Optimizing SIP Service Provisioning in Internet Connected MANETs

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***Abstract*— Recently, Mobile Ad-hoc Networks (MANETs) have gained a lot of attraction because they are flexible, self-configurable and fast to deploy. Systems beyond 4G are likely to consist of a combination of heterogeneous wireless technologies and naturally might comprise MANETs as one component. In order to provide multimedia services such as Voice over IP in such environment, support for Session Initiation Protocol (SIP) is essential. A MANET is a decentralized collection of autonomous nodes but a SIP infrastructure requires centralized proxies and registrar servers. In this paper, we first study the implications of using standard SIP architecture in internet connected MANETs. We analyze limitations of SIP service scalability when centralized proxies/registrars located in the Access Network are used by MANET nodes. Finally, we present alternative approaches to provide SIP services in such environment to avoid current limitations.**

I. Introduction

Mobile Ad-hoc networks (MANETs) will be an integral part of 4G networks and several projects are currently working towards prototypes (such as the EU IST-DAIDALOS project, http://www.ist-daidalos.org). A MANET is a collection of autonomous mobile nodes (MN) that communicate using wireless links without support from any pre-existing infrastructure network. For integration into 4G networks, internet connectivity is required for MANETs, which can extend the range of hotspots by providing multihop connectivity from MNs towards the internet through one or more gateway nodes utilizing packet forwarding capabilities of intermediate nodes via multihop paths. MANETs will be a key enabler for future Ubiquitous and Pervasive Communication and Computation (UbiComp) scenarios [1] and internet connectivity for MANETs makes them even more attractive. However, for providing deployable and scalable services, the Session Initiation Protocol (SIP) has been considered as key element [2]. SIP is a signalling, presence and instant messaging protocol and was developed to set up, modify, and tear down multimedia sessions, and to request and deliver presence and instant messages over the Internet. The SIP architecture is based on centralized proxies and registrars, typically owned by the network operator. As the MANET is an autonomous network, several problems arise when providing SIP services.

Our contribution in this paper is to analyze limitations imposed by the standard approach where SIP proxy/registrar is located in the Access Network. All SIP messages involving MANET nodes pass through this proxy/registrar, even if both endpoints are located within the MANET. In order to increase service scalability we propose several alternatives to provide SIP in internet connected MANETs. We conclude that the standard approach is unsatisfactory from the service point of view which confirms the need to study alternative approaches in more detail. The paper is organized as follows. In Section II, we review SIP architecture and describe the problems of using SIP services in internet connected MANETs. Section III details alternative approaches and analyzes impacts of selected approaches on SIP architecture. Concluding remarks are offered in the last section.

II. The Session Initiation Protocol (SIP)

A. Review of SIP

SIP [3] has been standardized within the IETF in the framework of multimedia multicast conferences and VoIP services. SIP is a request/response protocol where user agent client (UAC) is a SIP entity that generates requests and user agent server (UAS) receives those requests and returns responses. A SIP request, together with the responses it triggers, is referred to as a SIP transaction. SIP defines a set of messages (request and response messages) such as INVITE, ACK, BYE, OPTIONS, CANCEL, and REGISTER to setup sessions between user agents. These messages are routed trough SIP proxies that are deployed throughout the network. DNS service records help in locating SIP proxies responsible for the destination domain.

Signaling messages are exchanged between UAs and proxy servers to locate the appropriate services or endpoints for media exchange. For reasons of scalability, multiple proxies are used to distribute the signaling load [4]. A session is setup between two UAs through SIP signaling messages starting with INVITE (messages 1-2 in Figure 1), an 200 OK response (messages 3-4) and an ACK (message 5-6) to the response. The call setup is followed by direct media exchange using RTP without any proxy involvement. The session is closed through an exchange of BYE (messages 7-8) and 200 OK (messages 9-10).

B. Problems with SIP in MANETs

The SIP architecture is based on centralized entities, where registrars and proxy servers are the most important entities.

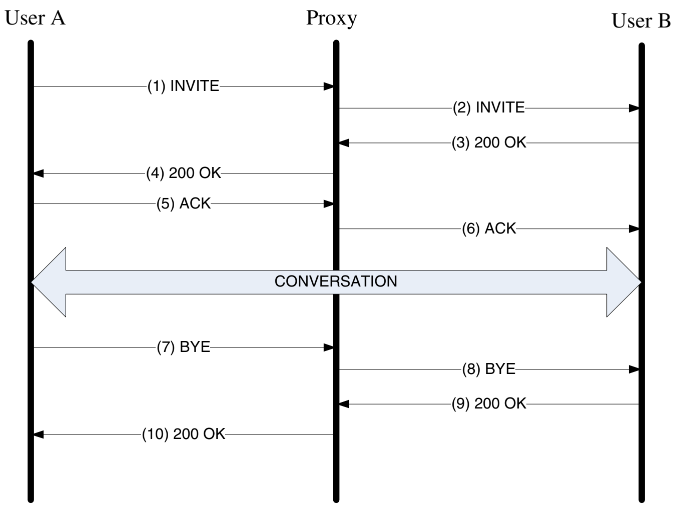

Fig. 1. Session establishment between two users

Usually, they are owned by network operator or service provider and are located in the access network. Registrars are necessary to register contact information once SIP users connect to the SIP enabled network. Usually, registrars IP address is pre-configured and well known to terminal software. In a standard registration scenario, a SIP UAC within the MANET node communicates to its registrar server the SIP user name of the user(s) using the device, referred to as SIP address of records (AOR) for that user, and the addresses where the user is reachable. For this work, we assume that MANET nodes autoconfigure a globally routable IPv6 address [5]. Proxy servers, that can be co-located in the same element as registrar server, are needed because SIP users do usually not know the current complete contact information of the callee but only its AOR. SIP proxy servers thus act on behalf of user agents forwarding or responding to the request received from other user agents or proxies.

The standard registration scenario, where a SIP UA uses services provided by registrars and proxies to locate the SIP proxy and to map SIP user name to a destination IP address involves centralized servers [3] that may not exist in an isolated MANET. Therefore, SIP cannot be deployed as is in isolated MANETs. In internet connected MANETs however, end points located in the ad hoc network can reach other parties located in the internet (and thus also SIP proxies and registrars) through gateway nodes, but when two nodes in the MANET need to communicate via SIP, any SIP signaling will traverse the gateway, which is a severe performance limitation. Therefore, alternative approaches are desirable.

III. SIP based Service Provisioning for Internet Connected MANETs

In order to enable and optimize SIP in internet connected MANETs, several approaches are considered in this paper.

A. *SIP Proxy/Registrar in Access Network*

Using standard SIP architecture, all SIP signaling exchanged between SIP MANET nodes (or a MANET node and an external node in the internet) need to pass through gateways that connect the MANET to the internet, even if communication parties are just one hop away within the same MANET. If two MANET nodes need to establish a SIP session, a SIP signaling message has to traverse the gateway even twice when the SIP proxy is located in the Access Network. Therefore, an optimization to enhance SIP service availability to internet connected MANET nodes is desirable. In order to overcome such limitation, we propose to add SIP proxy/registrar functionalities into MANET gateway nodes.

The proposed approach could be seen as an extension applied to internet connected MANETs, allowing MANET gateway nodes to act as SIP proxy/registrar server. It also changes the way MANET nodes find these SIP servers without modifications on standard SIP architecture. Instead of using pre-configured SIP outbound proxy server IP address (IP address of gateway acting as SIP proxy) in every MANET node, we propose the support of auto-configured SIP applications through the use of MANET gateway discovery mechanisms. A MANET gateway discovery mechanism is necessary in order to inform MANET nodes about internet connectivity capability which can be coupled with IP address auto-configuration of MANET nodes.

We adopt the strategy to add SIP proxy/registrar location information to this mechanism instead of using another auto-configuration protocol such as DHCPv6 [6]. The selected gateway discovery mechanism has a strong impact on the overall performance due to the number of messages exchanged versus latency [7]. An integration of the proactive approach based on prefix continuity [5], where the MANET is virtually divided into as many subnets as there are gateways would easily allow to deploy proxy/registrar functionality co-located with each gateway thus improving the scalability.

An extension to the gateway discovery mechanisms is required in order to convey the information that the MANET gateway which originated the gateway advertisement message can operate as a SIP proxy/registrar. We propose to reuse Jelgers gateway discovery mechanism [5], which is based on the GW_INFO message (Figure 2), and add a ”P”-bit in the reserved field which indicates gateway capability to act as a SIP proxy/registrar. A MANET node that receives such GW_INFO message with ”P” bit field set to 1 knows that the gateway who originated this message provides SIP proxy functionalities. As shown in Figure 2, the GW_INFO message format extension proposed for this approach does not impose more overhead to the network and also does not modify the protocol behaviour. According to [5], several algorithms exist for MANET nodes to select a proper gateway if the node receives different GW_INFO messages with different prefixes indicating several gateways that connect the MANET to the internet. If not all gateways implement SIP proxy/registrar functionalities, each node has now additional freedom to select

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  0                   1                   2                   3
  0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 | Option Length |P|  Reserved   |  Neighborhood connectivity    |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 | Prefix Length |    Distance   |        Sequence number        |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                                                               |
 |                    Gateway Global Address                     |
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Fig. 2. GW_INFO message extended with SIP proxyregistrar functionality co-located in GW (AR)

a gateway based on its SIP proxy/registrar capabilities.

The registration and session initiation processes follow the same behaviour as the standard SIP mechanisms. Figure 3 presents an illustration of the proposed approach, where gateways (AR1) operate as a SIP proxy/registrar server in an internet connected MANET. As all MANET nodes have learned SIP proxy/registrar IP address through gateway discovery mechanism, they can start the registration and session initiation process. As illustrated in Figure 3, all SIP control traffic (1. + 3. REGISTER, 2. + 4. INVITE, etc.) is exchanged between SIP MANET nodes (yellow circles) and gateways, and media traffic (green arrow) is exchanged directly between SIP MANET nodes (or between a MANET node and a node in the internet). Another advantage of the proposed proxy/registrar functionality co-located with the access router or gateway is the potential for easy integration into local and global mobility management mechanisms. Usually, MANET nodes register with Mobile IP foreign agents, which can be co-located with MANET internet gateways. Therefore, an integration of SIP proxy and mobility management at the gateway has the potential of significantly reducing signaling traffic in the MANET.

B. Distributed SIP and Integration with Routing

Registering SIP URIs and finding the location of callee is similar to MANET routing. Therefore, it seems natural to integrate the functions of SIP with MANET routing protocols or to use MANET multicast/broadcast routing protocols to distribute SIP registration information to all MANET nodes. Two solutions fall into this category: distributed SIP (dSIP) [8] and integration with cluster based routing where cluster heads take the responsibility of acting as SIP proxy/registrar servers [9]. As the role of cluster head might change over time due to mobility, this solution also requires that MANET nodes have limited server functionality.

In dSIP [8], all MANET nodes have proxy/registrar functionality. Fully distributed registration is achieved by broadcasting (or multicasting) a SIP REGISTER message in the MANET through ad hoc routing protocols. All nodes that receive a broadcasted REGISTER, process it using their local server modules. The binding of the registering user is cached by all nodes that receive the broadcasted message and a SIP 200 OK message containing the binding of the replying user is returned to the sending node. When a user wants to invite a peer to a distributed SIP session, an INVITE message is built by the caller user agent and forwarded to the local proxy module within that node, which maintains a cache for SIP URI bindings learned through broadcasted register. The INVITE is thus sent by the local proxy module, the logic of SIP has not changed, but the servers are decentralized and embedded in every MANET node. This requires to install middleweare on every MANET node to intercept SIP signaling.

To work in internet connected MANETs, [11] proposes a ”SIP gateway” for dSIP which hides the registration of ad hoc users from SIP servers outside the MANET. This solution mainly deals with mobile nodes using private addresses which are not globally reachable by internet nodes. Differentiation between callee inside or outside the MANET can be achieved through the extension ”.local” at the SIP address level. This solution can be used as a way to enable SIP sessions in internet connected MANETs, but it seems to be unpractical in a real scenario where a SIP user, reachable by its SIP address, could be located either inside or outside MANETs. Instead, we propose to provide interworking with nodes in the internet by enabling MANET gateways with proxy functionalities. MANET gateways will also receive the broadcasted SIP register messages and thus can act as supporting SIP proxies on behalf of MANET nodes. If a MANET node thus wants to invite a node located in the Internet, it looks up the cache but does not find a proper binding. It thus concludes that the callee is not located in the MANET and forwards the INVITE to the gateway, which in turn uses standard SIP proxy mechanism to locate the callee proxy.

Using cluster based routing protocol on the other hand reduces the number of transmitted messages in the MANET as SIP messages are integrated with cluster based routing protocol messages leading to improved bandwidth usage, decreased collision probability and improved scalability. However, cluster heads are single point of failure and the usage of specialized routing protocol limits the usability of the approach. Therefore, we do not consider it further.

C. Integration of SIP with Service Discovery Solutions

A service discovery framework can be used to discover SIP users either by finding out the bindings of users within reach

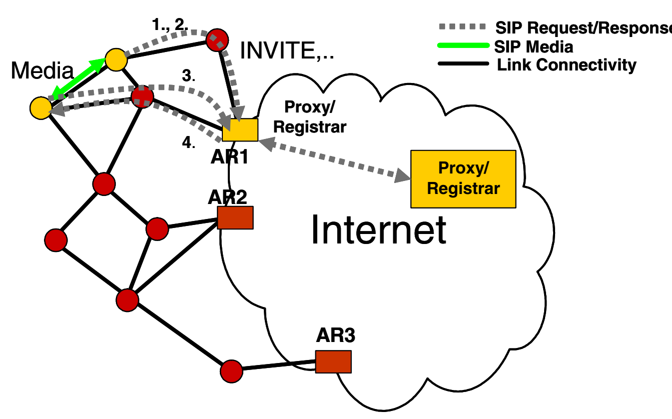

Fig. 3. SIP proxyregistrar server co-located with Gateway AR1

in the MANET or to discover the IP address of a user based on SIP AOR. Therefore, an integration of service discovery with SIP services seems to be beneficial. As an example, the Service Location Protocol (SLP) [10] was also used by [8] where the SIP location service is exploited by broadcasting SLP service request messages.

There are basically two different modes. In the server based approach, one of the devices in the MANET may have proxy/registrar functionality and can offer this service to the other users in the MANET looking for the service "SIP-registration". MANET nodes thus register with that node and use it for standard SIP processing. Thus, all SIP signaling passes through that MANET node. In the server-less mode, devices query for the service "SIP" and parameters contain the AOR of the user to contact as attribute filter. All devices in the MANET receive this request and the one that matches the attribute AOR returns the IP address of the service SIP on that host. When the server module within the MANET node receives the response it stores the IP address of the service in the cache. This step substitutes the registration procedures used in standard SIP, where bindings are received and maintained through periodic SIP REGISTER messages by SIP Registrars. In order to provide internet connectivity in this scenario, we propose now a solution where MANET nodes use the service discovery framework to find a SIP user. If the callee is located outside the MANET, the caller also issues a SLP query but it will not get a reply in the server-less mode. The caller then assumes that the callee is located in the Internet and the caller sends a SIP INVITE to the gateway, which is then processed similarly to distributed SIP (see section III-B).

In order to be reachable by external nodes in the internet, MANET nodes need to register its SIP clients at the gateway (which acts as SIP proxy/registrar) querying for the service "SIP-gateway-registration". In the server based approach, the callee is not registered with the MANET node that acts as SIP proxy so this MANET proxy then has to forward the INVITE to the MANET gateway.

Mutual interoperability is an important issue in this approach as all devices in the MANET must run the same service discovery framework in order to participate in SIP sessions. Also, the performance of SIP call setup then strongly depends on the performance of service discovery, which has some problems in MANET due to usage of broadcast messages [9].

D. Peer to Peer SIP

The term "Peer to Peer" (P2P) refers to a class of systems and applications that employ distributed resources to perform a function in a decentralized manner. In P2P SIP, a SIP system uses P2P mechanisms based on distributed hash tables (DHT) for management of e.g. user location [12] [13] [14].

In P2P SIP, the registration process is modified by changing where registration messages are sent to. The user agent constructs a SIP REGISTER message containing the contact information. The end point (in this case the user agent) hashes the username (e.g. callee@kau.se), and sends the SIP message embedded in a P2P message using the P2P overlay. Upon arrival at nodes registered in the P2P overlay network, the message is extracted and a reply is sent. Each node now serves as registrar and knows where parts of the users can be contacted. New nodes joining the system contact their neighbors and replicate the registrations and expiration times. When a caller wants to locate a callee, the caller node uses the same hash function to locate the callee in the overlay.

Interworking with nodes in the internet can then be achieved by constructing a hierarchy of P2P SIP networks, where MANET nodes are connected to local P2P SIP networks, which in turn are connected to the global SIP network through MANET gateways. MANET gateways thus have to act as P2P SIP Proxies [14] and have to be able to route SIP messages towards the Internet. Hence, a MANET gateway need to be registered with the P2P overlay network and is bound to a Fully Qualified Domain Name (FQDN).

P2P systems have the advantage of scaling more easily as the number of nodes increases, since each new node offers additional server-like functionality when it joins. However, the performance of P2P SIP in hybrid MANETs depends on the performance of the P2P overlay network and thus on DHT processing in MANETs which has several limitations [15]. Also, unlike O(1) lookup cost in classical client-server based systems, the P2P lookup cost can be much higher [12] leading to potentially increased call setup latency.

E. Impact of proposed approaches on SIP architecture

Table I gives an overview on new interfaces in MN and GW nodes for each proposed approach. Implementing SIP proxy/registrar into MANET gateway nodes, as discussed in section III-A, leads to a solution which does not modify standard SIP architecture. However, this approach proposes an extension of gateway discovery mechanism, which consequently modifies MNs and GWs architecture through the creation of new interfaces in order to manage and maintain SIP proxy/registrar information.

Distributed SIP and Integration with routing protocols (section III-B), Integration of SIP with Service Discovery Frameworks (section III-C) and Peer to Peer SIP (section III-D) approaches propose distributed ways to enable SIP in internet connected MANETs, where control is decentralized moving more intelligence to the MANET nodes and thus to SIP endpoints. As presented in table I, these last three approaches need to modify MN SIP stack introducing new modules to the architecture. This represents a considerable amount of modification in order to avoid limitation of standard SIP architecture. However, gateway discovery mechanism does not require modification, but gateway nodes need to implement some SIP proxy/registrar functionality to enable MANET nodes to interwork with internet nodes. An exception is the P2P SIP approach where MANET gateway need to bridge also P2P SIP with standard SIP.

IV. Conclusion

This paper presents limitations of using standard SIP architecture for providing SIP services in internet connected

TABLE I

COMPARISON OF PROPOSED APPROACHES

Parameters	Proxy-based	Distributed SIP	Service Discovery	P2P SIP with DHT
MN SIP Stack	Standard SIP Stack is sufficient	A SIP Proxy/Registrar module needs to be installed in some MNs	A Service Discovery module and SIP Proxy/Registrar module needs to be installed in MNs	DHT module needs to be installed in MNs. Extension to SIP REGISTER messages are required to transport DHT
GW Discovery	Extension needed (insert bit ”P” in GW_INFO message)	Standard GW Discovery is sufficient	Standard GW Discovery is sufficient	Standard GW Discovery is sufficient
Additional GW Functionalities	GW must implement SIP Proxy/Registrar functionality	GW must implement SIP Proxy/Registrar functionality	GW must implement SIP Proxy/Registrar functionality	GW must implement Peer-to-Peer SIP Proxy/Registrar [14]
Additional Interfaces Required	Interface between GW discovery module and SIP client module in each MN. Interface between GW discovery module and SIP Proxy/Registrar module at GWs	Interface between MANET Routing Protocol module and SIP Proxy/Registrar module at MNs and GWs	Interface between Service Discovery module and SIP Proxy/Registrar module at MNs and GWs	Interface between DHT module and SIP client module at MNs Interface between DHT module and SIP Proxy/Registrar module at GWs

MANETs and proposes alternative approaches. The application of decentralized solutions could improve the scalability of SIP services in internet connected MANETs. The alternatives presented show that for MNs to enable SIP communication with internet nodes, MANET gateways must have some SIP proxy functionality enabled. In order to make these alternatives practical, several improvement are still necessary and also implications with respect to security need to be studied in more detail. Simulation based studies can provide important insight in the performance of the alternative approaches under different mobility and traffic scenarios.

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