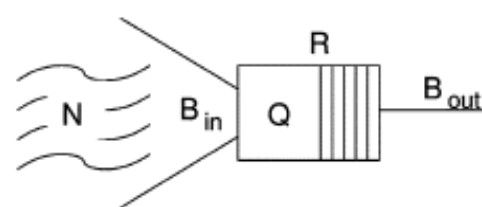
This part is based on a brief summary of the related work. The quality of VoIP sessions is assessed by using WiMAX verified and the transport protocols measured are UDP. The quality of WiMAX technology has been revealed in [1]. The network presentation of WiMAX is related with ADSL to check that which technology offers better video streaming. The performance of WiMAX and ADSL is assessed in terms of packet loss, delay, jitter, and throughput. Modulation systems in Mobile WiMAX have been assessed. Diverse modulation systems like BPSK, QPSK, 16-QAM and 64-QAM are compared with TCP and with different variations of TCP in terms of presentation Metrics like throughput and packet loss ratio. In earlier work, the presentation of transport layer protocols is examined by using VoIP applications. So in this thesis work, the performance of SCTP and UDP is analyzed by using CBR traffic over VoIP [2]. Several studies comprise the use of loss measurements to estimate network characteristics such as bottleneck buffer size and cross traffic strength [1]. We are also guided by Paxson's recent work [2] in which he supporter's rigorous standardization of network measurement tools. ZING is a tool for measuring end-to-end packet loss in one way between two participating end hosts [3]. ZING sends UDP packets at Poisson-modulated intervals with fixed mean rate and time. Savage developed the STING [3] tool to measure loss rates in both forward and reverse directions from a one host. STING uses a clever scheme for measuring a TCP stream to measure loss. Allman *et al.* explained how to estimate TCP loss packets rates from passive packet suggestions of TCP transfers taken close to the sender [3]. Throughput quality of SCTP is analyzed over WLAN in [2]. The quality is analyzed by using Varsity number of hops between sender and receiver and with Varsity SCTP window sizes at receiver side. The quality of SCTP is analyzed by through streaming video over CDMA2000 wireless topology. The performance is evaluated by analyzing quality of video accept using a specific buffer size. A study has been showed to evaluate the quality of SCTP and UDP for video traffic over Wi-Fi in [4]. Quality of Transport Layer Protocols for video traffic over fixed WiMAX is analyzed in [2].

7. PROBLEM SPECIFICATION

This project is for noble understanding of how VoIP operates in underlying on transport of the network to compare and analyze the wired network parameters which includes loss of packets, throughput, jitter and latency. The wired VoIP topologies are built by through NS2. In order to make the simulation more accurate as in the real world, the background traffic is included. The simulation is conducted with UDP and SCTP protocols. To have realistic understanding of the difference between the VoIP on UDP and SCTP protocol, packet loss, throughput, delay and jitter are calculated and analyzed. To better visualize the calculated results, the simulation results are plotted in to graph representation.

8. PROPOSED SYSTEM

The goal of the study is to understand the comparison VoIP on the wired network over the UDP and SCTP protocols how to accurately measure loss packets, throughput, jitter and latency in the wired network [3]. This is done through the ns-2 simulation tool. It creates a trace file which contains the records all about the transmission of the packets after the complete transmission. Whenthe trace file created, apply the AWK which is used to extract data from the trace file, it is power full tool used in Ubuntu to extract the measure characteristics which includes the loss of packets, jitter, throughput, and the amount of received packets.



Loss of packets in the router can caused due to limited buffer capacity, slow processor, and when the incoming or input bandwidth greater than the output link bandwidth. The above diagram shows: Fig: 03 A simple system representation and model of loss characteristics under consideration. It is Simple system model. With N flows on input links with collective bandwidth B_{in} strive for a single output link on router R with bandwidth B_{out} where $B_{in} > B_{out}$. The output link has Q of buffer capacity or temporary storage capacity.

9. NETWORK MODEL WITH UDP PROTOCOL

In my implementation, I assumed to use G7.11 codec which is a broadly use audio codec nowadays [3].This codec specifies with a 64kbps bit rate and with packet size of 160 bytes. Node 0 & Node 1 will be my VOIP traffic and Node 3 & Node 4 will be my background. There are two way of traffic for both background and VoIP and with the UDP protocol. The background traffic will be generating at a constant bit rate of 128kbps. The bandwidth

restricts for the routers (Node 4 & Node 5) is set at the same bitrate as the background traffic state [4]. The topology for the VoIP setup is shown in Figure 1, which has blue and red representing two-way traffic between the VoIP clients. Red and yellow represents two-way traffic between the background sources. The queue buildup occurs at the center link, serving as a bandwidth bottleneck.

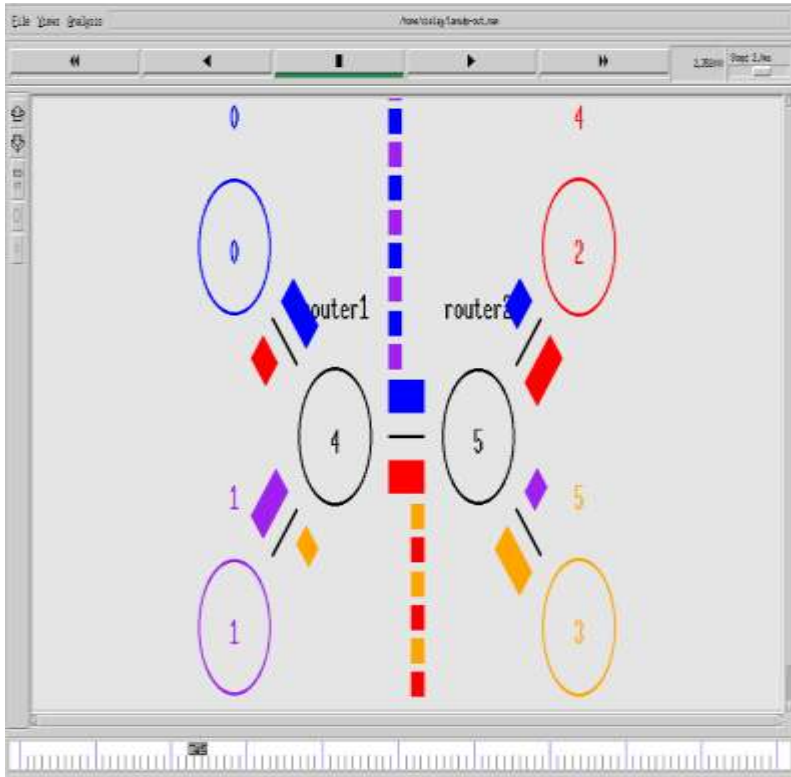


Fig 04: wired VoIP over UDP protocol

10. NETWORK MODEL WITH SCTP PROTOCOL

In this scenario, it designed like the UDP topology. This codec specifies with a 64kbps bit rate and with packet size of 160 bytes. Node 0 & Node 1 will be my VOIP traffic and Node 3 & Node 4 will be my background. There are two way of traffic for both background and VoIP and with the SCTP protocol. The background traffic will be generating at a constant bit rate of 128kbps.

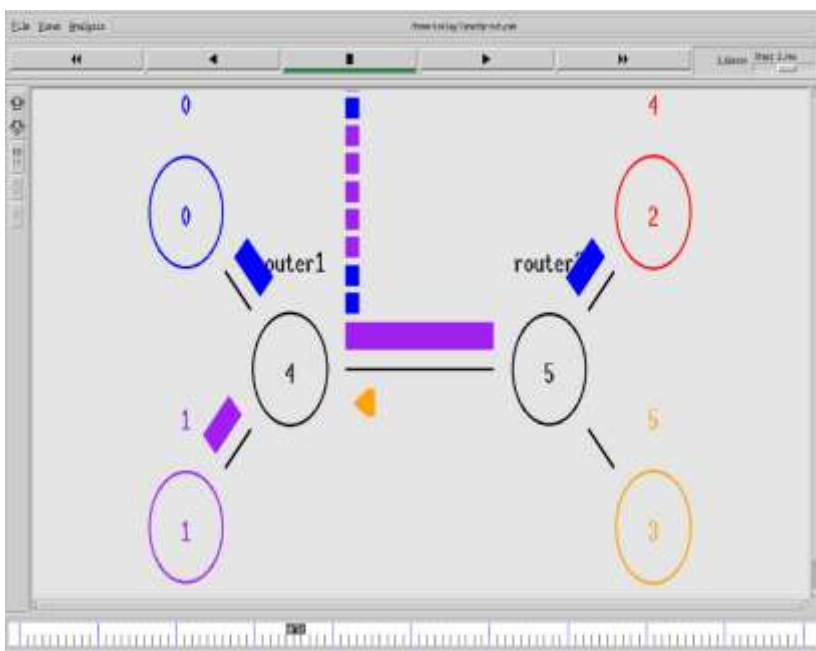


Fig 05: wired VoIP over SCTP protocol

11. SYSTEM FLOW CHART FOR THE SIMULATION OF BUFFER SIZE STRATEGY

System flowcharts are a method of presenting how data flows in a system and how decisions are complete to control events [3]. Note that system flow charts are very like to data flow charts [3]. Data flow charts do not contain decisions, they only just show the path that data takes, where it is held, managed, and then output. A flowchart is a kind of diagram that denotes an algorithm, workflow or process, showing the steps as boxes of various types, and their order by linking them with arrows [6]. This graphic representation demonstrates a solution model to a given problem. Flowcharts are used in examining, designing, recording or managing a process or program in various fields.

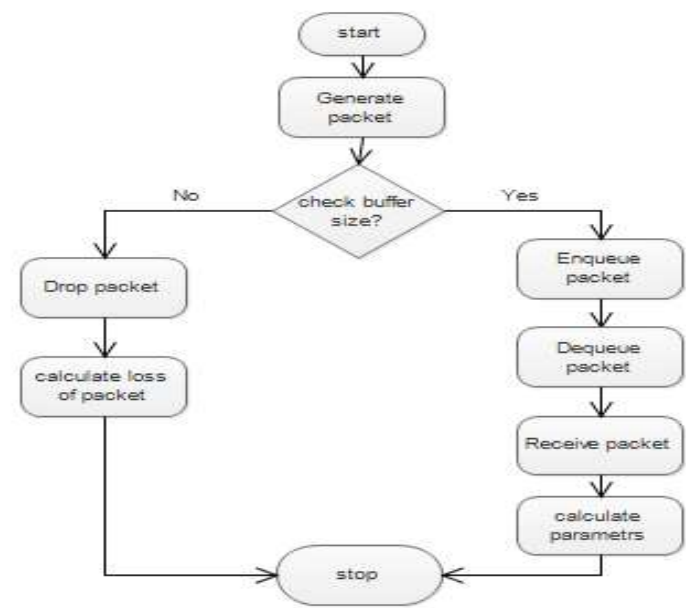


Fig 06: System flow chart for the simulation of buffer size strategy

12. PERFORMANCE CALCULATIONS

Throughput

Throughput is the mid value of effective packets permit over the communication link in a second. It calculated in bytes per second. In the trace file, “r” denotes receive in normal trace file. With old-style analog phone amenities, substituting them with Voice ended Internet Protocol VOIP, one of the examinations becomes allotting the right amount of bandwidth to your Internet phone capability. [13] How much of your current bandwidth is desired for good voice calls. This is inquiry we are requested every day by our customers. The bandwidth that our VOIP phone capability needs contingent on the number of concurrent calls you lack to make and common description, as well as proposed speeds for best performance. Best quality voice calls are the normal today but reliable excellence does necessitate some effort. One way to measure your VoIP volume is expending the Phone.com voice over internet protocol test. We can plot a graph from this trace file but we cannot plot a graph from the original trace file which represents the throughput or loss of packets generally which represent network parameter. The instantaneous throughput can be calculated as:

$$\text{Inst. throughput} = \frac{\text{The number of succeded packets reach on destination}}{\text{sec}}$$

The instant throughput will yield a graph exhibition the amount of information conservative by the destination node over individual second. This is valuable for assessing the immediate possessions of the contextual traffic on the pre-existing VoIP traffic [6]. The ordinary throughput wills outcome a single value exhibition the Midvale throughput for the entire duration of the simulation. The representation is as following:

$$\text{Average throughput} = \frac{\text{Total number packets reached on destination node}}{\text{simulation time}}$$

Packet loss

Packet loss occurs when one or more packets of data transporting through a network flop to reach its endpoint. Packet loss is primarily happened due to network congestion. Packet loss is derived as a percentage of packets lost with detail to packets directed. The collective packet loss refers to the entire number of packets failed in the simulation [11]. VoIP packet loss occurred when a vast amount of traffic on the network causes dropped packets. This hints to penalties in dropped conversations, a delay in getting the voice communication, or insignificant noise on the call. Packet loss within facts networks is both public and predictable. Large amount of data protocols, in fact, custom packet loss so that they know the weakness of the network and can decrease the number of packets they are transfer. When placing strict traffic on data networks, it is important to fix the quantity of packet loss in that network. Cisco Systems has been placing business-critical, time-sensitive traffic on data networks for numerous years, starting with Schemes Network Architecture (SNA) [9] traffic. Through protocols such as SNA that do not stand packet loss well, you need to shape a well-engineered network that can order the time-sensitive data gaining of data that can switch delay and packet loss. When putting voice on data networks,

it is significant to build a network that can positively transport voice in a consistent and timely manner. Also, it is supportive when you can use a mechanism to brand the voice somewhat genuine to periodic packet loss. Cisco Systems industrialized numerous qualities of service (QoS) tools that allow administrators to classify and achieve traffic through a data network. If a data network is healthy engineered, you can save packet loss to a minimum.

$$\text{cumulative packetloss} = \frac{\text{total of packet failed to reach the destination}}{\text{simulation runtime}}$$

Latency

Latency is an extent of time delay occurred in a scheme. [9] In this simulation delay is calculated in taking the time change between when packet is sent from basis node and when it touches its destination. The packet will be recognized by its packet id as it movements through each node in the network. [4] Latency or delay is the time that it essentials a packet to create its way over a network end to end. In telephony rappers, latency is the measure extent of time it needs the talker's voice to touch the listener's ear or end receiver. Huge latency values do not essentially damage the sound quality of a phone call, but the output can be a lack of organization between the talkers such that there are reluctances in the speaker' interactions. Generally, it is identified that the end-to-end latency must be less than 150 ms for best superiority of phone calls.

Understanding these plans makes it likely to test network latency with known levels of network deployment. To methodically calculate latency, it is good to model latency throughout the day in regular intermissions.

Instantaneous latency=Receive time end node-send time of sending node

This latency can be plotted graphically, wherever the end-to-end delay (in seconds, y-axis) for a careful packet is posted at the time it stayed known by the end node (x-axis) The Average Latency is a one value, thoughtful from the formula given below: cumulative

$$\text{Average Latency} = \frac{\text{the total summation of instantaneous latency}}{\text{simulation time}}$$

The normal latency delivers a single value which can be naturally used to carry out performance comparisons between different situations.

Jitter

Jitter is an improper name given to IP packet delay difference; [17] it is widely used in electronics and telecommunication. It is unexpected deviation from true periodicity of expected periodic signal in the network. Jitter is a standard problem of the connectionless networks or packet switched networks and connection oriented networks which mean the circuit switched network. Due to the evidence is separated into packets each packet can foldaway by a different path from the emitter to the receiver. Jitter is strictly the measure of the inconsistency over time of the latency across a network. Real time infrastructures (for example VoIP) usually have quality difficulties due to this effect. Overall, it is a problem in slow-speed relations or through congestion. It is expected that the surge of QoS (quality of the service) techniques like importance buffers, bandwidth reservation or high-speed networks (100Mb Ethernet) can decrease jitter problem in the future though it will keep on existence a problem for a long time. The finest solution is to custom jitter buffers. A jitter buffer is essentially to allocate a small buffer to obtain the packets and provides it to the receiver with a minor delay. If small amount of packet is not with in buffer (it is lost or it has still not arrived) when it is vital it is not taking into consideration. Characteristically in IP telephones (hardware and software) buffer sizes can be modified. If jitter buffers it become high turns out in less packet loss but extra delay. A decrease turns out in less delay but extra packet loss.

$$\text{Instantaneous Jitter} = \text{Present latency} - \text{earlier latency}$$

The average jitter can be calculated as:

$$\text{average jitter} = \frac{\text{sum(Present latency} - \text{earlier latency)}}{\text{simulation time}}$$

Instantaneous Jitter is will plotted into graph over the simulation runtime, and each data point will be plotted to its particular time in the Instantaneous Latency graph. This lets investigation of the stability of the network as the traffic pattern continually changes throughout the simulation runtime.

13. Simulation with NS-2

NS-2 is an object oriented simulator [13], written within C++ and with an OTcl interpreter as a frontend. The simulator chains a class hierarchy in C++ (also called the compiled order in this document), and a similar class hierarchy within the OTcl simulator. The two hierarchies are closely related to each other; from the end-user's perspective, there is a one-to-one communication between a class in the understood hierarchy and one in the gathered hierarchy. The foundation of this order is the class Tcl Object. Clients generate new simulator objects through the simulator; these objects are instantiated within the translator, and are closely mirrored by a consistent object in the compiled hierarchy. The understood class hierarchy is automatically well-known through methods defined in the class TclClass. User instantiated objects are reflected through approaches defined in the class TclObject. There are other orders in the C++ code and OTcl scripts; these other hierarchies are not revealed in the manner of TclObject. Ns use two different languages because simulator has two different kinds of things it desired to do. On other hand, full simulations of protocols needs a systems programming language which can efficiently function bytes, packet headers, and implementation algorithms that run over large data sets. For these tasks run-time speed is very important and turn-around time (run simulation, find bug, fix bug, recompile, re-run) is less important [13].

The Simulator applied in this research is Network Simulator- 2. This is the most extensively used episode driven simulator and is open source [6]. NS-2 is a distinct occurrence simulator future at networking research. NS-2 brings extensive provision for simulation of UDP, TCP, MAC, Ad-hoc routing, Sensor networks, multicast protocol over wired networks.

Trace file is an output data file from NS-2 simulation. Its file extension is .tr. I only use several types of actions in trace file. They are "+" symbolize "En queue", "-" symbolize "De queue", "r" symbolize "receive", and "d" denote "drop".

1	2	3	4	5	6
Abbr.	Time Stamp	Source node	Destination Node	Packet type	Packet size
7	8	9	10	11	12
Flags	Flow Id	Source address	Destination Address	Sequence number	Unique Packet Id

Table: The typical traces file structure

NS-2 has of the subsequent components:

- NS –network Simulator or Emulator
- Nam – Network Animator for Graphical demonstration of Ns result
- Pre-processing – This encompassed whichever handwritten TCL scripts or Topology producer
- Post-processing – Trace file will examining through Perl,TCL,AWK,MATLAB.

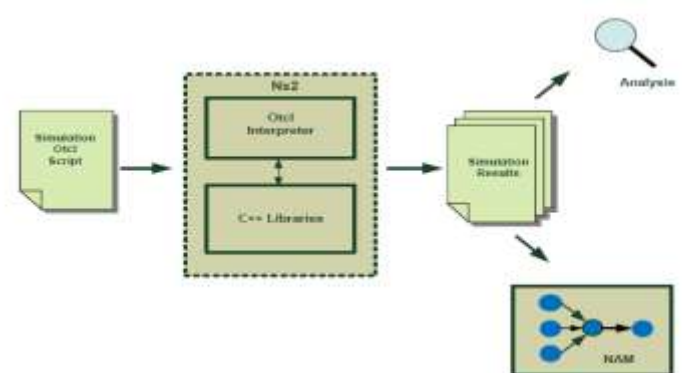


Fig 07: Structure of NS-2 [4]

14. RESULTS AND DISCUSSION

The left side is wired VoIP results of the UDP, and the right side is SCTP simulation results. The performance of wired VoIP network is compared for UDP and SCTP protocols. The background traffic starts by producing packet at time difference of 0.02ms into the simulation.

THROUGHPUT

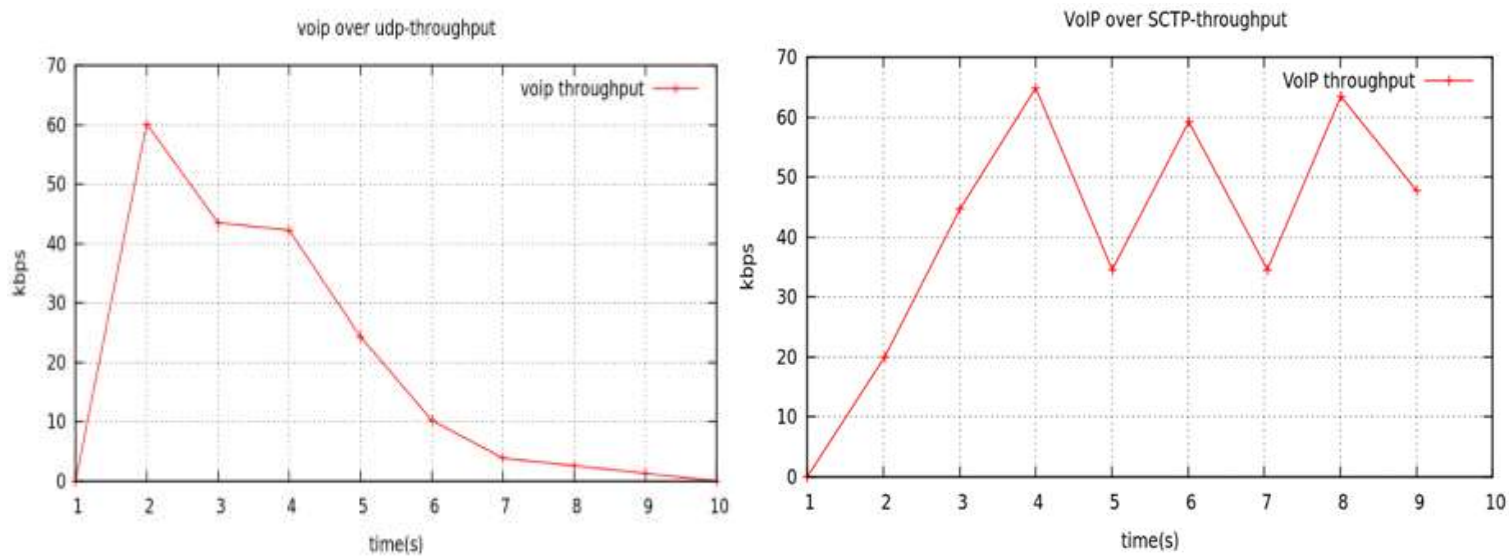


Figure 08 - Throughput - Wired VoIP over UDP VS SCTP

The subsequent resulting of the wired network VoIP between the UDP and SCTP protocols offers different outputs. As observed from the figure that results of the throughput, with the background traffic the throughput ranges 60kbps for the UDP network protocol but for the SCTP protocol it ranges around 65kbps, with assuring and delivering reliable voice traffic between the nodes. When used in the context of throughput in communication networks, such as Ethernet or packet radio, VoIP, throughput or network throughput is the rate of successful packets or messages delivery over a communication channel or Throughput is the number of messages that successfully delivered per unit time.

PACKET LOSS

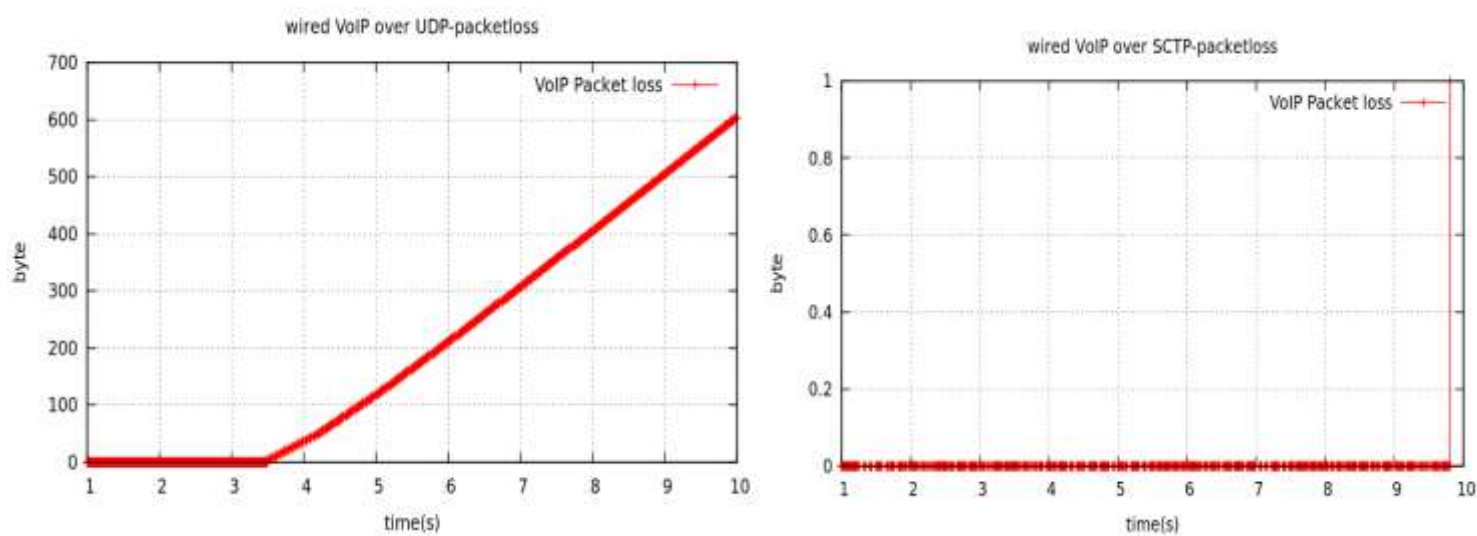


Figure 09- Packet loss - Wired VoIP over UDP VS SCTP

The packet loss is significantly more than for wired VoIP under the UDP protocol. In case of the SCTP protocol it is less than the UDP protocol. From the figure we observe that it almost provides no loss in case of the SCTP Protocol but in case of the UDP protocol the loss of packet strictly increases starting from around 3.5ms to the end of the transmission and from 1 to 3.5ms there is no loss of packets because in this demonstration I have designed the protocol with FIFO queue therefore due to limited buffer size in the queue it starts loss after some time.

LATENCY

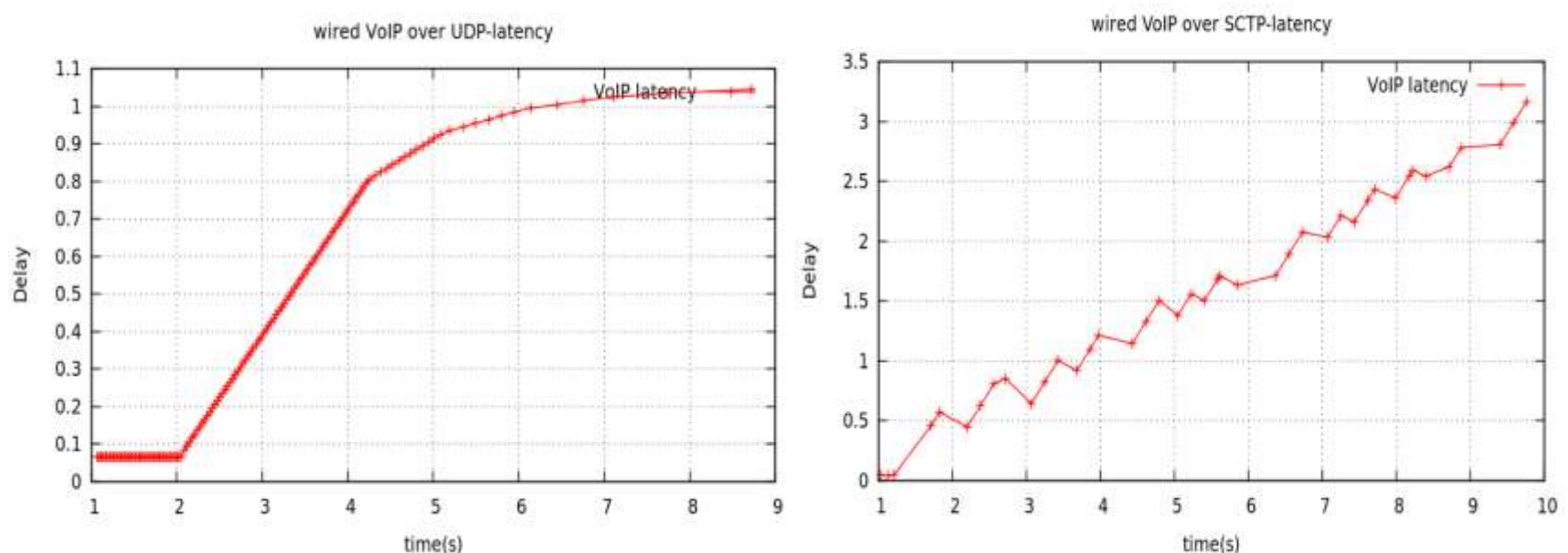


Figure 10- Latency - Wired VoIP over UDP VS SCTP

Latency is the time difference between the moment a voice packet it transmitted and the moment it reaches in its endpoint. It of course hints to delay and finally to echo. Network latency is an expression of how much time it consumes for a packet of data to get from one chosen point to another. In some environments latency is calculated by sending a packet that is returned to the sender, the round-trip time is reflected the latency. In VoIP terminology, latency denotes to a delay in packet delivery. From the demonstration we observe that the latency is high for SCTP protocol it reaches around 3.5 sec in maximum which is much more than the UDP protocol in case of the UDP protocol it reaches around 1.5 sec to maximum.

JITTER

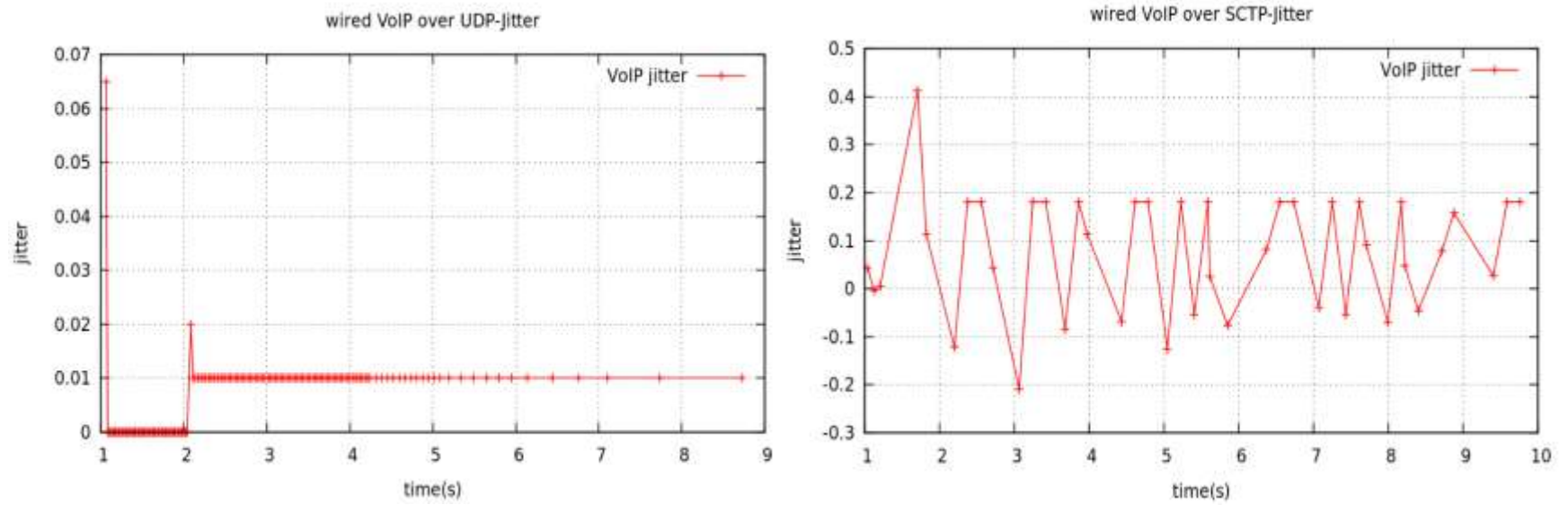


Figure 11- Jitter - Wired VoIP over UDP VS SCTP

Jitter is defined as a difference in the delay of received packets. The sending side sends packets in a continuous stream and spaces them consistently apart. Because of network congestion, improper queuing, or configuration errors, the latency between packets can differ instead of remaining constant, as shown in the figure. In case of the UDP protocol the jitter it reaches to maximum of 0.02 sec but in case of the SCTP protocol it reaches maximum of 0.04 sec which it matches with the theoretical concept.

The following table summarizes the output of the simulation. The average values can be used to compare the performances between wired networks for both on UDP and SCTP protocols.

Wired VoIP	UDP	SCTP
Throughput	13.696	41.0204
Latency	0.518272	1.56898
Jitter	0.00976636	0.0772927

Table 3: average Simulation Results of Wired VoIP

15. CONCLUSION

The objective of the project is to assess the performance change between wired VoIP over UDP and SCTP network protocols. The situations are formed to contain variability of real-world conditions, which includes for both UDP and SCTP network protocols, and with the adding of background traffic [5]. The outcomes are decisive for the characteristics of throughput, packet loss, latency, and jitter. In near proximity, the wired VoIP network, this had a fixed delay of 5ms for everyone with duplex links. Though, the delay and jitter is unbalanced for SCTP networks. From throughput, jitter, latency and packet loss I conclude that SCTP is better than UDP in case of the throughput and loss of packets it provides much more when we compare with UDP, but in case of latency and jitter it high for SCTP can considered as shortcoming. There is no doubt in the fact that SCTP is going to be the feature of VoIP and many other network technologies. But since this technology is under the process of evaluation so it may take some time for it to replace the older technologies like UDP and TCP. Generally SCTP is very high in throughput which means that it 3 times better than the UDP, but in case of latency UDP is better with the value of 3 times better than the

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UDP but the acceptable latency standard is 0 to 150ms in any VoIP application which means it is acceptable latency. And when we see jitter it is high in case of SCTP protocol which means that the variation of latency but it is with in the acceptable range of standard.so SCTP protocol is the future VoIP application rather than the UDP and TCP on the transport layer protocol.

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