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# A New SIP-Based Application Layer Protocol for VoIP in MANET

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***Abstract*—Internet telephony 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. This paper compares the session initiation protocol architecture in wired networks and in MANETs i.e mobile ad-hoc network. A MANET is a collection of independent nodes which is decentralized. 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, 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 collection 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].

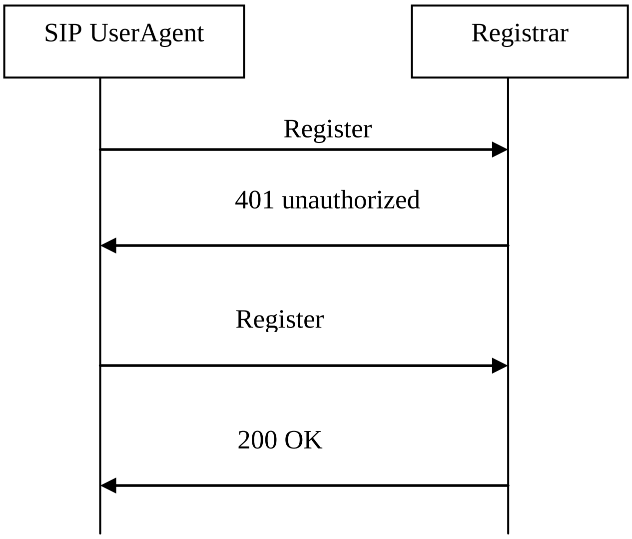


Fig. 1. Session Initiation protocol registration process in stable wired networks[5].

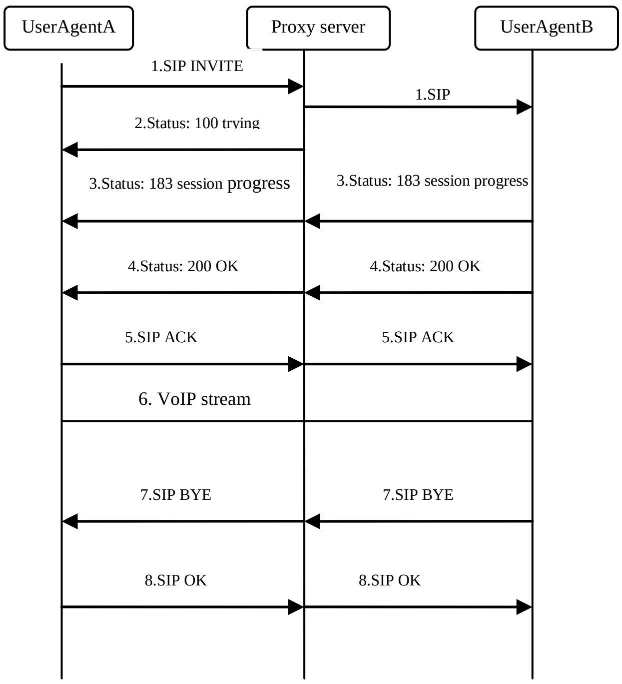


Fig. 2. Session Initiation protocol session set-up steps[6]

1. In Fig. 2 User Agent A invites User Agent B
2. Proxy locates User Agent B while responds to User Agent A that it is trying to build the session.
3. User Agent B responds to Proxy that the session is in progress - Proxy responds to User Agent A that the session is in progress.
4. User Agent B responds to Proxy that it establishes the session - Proxy responds to User Agent A that User Agent B has established the session.
5. User Agent A acknowledges the session with ACK message - Proxy responds to User Agent B that User Agent A has acknowledged the session.
6. Packet stream: it is the phone call/conversation itself.
7. User Agent B hang up and responds message BYE to proxy - Proxy responds message BYE to User Agent A.
8. User Agent A confirms with OK to the Proxy - Proxy responds to User Agent B that User Agent A has confirmed with OK.

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

SUBSCRIBE - Request notification of an event

NOTIFY - Transport of subscribed event notification

MESSAGE - Transport of an instant message body

PUBLISH - Upload presence state to a server

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 therequest, 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

VoIP calls in mobile ad-hoc network has been an active research area and many mechanisms to enable VoIP in MANET has been proposed.

### 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. The standard protocols for VoIP are robust and scalable a mandatory requirement for ad-hoc networks.

### 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 rediect 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.

### E. *VoIP over MANET (VoMAN): QoS & Performance Analysis of Routing Protocols for Different Audio Codecs[12]*

This paper analyses the performances of routing protocols (AODV, OLSR) in mobile ad-hoc network carrying VoIP data. VoIP is a biggest challenge due to Quality of Service (QoS) requirements. By simulation study the authors analyze and evaluate some QoS parameters like bandwidth, end-to-end delay and packet loss. Network Simulator (ns2) is used to study voice codecs to determine their effect on metrics QoS. VoIP channel QOS characteristic such as delay, bandwidth, packet loss are measured for varying hops and routing protocols. Different codecs used are G.711, G.723.1, G.729 and GSM.AMR.

## 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. The user agents cannot be reached by other nodes as they are not registered with the Registrar where the contact information is stored[15].

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.

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[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.

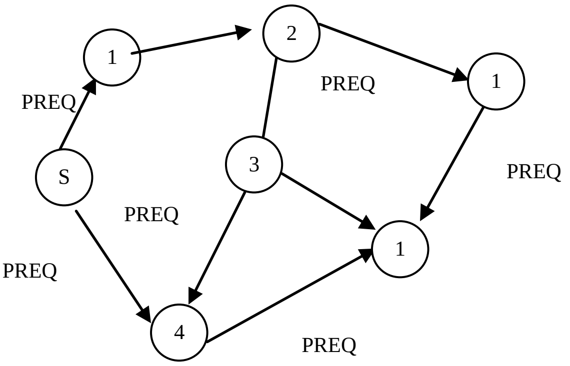


Fig. 3. Discovery phase 1: the path requesting phase(S-sender, R-Receiver)

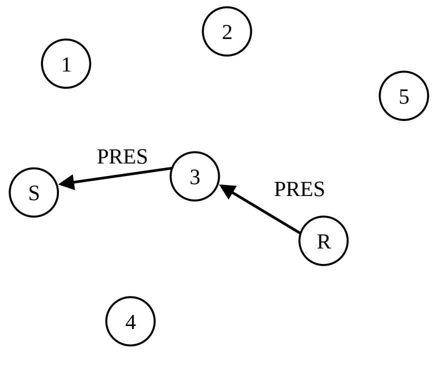


Fig. 4. Discovery phase 2: the path response phase

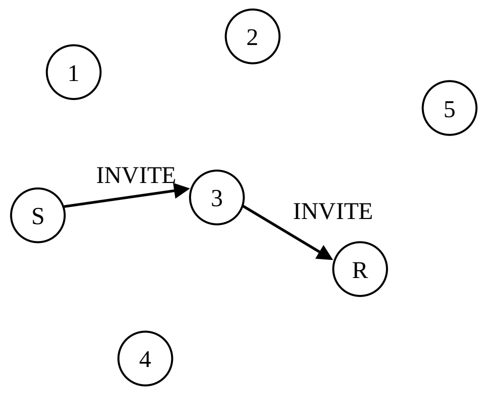


Fig. 5. The Inviting Phase: the call set-up phase

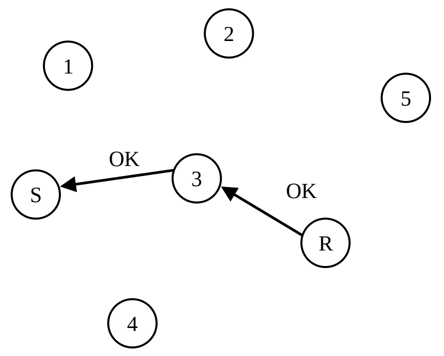


Fig. 6. The acknowledgement for the invitation phase

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 [19].

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 [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 in MANET, evaluating the various parameters for quality of service like delay, throughput, jitter and loss for VoIP packets in MANETs. The proposed model is SIP-based application layer protocol for setting up, managing and terminating sessions for VoIP calls in MANETs.

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