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QOS PERFORMANCE STUDY OF REAL-TIME TRANSPORT PROTOCOL OVER VOIP

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ABSTRACT

In recent years, Voice over IP (VoIP) has gained a lot of popularity and become an industry favorite over Public Switching Telephone Networks (PSTN) with regards to voice communication. This paper work consists of creating a VoIP network and testing for its known faults. Through this paper, we get a better understanding of the underlying layers of the network and see if and where improvements can be made. In implementation stage the Real-time Transport Protocol (RTP) packets for VoIP applications had been sent and compared with TCP/UDP packets to obtain results which are mainly related to Quality of Service (QoS) factors. The attained result approved that RTP consider to be better to reduce a packet loss than UDP and also approved that UDP/RTP are most reliable because they had a very small delay and jitter in contrast with TCP. Hence, we find that, UDP/RTP are more balance and prefer than TCP in real-time applications such as VoIP.

Keywords: Transport Control Protocol (TCP), UDP, RTP, VoIP.

INTRODUCTION

Voice over Internet Protocol, which is familiarly known as Voice over Internet Protocol (VoIP), is the technology which transmits the voice packet over the Internet by using packet switching technique. This technology is one of the most emerging technologies in the field of telecommunications. Today, VoIP applications are becoming very common among the young generation and among those whose computers are connected to the Internet. Text chat using mobile phones and the Internet is tedious for some people, instead voice chats is considered to be a quicker and comfortable way of communication. VoIP services enables people with lower income to communicate with their family members, friends and loved ones regardless of their geographical location [1].

Problem statement

VoIP is supported by several protocols and it has its own set of characteristics which are not common in other types of applications. These include the use of Transport Control Protocol (TCP), User Datagram Protocol (UDP) and, Real-time Transport Protocol (RTP). VoIP can use several types of codec to provide poor, average or high QoS, depending on the type of connection and technology available. VoIP is considered to be sensitive to delay, jitter and packet loss.

Switching a communication path from one end point to another in network is a critical challenge for end to end delay, jitter and packet loss in sensitive applications such as VoIP considering voice quality between networks implementing varies transmit protocols.

Proposed solution

To maintain VoIP quality during voice transfer process, Simulate the propose scheme in any network simulator. Evaluate and calculated the QoS enhancement parameters and finally compare the proposed scenario of RTP network with TCP and UDP. And estimate a more reliable protocol to transfer a voice.

LITERATURE REVIEW

The VoIP systems can be built up in numerous forms and these systems include mobile units, conferencing units and telephone handsets. Along with this equipment of end users [2], VoIP stands for Voice over Internet Protocol. It is also referred to as IP Telephony or Internet Telephony. It is another way of making phone calls, with the difference of making the calls cheaper or completely free. The 'phone' part is not always present anymore, as you can communicate without a telephone set.

VoIP has a lot of advantages over the traditional phone system. The main reason for which people are so massively turning to VoIP technology is the cost. VoIP is said to be cheap, but most people use it for free. Yes, if you have a computer with a microphone and speakers, and a good Internet connection, you can communicate using VoIP for free. This can also be possible with your mobile and home phone.

There are many ways of using VoIP technology. It all depends on where and how you will be making the calls. It could be at home, at work, in your corporate network, during a travel and even on the beach. The way you make calls varies with the VoIP service you use [3].

Real-time application

To add a special challenge for the deployment of a VoIP network, integration and detailed information about the characteristics of each, is to maintain the position of the phone. This information is often performed manually in a spreadsheet or database of some type. This information is required to keep on working hard and expensive, making the time to manage the VoIP deployment, represents a significant labor cost. Phones to connect to a network increases as the number of time and

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effort required to keep accurate information about the growth. As a result, the rule will no longer have to wait for new phones or change times will move to approve by an inability to control systems, and the entire process [2].

Pros and cons

When you are using PSTN line, you can usually use the time you pay a PSTN line manager company: more time for you to stay on the phone and the more you will pay. In addition, you cannot talk with the other person at a time. In opposite with VoIP mechanism you have in addition to, and as far as you want (money independent), each person you want (the other person is connected to the Internet at the same time) with and can talk all the time, at the same time you can talk to many people. If you are still not convinced that you can at the same time, you have people you have pictures, graphs, and sending videos, you can exchange data with and are speaking with, that may be considered.

- Leading edge (not bleeding anymore) technology
- QoS and capacity for network gear need to know (no matter what the sales guys)
- If the cost of driving force behind integration, do not fool yourself - you can spend a great deal, in some cases a complete set of services
- Redundancy and time must be carefully engineered: you can "email down", "Internet down", or how often you need to hear? You can "phone does not work" and how often you need to hear?

This is what you want them to run you over your phones more redundant networks, and to be consistent with the meaning [2].

VoIP over TCP

TCP stands for Transmission Control Protocol and is one of the main protocols used for data transmission over the Internet and LANs. It works together with the IP protocol to make the well-known TCP/IP protocol suite. Since other protocols like IP do not provide reliability over a network, TCP ensures that data transmission is reliable. It ensures that, during a transmission, there is no packet loss; there is an acceptable delay between the packets. TCP also bundles data into TCP packets. The data packets however do not contain an address for the source and destination machines, since IP packet stake care of the addressing and routing [4]. On the other side, TCP gives error and flow control [5] .The connection is formed via a handshake between two hosts with connection requests and acknowledgments. Once the connection is formed, the data being transmitted is broken into segments. Before the segment is transmitted, a header is attached which contains a sequence number. The receiver will respond to the arriving packet with an acknowledgement if no errors are found. If no acknowledgement arrives at the original sender after a certain timeout period, the sender will re-transmit the packet [6].

VoIP over UDP

The UDP is a simple protocol that passes data along from the application layer to IP to be transmitted. It performs none of the error checks that TCP does, and is therefore unreliable. A UDP header merely consists of an optional source port, a destination port, the length of the datagram, and a checksum [7]. UDP cannot provide the reordering or sequencing and cannot detect the damaged packet [8]. As previously mentioned, the main reason for using UDP over TCP in VoIP applications is the reduced delay. In general, the sporadic loss of packets in a conversation will not be as disruptive as excessively long delay times. In fact, a packet loss of about 5% is said to be tolerable depending on how the losses are distributed [9]. We will investigate how well a UDP-based VoIP network performs in contrast to its TCP counterpart.

Asodi *et al*., in [10] highlighted that Stream Control Transmission Protocol (SCTP) and UDP complete with each other under the considered quality metrics for voice transmission. The good performance of UDP in VoIP applications makes it a preferred transport layer protocol to carry voice packet from source to destination. However, it is likely that SCTP many perform better with some modification/extensions in the as observed transmission protocols, it performance is comparable to UDP in most of the cases. Camarillo *et al*., in [11] implements Session Initiation Protocol (SIP) over SCTP, UDP and TCP protocols under different network conditions and observe that UDP is good only for the light traffic. Under heavy traffic load, TCP and SCTP are better than UDP. However, SCTP has some advantage over TCP owing to its features as multi-streaming and multi-homing. In general, SCTP performance increases with worsening of network conditions [10].

Hence, the previous studies compare the performance of transport layer protocols is crucial for any IP based network application. We can evaluate the performance of RTP transport layer protocol in different network QoS factor for streaming media applications, e.g., VoIP and compared with various protocol.

DESIGN AND METHOD

VoIP over RTP

The RTP is an application layer protocol that attaches itself to UDP to provide added benefits for real-time applications. An RTP header includes a sequence number to help preserve the order of the transmitted packets. It also includes a timestamp, which is meant to provide information to the destination application so that it may compensate for problems such as delay or jitter if they arise. The optional companion protocol, Real-time Transport Control Protocol (RTCP) (specified in RFC 3550), is used as a means of exchanging information on session quality, which can include the number of lost packets or the average delay time. RTP is the protocol of choice for streaming media over the Internet and is widely used in VoIP applications [11]. RTP may be used with other suitable underlying network or transport protocols. If

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multicast distribution is provided by the underlying network, RTP can transfer data to multiple destinations [12, 13].

RTP is typically run on top of UDP to make use of its multiplexing and checksum functions. TCP and UDP are two most commonly used transport protocols on the Internet. TCP provides a connection-oriented and reliable flow between two hosts, while UDP provides a connectionless but unreliable datagram service over the network. UDP was chosen as the target transport protocol for RTP because of two reasons. First, RTP is primarily designed for multicast; the connection-oriented TCP does not scale well and therefore is not suitable. Second, for real-time data, reliability is not as important as timely delivery. Even more, reliable transmission provided by retransmission as in TCP is not desirable. For example, in network congestion, some packets might get lost and the application would result in lower but acceptable quality. If the protocol insists a reliable transmission, the retransmitted packets could possibly increase the delay, jam the network, and eventually starve the receiving application [14].

VoIP Quality factor

In such a mobile environment, typically, three main factors degrade VoIP quality:

End-to-end delay : End-to-end delay is the time it takes for a packet to travel from source to destination [15]. The equation below is used to calculate the delay on the packet:

$$D = T_f + l + \sum_{h \in Path} (T_h + Q_h + P_h) + D_{play} \qquad (1)$$

where: D is end-to-end delay, T_f is formation delay or default delay time, h is path of transmission or path between tow hop, l *is* time to ok look a header, T_h *is* transmission delay, Q_h *is* queuing delay, P_h is propagation delay and *Dplay* is playout buffer delay.

The end-to-end calculation is performed for all transmitted/received packets within a time interval and the resulting differences are averaged.

Jitter: Jitter more formally known as IP Packet Delay Variation (IPDV). Jitter is defined as the difference in end-to-end delay of the transmitted packets [14]. There are numerous ways of calculating this quantity.

$$Jitter = \frac{\left(\left(recvtime(j) - sendtime(j)\right) - \left(recvtime(i) - sendtime(i)\right)\right)}{(j - i)}; j > i \qquad (2)$$

where *recvtime* is packet received time, *sendtime* is packet send time, i is the last received packet sequence and j is the current received packet sequence.

Packet loss: Packet loss is a measure of the amount that lost between the source and the destination in the network. Some of the voice packets may be dropped by network routers or switches that become congested, such packets are called lost packets [15]. Two measures we were taken:

a) Instantaneous packet loss: describes how many packets are lost at each time interval.
b) Cumulative loss: describes the total number of packets loss of all time.

Design module

In a standard circuit-switching network, an analog voice signal must be sampled at twice its maximum frequency at 8 bits per sample. Standard human speech reaches about 4000 kHz. Thus, a bandwidth of 64kbps is required. The advancement of codec technology has improved bandwidth efficiency in telephony by only transmitting information when a person is talking [15]. Therefore, a variable bit rate on each end is required to accurately simulate a VoIP call.

NS-2 is the simulator tool used for designing and simulating the network and deploying VoIP technology view in the Figure-4 [17]. In our implementation, we assume the commonly used G.711 codec, which transmits information at a rate of 64kbps [14]. The size of the transmitted packets was chosen to be 128 bytes for RTP. The phone call is established between Nodes 0 and 4 as shown in Figure-1. Node 0 transmits data with an average “on” time of 1200ms and idle time of 800ms. Node 4 is setup to transmit fewer packets over the 60 second simulation with an average “on” time of 800ms and idle time of 1200ms.

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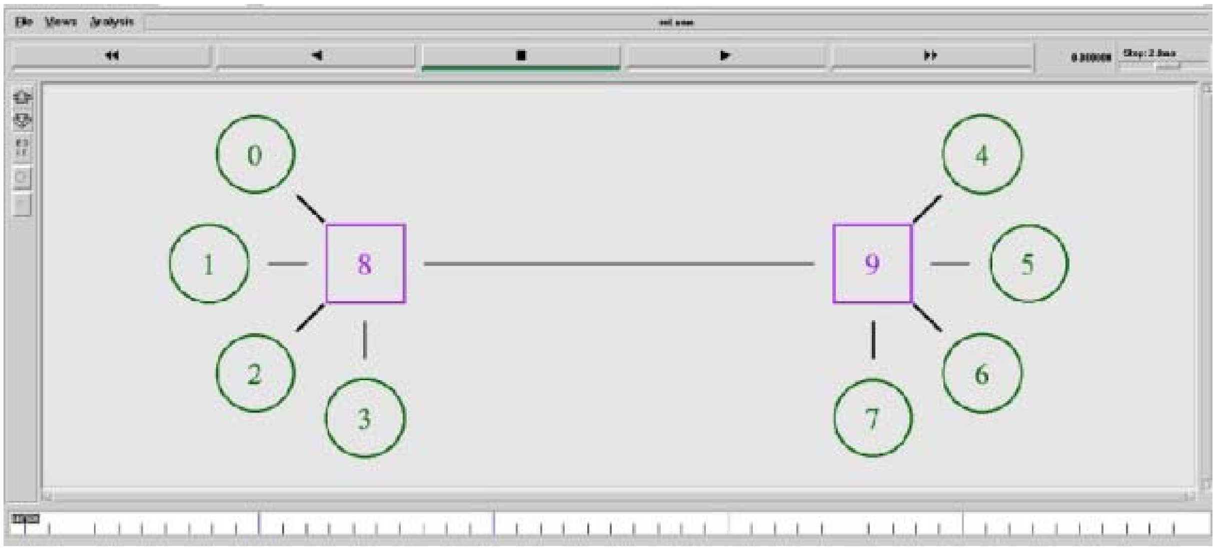

Figure-1. NS-2 simulated VoIP network topology.

The background traffic of the network is supplied by Nodes 1, 2, 3, 5, 6, and 7 at constant bit rates. As the simulation begins, Nodes 1 and 5 create background traffic at a rate of 25.89 mbps, providing a sub-maximal load for the duplex link between the two routers (8 and 9). Then from 20s to 40s, Nodes 2 and 6 are turned to provide background traffic of 25.91 mbps each. This was chosen to slightly overload the link's capacity and thus cause congestion within the network. Finally, when the simulation reaches 40s, Nodes 3 and 7 are tasked with providing the background traffic at a rate of 25.92mbps each, greatly exceeding the network's bandwidth. At this point, it is expected that the queues become full and resulting in many dropped packets.

RESULTS AND DISCUSSIONS

The performance of RTP is identical to that of UDP in every aspect. RTP merely attaches additional data to the UDP stream to provide valuable information to the application at the other end. It does not directly prevent issues such as jitter, however the information it provides can warn the application that such issues are present so that the application can take whatever preventative actions it needs to.

End-to-end delay

Figure-2 showing the delay for TCP, UDP and RTP. Evaporate delays varying between (2.5 to 48.5) second of simulation time for TCP, the TCP delay from 219ms to 810ms. This delay for node (0) to node (4). This greatly exceeds the recommendation of 150ms.

The delay for UDP is around 94ms which is well below the recommended limit of 150ms specified earlier in UDP. Note that the delay varies very little regardless of increasing background traffic, the delay time between 3s and 55s.

The delay for RTP is around 94ms which is well below the recommended limit of 150ms specified earlier in UDP. Note that the delay varies very little regardless of increasing background traffic, the delay time between 13s and 24s.

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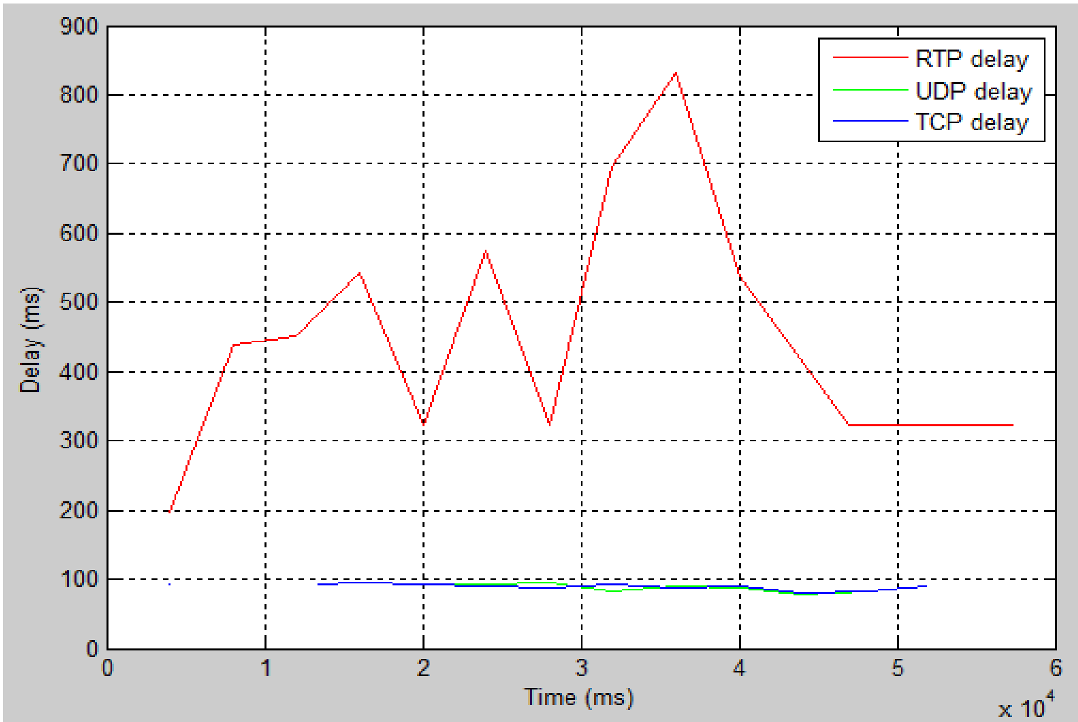

Figure-2. End-to-end delay.

Jitter

Figure-3 showing the jitter for TCP, UDP and RTP. The limits of variation jitter is almost between 5s to 58.5s of simulation time with the jitter values occurred between 40s to 350s.

A very low variation in the end-to-end delay times for UDP during a first 4^{th} second, which is ideal for VoIP. Stating from 4^{th} second to 13^{th} second, the jitter increased. The high variation refer to high background traffic causes increase and decrease for end-to-end delay.

Very low variation in the end-to-end delay times for RTP between 4^{th} and 16^{th} second, which is ideal for VoIP. Stating from 16^{th} second to 44^{th} second, the jitter increased the high variation it above than 45 refer to high background traffic causes increase and decrease for end-to-end delay.

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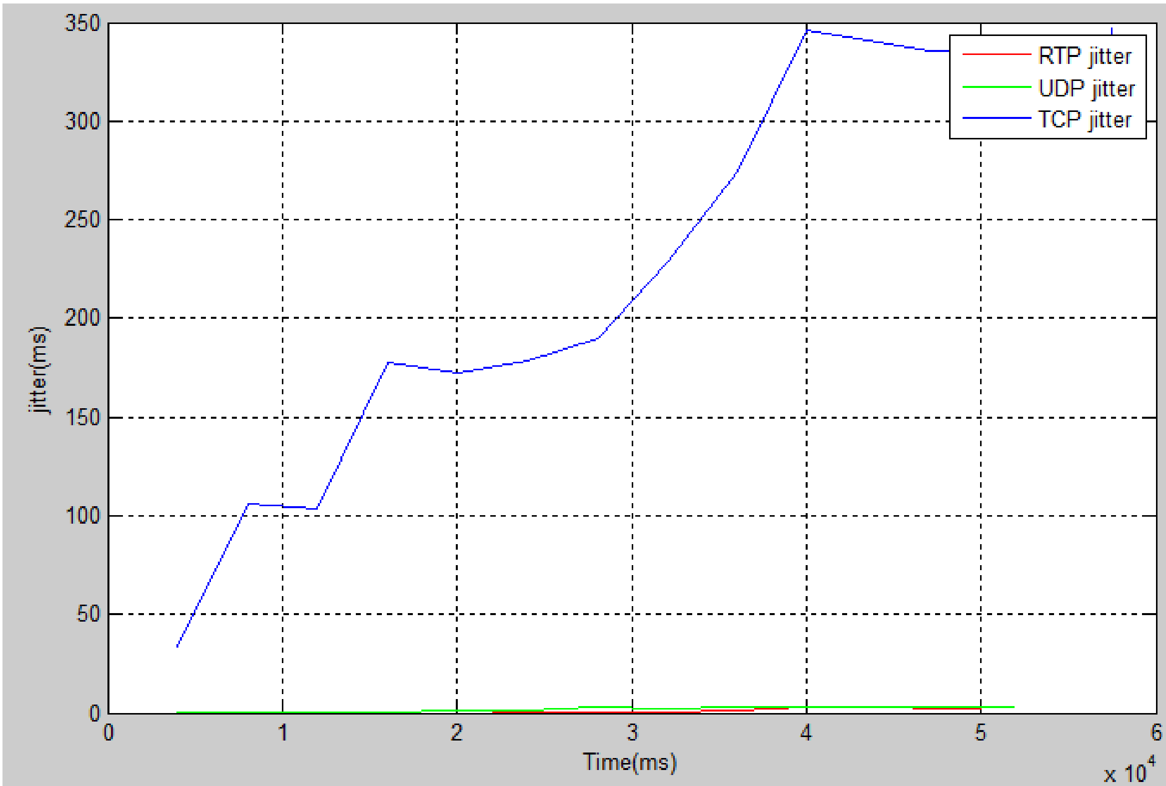

Figure-3. Jitter.

Packet loss

Figure-4 is showing the packet loss for TCP, UDP and RTP. In Figure-4, TCP shows the lowest packet loss. For TCP, the first 40 seconds of simulation shows a fair bit of dropped packets. However, since the throughput did not drop during this interval may conclude that the only packets lost are ACK packets, which are small in size and are not of real concern. Packets containing actual voice data are lost only after 3 seconds of simulation when the network is under a maximal load. This is because the packet size is larger.

The background traffic is upped at 20 seconds and beginning to see the consequences at around 30 seconds with the dramatic increase of lost packets in UDP protocol. At this point, the queues are growing exceedingly large causing many packets to be dropped. Since the background traffic remains constant, the queues do not have time to relieve themselves and as the background traffic is further increased, more packet are consistently dropped.

For the RTP, in beginning, some packet losses due to the sudden congestion caused by a combination of background traffic and both users speaking at the same time. The background traffic is upped at 3^{th} second from node 0 to node 4 and at 10^{th} second from node 4 to node 0, the consequences at around 54 seconds with the dramatic increase of lost packets.

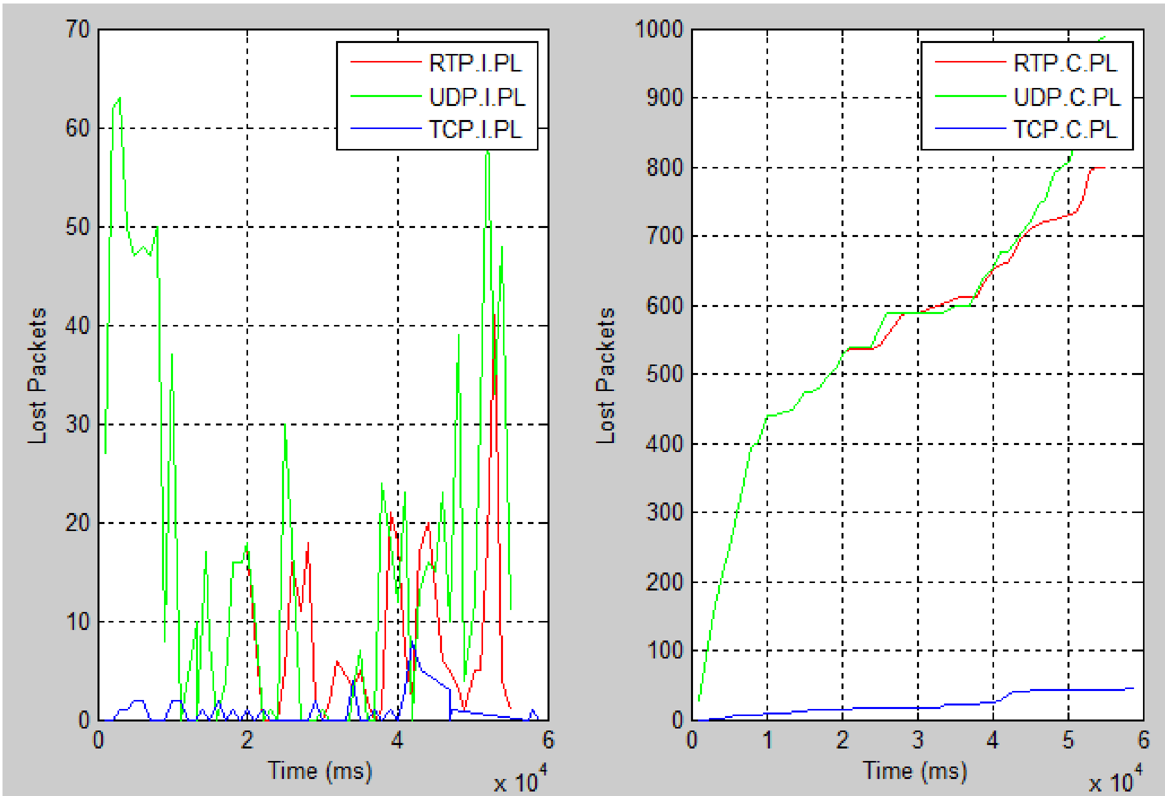

Figure-4. Packet loss.

CONCLUSIONS

Depending on the design scenario implementing in NS-2, results are achieved. The QoS factors are captured graphically and regarding to our criteria of low packet loss to kept voice quality high, we note that TCP outperform UDP and RTP.

Packet losses were not obtained for TCP unless the background traffic was at maximum load. Low delay and jitter are higher priority in contrast with loss for RTP's quality. TCP's stagnation delay and large jitter is undesirable for VoIP applications. On the contrary, the simple best effort characteristics for RTP allows for very small delay and jitter, with an acceptable voice quality during low/moderate background traffic which makes RTP or UDP the most preferred protocols for VoIP application. Generally, RTP is always used over UDP because of its identical performance and additional features. Finally, this implementation concludes that RTP is the best among the transmission protocols.

As a future work, we will compare our topology in real environment instead of simulation results to obtain more accurate results of quality factors.

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REFERENCES

[1] D. Keromytis. 2009. Voice over IP: Risks, Threats and Vulnerabilities. In: Proc. Cyber Infrastructure Protection (CIP) Conference.

[2] A.-A. Mousa and I. A. Momani. VoIP Quality Assessment Technologies.

[3] R. Swale. 2001. INTRODUCTION TO VoIP. Voice over IP: Systems and Solutions. p. 1.

[4] A. Jasin, R. Alsaqour, M. Abdelhaq, O. Alsukour and R. Saeed. 2012. Review on Current Transport Layer Protocols for TCP/IP Model. International Journal of Digital Content Technology and its Applications. 6: 495-503.

[5] J. Postel. 1981. Transmission control protocol.

[6] C. Demichelis and P. Chimento. 2002. IP packet delay variation metric for IP performance metrics (IPPM).

[7] I. Widjaja. 2004. Communication Networks Fundamental Concepts and Key Architectures. ed: Mc Graw Hill.

[8] M. I. Öztürk and A. Üniversitesi. 2004. Analysis of Delay and Delay Jitter of Video Traffic in the Internet: Atılım Üniversitesi.

[9] D. Collins. 2003. Carrier grade voice over IP vol. 2: McGraw-Hill New York.

[10] S. Asodi, S. Ganesh, E. Seshadri and P. Singh. 2009. Evaluation of transport layer protocols for voice transmission in various network scenarios. in Applications of Digital Information and Web Technologies, 2009. ICADIWT'09. Second International Conference on the. pp. 238-242.

[11] G. Camarillo, R. Kantola and H. Schulzrinne. 2003. Evaluation of transport protocols for the session initiation protocol. Network, IEEE. 17: 40-46.

[12] O. Hersent. 2011. IP telephony: deploying VoIP protocols and IMS infrastructure: John Wiley and Sons.

[13] V. Jacobson, R. Frederick, S. Casner and H. Schulzrinne. 2003. RTP: A transport protocol for real-time applications.

[14] C. Liu. 2000. Multimedia over ip: Rsvp, rtp, rtcp, rtsp. Handbook of Communication Technologies: The Florida.

[15] S. Vadivelu. 2011. Evaluating the Quality of Service in VOIP and comparing various encoding techniques.

[16] T. Christiansen, I. Giotis and S. Mathur. Performance Evaluation of VoIP in Different Settings.

[17] T. Issariyakul and E. Hossain. 2011. Introduction to network simulator NS2: Springer.