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Adapting SIP for Enabling Voice Calls in MANET

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Abstract— Voice calls using internet is growing very fast, due to this reason the phone calls over the internet are becoming cheaper. There are very few protocols to support voice over IP calls at present. The SIP, the session initiation protocol is a signalling protocol, is evolving as a standard for Voice over IP communications. A MANET is a network of independent nodes which is decentralized. Mobile Ad-Hoc Networks are easy to configure, deploy and are flexible. But SIP infrastructure needs centralized proxies and servers for registration. In this paper, we first study about the Session Initiation Protocol standard, how it is configured in stable networks, how SIP can be adapted to MANETs. We propose a new decentralized architecture for deploying SIP to support VoIP calls in MANETs.

Keywords— VoIP, SIP, MANETs

I. INTRODUCTION

A Mobile adhoc Network (MANET) is a network of independent mobile nodes that communicate with each other using wireless links without support from any pre-existing infrastructure network[1]. For providing deployable and scalable services in MANETs, the Session Initiation Protocol (SIP) has been considered as key factor. SIP is a signaling and instant messaging protocol. It was developed to set up and modify multimedia sessions. SIP is used to request and deliver voice and instant messages over the Internet. The SIP architecture uses centralized proxies and registrars, which is owned by the network operator. Registrars are SIP entities where SIP users register their required contact information once they connect to the SIP network. In a registration scenario, a SIP user agent communicates to its registrar server. The registrar IP address is usually pre-configured at the SIP user agent. SIP address of records (AOR) for a user is the SIP user name of the user(s) using the device and the addresses where the user is reachable. Contact information is stored in the form of IP addresses or resolvable names, but can also contain other kinds of contact information like telephone numbers. The registrar servers are responsible for managing user location information while the proxy servers enable routing of SIP messages[2].

In a MANET, centralized servers are not available. New mobile clients are becoming smaller and cheaper. It is attracting increased attempts to deploy Internet-based applications like voice-over-IP (VoIP) [7]

over wireless Mobile Ad-hoc Networks (MANETs). For example, the voice-over-IP (VoIP) services are very popular. Many people would like to use VoIP over a MANET for communications. MANET can help for example, it can act as an emergency response at a site where a disaster has occurred and the network infrastructure is broken down. The conventional Internet-based applications are dependent on the stable internet infrastructure and server-based like routers and DNS servers components to run. MANET is dynamic in nature and hence internet-based applications cannot be run in the MANETs straightforwardly.

In this paper we present the general SIP architecture in stable wired networks, the SIP architecture that we have proposed for MANET.

II. SIP IN WIRED STABLE NETWORK

A basic SIP session involves the calling user agent contacting the calling side proxy server, which 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[2].SIP Architecture: SIP has two main components, user agents and SIP servers. User Agent: An user agent is a logical entity which sends SIP requests and receives answers to those requests. There are 2 different SIP Servers Proxy Server: The user agent sends requests to the proxy server. Proxy server is similar to HTTP proxy. It resends the request to appropriate node. The proxy server is like an intermediate routing node between the sender and the receiver. Registrar Server: Each SIP user agent must register with the registrar before it communicates with other user agents. It has a location and address translation service[2][4].

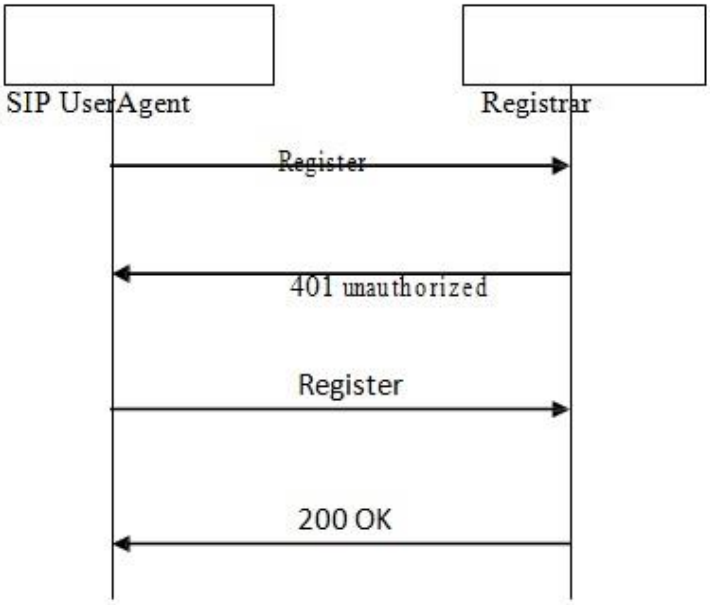

Fig. 1. Session Initiation protocol registration process in stable wired networks[5].

SIP Methods[5] are: INVITE - Session setup
ACK - Acknowledgment of final response to INVITE BYE - Session termination

CANCEL - Pending session cancellation REGISTER - Registration of a user's URI

OPTIONS - Query of options and capabilities INFO Mid - call signaling transport

UPDATE - Update session information

SIP Response Code Types[4] are:

1xx Provisional or Informational — Request is progressing but is not yet complete

2xx Success — Request completed successfully

3xx Redirection — Try the request at another location

4xx Client Error — Request was not completed because of an error in the request, retry after correction

5xx Server Error — Request was not completed because of an error in the recipient, can be retried at another location

6xx Global Failure — Request has failed do not retry again[6]

III. RELATED WORK

A. VoIP applications over MANET CODEC performance enhancement by tuning routing protocol parameters[8].

In this paper, the authors are proposing VoMAN (voice over MANET) voice streaming between nodes relying on multi-hop by means of simulation. Various other parameters are used for simulating like CODEC, routing protocol fine tuning. No emphasis is laid on choosing the application layer protocols like SIP or H.323 signaling protocols.

B. Proposition of a new approach to adapt SIP protocol to Ad hoc Networks[9].

In this paper, the main idea proposed is named VNSIP (Virtual Network for Session Initiation Protocol), that self- organize the ad hoc network using a virtual backbone. This virtual network will be used as architecture to ensure roles like proxy, registrar or redirect server of SIP entities. The disadvantage is that the virtual backbone network construction provides additional overhead. VoIP traffic is real time and any delay in transmission is immediately noticeable. Further the centralized entities like proxy server and registrar does not provide feasibility in a decentralized ad-hoc network

C. Enabling SIP-Based Services in Ad Hoc Networks[10]

In this paper, a middleware that integrates with AODV reactive MANET routing protocol in order to minimize the signaling overhead as well as enable SIP applications in MANETs is proposed. The middleware manages the SIP server functions. The middleware calls the underlying network routing module while discovering targets and delivering SIP messages. The middleware takes the SIP server's place.

D. SIPHoc: Efficient SIP Middleware for Ad Hoc Networks[11]

In this paper SIPHoc a middleware infrastructure for session set up and management in MANETs. SIPHoc provides the same interface as the SIP standard but in order to suit MANETs its implementation is fully decentralized. As a single node in the MANET has Internet access SIP session establishment is possible. The paper presents the architecture of SIPHoc and evaluates its performance.

IV. THE PROPOSED SIP ARCHITECTURE IN MANET

The SIP architecture is not applicable to MANETs. In MANETs the nodes join or leave the network dynamically. SIP requires fixed proxies and registrars[13][14]. The proxies and registrars are centralized entities. Hence the SIP protocol cannot be deployed as it is in isolated ad-hoc networks. SIP user agents in MANETs cannot reach other nodes, as they do not support centralized proxy servers. Since MANET is a decentralized network of nodes, the SIP functionality includes the following:[16]

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

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

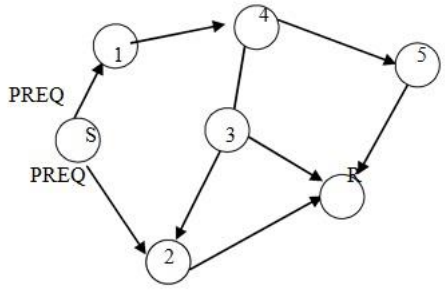

Fig. 2. Discovery phase 1: the path requesting phase(S-sender, R-Receiver)

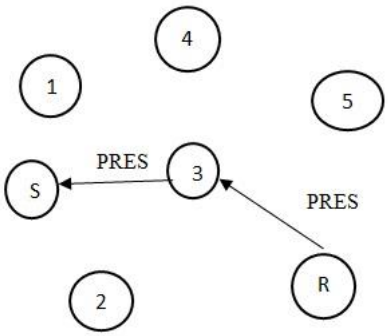

Fig. 3. Discovery phase 2: the path response phase(S-sender, R-Receiver)

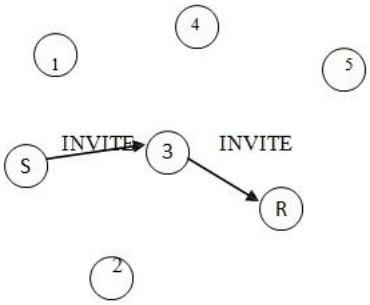

Fig. 4. The invitation phase

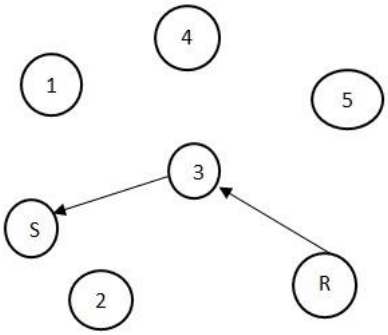

Fig. 5. The acknowledgement for the invitation phase

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 neighbours 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[17].

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[18]. The neighboring nodes further broadcast to their neighbors and so on till they reach the receiver node R.

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 [20].

V. FUTURE WORK AND CONCLUSION

In this paper we 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. We propose an architecture for SIP in decentralized MANETs.

The future work includes simulation of the proposed system and recording the various quality of service parameters like throughput, delay and jitter.

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