Performance Analysis of VoIP in Wireless Networks

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***Abstract*—Voice over Internet Protocol (VoIP) is important technology that's rapidly growing in the wireless networks. The Quality of Service (QoS) and Capacity are two of the most important issues that still need to be researched on wireless VoIP. The main aim of this paper is to analysis the performance of the VoIP application in wireless networks, with respect to different transport layer protocols and audio codec. Two scenarios used in the simulation stage. In the first scenario VoIP with codec G.711 transmitted over User Datagram Protocol (UDP), Stream Control Transmission Protocol (SCTP), and Real-Time Transport Protocol (RTP). While, in the second scenario VoIP with codec G.726 transmitted over UDP, SCTP, and RTP protocols. Network simulator NS2 is used in all scenarios. In addition, several QoS criteria such as throughput, end-to-end delay, jitter, and packet loss has been considered to evaluate the performance of VoIP. The result shows that SCTP throughput performs better for VoIP application when compared to other protocols and also approved that VoIP has less end-to-end delay and jitter over RTP and UDP contrast with SCTP.**

Keywords-Wireless Network, VoIP, UDP, SCTP, RTP.

I. INTRODUCTION

In a wireless network, the bandwidth is limited contrast to wired networks. Also, a wireless channel is error-prone and data packets can be lost in transmission due to wireless network errors such as signal fading or interference. Nowadays, the most popular wireless network standards are the IEEE 802.11b, 802.11a, and 802.11g, which can theoretically support data rates up to 11 MB/s, and 54 MB/s. [1]. However, they are used for data transmission and not designed to support voice transmission. Voice packets are small in size when Compare to the data packet. Due to the large overhead involved in transmitting small packets, the bandwidth available for VoIP traffic is far less than the bandwidth available for data traffic. [2].

VoIP is an IP telephony term for a set of facilities used to controlling the delivery of voice information over the Internet. VoIP includes sending voice information in digital form in discrete packets rather than by using the traditional packet, and circuit-committed protocols of the Public Switched Telephone Network (PSTN). A major advantage of VoIP is that it provides communication for long distances cheaply, the flexibility of using different compression technologies, bandwidth efficiency, and trouble free. [3]

Furthermore, VoIP is supported by different transport protocols, and it has special characteristics which are not common in other types of applications. These include the use of User Datagram Protocol (UDP) [4], Stream Congestion Transport Protocol (SCTP) [5], and Real-time Transport Protocol (RTP) [6]. VoIP application can use several types of audio codec to provide low or high QoS. VoIP is considered to be affected by delay, jitter, throughput, and packet loss.

This paper analysis the performance of VoIP over the wireless network by taking into account various Transport layer protocols and voice encoding. The network model is implemented using NS2.35 network simulator. Moreover, different metrics that indicate the QoS like end-to-end delay, throughput, jitter, packet loss are measured and analyzed in wireless network scenarios. So, based on this evaluation Selection transport protocol and voice codecs consider the best choice for VoIP.

II. RELATED WORK

In recent years, many researchers have focused on analysis the performance of VoIP application in different networks. In [7], the authors have compared and analyzed the performance of the VoIP application over mobile and fixed Wi-MAX, with respect to various voice codecs. Different QoS parameters are considered. In [8], the authors implement and test VoIP networks. VoIP application packets had been sent and compared over RTP, TCP, and UDP protocols to obtain results which are related to Quality of Service (QoS) metrics. The author in [9] Analysis the performance of VoIP application in wireless LAN/WAN with taking into account the various voice encoding schemes. The network model was simulated by using OPNET. Different parameters that indicate the QoS such as MOS, jitter, and received traffic are measured and analyzed in Wireless LAN/WAN. In [10], the authors measured and compared CBR traffic and VoIP traffic in the Wi-MAX network against several routing protocols using QualNet 4.5.1 network simulator. The obtained result shows that VoIP application can be best served with Wi-MAX rather than the CBR. In [11] author evaluate performance of UDP, DCCP, SCTP, and TFRC transport protocols for VoIP traffic in wired networks. Network simulator NS-2.35 used for simulating the VoIP network.

III. SIMULATION NETWORK AND SCENARIOS

The performance of VoIP is studied under varying Transport layer protocols and voice codec. The Implementation will be carried out using renowned network simulator NS-2 [12]. Moreover, AWK scripts and Gnuplot will be used to present the results.

The network topology will consist of two VoIP clients. Each client sends two-way traffic to another in order to simulate real VoIP communication. At the start of the simulation, VoIP traffic will be transmitted after ten second background traffic will be sent via different nodes to study its effect on the VoIP traffic when sharing the transmission path. The topology of VoIP wireless networks is shown in Figure 1.

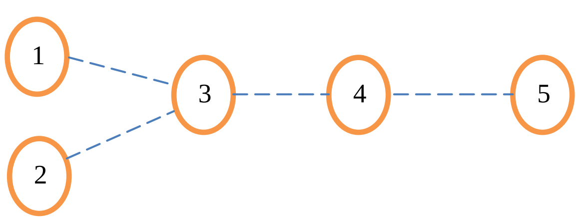

Fig. 1. VoIP wireless network topology.

Simulation of network environment requires Adjusting of the simulation parameters, Details of these parameters are depicted in Table 1.

TABLE 1. SIMULATION PARAMETERS

Parameters	Value
Simulation area	200 *200
Simulation time	50 Sec.
No. of Nodes	5
Channel Type	Wireless
Routing Protocol	DSDV
Transporting Protocol	UDP, SCTP, RTP
Packet Size	160Bytes
Audio Codec	G.711, G.726
Application Traffic	VoIP

Two scenarios used in the simulation to analysis the VoIP performance over wireless networks are as follows:

- VoIP with G.711 codec over UDP, SCTP, and RTP transport protocols. It is used to study the effect of different transport protocols with high data rate on VoIP performance
- VoIP with G.726 codec over UDP, SCTP, and RTP protocols. It is used to study the effect of different transport protocols with low data rate on VoIP performance.

Table 2 lists some characteristics of the G.711 and G.726 codecs

TABLE II. FEATURES OF G.711 AND G.726 CODEC

IUT-Codec	Bit Rate (Kbps)	Packet Size (Bytes)	Interval
G.711	64	160	0.020
G.726	32	80	0.020

IV. RESULT ANALYSIS

To evaluate the performance of VoIP application various quantitative metrics are measured. In this paper, four different performance metrics have been used to analysis the performance of VoIP against transport layer protocols and audio codecs

A. ***Throughput***

Throughput defines as the Total number of packets successfully delivered over a network. It is measured in bits/second or bytes/second [13].

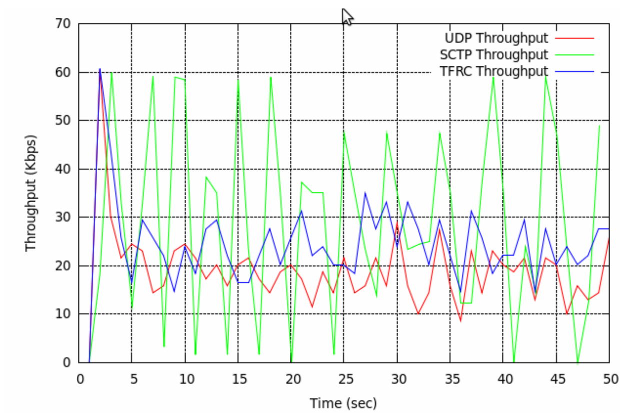

Fig. 2. Throughput for VoIP with G.711 codec

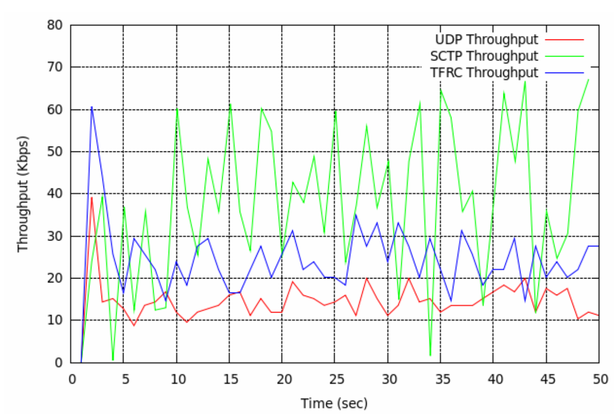

Fig. 3. Throughput for VoIP with G.726 codec

From the resulting throughput shown in figure 2 and figure 3, we have observed that in both G.711 and G.726 code the SCTP Protocol has a maximum throughput compared to other Protocols. This is due to SCTP features like multi-streaming and multi-homing

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B. ***End-to- End Delay***

End-to-End delay indicates the time taken by a packet to reach from source node to destination node over the network, and it is measured in seconds [14]. The graph of the end-to-end delay for VoIP G.711 and G.726 are shown in Figure 4 and Figure 5 respectively.

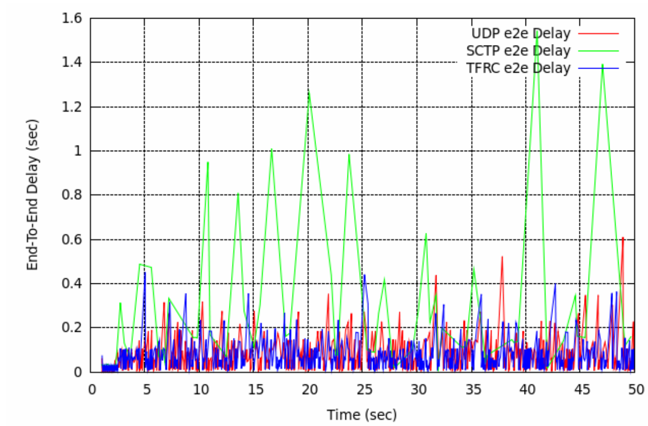

Fig. 4. End-to-End for VoIP with G.711codec

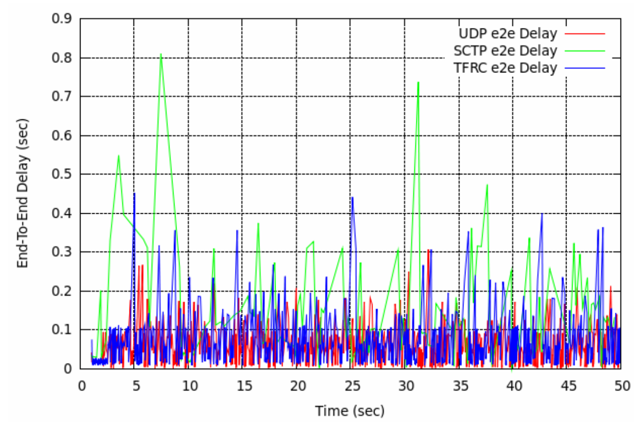

Fig. 5. End-to-End for VoIP with G.726 codec

In the first scenario, the end -to-end delay is shorter for the RTP protocol than SCTP and UDP protocols. On the other hand, UDP has a less end-to-end delay in the second scenario. However, the fluctuation of the end-to-end delay is significantly lower for RTP and UDP contrast SCTP, which is a very important issue for real time application like VoIP.

C. ***Jitter***

Jitter is defined as the difference in end-to-end delay of transmission packets, also known as Packet Delay Variation [13]. The jitter, measured in seconds, is an important metric to evaluate the QoS for real-time application. Figure 6 and Figure 7 represent jitter for VoIP with G.711 and G.726 respectively. We notice that UDP and RTP have a minimal jitter for two scenarios. On the contrary, SCTP has a high unsmooth jitter.

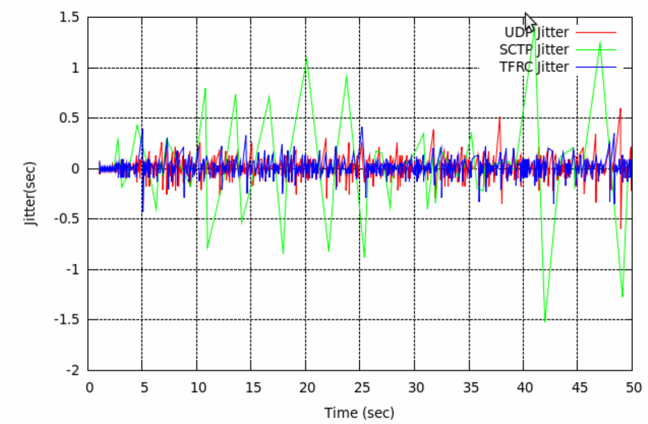

Fig. 6. Jitter for VoIP with G.711 codec

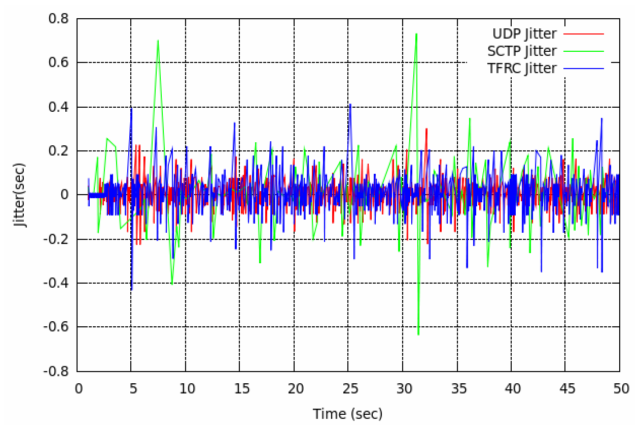

Fig. 7. Jitter for VoIP with G.726 codec

D. *Packet Delivery Ratio*

Packet loss occurs when transmitted packets across a network never reach its destination [13]. The packet delivery ratio is an important parameter affecting the performance of VoIP. However, it is expressed as a ratio of the number of packets received in the total number of packets sent.

Table III. PACKET DELIVERY RATIO OF SCTP, UDP, AND RTP

Protocol	**Packet Send**	**Packet Received**	**Packet Delivery Ratio**
SCTP	2451	2449	0.99918
UDP	2130	2070	0.97183
RTP	2451	1866	0.76132

The values shown in Table 3 depict a comparison of the packet delivery ratio of VoIP over SCTP, UDP, and RTP. SCTP and UDP reflect a higher packet delivery ratio as compared to RTP.

V. CONCLUSION

In this work, the analysis of the performance of VoIP over the wireless networks is explained. The performance metrics like throughput, end-to-end, jitter and packet delivery ratio has been evaluated using two different simulation scenarios. Results indicate that SCTP exhibit better throughput performance than UDP and RTP. Both RTP and UDP are reasonably fair in end-to-end delay among the simulation as compared to SCTP. It is obtained that in the case of RTP jitter is minimum as compared to other protocols. Further, the evaluation of the SCTP shows high values of the packet delivery ratio and hence good affects the VoIP quality.

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