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Performance Analysis of VoMAN using Routing Protocols to Improve QoS of VoIP flows

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Abstract- Voice over Internet protocol (VOIP) or IP Telephony is a technology that acts as an alternative approach to voice communication over Public Switched Telephone Network (PSTN). It supports the transmission of voice by encapsulating and routing voice packets over Internet Protocol. Its main objective is to cut off the cost for making long distance telephone calls. This paper discusses the overview, issues and challenges of VoIP and presents the comparative performance analysis of VoIP using EMODE-1 codec with respect to QoS metrics Packet Delivery Ratio, End-End-Delay, Packet Dropping Probability, Throughput, and Residual Energy. It compares three MANET routing protocols AODV, DSR, OLSR and analysis shows that OLSR is moderate than DSR and AODV with respect to QoS metrics End-end-delay and Packet Dropping Probability, supporting high residual energy.
Keywords : VoIP, PSTN,TCP,EMODE-1,QoS, RTP

I. INTRODUCTION

MANET is a network of mobile nodes such as Laptop, sensors, etc., interfacing without centralized infrastructure (Access point, bridges, etc,.). Mobile nodes move in and out of the network randomly. Thus the topology and link between nodes changes frequently. This link breakage causes packet loss and latency problem in the network and it degrades the performance. Each node in MANET acts both as a host and as a router to forward messages.

VoIP has been identified as a widely accepted technology since of the emergence of Voice and data network into single unit. VoIP communication over mobile network involves four steps namely, 1. Call Setup or establishing communication between caller and called node. 2. Data processing or data transfer between caller and called node. 2. To setup call and to terminate the call it requires a signaling protocol [1]. 3. To encode and decode on the sender and receiver side, it uses Codec. 4. To transfer voice data and to manage it over the media it requires media protocol. It uses popular signaling protocols include SIP, H.323, etc., and media protocols include Real Time Transport Protocol (RTP), RTP Control Protocol (RTCP). These RTP packets are encapsulated into transport protocols to transmission over IP. UDP is the most preferable protocol for transferring voice over IP network.

Quality of Service (QoS) is set of service requirements that are met by the network while transferring packet streams from source to destination. It is a significant component to improve the performance of VoIP flows. Metrics for QoS could be defined in terms of one or set of parameters. Examples: delay, bandwidth, packet loss, delay-jitter, etc. Each metric has constraints based on which routing protocols should react. The violation of the constraints of these metrics degrades the performance of VoIP application to an unacceptable level especially for the metrics End-end-delay and Packet loss. Protocols can be classified as single constraint or multiple constraints based on the idea of whether they support single or multiple constraints. Savithri et.al [11] presents the detailed classification of QoS constraints based routing protocols in MANET.

CHALLENGES OF VOIP SYSTEM

- The main objective of the VoIP system is to deliver voice with the same quality as it is produced by the caller. Hence the key issues of VoIP system are end-end delay or latency, Delay Variance (Jitter), Packet Loss. Preventing voice packets from packet loss can be accomplished by providing an advanced queue management approach that monitors the incipient congestion over the line and drops the incoming packets prior to the performance of VoIP.

- Another important issue is reducing end-end delay or latency. Latency is the time taken by the packet to reach from caller to callee. For each packet the latency or end-end delay is calculated and it is compared with the maximum life time of the packet to know whether the voice packet has its life time within acceptable limit or not. If the packet crosses its lifetime then it is discarded before reaching the callee without affecting the quality of VoIP packet as in [2].

- Preventing voice packets from intruder or providing a secured communication between the caller and callee.

- Reducing the time taken to set up call between the caller and callee. This is more challenging issue in case of mobile VoIP systems where session setup latency is affected by mobility parameters and threshold. This can be reduced by prefetching location information of mobile nodes from location server and if the callee is present at the same location, communication is established. Otherwise a time out event occurs and the conventional session set up is made between the caller and the callee[3].

The rest of the paper is organized as follows. Section II we overview related work . In Section III, we discuss the routing protocols. Section IV describes the features of VoIP simulation methodology. Section IV discusses the simulation results and comparative analysis in terms of QoS metrics using XGraph. Finally, we conclude the paper in Section VI.

II. VOIP OVER MANET

Figure.1 shows the block diagram involved in the communication of VoIP system as discussed in [4]. Each speech source is an alternating sequence of talking and silence period and it is exponentially distributed. It undergoes a digitalization process that includes sampling, quantization and encoding. First step involved in voice communication is to convert the analog signal to digital signal using voice codec on the sender side. It compresses the voice data without affecting its quality. It is packetized and encapsulated into IP packets and passed over IP network. The receiver on the other side has play out buffer to store voice packets; it is depacketized and decoded using codec which does compression/decompression of voice packets to preserve the bandwidth of the network.

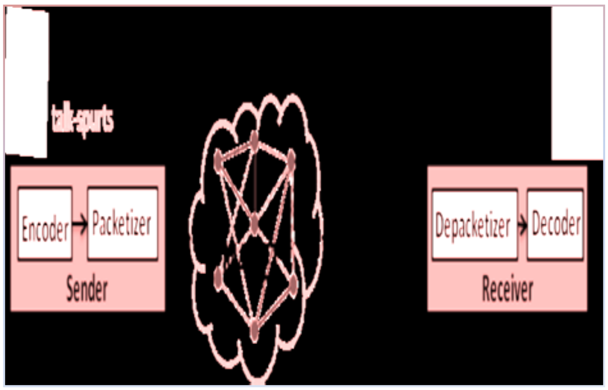

Fig 1. VoIP System Block Diagram

Sumit Mahajan et. al [5] dicussed the performance of VoIP over MANET with scalability using routing protocols and concluded that the overall performance of OLSR is better for small and large networks and the performance of AODV is poor for high populated network. Jae-Yul Yoon et. al. [6] discussed about a new method of improving the speech quality of VoIP by assigning a priority to each packet at the edge router of a DiffServ-based network in an environment prone to packet loss. [7] proposed two MOS measurement procedures for mobile devices which are use for measuring the quality of VoIP flows. In the PESQ MOS measurement procedure, a lightweight real-time table lookup solution significantly reduced the computation time of PESQ MOS from 315.4 s to 3 s. In the E-model MOS measurement procedure, MOS value can be accurately computed in 5.35 s. These results indicated that the proposed approaches can effectively provide the real-time VoIP quality measurement. Arlen Nascimento et. al [8] discussed on the security of VoIP call by establishing calls between mobile stations in BSS and PICONET and concluded that QoS tool based on E-Model is the efficient tool for providing secured VoIP call in BSS and PICONET environment. [9] discussed about the performance of VoIP using four routing protocols AODV, DSR, OLSR and TORA over Hybrid MANET and concluded that the overall performance of OLSR is best compared to other three routing protocols.

III. ROUTING PROTOCOLS

Routing is a core part of MANET in which communication takes place using protocols of different nature. Routing in MANET is broadly classified into three types, Proactive or table driven, Reactive or On-Demand, Hybrid. In case of proactive routing, routing information is static and it is used to discover the route. Reactive routing discovers route dynamically to route packet. Proactive routing is best for network that supports low mobility or where the transmission is frequent. Examples of proactive routing protocols are DSDV, OLSR(Optimized Link State Routing). Reactive/On-demand routing Protocols include AODV(Ad hoc On-Demand Distance Vector Routing (AODV),TORA (Temporally Ordered Routing Algorithm), DSR(Dynamic Source Routing). This article discusses about three routing protocols AODV, DSR, OLSR.

Dynamic Source Routing (DSR)

DSR uses source routing as the key feature. That is, sender knows the complete hop-by-hop route to the destination. This information is stored in route cache. Data packets carry the source route in the header of the packet. When a source sends packets to the destination for which the route is not known, it uses route discovery process to determine the route dynamically. Route discovery works by flooding the route request (RREQ) packets. Each node receiving RREQ, broadcasts it to other nodes if it is not the destination. If the receiving node is the destination, it replies to RREQ with RREP packets. The route carried by RREP is stored at route cache for further use.

Ad hoc On-Demand Distance Vector Routing (AODV)

AODV is an improvement of DSDV and DSR. It is reactive in nature and uses routing table that has one entry per destination. An important feature of AODV is that a routing entry which is not recently used is expired. That is, if the routes are not used for some period of time, they are considered to be no longer valid and the corresponding entries are removed from the table. Another important feature is that it notifies route breakage to its neighbors [5]. It uses the concept of route discovery and route maintenance of DSR and the concept of sending periodic hello messages and sequence numbers from DSDV. Sequence numbers are used to determine whether routing information is up-to-date and to prevent routing loops. When a source wants to communicate to some destination, the protocol starts route discovery by sending a route request message to its all its neighbours. The neighbour node on the active path sends route reply message to the route request message initiator. A unique id is assigned, to avoid duplicate route request message. When a node receives, it will check this id and the address of the initiator and discarded the message if it had already processed that request. Node that has information about the path to the destination sends route reply message to the neighbour from which it has received route request message. When a route reply message reaches the source it can start sending data packets.

Optimized Link State Routing(OLSR)

It is a proactive routing protocol that minimizes flooding overhead by selecting nodes called Multi Point Relay nodes(MRP$_s$) to transmit Topology control (TC) packets in an optimized way. This reduces the retransmissions required to flood a message to all nodes. The TC packet contains list of his neighbors which enables all nodes to update their routing tables. OLSR uses also periodic broadcast of hello packets at one hop to sense the neighborhood of a node, to verify the symmetry of a radio link, and to select the MPR nodes. It requires only partial link state to be flooded to select shortest path between nodes. The minimal set of link state information is required for each MPR node to declare their links to their MPR selectors. The protocol reduces the time taken for transmitting periodic control message. It maintains routes between all destinations in the network. Hence the protocol is best suited for large and dense networks.

IV. Simulation Results And Analysis

The performance of the above discussed routing protocols during the transmission of voice is simulated using NS-2. The QoS metrics include the parameters such as Packet Delivery Ratio, Packet Dropping Probability, Throughput, End-end-delay, Routing overhead. The following table shows the simulation parameters.

Table 1. Simulation Parameters

Parameters	**Value**
Routing Protocols	AODV,OLSR,DSR
MAC Layer	802.11
Packet Size	512bytes
Terrain Size	800X800
Nodes	50-100
Mobility Model	Random Waypoint model
Data Traffic	TCP
Simulation Time	100
Codec used	EMODE-1

Packet Delivery Ratio: It is the ratio of the packets received by destination to those generated by the sources. TCP traffic type is used by source. It specifies the packet loss rate, which limits the maximum throughput of the network. The routing protocol which has better PDR is more complete and correct. This reflects the usefulness of the protocol.

$$PDF = \frac{\text{Number of received packets}}{\text{Number of sent packets}}$$

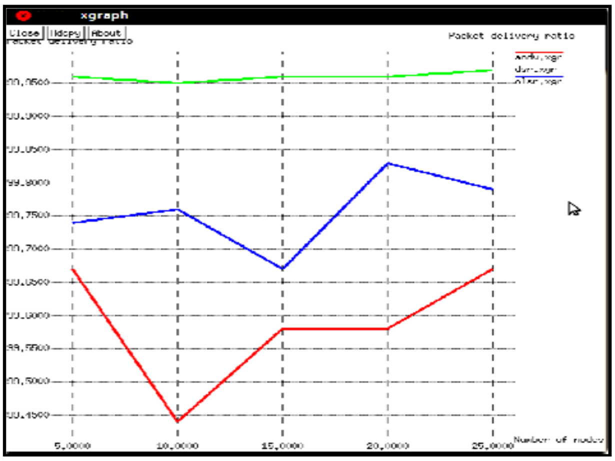

Fig 1. Number of nodes Vs Packet Delivery Ratio

Throughput

It is the number of packets passing through the network in a unit of time. It is measured in kbps.

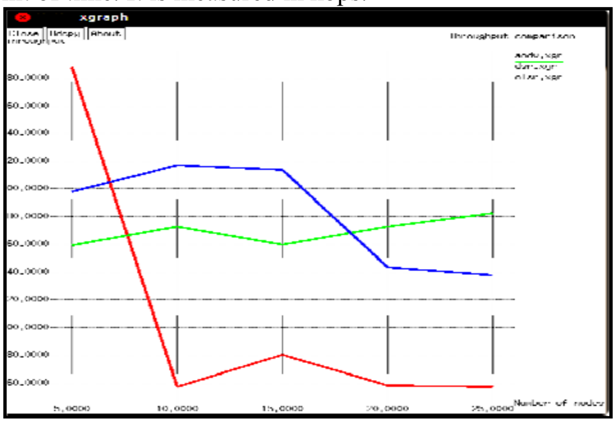

Fig 2. Number of nodes Vs Throughput

Packet Dropping Probability

The number of data packets that are not successfully sent to the destination are known as dropped packets. There are various packet dropping schemes available in wired network to reduce packet loss and to improve the performance of queue management as discussed in [10]. In MANET, it is still in its early stage.

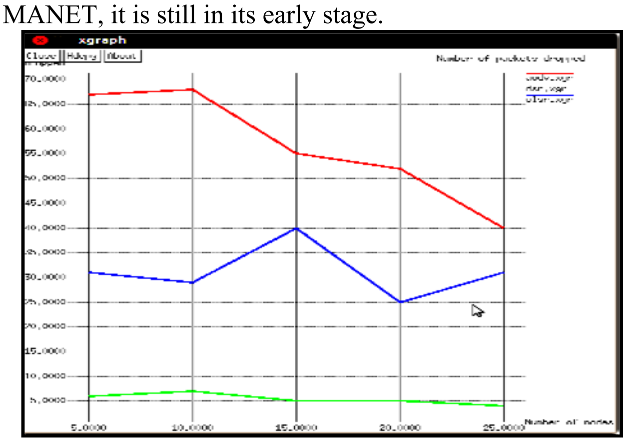

Fig 3. Number of nodes Vs Packet Dropping Probability

Average End-End Delay

Average End-to-end delay is the average time delay for data packets from the source node to the destination node.

To find out the End-to-end delay the time difference of packet sent and received was stored and then dividing the total time difference over the total number of packet received gives the average End-end delay for the received packets. The performance of the protocol is better when packet End-to-end delay is low.

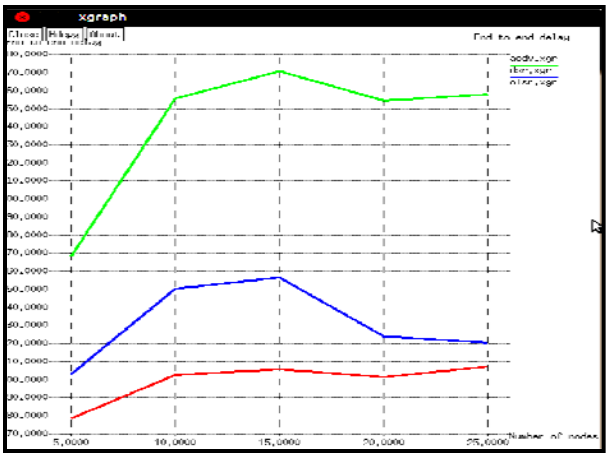

Fig 4. Number of nodes Vs Average End-end-delay

$$AED = \frac{\sum(\text{time received} - \text{time sent})}{\text{Total data packets received}}$$

Residual Energy

Mobile nodes in MANET work with battery power. Energy is consumed by the node as they move and transfer and receive data. Residual energy is the energy remaining at node after a particular data transfer. The following graph shows that the residual energy of OLSR is high, medium for DSR, low for OLSR.

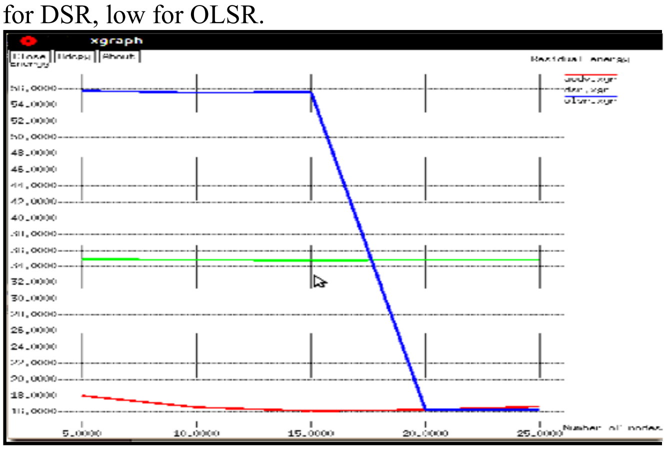

Table 2. Performance of AODV,OLSR,DSR for voice data transfer

Routing Protocol	**AODV**	**DSR**	**OLSR**
Throughput	Low	Medium	High
Packet Delivery Ratio	Low	High	Medium
Packet Dropping Probability	High	Low	Medium
End-end-delay	Low	High	Medium
Residual Energy	Low	Medium	High

V. CONCLUSION

From the analysis and graphical results, the following is concluded as final result. With respect to End-end-delay of VoIP, the performance of AODV is best. With respect to Packet dropping Probability, Packet Delivery Ratio and routing overhead of VoIP, the performance of DSR is best and as far as throughput is concerned, the performance of DSR is moderate and it is better for OLSR. Since for voice data transfer, QoS can be improved by reducing packet loss and End-end-delay, the protocol that satisfies the constraints of these two metrics are known to be best. From the resulting analysis it is clear that DSR has to be improved for VoIP flows to support reduced End-end-delay, Packet Dropping Probability of AODV has to be improved. The performance of OLSR is medium for both End-end-delay and Packet Dropping Probability, supports high residual energy. The next step for the future work would be to modify OLSR protocol in VoMAN environment to improve the performance of VoIP flows using EMODE-1 Codec.

REFERENCES

[1] JianqiangXin,jqxin@eng.iastate.edu http://www.sans.org/reading_room/whitepapers/honors/security-issues-countermeasure-voip1701,SANS Institute 2007.

[2]Erika P. Alvarez-Flores., Juan J. Ramos-Munoz., Jorge Navarro-Oriz., Pablo Ameigeiras., Juan M. Lopez-Soler., "User-level Quality Assessment of a "Delay-Aware Packet Dropping Scheme for VoIP". Network Protocols and Algorithms. Vol.3, No.3, December 2011. Available at: http://www.macrothink.org/journal/index.php/npa/article/view/1113/941

[3] Sangheon Pack, Gangwoo Park, Younghyun Kim, and Wonjun Lee, "Reducing Call Setup Latency in Mobile VoIP Systems". IEEE Communications Letters, vol. 15, no. 10, October 2011.

[4] S. El Brak, M.Bouhorma, M.El Brak, A.A.Boudhir, " VoIP applications over MANET : Codec Performance enhancement by tuning routing protocol parameters", JATIT, vol.50, No.1, April 2013.

[5]Sumit Mahajan, Vinay Chopra, "Performance Evaluation of MANET routing protocols with scalability using QoS metrics of VoIP Applications", (ijarcsse) , Vol.3,Issue 2, February 2013.

[6] Jae-Yul Yoon and Hochong Park, "Improving the Speech Quality of VoIP by Packet Prioritization", IEEE Signal Processing Letters, VOL.18, NO. 12, December 2011.

[7]Whai-En Chen, Pin-Jen Lin, Yi-Bing Lin ," Real-Time Voip Quality Measurement For Mobile Devices",IEEE Systems journal,vol. 5, no. 4, december 2011.

[8]Arlen Nascimento, Alexandre Passito, Edjair Mota Edson Nascimento, Leandro Carvalho, "Can I add a secure VoIP call?", (2006) IEEE, Proceedings of 2006 International symposium on a world of wireless, Mobile and Multimedia Networks (WoWMoM'06).

[9] Rajneesh Kumar Gujral, Manpreet Singh,"Performance Analysis of Adhoc routing protocols for Voice Communication support over Hybrid MANET,", (IJCA)International Journal of Computer ApplicationsVol.22 , No.3, May 2011.

[10]V.Savithri, Dr.A.Marimuthu , "A study on RED based AQM dropping packet selection schemes to improve the quality of VoIP flows mixed along with TCP and other UDP flows", (2012) MMASC, International Conference on Mathematical Modeling and Applied Soft Computing.

[11] V.Savithri, Dr.A.Marimuthu , "A Survey on QoS Constraints based Routing Protocols in Mobile Adhoc Network", (2010)GCOMPNETS, 2nd National Conference on Green communication, Computing And Networks.