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RESEARCH ARTICLE

QoS Enabled SIP for VoIP in MANET

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Abstract— Voice over Internet Protocol in Mobile ad-hoc Network (MANET) is challenging because MANET is an infrastructure-less network. VoIP requires a signalling protocol like the Session Initiation Protocol (SIP) for session establishment and management. In infrastructure network, SIP utilises the centralised proxy, register and redirect servers for session establishment. MANET is a decentralised network of mobile nodes. Hence SIP cannot adopted as it is to MANETs. In this paper, we have described the SIP implementation for MANET using network simulator 2 ns2. We have also included CODEC (Coder-Decoder) support for VoIP. CODEC 723.1 recommended by ITU-T is supported for the VoIP calls. The Quality of Service (QoS) objective for VoIP calls is determined by calculating throughput, packet delivery ratio (PDR), jitter and delay.

Keywords— VoIP, SIP, MANET, QoS, ITU-T, CODEC

I. INTRODUCTION

VoIP is an emerging technology and is becoming popular because VoIP provides the users to make low cost calls enabled by IP telephony. The Session Initiation Protocol is an application layer signalling protocol for session establishment and management. The SIP is used in infrastructure networks as a signalling protocol. The servers in SIP are: Since MANET is a decentralized network of nodes, the SIP functionality includes the following:

Discovery phase: identifying the nodes that are present in the MANET

Inviting phase: setting up sessions between nodes and session management.

In a MANET, discovering the nodes is difficult because each node does not have any idea about all the nodes in the network. Each node can only communicate with its neighbors which are in its coverage area. For two nodes to communicate with each other, the packet needs to pass through multiple nodes to reach the destination, if the source and destination are not neighbors.

The source node does not know the path to reach the destination node. In our proposed method, we are suggesting using the HELLO broadcast packets to discover the neighbors. The routing table at each node then maintains a list of its immediate neighbors. MANET routing protocol AODV is used for routing the packets[1]. We introduce two new SIP headers PREQ i.e Path Request to request path information and PRES i.e Path response to send back the path information. These headers can be embedded in the SIP INFO message. In fig 3 , if a node S say the sender wants to communicate with the receiver node R, node S broadcasts PREQ(Path Request) packets to its neighbors[2]. The neighboring nodes further broadcast to their neighbors and so on till they reach the receiver node R.

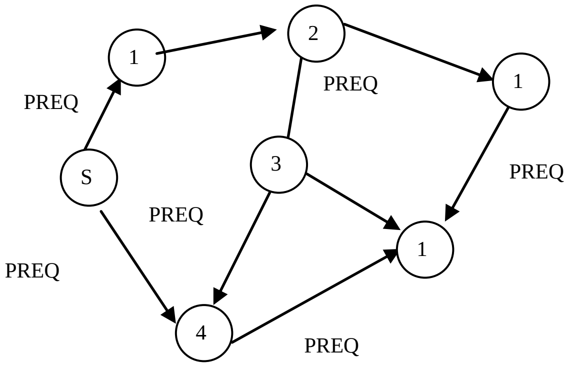

Fig. 1. Discovery phase 1: the path requesting phase(S-sender, R-Receiver)

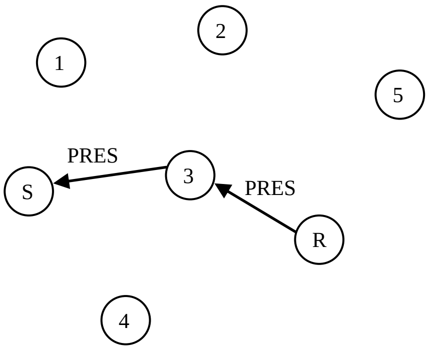

Fig. 2. Discovery phase 2: the path response phase

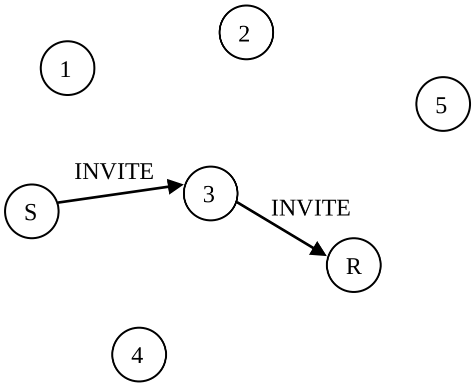

Fig. 3. The Inviting Phase: the call set-up phase

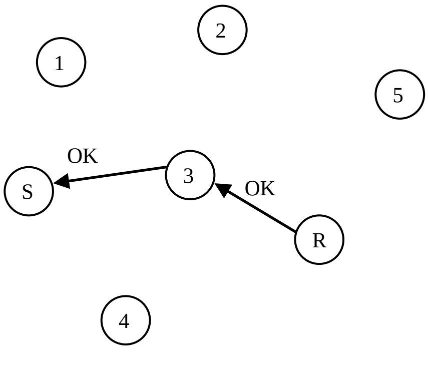

Fig. 4. The acknowledgement for the invitation phase

The receiver node has a timer which does not accept PREQ packets after the timeout period in order to eliminate the longer routes. The Receiver node R then examines all the received PREQ packets to determine the shortest cost path. The shortest cost path considers various factors like cost, distance and bandwidth. Once the shortest cost path is determined, the receiver node sends back the response PRES to the sender by routing the packet through the intermediate nodes in the path that is determined shortest as in fig 4. When the sender receives the PRES response packets, it then sends an SIP INVITE message to the receiver through that path to setup a session with the receiver as in fig 5. The receiver then sends a SIP OK message back to the sender as in fig 6. After the receiver receives SIP ACK message from the sender, both the sender and receiver can start communicating. The SIP session is setup between the sender and the receiver. They start communicating with each other. The routing protocol that we have chosen is AODV.

Ad hoc On-demand Distance Vector (AODV) is a reactive routing protocol in MANETs. AODV creates routes in on-demand basis and hence it minimizes the number of required broadcasts. Destination sequence numbers are used to ensure freedom from loops at all times.

II. SYSTEM ARCHITECTURE

The fig. 3.1 shows the overall system overview. The MANET network consists of mobile nodes. SIP session establishment to facilitate VoIP calls is a peer to peer communication between two nodes. The system architecture is as shown in the above figure 3.2. Since SIP follows a peer-to-peer communication the sender is the SIP caller who wants to set-up a VoIP session and the receiver is a SIP callee.

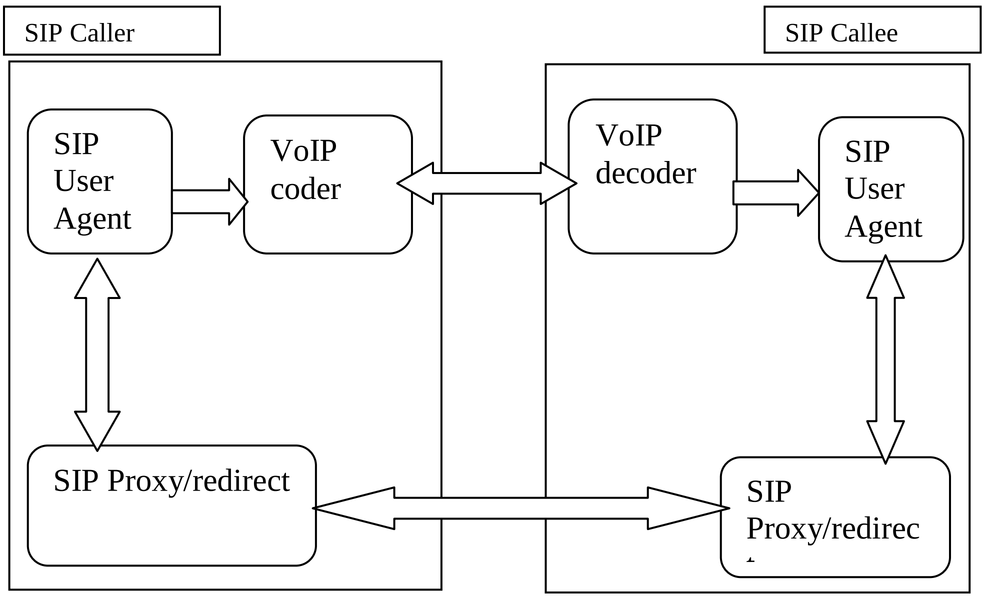

Fig. 5 SIP for VoIP in MANET System architecture

Module 1: The SIP user agent module is responsible for populating the SIP header fields like to address, from address, other contact information and forming the SIP packet. The SIP request then is forwarded to the SIP Proxy/Redirect module.

Module 2: The routing table is maintained by the SIP Proxy/Redirect module. This module is also responsible for routing the SIP packet to the appropriate next hop neighbouring node and at the callee end to forward it to the SIP user agent module. The SIP User Agent module at the callee end then sends an acknowledgement back to the SIP Proxy/redirect module which routes it back to the caller SIP user agent module. The SIP session is set-up [4].

Module 3: The VoIP coder/decoder module is responsible for compressing the voice data at the caller end and decompressing at the callee end. The codec that is adopted is G. 723.1. VoIP communication then takes place between the caller and the callee.

III. ANALYSIS OF THE QOS RESULTS

A NAM simulation window has a MANET consisting of 50 nodes. Node 1 is the sender/caller and node 2 is the receiver/callee. The SIP session establishment and the VoIP communication is seen in the simulation. The trace file has a log of all the communication that happens between the node 1 and node 2. The awk script parses the file to retrieve appropriate data to generate graphs. This data is fed to Gnuplot utility to generate graph for QoS parameters like throughput, jitter, delay and packet delivery ratio. The results are analysed. Throughput for VoIP with codec and without codec [5].

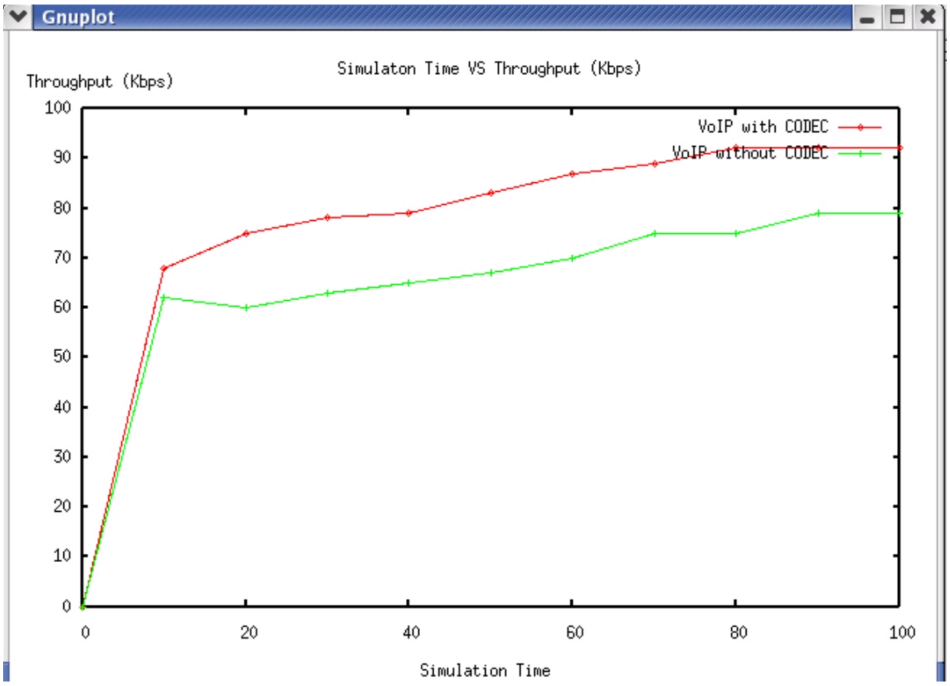

Fig. 6 Throughput for VoIP in MANET without codec and with codec

Simulation time (Seconds)	Throughput for VoIP without codec(Kbps)	Throughput for VoIP with codec	Percentage increase in throughput
20	60	75	15%
40	65	79	14%
60	70	87	17%
80	75	92	17%
100	79	92	13%

Table. 1 Throughput for VoIP in MANET

The throughput for VoIP call without codec and with codec is calculated and depicted in the graph in the fig. 6. As seen in the figure 6 the network throughput for VoIP call increases with codec compared to that of VoIP without codec. The analysis is as shown in table. The Throughput (Threshold) obtained without codec is 79 Kbps and with codec is 92 Kbps. The percentage increase maximum that is obtained is 17%. There is a steady increase in Throughput for VoIP with codec as compared with VoIP without codec as seen in table 1.

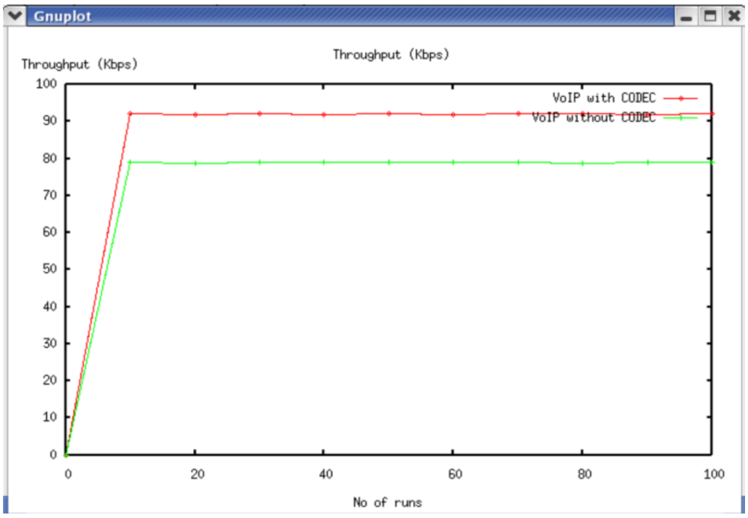

Fig. 7 Throughput for VoIP for 100 runs

As shown in the table the throughput of VoIP with codec is higher than that of the throughput of VoIP without codec. The simulation was run for 100 times to observe the variation in throughput as in figure. The results are tabulated in table 2[6].

No of runs	**Throughput for VoIP without codec(Kbps)**	**Throughput for VoIP with codec(Kbps)**	**Percentage increase in throughput**
20	78.8	91.8	13%
40	79	91.9	12.90%
60	79	91.8	12.80%
80	78.8	92	13.40%
100	79	92	13%

Table 2 Throughput for VoIP when the simulation is run 100 times

The packet delivery ratio is computed and plotted for VoIP without codec and VoIP with codec. The graphical representation for PDR is as in fig 8. The table shows the computational results for PDR [3].

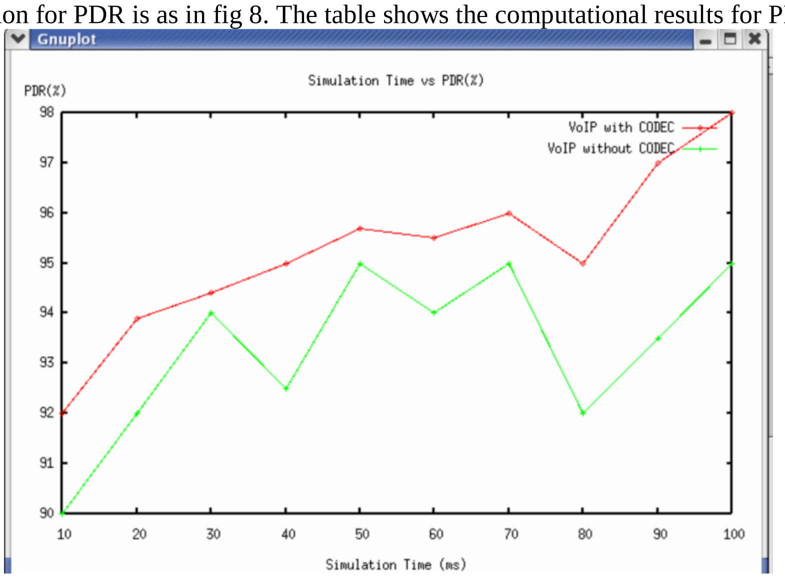

Fig. 8 PDR for VoIP without codec and VoIP with codec

Table 3 PDR for VoIP with codec and without codec

Simulation time (Seconds)	**PDR for VoIP without codec (%)**	**PDR for VoIP with codec (%)**	**Percentage increase in PDR**
20	90	93.9	3.90%
40	92.5	95	2.50%
60	94	95.5	1.50%
80	92	95	3%

The PDR for VoIP call without codec and with codec is calculated and depicted in the graph in the fig. 3. As seen in the figure 7.3 the network PDR for VoIP call increases with codec compared to that of VoIP without codec. The analysis is as shown in table 3. The PDR ($_{\text{Threshold}}$) obtained without codec is 94 Kbps and with codec is 95.5 Kbps. The percentage increase maximum that is obtained is 1.50%. There is a steady increase in PDR for VoIP with codec as compared with VoIP without codec as seen in table 3[7].

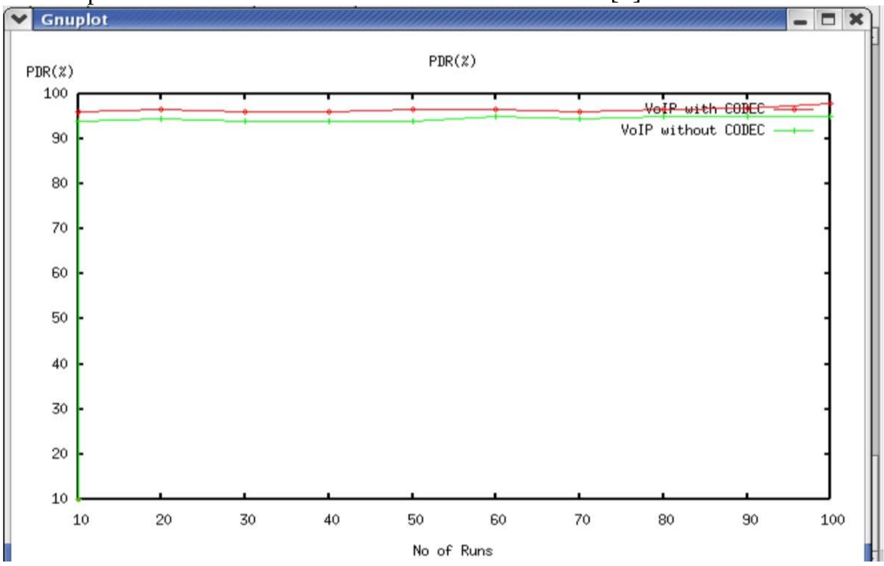

Fig. 9 PDR for VoIP without codec and VoIP with codec for 100 simulation runs

Table 7.4 PDR for VoIP without codec and VoIP with codec for 100 runs

No of Runs	**PDR for VoIP without codec (percentage)**	**PDR for VoIP with codec (percentage)**	**Percentage increase in PDR**
20	94.5	96.5	2%
40	94	96	2%
60	95.0	96.5	1.50%
80	95.0	96.5	1.50%

When the simulation is run 100 times, the PDR $_{\text{threshold}}$ obtained for VoIP without codec and with codec is 95% and 96.5% respectively. Therefore there is a steady increase of PDR for VoIP with codec.

The delay in packet delivery as in fig 10 is computed from the trace file and tabulated for a simulation time of 100 sec for VoIP without codec and VoIP with codec [8].

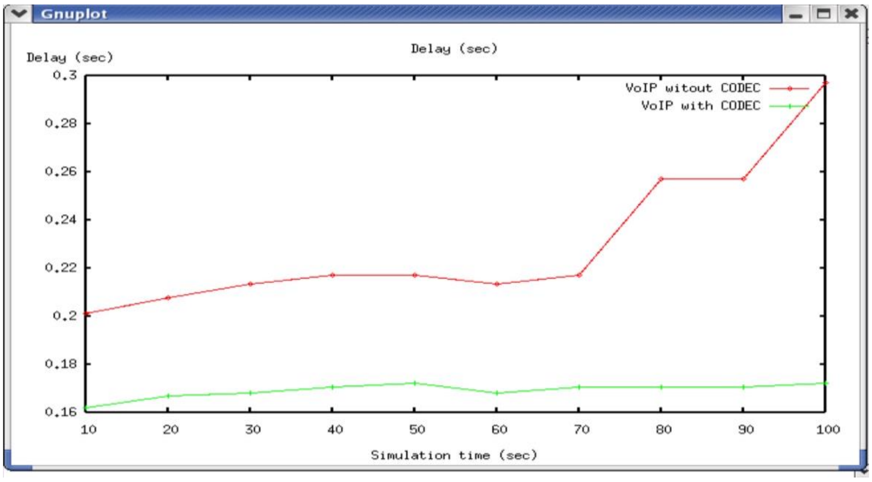

Fig. 10 Packet delay for VoIP without codec and VoIP with codec

Table 5 Packet delay for VoIP without codec and VoIP with codec

Simulation time (Seconds)	**Delay for VoIP without codec**	**Delay for VoIP with codec**	**Decrease in delay**
20	0.2078	0.1668	0.041
40	0.217	0.1704	0.046
60	0.2132	0.1682	0.045
80	0.257	0.1704	0.086
100	0.297	0.1721	0.124

The simulation is run for 100 times and the delay computed for each time. There is a decrease in delay for VoIP with codec.

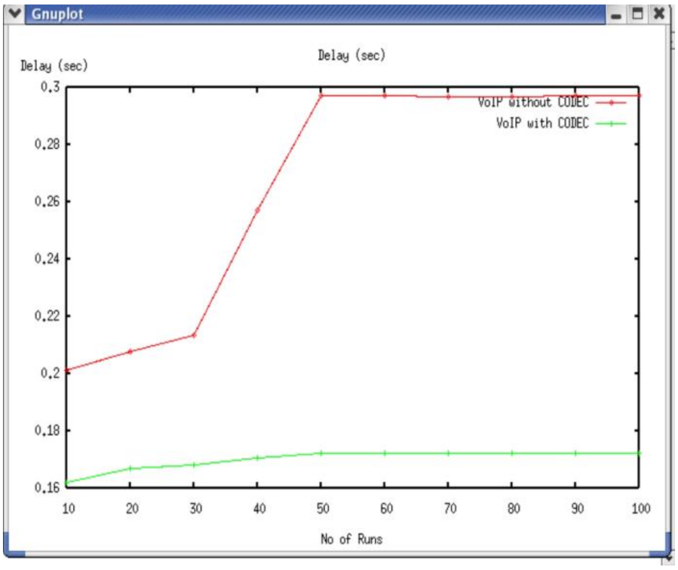

Fig.11 Packet delay for VoIP without codec and VoIP with codec when simulation is run 100 times

Table 6 Packet delay for VoIP without codec and VoIP with codec when simulation is run 100 times

No of Runs	**Delay for VoIP without codec(sec)**	**Delay for VoIP with codec (sec)**	**Decrease in delay**
20	0.2078	0.1668	0.041
40	0.257	0.1704	0.0866
60	0.297	0.1721	0.1249
80	0.2967	0.1721	0.1247
100	0.2972	0.1721	0.1251

The delay in delivering packet is computed for VoIP with codec and VoIP without codec. The simulation is run 100 times and the delay is computed. The delay (max) for VoIP without codec is 0.2972 seconds and VoIP with codec is 0.1721 seconds. It is observed that that the delay for VoIP with codec is less than the VoIP without code. Hence achieves the QoS objective.

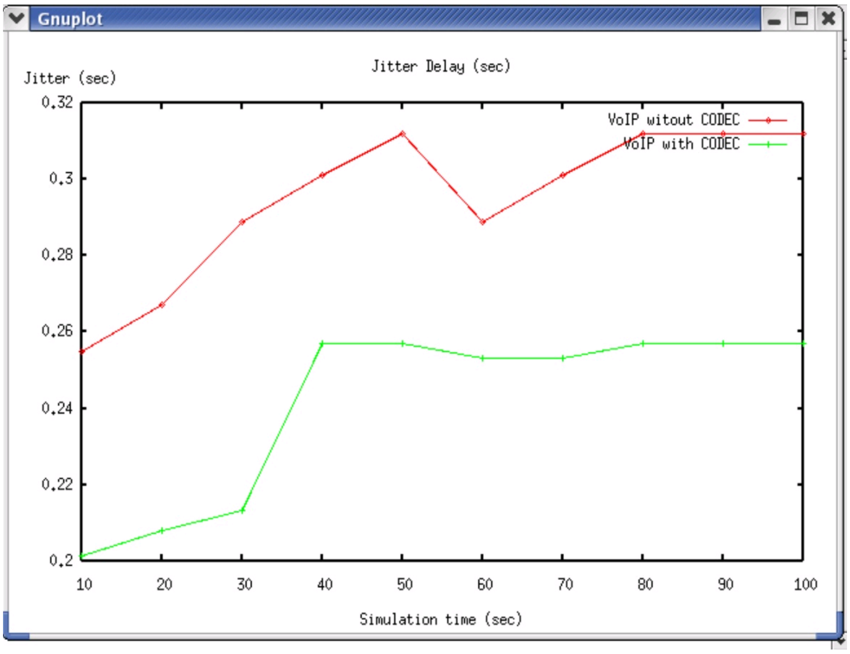

Fig. 7 Jitter for VoIP without codec and VoIP with codec

The jitter is a variation in delay in the delivery of packets and is computed as shown in table 7.7. Jitter is variation in delay in delivery of packets from source to destination. For a simulation time of 100 seconds, the jitter maximum obtained is 0.312 seconds for VoIP without code and 0.257 seconds for VoIP with codec.

Table 7 Jitter for VoIP without codec and VoIP with codec

Simulation time (Seconds)	Jitter for VoIP without codec(sec)	Jitter for VoIP with codec (sec)	Decrease in jitter (sec)
20	0.267	0.207	0.06
40	0.301	0.257	0.04
60	0.289	0.253	0.03
80	0.312	0.257	0.05
100	0.312	0.257	0.06

The VoIP communication in MANET is implemented without codec and with codec. Various QoS parameters like throughput, packet delivery ratio, jitter and delay are computed from the values obtained from the trace file after running the simulation for 100 seconds. The simulation is run for 100 times an all the above mentioned QoS parameters are computed for each run and computed. The results and metrics show that VoIP in MANET with codec has higher throughput and PDR and lower delay and jitter as compared to VoIP without codec. Hence VoIP with codec achieves the QoS objective than VoIP without codec.

IV. CONCLUSION

The Session Initiation Protocol which is essential for session establishment for VoIP in MANET is proposed and implemented in this project. The SIP is decentralized as MANETs are also decentralized networks. Once the SIP session is established the VoIP communication takes place between the caller and the callee. The QoS objective is very essential as VoIP traffic is a real time traffic. Hence the VoIP QoS objective is achieved by implementing the VoIP codec 723.1 which encodes the VoIP data before communication and at the callee end decodes back the VoIP data [9][10]. The comparison study of VoIP communication as in chapter shows that the VoIP data after using the codec has higher throughput, low delay and jitter and higher packet delivery ratio. Hence SIP for VoIP using codec in MANET has achieved the QoS objective of higher throughput and PDR and lower delay and jitter for VoIP packets.

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