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QoS Evaluation of SIP Signalled VoIP Network Routed using MANET Routing Protocols

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Abstract- A Mobile ad hoc network (MANET) is a type of network which consists of group of mobile nodes which are wireless and do not have fixed architecture. The nodes act as a router and depict the nature of dynamism. The three different classification of protocols in MANETS supports different applications. But to support real time applications like voice signalling and video signalling, we require the most efficient protocol that gives the QoS mechanism. Voice and video signalling demand to know the performance of different metrics in the network such as end-to-end delay, overall throughput of network and jitter of the network. This paper works on identifying and analyzing the performance of various protocols like AODV, DSR, OLSR and TORA which would help in fulfilling the mentioned need. Voice over Internet Protocol (VoIP), also known as IP telephony is a class of technologies used to deliver voice and multimedia sessions over internet protocol networks.

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GJCST-E Classification: C.2.2

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Abstract- A Mobile ad hoc network (MANET) is a type of network which consists of group of mobile nodes which are wireless and do not have fixed architecture. The nodes act as a router and depict the nature of dynamism. The three different classification of protocols in MANETS supports different applications. But to support real time applications like voice signalling and video signalling, we require the most efficient protocol that gives the QoS mechanism. Voice and video signalling demand to know the performance of different metrics in the network such as end-to-end delay, overall throughput of network and jitter of the network. This paper works on identifying and analyzing the performance of various protocols like AODV, DSR, OLSR and TORA which would help in fulfilling the mentioned need. Voice over Internet Protocol (VoIP), also known as IP telephony is a class of technologies used to deliver voice and multimedia sessions over internet protocol networks. The terms internet telephony, broadband telephony provides provisions over public internet networks rather than public switched telephone networks. Since the smart phones have evolve, VoIP has more popularity and its performance optimization has become a research interest. The VoIP network consisting of wireless nodes which are signalled through SIP are simulated with the help of OPNET Modeller 17.5. For the mentioned protocols which are useful in VoIP applications, their performance evalutions have been experimented and various conclusions have been put forth.

Index Terms: MANET, Router, QoS, AODV, DSR, OLSR, TORA, VoIP, SIP.

I. Introduction

The MANET is a group of wireless nodes which are mobile in nature. They do not contain a central access point or any established infrastructure. Every node act as a router in order to establish communication between other entites in the network. These networks reflect dynamism which results irregular topology causing a complicated traffic among the nodes. The different protocols available are classified as reactive, proactive, hierarchical, flat, adaptive and geographical. Each of the above-mentioned category have their own set of protocols. Based on algorithmic designs the proactive and reactive protocols are most known. Each protocol has a unique nature and are designed differently. Routing efficiency has become a major issue in MANETS as they have mobile nodes. So, any protocol selected should be efficient in facing the challenges posed by the network. Since each protocol is designed differently, they provide one or more than one solution to the challenges faced by the network Voice over Internet Protocol has been seen to gain immense popularity and is most common to most of the applications. The use of VoIP application is to such an extent that it has replaced most of the conventional telephone systems in the developed nations as VoIP has been found to be not so expensive and is compatible for systems to switch to new technologies. As understood, VoIP makes use of public internet for its communication so the input voice data is transformed to IP packets which are transmitted from source node to destination node through a secure channel using the protocol selected for its routing over the internet.

Various factors determine the VoIP QoS performance over MANET routing protocols which include mobility of nodes, voice codec, voice quality and distance between communicating pair, hop count, node capability, wireless LAN technology and duration of calls. The VoIP with GSM quality voice codecs has been considered, which has good quality of voice and performance over large varieties of systems and applications.

The main aim of paper is examining the QoS in SIP signaled VoIP application that uses MANET reactive routing protocols for its routing. The WLAN technology considered here is IEEE 802.11n which serves as physical layer technology. The nodes have been addressed through IPv4. The OPNET Modeller 17.5 is used for simulations. Both TCP and UDP based signalling of SIP has their impact on QoS in VoIP applications.

a) SIP Signalling

Session Initiation Protocol is one of the common protocols used in VoIP technology. It belongs to application layer protocol that works in conjunction with other application layer protocols for the purpose of signalling and controlling multimedia applications like voice and video calls. The messages are sent between communicating pair i.e., the nodes to establish and terminate the calls among them. It is similar to HTTP and SMTP which involve message requests and message responses. So it is known as a text based protocol. It was defined by SIP working group and was published as IETF (RFC 2543). A SIP session may include more than one participant or application as it has internal

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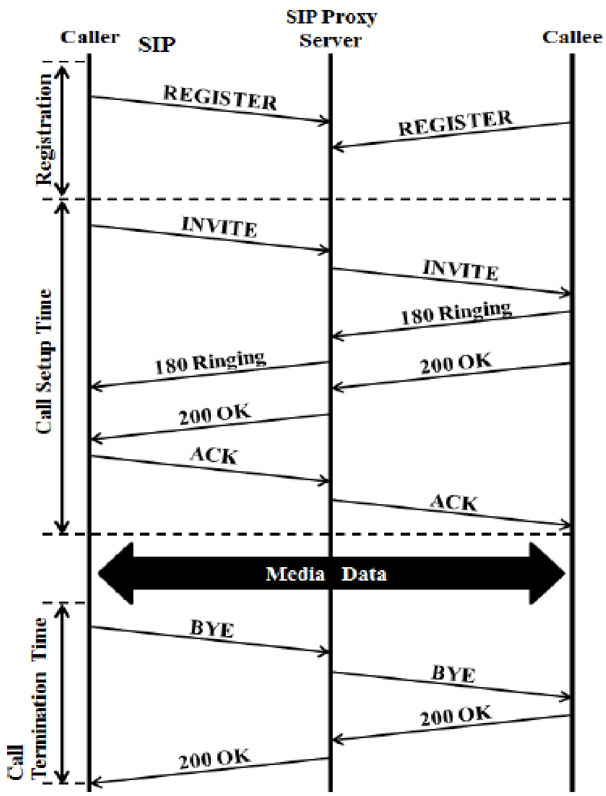
functionality to allow extensions and modifications. The various elements of SIP session are replicated by the changes in the code.



This protocol is dependent on internet protocols but independent of transport layer. A SIP based session or application consists of three stages. i. The Registration ii. The Initiation and iii. The Termination. The working of these stages depends on SIP proxy server for connectivity between nodes. The application performance is mostly affected due to delays which occur in the process of these stages. The acceptable average delay in a SIP system is in the range of [0.145, 0.345] seconds.

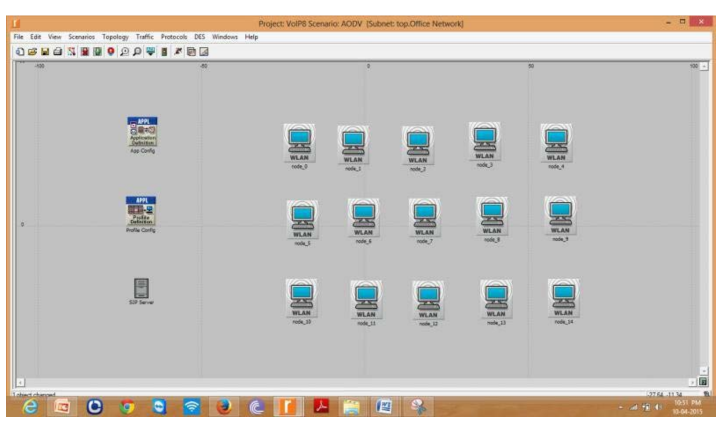
b) VoIP Applications

VoIP application is similar to using a microphone to record an audio message and storing it in a memory. In VoIP the message is not stored in a memory, rather it is disintegrated and transformed into IP packets which are transmitted over IP network. VoIP calls support any kind of device like a computer, a smart phone or a traditional telephone. The process of fragmentation into IP packets and then their transmission leads the packets to arrive in an arbitrary order. This reordering of packets is the issue as it may cause some of the packets to drop leading to silence in the calls for short time. The quality of VoIP calls depends on jitter, end-to-end delay, MOS value, throughput and coding schemes. A lot of research is being carried out in order to improve the reliability and quality of VoIP application. This paper discusses the different parameters that deteriorate the call quality.

c) MANET Routing Protocols

MANET is one of the growing and eminent technology in the field of telecommunication. It is self configured, infrastructure less, wirelessly connected without any central access point. In basic sense routing protocols are divided into Flat, Hierarchical, Geographic position assisted types. The flat routing protocols are further divided as proactive and reactive protocols. The Reactive routing protocols are the On-demand routing protocols which calculate the the routes when needed. These are AODV and TORA known as source-initiated route discovery protocols. On the other hand, proactive routing protocols calculate the shortest paths between nodes depending on updates on the routing tables. It includes OLSR and DSDV. The hybrid routing protocols contain the features and functionality of both proactive and reactive protocols.

II. VoIP Network Topology



Our main concentration is towards VoIP QoS, so mobility in nodes is ignored and are restricted to static model. The entire network is configured to function as VoIP network with GSM application and voice codec as G711. Different scenarios have been created for different MANET routing protocol but the topology, application configuration and other parameters have been kept constant so that we obtain an ideal comparision among the protocols used.

III. Simulation Parameters in Opnet

Simulation Duration		10 Minutes	
Mobility Model		Static	
MANET Routing Protocols		DSR,AODV,OLSR,TORA	
No. of Nodes	16	Area Dimension	100 m x 100 m
WLAN Physical Characteristics		IEEE 802.11n	
Data Rate		13 Mbps	
Frequency Range	2.4 GHz	Transmission Power	0.001 W
Packet Size	512 B	Buffer Size	32 Kb

OLSR Parameters

Hello Interval (Seconds)	3	Neighbour Hold Time (Seconds)	6
TC Interval (Seconds)	5	Topology Hold Time (Seconds)	15
Duplicate Message Hold Time (Seconds)	10	Addressing Model	IPv4

AODV Parameters

Hello Interval (Seconds)	Uniform (1,1.1)	Active Route Timeout (Seconds)	3
Allowed Hello Loss	2	Node Traversal Time (Seconds)	0.04

Route Error Rate Limit	10	Timeout Buffer	2

DSR Parameters

IV. Simulations

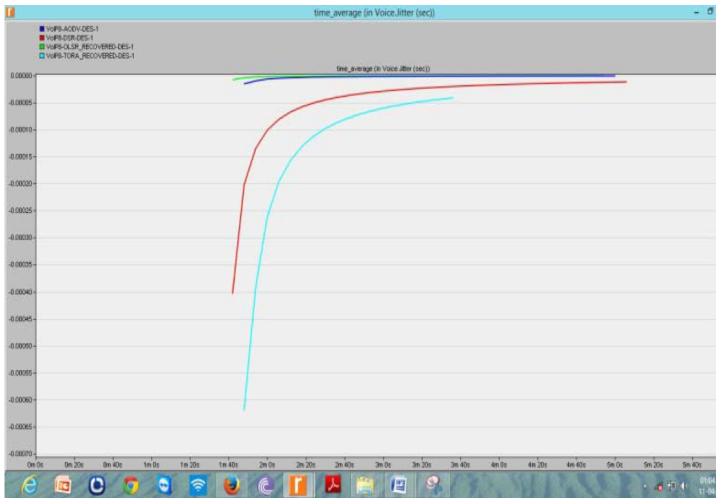
The software used for simulations of VoIP network is OPNET Modeller 17.5. the simulation consists of an office setup with dimension of 100x100 metres. The technologies selected from the dropdown menu are MANET, SIP and Voice Signalling. The wireless LAN workstations are selected from the object palette as user nodes and are placed in the workspace provided. A SIP proxy server is selected as network node. The application configuration attributes are edited and are set to voice application. The attributes are changed to GSM voice application with voice codec G711. The user profiles are created using profile configuration.

V. Results and Evaluations

The simulation results are provided in two groups as – the Voice statistics and the wireless LAN characteristics. The parameters jitter, MOS value and packet end-to-end delay come under the voice statistics and the throughput and delay come under WLAN characteristics. All statistics are found under global statistics of the modeler. The traffic sent and the traffic received are also analysed through the simulation. In the graphs obtained, the horizontal axis depicts the simulation time in seconds and vertical axis depict the values of evaluated statistics which include jitter, throughput, MOS value, etc. The total simulation time is set to 600 seconds, in which the initial results of first 150 simulation seconds are difficult to analyse. Therefore, only rest of the 450 simulation seconds are taken into consideration for estimation of performance of application under different conditions.

a) Voice Signalling Statistics:

Jitter (Seconds)

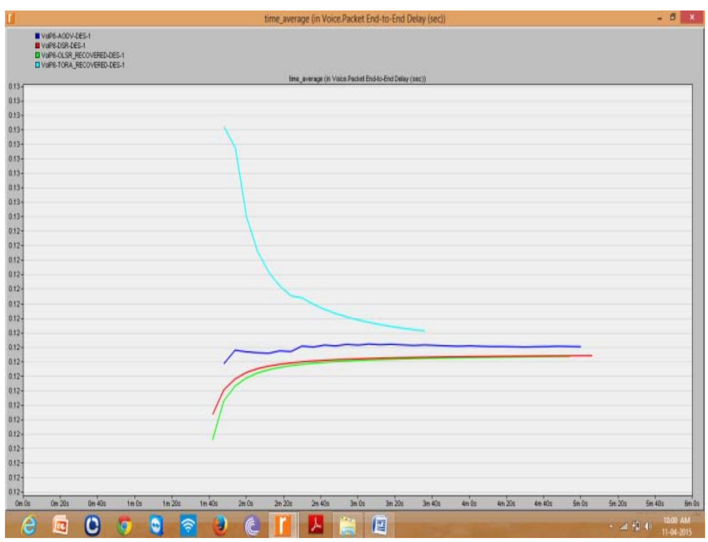


The following graphs show the time averaged jitter values of the VoIP signal with the different MANET protocols.

If two packets leave the source node at time interval t1 and t2, the same packets replay at the receiver at time interval t3 and t4 respectively, then the jitter is (t4-t3)-(t2-t1). Negative jitter indicates that the time difference between the packets at destination loads is less than that at the source node.

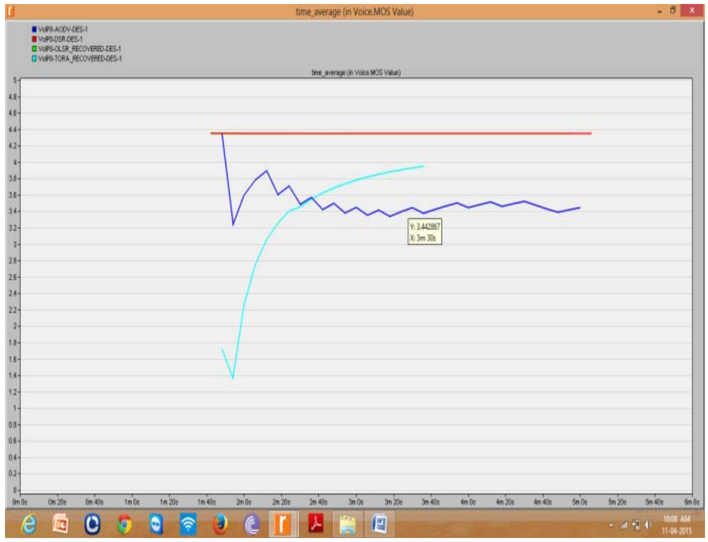
Packet end-end delay

The following graph represents the time averaged packet end to end delay of the VoIP network over the MANET routing protocols.



MOS Value

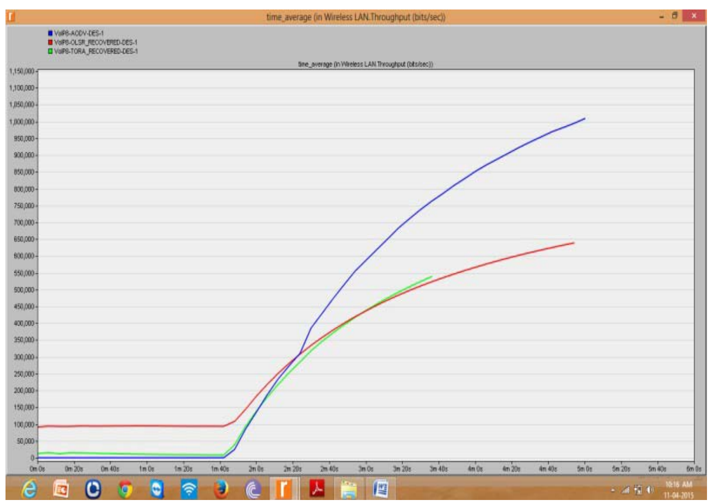
The following graph represents the time averaged MOS value of the VoIP network over the MANET routing protocols.



b) WLAN Statistics

Throughput

The following graph represents the throughput values of the VoIP network when routed through the MANET Routing Protocols.



VI. Conclusions

The above study gives a clear idea of various parameters of VoIP network routed through three reactive protocols: DSR, AODV and TORA and a proactive protocol: OLSR. This segment of paper talks about the behavior of each protocol atdifferent simulation time intervals.

To start with jitter values, we observe that DSR and TORA have negative jitter values whereas OLSR and AODV have almost zero jitter value. This tells us that frequency of IP packets at receiver end is small compared to source end. The receiver end is also noticed to face difficulty in synchronizing and reattaching the received packets. In terms of real time appications, jitter is found to be not a good parameter to be considered for knowing the quality of voice. Keeping in mind the above problem, OLSR and AODV dominate the other protocols.

In the initial time period of 120 seconds, the packet end-to-end delay was found to be low when DSR, AODV and OLSR were used. But after 120 simulation seconds, it increased and maintained constant value. On the contrary, in case of TORA same statistic was high in the initial and de- creased as simulation came to an end. As delay cannot be entertained in voice applications, use of TORA has been avoided. Among the other three protocols we cannot conclude on the better among them as they have minimal difference.

Higher is the Mean Opinion Score that better is any application. From the graphs, it is very clear that the protocols DSR and OLSR have an unbeatable MOS value compared to that of AODV and TORA. The AODV initially maintained a good score but that didn't maintain longer. The TORA routed network performed the least among all the protocols considered. In this context, the usage of AODV and TORA are strictly not recommended. Taking all the above three observations into consideration, among the four MANET routing protocols considered, from the reactive routing protocols, the DSR is observed to be the optimal one to use for the voice applications and on the other side, among the proactive routing protocols, the only considered Optimized Link State Routing protocol managed to perform equally as the DSR. Hence, the paper suggests the usage of both the DSR and the OLSR based on the requirements of the application end.

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