Demo Proposal – VoIP for isolated and Internet-connected Mobile Ad Hoc Networks

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ABSTRACT

SIPHoc is a middleware infrastructure for session set up and management in MANETs. SIPHoc is compliant to the SIP standard, does not require any centralized components, is message efficient (through routing message piggybacking), and independent of the routing protocol (currently, SIPHoc supports both AODV and OLSR). Moreover, SIPHoc allows seamless interaction with the Internet by treating nodes connected to the Internet as *gateway services* and making this information known across the MANET.

In this demo we will show how SIPHoc can be used to build a wireless multihop VoIP network. Using iPAQ handhelds, participants of the demo can set up VoIP calls between each other, as well as to and from the Internet if a gateway can be discovered. As part of the demo we will further use a protocol analyzer to show how the SIP contact information is piggybacked onto routing messages, for both AODV and OLSR.

1. INTRODUCTION

Mobile Ad hoc Networks (MANETs) are envisioned as playing a significant role in situations where no network infrastructure is available. VoIP for MANETs is of particular interest in settings such as emergency response [10] [7] [4]. Traditional VoIP applications, however, cannot easily be deployed in MANETs since setting up a VoIP session involves centralized components. Numerous attempts have been made to adapt SIP [11] – the de facto standard for VoIP session establishment – to MANETs [9, 5, 8]. Unfortunately, none of these proposals has been implemented and often they work only on either isolated MANETs, or MANETs permanently connected to the Internet. Moreover, all existing solutions impose limitations on the network topology and/or the routing protocol. SIPHoc [12] is a middleware platform for session establishment and management in MANETs that does not suffer from any of the limitations of previous proposals. The purpose of this demo is to show how SIPHoc can be used to provide VoIP in a fully decentralized wireless multihop network. Since SIPHoc is SIP compliant, any SIP compatible VoIP application can be used seamlessly in MANETs. In our demo, we will demonstrate this by using two different, unmodified VoIP clients, *Kphone* [1] and *Twinkle* [2].

2. SIPHOC

The SIPHoc architecture is shown in Figure 2. It is based on four components running as independent operating system processes within a node in the MANET.

- A *SIPHoc Proxy* with a standard SIP interface but implementing MANET specific functionality. Each *SIPHoc Proxy* serves as an outbound SIP proxy for local SIP applications.
- A *MANET SLP* layer providing a regular SLP (Service Location Protocol) interface but implementing efficient and decentralized service lookup functionality.
- A *Gateway Provider* that turns the node into a gateway if the node has Internet access.
- A *Connection Provider* that manages connections of the node to the Internet when there is a gateway in the MANET.

2.1 MANET SLP

MANET SLP is a fully distributed service discovery platform for MANETs. MANET SLP provides a regular SLP [6] interface over UDP for service registration and lookup. MANET SLP works by piggybacking service information onto routing messages. This is done by capturing routing messages (using the *libipq* library under linux) and extending them with service information. To assure generality, the routing specific functionality is encapsulated within a *routing handler*. The routing handler is a software module that receives raw routing packets as input and generates altered packets that include the piggybacked service information. The advantage of this approach is is that, routes are established simultaneously with service discovery, and routing protocols do not have to be modified to be used in SIPHoc.

2.2 SIPHoc Proxy Example

Assume two users (Alice and Bob) in the MANET, both running a *SIPHoc Proxy* and an unmodified SIP-based VoIP

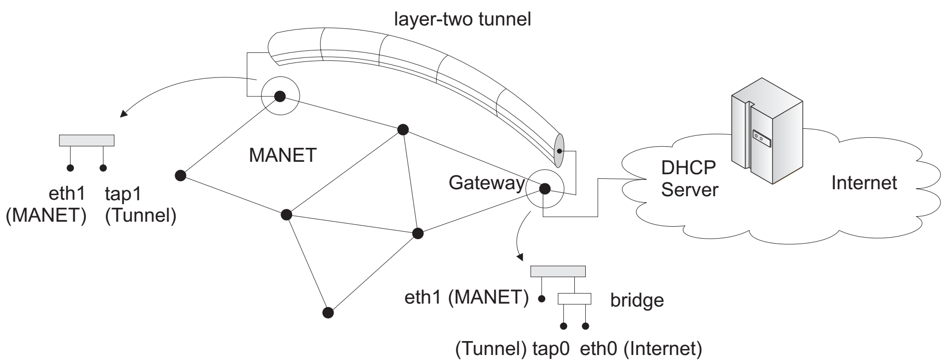

Figure 1: Network state after a node has successfully connected to a gateway

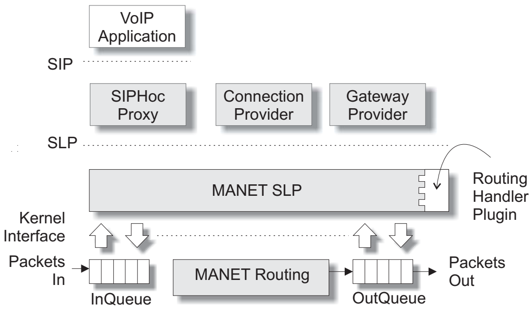

Figure 2: Architecture Overview: *SIPHoc* Processes on a MANET node

application (e.g. Kphone). When the VoIP application is started, it sends a SIP REGISTER message to its outbound proxy (which is the local SIPHoc proxy in our case) in order to announce its current contact address. The local SIPHoc proxy for each of them will then contact the underlying MANET SLP and register the entry so that it is advertised by the MANET SLP module. To establish a SIPHoc session between Alice and Bob, Alice sends a SIP INVITE message to the *SIPHoc* proxy. The proxy then requests the entry from the MANET SLP layer. Once the contact address of the proxy of the user Bob is found, the SIP INVITE message is forwarded to Bob.

SIPHoc is fully agnostic about the routing protocol used since it allows protocol specific *routing handlers* to be inserted into the system like a plug-in (see section 2.1). Currently, we have implemented a routing handler for AODV (an on-demand routing protocol) and OLSR (a pro-active routing protocol). In an on demand routing handler, a lookup request from the SIPHoc proxy results in a *route request* from the routing protocol with the SIP information piggybacked to it. In a pro-active routing handler, the routing messages are exploited to constantly disseminate and maintain information across all MANET SLP modules.

2.3 Attaching SIPHoc to the Internet

The ultimate vision of a SIP infrastructure that works in both isolated and Internet connected MANETs is that clients can use their officially registered SIP accounts[1] transparently in the MANETs. In other words, given that Bob's SIP account is officially associated with the SIP provider at *ethz.ch*, we would like calls to and from the Internet to become possible as soon as the MANET is connected to the Internet. On the other hand, Bob should always be able to call any SIP user within the MANET – and vice versa – even if the MANET is currently disconnected from the Internet. We have extended the *SIPHoc Proxy* described in the last section to implement this vision, see [12] for details. Internet connectivity to nodes in the MANET is provided through a special gateway that is discovered dynamically. If a node has Internet connectivity it makes this information available to other nodes by publishing an SLP *gateway* service. It also starts a layer two tunnel server ready to accept connections. Nodes willing to have Internet access may look up a *gateway* service and open a layer two tunnel connection to the node offering the tunnel server. Since the gateway node will directly forward all the traffic it receives on the tunnel interface to the Internet, any node with a tunnel connection is automatically attached to the Internet as well. A more detailed perspective on how components the network looks after a node has successfully discovered a gateway is illustrated in Figure 1.

3. SYSTEM SETUP

Our MANET deployment to demonstrate the VoIP system described above consists of 4 iPAQ handhelds (Figure 3b) and two laptops with the following configuration: all of the devices run a Debian Linux. Each iPAQ is equipped with an 802.11 wireless card. One of the laptops is equipped with both an 802.11 wireless card and a wired network interface (laptop A). The other laptop is only equipped with a wired network interface (laptop B). All of the iPAQs and laptop B run SIPHoc. Additionally these devices have both an AODV and an OLSR routing protocol installed. We will use a special firewall configuration to artificially enforce multihop communication among all wireless devices. In our demonstration, the wired network interface of laptop A may from time to time be connected to laptop B while laptop B runs a DHCP server. Laptop B and the wired network interface of laptop A are considered to emulate the Internet. All devices involved in the demonstration run the Kphone

[1]A SIP account associated with some official SIP provider in the Internet

(a) Turning an iPAQ into an ad hoc telephone (b) iPAQ handhelds used for the demo

Figure 3: Equipment used for the demo

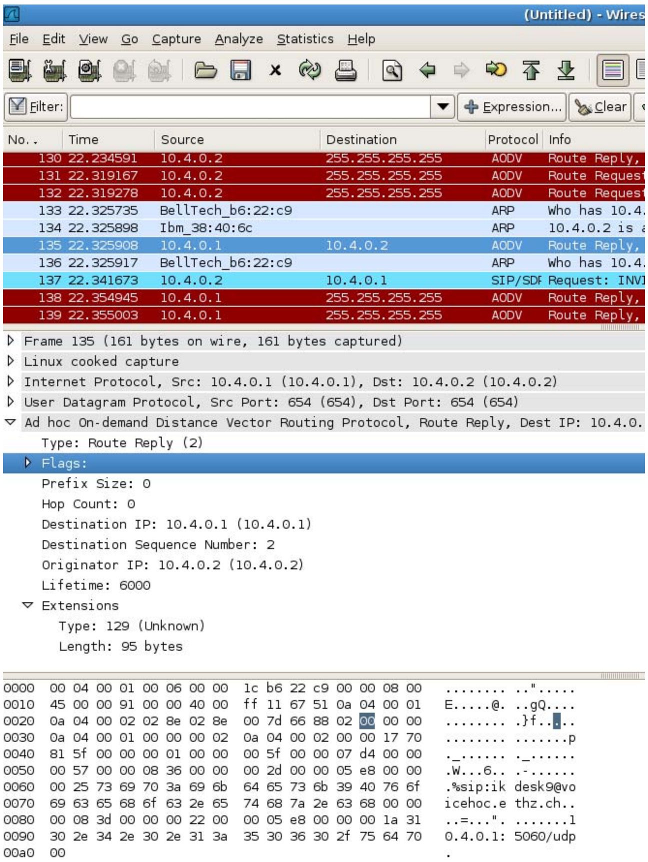

Figure 4: Snapshot of a packet analyzer showing an AODV route reply with encapsulated SIP contact information

and the Twinkle VoIP application. The overall setting corresponds to the setting illustrated in Figure 1. The setup time will be one hour maximum.

4. DEMONSTRATION

In the demo, we turn the iPAQ handhelds into VoIP telephones (Figure 3a). Participants of the demo can set up VoIP calls to each other without relying on any fixed infrastructure. Moreover, participants can set up calls to and from the fixed network depending on whether laptop A currently is attached to laptop B or not. The dynamics of the system is demonstrated by periodically plugging and unplugging the network cable at laptop A, which is serving as a gateway node. One aspect of *SIPHoc* is that service information is propagated in the network using routing layer piggybacking. We will use the Wireshark [3] protocol analyzer on laptop A to show how SIP contact information is piggybacked onto routing layer messages (see Figure 4).

5. REFERENCES

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