SIP based Service Provisioning for hybrid MANETs

Marcel C. Castro, Andreas J. Kassler
Karlstad University
Department of Computer Science
SE-651 88 – Karlstad - Sweden
(Marcel.Cavalcanti, Andreas.Kassler)@kau.se

***Abstract*— Traditional Voice over IP (VoIP) systems is based on client/server architecture, which is not applicable to Mobile Ad Hoc Networks (MANETs), which are a decentralized collection of autonomous nodes. However, internet connectivity for MANETs becomes important as internet connected MANETs can serve as hot spot extension in 4G scenarios. Here, MANET nodes can reach any wired node thus potentially registering with SIP proxies in the fixed network becomes a viable solution. In order to study the implications of using VoIP systems in internet connected MANETs we present in this paper simulation result of SIP service scalability when centralized proxies/registrars located in the Access Network are used by MANET nodes. Alternative approaches to provide SIP services in such environment are also discussed to improve performance.**

***Index Terms* — VoIP, SIP, MANET, Internet Connectivity, P2P, Service Discovery**

I. INTRODUCTION

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 [1], internet connectivity is required, which then 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 [2] 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 [3]. SIP is a signaling, 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 in internet connected MANETs.

Our contribution in this paper is a study of several alternatives to provide SIP services in internet connected MANETs. The use of alternative approaches is motivated through simulation results which address scalability limitations for the standard SIP approach where the SIP proxy is located in the access network and all SIP communication goes through this proxy, even if both SIP endpoints are located in the MANET. We conclude that the standard approach is unsatisfactory from the performance point of view which confirms the needs to deploy alternative approaches in such scenarios.

The paper is organized as follows. In Section II we present the problems related to the deployment of standard SIP approach in internet connected MANETs, where performance simulation is presented in Section III. Section IV describes alternative approaches, exploring advantages and disadvantages of each one. Section V concludes this paper and discusses future work.

II. SIP SERVICES IN HYBRID MANETS

SIP is a request/response protocol and SIP users normally register their contact information with Registrars once they connect to the SIP enabled network. Contact information is comprised of the SIP user name of the user(s) using the device, referred to as SIP address of records (AOR), and the IP addresses where the user is reachable. Proxy servers are needed because SIP users typically do not know the current complete contact information of the callee but only its AOR. 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 proxy server for the callee retrieves from the called side registrar (i.e. utilizes the SIP location service) the bindings for the callee and eventually delivers the request to the intended recipient. SIP can also be operated in a serverless mode which however requires the user to enter the contact address directly.

SIP messages can be carried over UDP or TCP. When SIP is transmitted over TCP, the transport layer provides reliability. But when SIP is carried over UDP, SIP takes care of reliability itself as SIP requests are retransmitted after *Tr(1)* seconds if no response is received, and the timer *Tr(1)* doubles after each retransmission following an exponential backoff behavior. *Tr(1)* should resemble an estimation of the round-trip time with default value of 500ms [4]. The retransmission ceases upon the reception of adequate responses or after seven transmissions of the INVITE request. The SIP response retransmission scheme follows the same concept of the SIP request. Although the retransmission is useful for maintaining the reliability, the retransmission increases load and can cause performance degradation of SIP signaling network.

MANETs are dynamic networks formed by peer nodes which impose limited applicability of standard SIP architecture as registrars and proxies are fixed, static and centralized entities. Therefore, the SIP protocol 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. PERFORMANCE EVALUATION OF STANDARD SIP ARCHITECTURE IN INTERNET CONNECTED MANETS

In order to analyze performance limitation of standard SIP architecture in internet connected MANETs, we plot in this section results based on the usage of centralized SIP proxy/registrar located in the access network. Simulations were performed using ns-2 in a static wireless MANET network topology with 100 nodes. The transmission rate of the nodes was set to 24Mbps, and the nodes are placed on a 10x10 grid at a distance of 200 m with 250m of transmission range and 500m of carrier sense using two ray ground as a radio propagation model.

AODV-UU routing protocol [6] was adopted enabling MANET nodes to discover routes on demand. If the destination is located in the internet, gateways respond with a special proxy RREP. Mobile nodes then using RREQ/RREP phase create tunnels towards the gateway for all traffic destined to the Internet. MANET gateway is connected to the fixed network through wired links and offers wireless ad hoc internet connectivity. The wired links are specified with 5Mbps bandwidth and 40ms delay. SIP proxy/registrar server is located in the same access network as the gateway.

In our scenario, MANET nodes try to establish SIP sessions to other nodes within the MANET or the Internet using the ns-2 SIP extensions provided by [8]. We measure average SIP call setup delay and SIP call blocking probability under different number of background flows, which are modeled as bi-directional exponentially distributed traffic with mean values for talk/silent time of 350ms/650ms, approximating G.729 voice codec [7]. Call setup delay is measured as the time between user agent sends an INVITE request until it gets the 200 OK response while SIP call blocking probability presents the percentage of SIP sessions that were not established within 5 seconds [5]. Results on capacity of voice calls are available from [9].

In this simulation, sources and destinations have been selected randomly in order to generate 100 SIP call attempts where all callers are located inside the MANET and 75% of callees are inside and 25% outside the MANET. For the background flows always 100% of voice sources are inside the MANET and 75% of the voice sinks are located inside the MANET and 25% in the Internet. As an example for 24 background flows, sources are always MANET nodes, 6 sinks are outside the MANET and 18 sinks are also MANET nodes. Voice sources and sinks picked differently for different number of hops between two nodes (or a node and the gateway) varying from 2 to 7 hops.

Fig 1 shows the average SIP call setup delay as number of background flows increases, for different number of hops. As expected, raising the number of background flows increases the SIP call setup delay of the new call attempts independent of the number of hops between source and destination. There are several factors contributing to the overall call setup delay. SIP messages have to compete with background traffic in the ad hoc network thus leading to increased packet loss probability for SIP signaling. If a SIP message is lost or the answer from the callee/proxy does not arrive until *Tr(1)* expires, it will be re-transmitted following an exponential backoff procedure. If the answer from the proxy arrives later, the re-transmission was not necessary further contributing to the congestion. Also, SIP messages compete at the MANET gateway with background flows for buffer space and might be dropped. As the SIP proxy is located in the access network, the delay imposed to SIP messages to reach the proxy in the access network also contributes to the increase of SIP setup delay together with SIP processing delay at the proxy. Increasing the number of hops for background flows also leads to high SIP setup delays due to the increase in channel utilization and contention for the wireless medium. Also we observed increased number of retransmission due to Medium Access Control (MAC) data frame collisions which in turn also increases channel utilization.

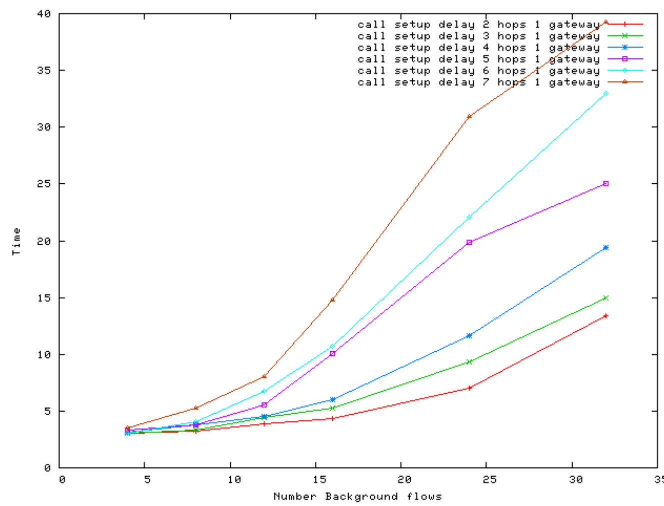

Fig 1: SIP call setup delay versus network load

As can be seen from Fig 1, the worst case is seen for seven hop paths between source and destination. Also, in some source/sink combination, all SIP signaling must traverse the gateway twice (to reach the SIP proxy/server in the access network plus to be forwarded to the callee located in the MANET). Here, the average time to establish a new SIP call can reach almost 40 seconds under 32 background flows. Even for 12 calls we observed more than 8 seconds call setup delay. Therefore, as the number of hops and background voice calls increases, the call blocking probability for new SIP calls increases as shown in Fig 2. Even if we choose the best case where source and destination is 2 hops away only 25% of the new SIP calls are completed within 5 second when 32 background flows have been established, resulting in a blocking probability of 75%, which is high for voice communication systems [10].

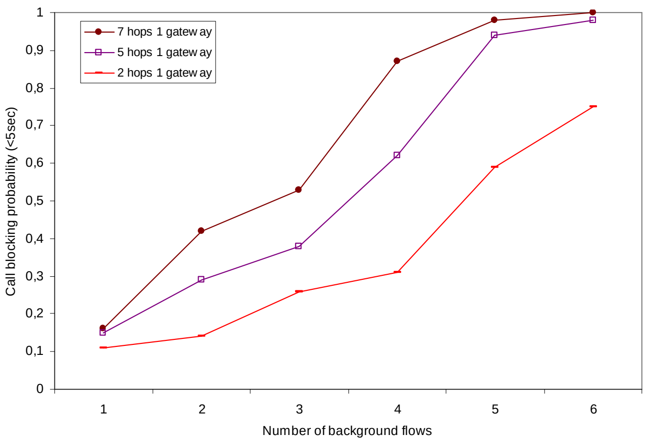

Fig 2: Blocking probability versus network load

In order to evaluate the reasons for the high SIP call setup delay, we plotted in Fig 3 the number of invitation messages (SIP INVITE) generated for the 2 and 5 hops scenarios. We differentiated the number of invitation messages generated in three categories; (1) original 100 invitations generated to establish the 100 SIP calls, (2) invitations re-sent due to SIP messages dropped (e.g. SIP 200 OK dropped due to network congestion), and (3) invitations re-sent due to SIP timeout (e.g. SIP timer expires just before SIP 200 OK arrives). Fig 3 presents case 2 and 3, where the number of re-sent invitations increases when number of background traffic increases. Furthermore, it shows that if background traffic is low not many SIP messages are dropped so only a few invitations are sent due to packet loss. As more background flows are added, more packets get dropped leading to more retransmission of INVITE due to missing SIP packets. However, even when small number of background flows and few hops, the number of re-sent INVITEs due to timeout is significant. This is due to the bad SIP timer configuration, which times out after 500ms following an exponential backoff strategy. The reactive nature of AODV combined with the additional delay to reach the proxy in the access network leads to frequent timeout and thus unnecessary retransmissions. Therefore, a better timeout strategy needs to be deployed for SIP over UDP in internet connected MANETs.

IV. SIP BASED SERVICE PROVISIONING FOR INTERNET CONNETECD MANETS

In order to enable SIP in internet connected MANETs, several alternative approaches are described in this section.

A. SIP Proxy/Registrar co-locates at Gateways

As discussed in section II, the use of standard SIP architecture where all SIP signaling exchanged between SIP MANET nodes (or a MANET node and an external node in the Internet) needs to pass through gateways that connect the MANET to the Internet brings performance limitation to SIP services. 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 of 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 [11]. 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.

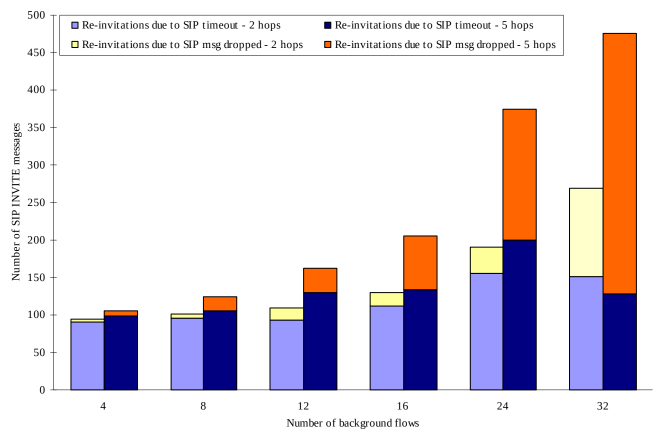

Fig 3: Invitation Attempts versus network load for 1 gateway and 5 hops

We adopt the strategy to add SIP proxy/registrar location information to this mechanism instead of using another auto-configuration protocol such as DHCPv6 [12]. The selected gateway discovery mechanism has a strong impact on the overall performance due to the number of messages exchanged versus latency [11]. An integration of the proactive approach based on prefix continuity [13], 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 [13], which is based on the GW_INFO message, adding 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 Fig 4). GW_INFO message format extension proposed for this approach does not impose more overhead to

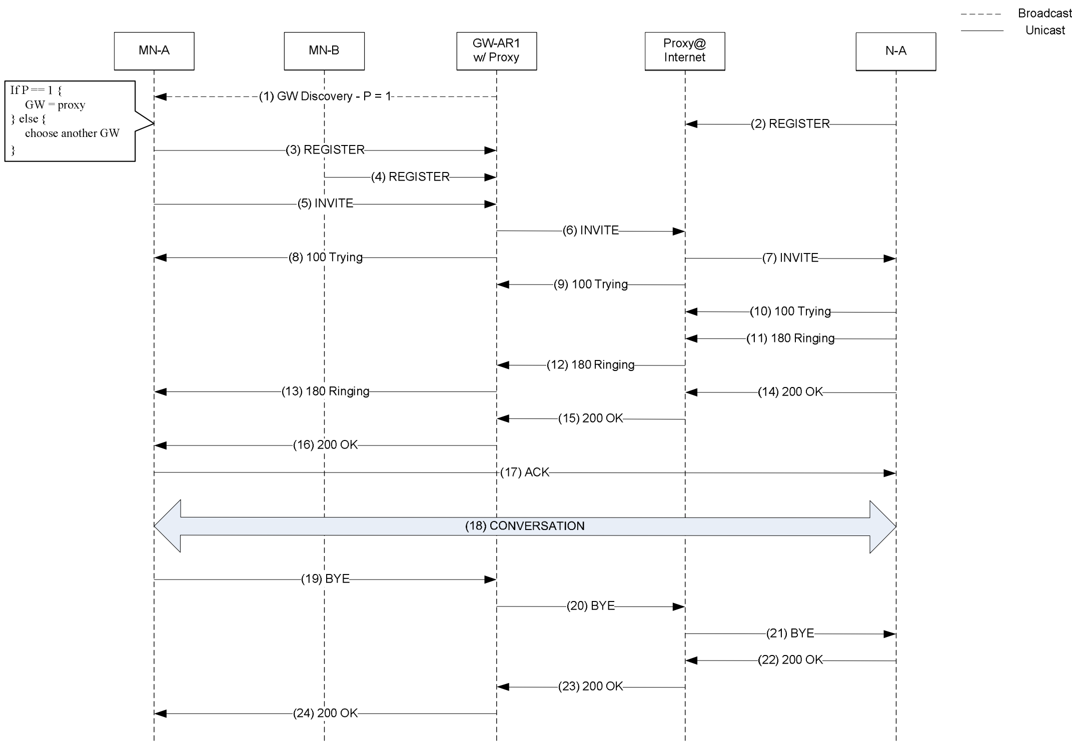

Fig 4: MSC of proposed approach

the network and also does not modify the protocol behavior. According to [13], 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 a gateway based on its SIP proxy/registrar capabilities.

The registration and session initiation processes follow the same behavior as the standard SIP mechanisms. Fig 4 presents a MSC of the proposed approach, where gateway GW-AR1 operates as a SIP proxy/registrar server in the internet connected MANET. As all MANET nodes (MN-A and MN-B) have learned SIP proxy/registrar capability (GW-AR1 IP address) through gateway discovery mechanism (message 1 in Fig 4), they start the registration process (message 3 and 4) in order to enable SIP service. In this example, MN-A calls user N-A using N-A's SIP URI located in the Internet and registered at Proxy@Internet. As shown in Fig 4, INVITE (message 5-7) is used to request establishing a session between users. User N-A receives an INVITE and returns a provisional response 100 Trying (message 8-10) immediately indicating the receipt of the INVITE and call progress. After parameters confirmation such as codec to be used, user N-A returns a response 180 Ringing (message 11-13). When N-A picks up the phone, it sends a response 200 OK (message 14-16). Finally user MN-A receives the 200 OK and returns and ACK (message 17) to user N-A. Then the session is established and the call setup is followed by direct media exchange using RTP without proxy involvement. The session is closed through an exchange of BYE (message 19-21) and 200 OK (message 22-24). It can be seen from Fig 4 that the proposed approach is an extension of standard SIP through the insertion of gateway discovery message.

An 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. However, using the proxy co-located at the gateway could have drawbacks, because it is then difficult to offer 3GPP/IMS conform services. Integration into 4G networks architecture is currently under study within the IST project DAIDALOS [1]. An evaluation of proposed method is available from [9].

B. Distributed SIP and Integration with routing protocol

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) [14] and integration with cluster based routing where cluster heads take the responsibility of acting as SIP proxy/registrar servers [15]. 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 [14], 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, where the logic of SIP has not changed but the servers are decentralized and embedded in every MANET node. This requires to install middleware on every MANET node to intercept SIP signaling.

To work in internet connected MANETs, [16] 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 Frameworks

A service discovery framework can be used to discover SIP users either by finding out the bindings of users within reach in the MANET or to discover the IP addresses of a user by SIP AOR. Therefore, an integration of service discovery with SIP services seems to be beneficial. Service Location Protocol (SLP) [17] was used by [14] 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 normal SIP processing. Thus all SIP signaling goes 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 substitute the registration procedures used in standard SIP, where bindings are received and maintained through periodic SIP REGISTER messages by SIP Registrar.

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. We propose that 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 B). In the server based approach, the callee is not registered with the MANET node that acts as SIP proxy so this proxy then has to forward the INVITE to the MANET gateway.

The main problem is mutual interoperability 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 broadcast messages [13].

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 Peer to Peer SIP, a SIP system uses P2P mechanisms based on e.g. distributed hash tables (DHT) for management of distributed functions such as user location [18].

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 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 [18] 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).

TABLE 1: 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
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

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 of DHT processing in MANETs which has some limitations [19]. Also, unlike *O(1)* lookup cost in classical client-server based systems, the P2P lookup cost can be much higher [18] leading to potentially increased call setup latency.

E. Impact of proposed approaches on SIP architecture and functionalities

Table 1 gives an overview on new interfaces required in MANET nodes (MN) and gateway nodes (GW) for each proposed approach. Implementing SIP proxy/registrar into MANET gateway nodes, as discussed in section 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 trough the creation of new interfaces in order to convey this information to the SIP stack.

Distributed SIP and Integration with routing protocols (section B), Integration of SIP with Service Discovery Frameworks (section C) and Peer to Peer SIP (section 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 1, 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 gateways need to bridge also P2P SIP with standard SIP.

V. CONCLUSION

This paper analyses the limitations of using standardized SIP infrastructure for providing SIP services in internet connected MANETs and demonstrates several 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 should have SIP proxy functionality enabled. In order to make these alternatives practical, several improvements are still necessary and a detailed comparison is required for the different approaches. This should pave the way for efficient SIP support for future wireless network.

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