EVALUATION OF VOIP TRAFFICS OVER TIKRIT UNIVERSITY NETWORKS

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Abstract
In recent years, VoIP is a rapidly evolving technology that could possibly revolutionize the telecommunication industry. When implemented on wireless data networks, VoIP could prove to be instrumental in the convergence of existing fixed and cellular telephony networks with the fast growing wired and wireless data networks. Whereas, the growth of the VoIP applications with additional mobile devices in the institution of higher education and increasing number of users led to slow VoIP services. In addition, the throughput and latency (or delay, jitter, and jabber) are two of the most important issues that need to be resolved before the commercial deployment of wireless VoIP. The trace composes of two parts: firstly, collecting of data, secondly, performing the traffics analysis. Therefore, we have analyzed the performance offered by SCTP, TCP and UDP over VoIP at Tikrit University, Iraq. In this paper, we can see that the kind of performance given by SCTP is best then UDP, but the latency is decreased it performance significantly compared to UDP and the SCTP throughput is looks in a better manner. The measurements were verified by simulating the VoIP traffics using Network Simulator 2

Keywords:- Simulation, SCTP – UDP - TCP, Comparison, VoIP, Traffic analysis.

1. Introduction
With the acceptance that Public Switched Telephone Network (PSTN) and Internet Protocol (IP) based telephony are to co-exit for a fairly long period of time, telecommunication service providers tend to transform their traditional circuit-switched networks into packet-based networks. IP base networks that smooth the progress of new services that put altogether data, voice, and video information . Internet telephony is a transport of telephone calls over internet, base on IP and Voice over IP is IP telephony. In this procedure voice calls are transmitted over a packet switched network. The most noteworthy advantage of Internet Protocol Telephony (IPT) is the saving of money and easy implementation of innovative services (value added services). In money saving low capital and operating costs are considered [1]. The efficient use of bandwidth always remained an advantage of VoIP. Internet Telephony Service Providers (ITSP) uses a single transportation for providing both, Internet access and Internet telephony. Only data oriented switches could be installed for switching data as well as pocket-sized voice. Multiplexing of data and voice, result in better bandwidth utilization. Not only the service providers but clients also take advantages of lower cost in IP telephony. The main motivating force of this deployment is that it is economical and also associated with converged data and voice networks. In packet switched networks several user share the same physical connection [2, 3]. The communication channels are broken down in small chunks of data called packets in IP networks and cells in ATM networks. IP packets are routed all the way through a network based on the destination address contained within the packet header. As shown in Figure 1: is explain how Voice over IP works.

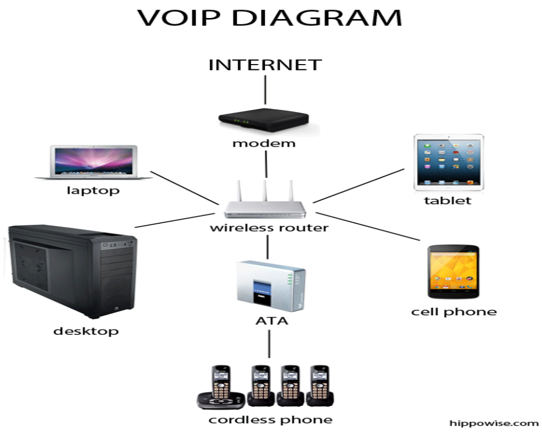

Figure 1: How Voice Over IP works.

Moreover, 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 [1] .Voice over IP (VoIP) applications are gaining an ever increasing popularity in the Internet community, favored by the massive deployment of wireless access technologies [4].
The University of Tikrit is an Iraqi university. It was established in 1987. It is one of the largest universities in Iraq with over 29,500 users. The wireless network consists from one switch that is linked to the Core switch in the Computer Center and the Core Switch is linked to the Router Board 1100. In addition, the Router Board 1100 is linked to the Real IP switch that is linked to the ISP in the Postal Centre. Figure 2: shows the Network in Tikrit University.

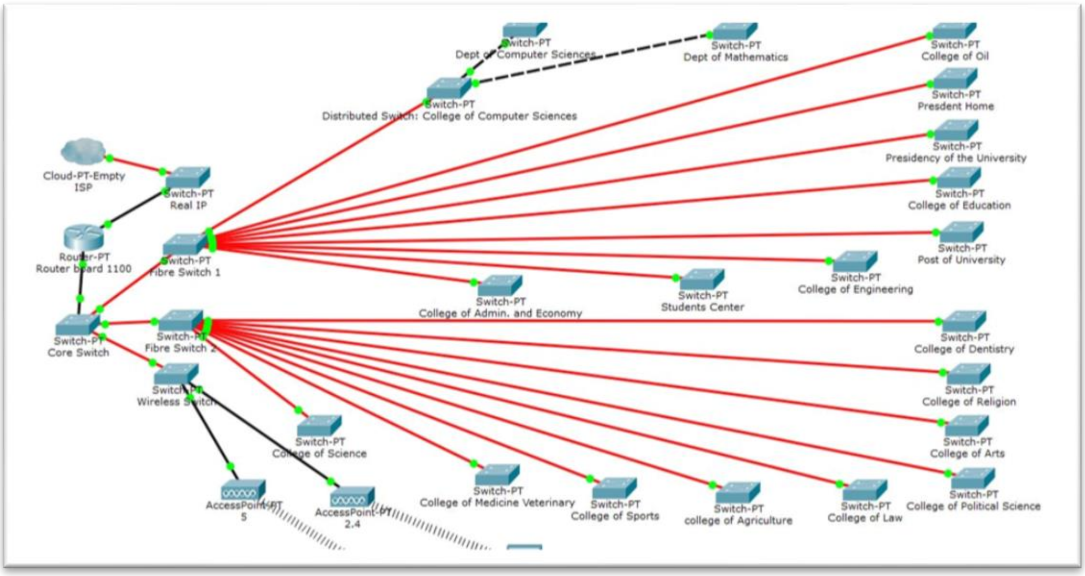

Figure 2: Tikrit University Network.

Recently everybody has noticed that most of people and a huge part of them are Tikrit University (TU)users , are spending most of their free time using electronic devices such as phones and iPads and PCs , etc, which are (electronic devices) definitely connected to the internet using many services such as audio and video calls over the internet using the voice over IP (VoIP) (for example: Viber , Tango, Skype ,and WhatsApp) which has affected badly upon the network, made it slower and also weaker than before due to the single path of network in our university , this led to decrease the Quality of service (QoS) besides loss of "throughput". The major factors affecting QoS in a wireless network are throughput, packet loss, packet delays and jitter. The wireless channel changes with time and hence it is possible that a link between two nodes could break in midst of a voice session. This makes it difficult to achieve an acceptable QoS in a wireless environment [2].

This paper will deal with the quality of service by simulating the performance of the transport protocols over the VoIP in university of Tikrit by using Network Simulator (NS2).

2. Internet Telephony

Also known as Voice over Internet Protocol (VoIP) or Internet Protocol Telephony (IPT) is the real-time delivery of voice and possibly other multimedia data types. Between two or more parties, across networks using the Internet protocols and the exchange of information required to control this delivery. Internet telephony offers the opportunity to design a global multimedia communications system that may eventually replace the existing telephony infrastructure, without being encumbered by the legacy of a century old technology [1, 3, 5]. Internet telephony requires a range of protocols, ranging from those needed for transporting real-time data across the network, to quality-of-service-aware routing protocols, to resource reservation, QoS-aware network management and billing protocols. In addition, Internet telephony, defined here as synchronous voice or multimedia communication between two or more parties, requires a means for prospective communications partners to find each other and to signal to the other party their desire to communicate. We refer to this functionality as Internet telephony signaling [1]. The need for signaling functionality distinguishes Internet telephony from other Internet multimedia services such as broadcast and media-on-demand services .IP tel signaling as we understand it creates and manages calls. We define a call as a named association between applications that is explicitly set up and torn down. Examples of calls are two-party phone calls, a multimedia conference or a multi-player game. A call may encompass a number of connections, where a connection is a logical relationship between a pair of end systems in a call, as figure 3.

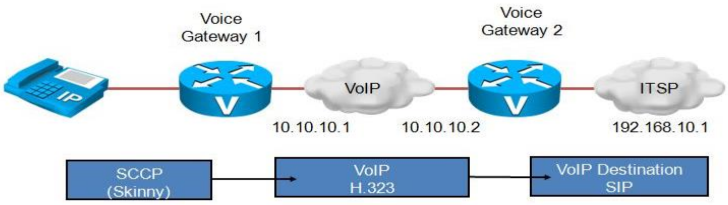

Figure 3: Internet Telephony or (VoIP).

3. TCP, UDP and SCTP

Stream Control Transmission Protocol (SCTP) is a general-purpose transport layer protocol providing a service similar to TCP plus a set of advanced features to utilize the enhanced capabilities of modern IP networks and to support increased application requirements. SCTP is developed by IETF SIGTRAN Working Group to carry telephony signals on IP from networks like SS7 [3]. The main propose of design consideration was to beat the limitations of Transmission Control Protocol (TCP) and User Datagram Protocol (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. UDP provides unreliable and untrustworthy datagram service [6], and relies on the application layer for error control, detection of message repetition, duplication, and retransmission of lost messages. In addition, it is a simple transport protocol that extends the host-to-host delivery of packets of the underlying network into a process-to-process communication. Since there are many processes running on a given host (e.g. multiple Internet browsers), UDP needs to add a level of demultiplexing, allowing multiple application processes on each host to share the network.

On the other side, TCP gives error and flow control [7]. 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 TCP, UDP and SCTP.

4. Network Simulator (Ns2)

NS2 is a discrete event simulator meant for networking research [2,9]. Simulation is the execution of a system model in time to yield useful information about a system which is being investigated. Events in the modeled system occur at discrete points of time. When the number of such events is finite, we call it discrete event. A discrete event simulator consists of a set of discrete events & a Simulator object which that executes these events in a total order in time.

NS2 is written in OTcl(Object-Oriented Tool Command language) and C++. OTcl is its primary Command and Configuration Language. It implements network protocols such as TCP and UDP over wired and wireless (both local and satellite) networks, and also traffic source behavior such as FTP, Telnet, Web, CBR (constant bit rate) & VBR (variable bit rate) [9]. Router queue management mechanisms such as Drop Tail, RED and CBQ and routing algorithms such as Dijkstra and more have also been implemented. NS also implements multicasting and some of the MAC layer protocols (which include 802.11b MAC layer specification) for LAN simulations. Also tools for analysis and display of the simulation results are now a part of NS2. These tools include NAM (Network Animator).

5. Materials and Methods

In this section, we first describe the network phase configuration of the VoIP network from which we collected our data, and then the evalution phase of the network for describe our methodology for analysis of data, presentation performance metrics, and interpretation [10]. The researcher follows Jain and Hassan steps as shown in Figure 4.

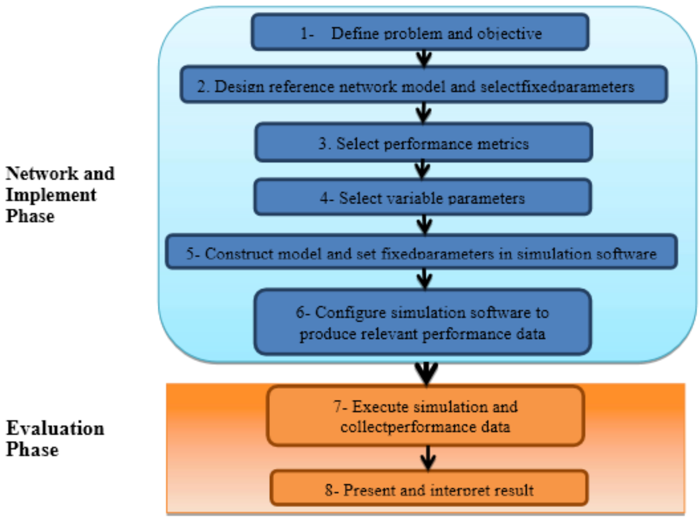

Figure 4: Phases of Methodology [10-12].

5.1 Network And Implementation Phase

Firstly, define problem and objective, presence many subscribers in WLAN' linked with ISP through hotspot or whatever WLAN' connecting techniques its need trusted level of performance indeed, especially many able to take vary amounts of users .Performance metric which are chosen to achieve the simulation over Internet Protocols, such as throughput and latency [12]. Then constructing the network model and set fixed parameters in simulation model, build references networks model designed with the second version Network Simulator software application, each of this models will be measured with increases during the simulation processes several factors such as excess numbers of nodes , increase in time and augmented with some randomize factors regarding each model [13, 14]. Set of scripting lines which comprise the TCL file, these lines assigned with various parameters and those parameters give the models own flavor regarding the model creation to simulate with fact ,so these facts indeed will adopted as a parameters inside TCL file, as well as increase the number of nodes inside each models working to clarify the attitude of performance in terms of measuring how close to reality, so most researcher and studies introduced increased number of nodes gradually and analysis the outputs for highlighting the behavior of models in terms of metric performance measures such as delay ,throughput and packet loss also it should be noted that there are many metrics remaining which its various.

Implying random factors inside the model per se presenting infrastructures is closer to the reality, which leads to Random Variables with various shapes which can be implemented in NS2. They have a significant place to play in traffic modeling and in network simulation. About boosting the time, these is due to following reasons simulating environments need to warm up technically inside the TCL file which represent the model reference with libraries inside the network simulator, so it needs few moments before establish executions process[11] [12].

5.2 Evaluation Phase

In conclusion, this phase consists from the last three steps of paradigm 2 of methodology; the Configure simulation software is to produce relevant performance data. This step consider as stage clutch among network phase forwarding evaluation phase, so it should be noted that the paradigm steps it not compulsory followed. Regarding to this phase it show the steps to combine so as to configure the simulation software in other to produce relevant performance [11]. This step requires knowing the theoretical part and immersion in the work, being its own due to the representing parameters inside simulation environment.

Construct through know-how it's act in theory and on the other hand how's personification of parameters inside the simulation model , regarding the sixth step of the paradigm of methodology, will be divide into two parts according to accommodate this study. As for the Execute simulation, collect performance data, present and interpret result then confirm the parameters in sixth step, absolutely these parameters follows the simulation environment. However this study through parameter setting regarding the Simulation models also note the effective performance factors after combing the parameters correctly during the sixth step of paradigm of methodology [12]. This study need to execute those models which is substantially executable scripts in the NS2.

6. Results

This section will present the results regarding both **Table 1** and **Table 3** models after each of which executed in NS2, the various parameters setting which have been used inside each scenario code as a TCL script, in per se pose flavor in relation between Scenario 1 and Scenario 2.

6.1 Scenario 1.

This study reports our analysis of the traffics based on illustrated it in the table 1 to Scenarios below the Tables 2 of results after have been performed with NS2.

Table 1: Simulation parameters.

Parameter	Value	Unit
Simulation Time	1000	Second
Packet size	512	bytes
Interval	0.050	ms
Transfer Protocols	UDP	
Client Time out	60	Second
Bandwidth	1024	Mbps

Table 2: UDP: Scenario result.

Property Name:	Value:
Simulation length in seconds:	1000.350000
The number of nodes	4
Number of sending nodes:	4
Number of receiving nodes:	2
Number of generated packets:	4155
Number of sent packets:	3713
Number of received packets:	2387
Packet loss:	1326

Packet Delivery Ratio (PDR) came from the percentage result of dividing total number packet receive on total number packet sent, hence the equation for calculate the PDR is:

Packet Delivery Ratio = (Number of receiving packets /Number of sending packets) %

As the following results:

Number of receiving packets=2387

Number of sending packets=3713

Packet Delivery Ratio (PDR) = (2387/3713) %

Packet Delivery Ratio (PDR) =63.318 %

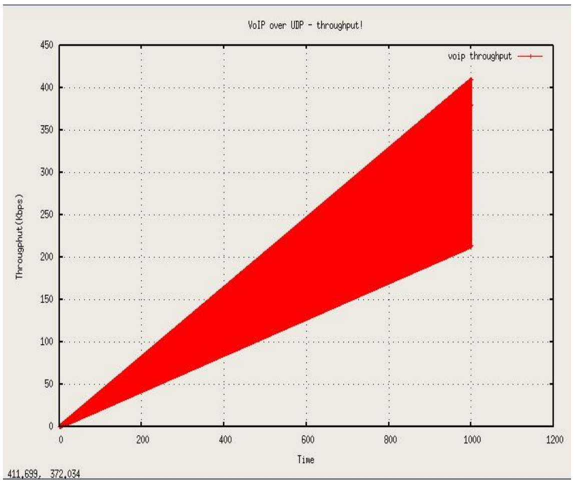

Figure 5: UDP Throughput (Kbps).

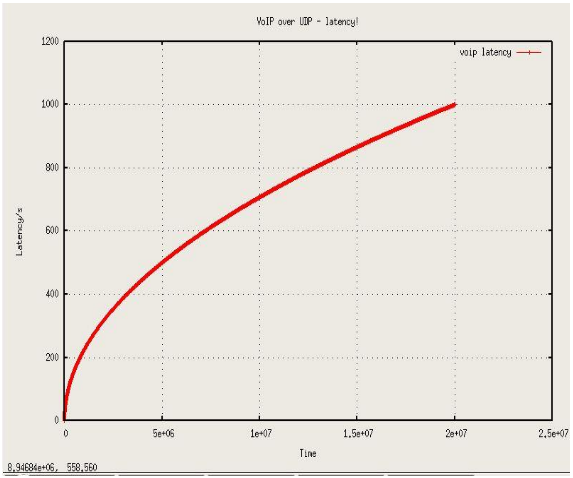

Figure 6: UDP latency.

6.2 Scenario 2

This study reports our analysis of the traffics based on illustrated it in the table 3 to Scenarios below the Tables 4 and Tables 5 of results after has been performed with NS2 are shown below, according to values of parameters that appears in the tables.

Table 3: Simulation parameters.

Parameter	Value	Unit
Simulation Time	1000	Second
Packet size	512	bytes
Interval	0.050	ms
Transfer Protocols	SCTP,TCP	
Client Time out	60	Second
Bandwidth	1024	Mbps

Table 4: SCTP: Scenario result.

Property Name:	Value:
Simulation length in seconds:	1000.350000
The number of nodes	4
Number of sending nodes:	4
Number of receiving nodes:	2
Number of generated packets:	3984
Number of sent packets:	3565
Number of received packets:	3311
Packet loss:	254

Packet Delivery Ratio (PDR) came from the percentage result of dividing total number packet receive on total number packet sent, hence the equation for calculate the PDR is:

Packet Delivery Ratio = (Number of receiving packets /Number of sending packets) %

As the following results:

Number of receiving packets=3311

Number of sending packets=3565

Packet Delivery Ratio (PDR) = (3311/3565) %

Packet Delivery Ratio (PDR) =92.875 %

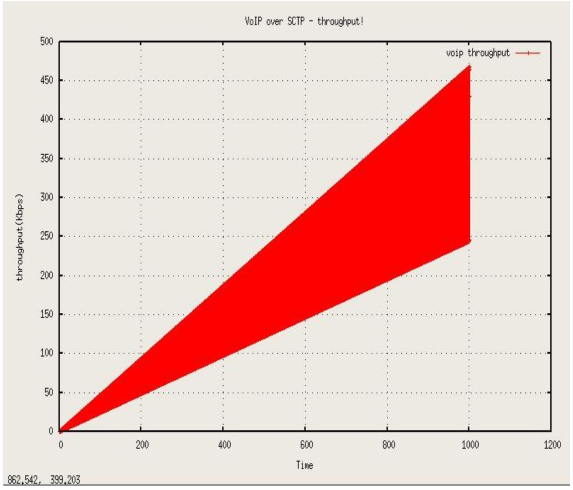

Figure 7: SCTP Throughput (Kbp

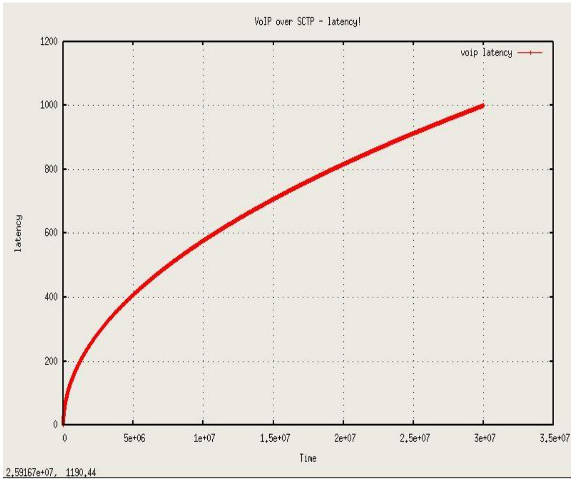

Figure 8: SCTP latency.

Table 5: TCP: Scenario result.

Property Name:	Value:
Simulation length in seconds:	1000.350000
The number of nodes	4
Number of sending nodes:	4
Number of receiving nodes:	2
Number of generated packets:	3724
Number of sent packets:	3430
Number of received packets:	3261
Packet loss:	169

Packet Delivery Ratio (PDR) came from the percentage result of dividing total number packet receive on total number packet sent, hence the equation for calculate the PDR is:

Packet Delivery Ratio = (Number of receiving packets /Number of sending packets) %

As the following results:

Number of receiving packets=3261

Number of sending packets=3430

Packet Delivery Ratio (PDR) = (3261/3430) %

Packet Delivery Ratio (PDR) =95.072 %.

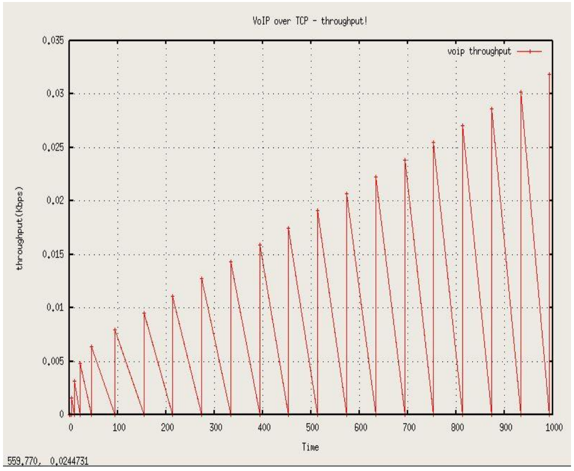

Figure 9: TCP Throughput (Kbps).

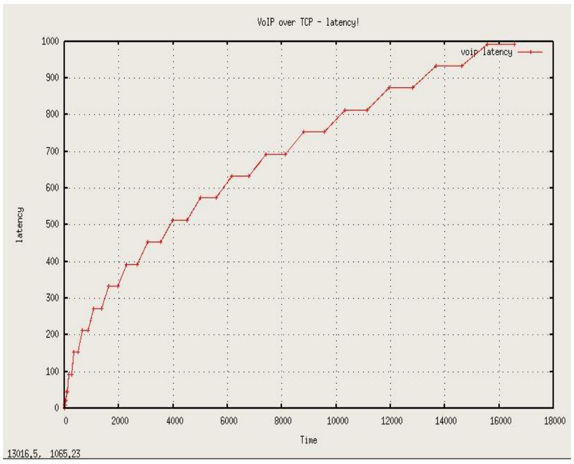

Figure 10:TCP latency.

7. Discussion

The table 6 below shows the comparison of SCTP, UDP and TCP with various parameters.

In addition, the figures: 5, 6 (UDP), 7, 8(SCTP) and 9, 10 (TCP) above show the comparison of throughput and latency. Based on that, we can see that UDP latency is increasing up till last traffics (1000 ms) and the throughput is reached to 420 kbps and SCTP latency is increasing slowly till last traffics (1000 ms) and the throughput is reached to 470 kbps. However, TCP latency is very high and the throughput is not stabling till last traffics. Moreover, the 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 when comparing the results we can see that the kind of performance given by SCTP is best then UDP, but the latency is decreased it performance significantly compared to UDP and the SCTP throughput is looks in a better manner.

Table 6: Comparison of SCTP, UDP and TCP.

Protocols	Sent	Receive	Packet loss	PDR
SCTP	3565	3311	254	92.875 %
UDP	3713	2387	1326	63.318 %
TCP	3430	3261	169	95.072 %

8. Conclusion

In this paper we have the opportunity investigate the performance of VoIP traffics of one of the institution of higher learning in Iraq by using the NS2. Unlike previous studies of transport traffics, our traffics were collected depending on the the bandwidth and other prameters for our university. The above study was conducted to compare the performance of UDP, SCTP and TCP with traffic analysis. The researcher kept the packet size of 512 bytes and run the simulation time of 1000 sec. Moreover, the constant bit rates over all transport protocols and we had the bandwidth 1024 Mbps. As observes in table 6, 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 as compared to that of UDP. In addition, 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, we sees that the UDP packet loss is more as compared to that of SCTP and TCP. 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 loss has not more effect on its application. Finally, when comparing the results we can sees that the kind of performance given by SCTP is best then UDP and TCP because its multi homing and multi association but its packets sent acknowledgement is time consuming as compared to that of UDP.

9. References

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تقييم أداء الـ VoIP في شبكة جامعة تكريت

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الملخص

في السنوات الأخيرة، VoIP هي تكنولوجيا تطورت بسرعة وأحدثت ثورة في مجال الاتصالات السلكية واللاسلكية. وخصوصاً مع الشبكات اللاسلكية Wireless. في حين، أدى نمو تطبيقات VoIP و الأجهزة المحمولة في مؤسسات التعليم العالي، إضافة إلى تزايد عدد المستخدمين إلى إبطاء خدمة الـ VoIP. بالإضافة إلى ذلك، فإن الإنتاجية (Throughput) والتأخير (Latency) هي من أهم القضايا التي تحتاج إلى حل قبل الانتشار التجاري لخدمة VoIP.

تتألف منهجية الدراسة من جزئين هما: أولا: جمع البيانات، وثانيا: إجراء تحليل للبيانات. ولذلك ، قمنا بتحليل الأداء المقدم بواسطة بروتوكولات الـ TCP , UDP, SCTP عبر الـ VoIP في جامعة تكريت، العراق. في هذه الدراسة، نحن نلاحظ بأن الأداء المقدم بواسطة بروتوكول الـ SCTP هو الأفضل ثم بروتوكول UDP ، ولكن التأخير (Latency) قلل من الأداء مقارنة ببروتوكول الـ UDP، وإنتاجية (Throughput) بروتوكول الـ SCTP تبدو بشكل أفضل . وتم التحقق من القياسات من خلال محاكاة بيانات الـ VoIP باستخدام برنامج المحاكاة NS2.