SIP Over AODV Routing Protocol

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Abstract — **The session initiation protocol is a signaling protocol, is evolving as a standard for Voice over IP communications. A MANET is a collection of independent nodes. Mobile Ad-Hoc Networks have gained importance because they are easy to configure, deploy and are flexible. But SIP infrastructure needs centralized proxies and servers for registration. In this paper, Session Initiation Protocol standard is study, to support VoIP calls in MANETs and also it is configured a stable networks.**

Key Words — **Ad hoc network, VOIP call, SIP signaling, AODV protocol**

I. INTRODUCTION

MANET is one of the most common wireless network systems, which provides dynamic distributions for mobile devices with different types of mobility systems. Mostly common MANET Reactive and Proactive routing protocols; the Ad hoc On Demand Distance Vector (AODV), and the Optimized Link State Routing Protocol (OLSR). On the other hand, Voice over IP (VoIP) is considered one of the most common applications over different types of network systems [1] .In group communication the common problem is to enable a sender to securely transmit messages to a remote cooperative group. A solution to this problem must meet several constraints. First, the sender is remote and can be dynamic. Second, the transmission may cross various networks including open insecure networks before reaching the intended recipients. Third, the communication from the group members to the sender may be limited [2,7]. The currently used VoIP systems shows how they are characterized by a set of fixed nodes (stateless/stateful proxies and registrar servers), which act as intermediaries between their endpoints or provide registration and localization of nodes. This kind of approach has some disadvantages that make it unsuitable when system dimensions grow: low fault-tolerance (if a proxy is damaged, the whole system will not work), low scalability in the number of supported parallel Calls. In this paper, Session Initiation Protocol standard is proposed for supporting a VOIP calls and it is configured a stable network. Optimized link state routing (OLSR), where each node behaves actively and passively for routing operations. There is some functionality managing the construction of the lowest-cost path from source to destination, dynamic codec selection strategy, call admission control (CAC) procedure [3,6]. a new geographic multicast protocol scheme that is still inspired by the EST and addresses the problems outlined above in terms of overhead reduction, scalability, and energy efficiency[4]. it is unlikely that the base stations cover the entire city to maintain sufficiently strong signal everywhere to support an application requiring high link rates, the vehicles themselves can form a MANET to extend the coverage of the base stations, providing continuous network connections[5,8].

II. BACKGROUND

A basic SIP session involves the calling user agent contacting the calling side proxy server, this in turn will forward the message to the proxy server responsible for the domain of the called user agent. The called side proxy server retrieves the bindings for the called user from the called side registrar (i.e., utilizes the location service) and then delivers the request to the intended recipient. On the wired network SIP used two main component user agent and sip server. The Session Initiation Protocol has emerged in IETF as the session management protocol, and it was also chosen as the signaling protocol in the 3GPP framework. SIP relies on centralized entities, namely servers, maintained by organizations or service providers. SIP servers handle users' registration, their location, and forward SIP messages to the location, where the recipient is reachable.

III. RELATED WORK

Mazin Alshamrani et al. [1] proposed SIP based VoIP calls using GSM voice codec system over MANETs with Static, Uniform, and Random mobility models. This was considered AODV as a reactive routing protocol and OLSR as a proactive routing protocol over both IPv4 as well as IPv6. The SIP signaling examined call setup time, number of active calls, number of rejected calls and calls duration. It gave the outcome that; IPv4 has better performance over different types of mobility models, while IPv6 upholds longer delays and poor performance over Random mobility models. Iwao Sasase et al.[2] propose a gateway selection scheme that considers multiple QoS path parameters such as path availability period, available capacity and latency, to select a potential gateway node. It improve the path availability computation accuracy, we introduce a feedback system to updated path dynamics to the traffic source node and also propose an efficient method to propagate QoS parameters. This scheme, show that our gateway selection scheme improves throughput and packet delivery ratio with less per node energy consumption. It also improves the end-to-end delay compared to single QoS path parameter gateway selection schemes. Floriano De Rango et al.[3] proposes a new protocol, capable of ensuring a quality of service (QoS) level for VoIP calls over a MANET and to manage a large number of calls in the system. Novel metric and utility functions are proposed to perform the best path selection from source to destination nodes, respecting the QoS parameters for VoIP quality. In particular, an objective metric such as R-factor is considered, and a flexibility index is defined in order to maximize the number of acceptable VoIP calls. Performance shows that the proposed approach led to better network management in terms of admitted calls and respected QoS constraints. Laura Galluccio et al. [4] propose a new geographic multicast protocol denoted as GEM, which is inspired by the Euclidean Steiner Tree (EST) theory and does not require any information exchange for routing purposes. Therefore, it is very efficient and scalable in wireless networking these schemes achieve low performance, in terms of energy consumption. They also derive some key properties of GEM which is apply to wide range of algorithm, that allow us to characterize the protocol performance. Qianhong Wu et al. [5] proposed novel key management paradigm. In this system, each member maintains a single public/secret key pair. Upon seeing the public keys of the members, a remote sender can securely broadcast to any intended subgroup chosen in an ad hoc way. Following this model, it instantiate a scheme that is proven secure in the standard model. Even if all the non intended members collude, they cannot extract any useful information from the transmitted messages. After the public group encryption key is extracted, both the computation overhead and the communication cost are independent of the group size. This scheme efficient member deletion/ addition and flexible rekeying strategies. Its strong security against collusion, its constant overhead.

IV. Existing Methodology

A) SIP based VoIP; this method is depending on three main stages: registration, call initiation, and call termination. These stages depend on the SIP Proxy Server to relay the connectivity between different callers. The delay of SIP signaling in all stages affects the performance of VoIP calls. As SIP is a TCP based application layer signaling system, all the TCP timers, retransmission and Round Trip Timer (RTT), are important factors for the overall structure of SIP connectivity system. This was used in all mobility models of this study. In the Static model, MANET's nodes are stable and do not move. In the Uniform model, all nodes move in the same direction with different speed ranges including the SIP server. In Random model, the nodes move in different directions, but the SIP server is stable in the center of the simulation area. In Random mobility model, every node in the simulation, except the SIP Server, has its own mobility direction and speed depending on the identified random functionality of the node parameters. In Random All model, all nodes move in different directions, including the SIP server. The reason for examining the Random mobility using two different models is to study and evaluate the effect of SIP server mobility for VoIP applications and the signaling QoS. The assumptions also considered the GSM voice codec for VoIP applications because of its efficiency and good performance compared with other voice codec's.

B) E-model based metric and lexicographic ordering path selection. This method is based on an extension of the SIP protocol, due to its robustness and broad diffusion. A distributed scenario has been considered each node can communicate with its neighbors (a neighbor has a direct radio coverage in the considered environment) and, if a remote communication is requested, a multi-hop path is mandatory, so a key issue is the discovery of neighbor nodes. This problem can be simply avoided by using Hello packets when a node receives a Hello packet, containing the identity of neighbor nodes, it can easily identify the nodes under its coverage. To manage a new call properly, the link between source and destination must be bidirectional, forwarding protocols belonging to the OLSR family put tags on the links from a node to its neighbors, in order to know if it is unidirectional or bidirectional. Each node inserts the list of its neighbors in the Hello packet that it is going to send. When a node finds its identity in the received HELLO packet, it can tag the link with its neighbor as bidirectional.

V. Analysis And Discussion

The node capacity, position and SIP server mobility all affect the VoIP performance over MANET reactive routing protocols. The TCP signaling system and the header size have a direct effect on the performance of IPv6 MANET. In addition, SIP timers and retransmission timers in particular, have an impact on SIP based applications. All SIP performance factors could affect SIP based applications over IPv6 MANET. One of these factors is the retransmission timers of the SIP signaling that controls calls initiation, termination and modification stages of VoIP applications. In addition, using some header compression systems like Robust Header Compression could enhance the performance of the SIP based VoIP applications for IPv6 MANET.

When the proposed novel metric is used (based on theE-Model) the admission percentage is always 100 percent: No chosen path blocks any sector of the system, and the fixed nodes are used only when necessary. Also the R-factor is considered for the traditional metric a value of 85.78 is obtained, while for the proposed LexModel metric a value of87.12 are observed with the classical approach, for a couple of calls between workers of the same team, on average, the fixed node are used, while for the other calls longer paths are used. Using the proposed metric, team calls are forwarded on longer paths, while for calls directed to the head quarters fixed nodes are used. Proposed metric reduces the probability of a block in the network to the minimum and increases the number of admitted calls in the system.

VI. Proposed Methodology

It is introduced two new SIP headers PREQ that is 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 1, if a node S says the sender wants to communicate with the receiver node R, node S broadcasts PREQ (Path Request) packets to its neighbors. The neighboring nodes further broadcast to their neighbors and so on till they reach the receiver node R.

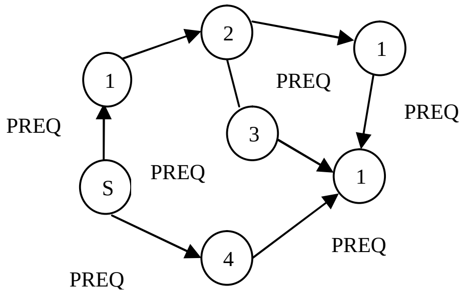

Fig1: Discovery phase 1: the path requesting phase(S-sender, R-Receiver)

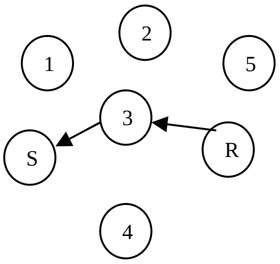

Fig2**:** Discovery phase 2: the path response phase

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 2.

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 3. The receiver then sends a SIP OK message back to the sender as in fig 4. 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 protocols that have chosen are 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. When a node needs a find a route to a destination node it broadcasts a Route Request (RREQ) message. The RREQ message is spread throughout the network and as soon as the Message the destination node itself, a Route Reply (RREP) message is unicast back to the requesting node. AODV protocol provides low overhead, adapting quickly to dynamic link conditions and memory overhead. It allows the mobile nodes to obtain routes quickly to new destinations. Moreover, AODV enables mobile nodes to respond to link breakages and changes in the network topology in a timely manner. Route tables are used in AODV to store applicable routing information. Invalid routes are quickly detected through the use of route errors (RERR) messages.

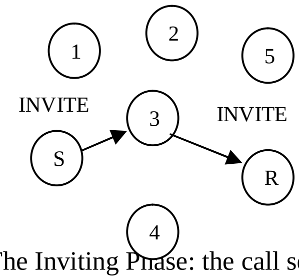

Fig3: The Inviting Phase: the call set-up phase

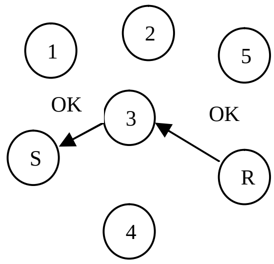

Fig 4: The acknowledgement for the invitation phase

VII. Possible Outcome

In this paper the result might be show that, form the stable network. It also detected the invalid route with the help of route error, and the cost of session was reduced. Packet delivery ratio was increased

Conclusion

In this paper author are proposing a novel way of adapting session initiation protocol i.e. SIP for VoIP in MANETs. The proposed model extends the SIP protocol which is present for wired stable network, because it is robust and an established standard for VoIP communications.

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