Enabling Multimedia Services over Wireless Multi-Hop Networks
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Enabling Multimedia Services over Wireless Multi-Hop Networks

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Abstract

With the constant development of wireless technologies, the usage of wireless devices tends to increase even more in the future. Wireless multi-hop networks (WMNs) have emerged as a key technology to numerous potential scenarios, ranging from disaster recovery to wireless broadband internet access. The distributed architecture of WMNs enables nodes to cooperatively relay other node's packets. Because of their advantages over other wireless networks, WMNs are undergoing rapid progress and inspiring numerous applications. However, many technical issues still exist in this field. In this thesis we investigate how Voice over IP (VoIP) and peer-to-peer (P2P) application are influenced by wireless multi-hop network characteristics and how to optimize them in order to provide scalable communication.

We first consider the deployment of VoIP service in wireless multi-hop networks, by using the Session Initiation Protocol (SIP) architecture. Our investigation shows that the centralized SIP architecture imposes several challenges when deployed in the decentralized wireless multi-hop environment. We find that VoIP quality metrics are severely degraded as the traffic and number of multiple hops to the gateway increase. In the context of scalability, we further propose four alternative approaches which avoid current limitations.

In the second part of this thesis we tackle the network capacity problem while providing scalable VoIP service over wireless multi-hop networks. The performance evaluation shows the influence of intra and inter-flow interference in channel utilization, which directly impacts the VoIP capacity. In order to avoid the small VoIP packet overhead, we propose a new adaptive hop-by-hop packet aggregation scheme based on wireless link characteristics. Our performance evaluation shows that the proposed scheme can increase the VoIP capacity by a two-fold gain.

The study of peer-to-peer applicability over wireless multi-hop networks is another important contribution. A resource lookup application is realized through structured P2P overlay. We show that due to several reasons, such as characteristics of wireless links, multi-hop forwarding operation and structured P2P management traffic aggressiveness the performance of traditional P2P applications is rather low in wireless multi-hop environments. Therefore, we suggested that a trade-off between the P2P lookup efficiency and the P2P
management traffic overhead can be achieved while maintaining the overlay network consistency in wireless multi-hop networks.

**Keywords:** Wireless Multi-hop Networks; Mobile Ad-hoc Networks; Wireless Mesh Networks; Voice over IP; Session Initiation Protocol; Peer-to-Peer Overlay Networks.
Para minha esposa
Roberta Martins Agostini
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Karlstad, May 2009

Marcel C. Castro
List of Appended Papers

This thesis is comprised of the following four peer-reviewed papers. References to the papers will be made using the Roman numbers associated with the papers such as Paper I.


Comments on my Participation

Paper I: The SIP service simulations in MANETs was the result of a master's student thesis that I supervised. It motivated me to develop the main idea of the paper, four alternative approaches to increase SIP service scalability in MANETs. Prof. Andreas J. Kassler took part in the discussions regarding the proposals' functionality and the theoretical analysis. He also supervised the work.

Paper II: I am the main author of this paper, and I participated in all parts of the work behind it. I conducted all experiments. Prof. Andreas J. Kassler supervised the work by discussing and reviewing the paper contents.

Paper III: I co-authored this paper. Its main contribution, a new adaptive hop-by-hop aggregation scheme based on link quality metrics, was the result of a master's student thesis, which I partially supervised with Prof. Andreas J. Kassler. The writing was a collective effort of the authors, where I specifically wrote the performance evaluation section.
Paper IV: I am the main author of this paper. I developed the idea behind the paper and conducted the experiments together with my co-authors. The idea of this paper also produced a master thesis defended by my co-authors Eva Villanueva and Iraide Ruiz and supervised by me. Prof. Andreas J. Kassler and Susana Sargento took part in the discussions regarding the theoretical analysis and the experiment results.

Other Papers

The following publications, although not included in this thesis, contain material that is related to the aforementioned contributions:


   This paper studies the implications of using standard SIP architecture in internet connected MANETs. It analyzes limitations of SIP service scalability when centralized proxies/registars located in the Access Network are used by MANET nodes and proposes alternative approaches to such limitations. Paper I is based on this work but extends it with a standard SIP performance evaluation over internet connected MANET scenarios and an extended description of the alternative approaches.


   A gateway based approach is proposed in this paper in order to minimize the impact of SIP service scalability in hybrid MANET scenarios. The paper describes the benefits of using MANET's gateways with SIP proxy functionality in hybrid MANETs. Paper II is based on this work but extends it by incorporating the influence of multiple gateways in the SIP service performance.


   This paper presents the benefits of using VoIP packet aggregation in wireless multi-hop networks. It proposes a static hop-by-hop packet aggregation mechanism that significantly enhances capacity of VoIP in wireless mesh networks while still maintaining satisfactory voice quality. Paper
III is based on this work but extends it with regards to a new adaptive hop-by-hop packet aggregation mechanism based on wireless link characteristics information exchanged among neighboring nodes.


This paper carries out a feasibility study of VoIP in a dual radio mesh environment. It presents the design of a wireless mesh testbed and methodology for performing the measurements, and also a simulated evaluation of VoIP scalability. The static hop-by-hop packet aggregation idea described in Paper III is part of the contributions of this paper.


This book chapter provides a comprehensive survey on recent research on mechanisms to provide peer-to-peer services in wireless multi-hop networks. Various approaches are presented that couple in several ways the interactions between the peer-to-peer overlay and the wireless multi-hop network. The approach presented in Paper IV, where a DHT is deployed on top of a broadcast-based ad-hoc routing, is one of the approaches discussed in this paper.


The Reference Path Ad-hoc Routing (REPAR) mechanism, proposed in this paper, optimizes the performance of reactive ad-hoc routing protocol by adopting a constrained flooding mechanism during route maintenance. The benefits of REPAR is complementary to Papers I-IV within mobility scenarios.

This paper proposes the application of a Policy-Based Network Management (PBNM) for dynamic reconfiguration of ad-hoc networks. The paper contribution comprises the proposal of an ad-hoc policy information model based on DEN-ng policy model for definition of ad-hoc specific policies focusing on dynamic routing protocol parameters configuration management. The proposed policy-based model is complementary to the approaches presented in Papers I-IV.
## Contents

Abstract

Acknowledgements

List of Appended Papers

Introductory Summary

1 Introduction

2 Internet Service Provisioning over Wireless Multi-hop networks
   2.1 General Characteristics of Wireless Multi-hop Communication
   2.2 Multimedia-based service in WMNs
   2.3 P2P computing in WMNs

3 Research Questions

4 Research Method Used

5 Contributions

6 Summary of Papers

7 Conclusions and Future Work

Paper I: SIP based Service Provisioning for hybrid MANETs

1 Introduction

2 SIP Services in Hybrid MANETs

3 Performance Evaluation of Standard SIP Architecture in Internet connected MANETs

4 SIP based Service Provisioning for Internet Connected MANETs
   4.1 SIP Proxy/Registrar co-located at Gateways
   4.2 Distributed SIP and Integration with routing protocol
   4.3 Integration of SIP with Service Discovery Frameworks
   4.4 Peer to Peer SIP
   4.5 Impact of proposed approaches on SIP architecture and functionalities

5 Conclusion
CONTENTS

6 Acknowledgment 45

Paper II: Challenges of SIP in internet connected MANETs 49

1 Introduction 51

2 SIP Service in Internet Connected MANETs 52

3 Optimizing SIP Service Provisioning in Internet Connected MANETs 54

4 Performance Results of SIP in Internet Connected MANETs 56
   4.1 Simulation Description ........................................... 56
   4.2 SIP Proxy co-located at GWs versus SIP Proxy at ANs ....... 57
   4.3 Multiple gateways ............................................. 59
   4.4 VoIP capacity .................................................. 61

5 Conclusion 63

6 Acknowledgment 64

Paper III: VoIP Packet Aggregation on Link Quality Metric for Multihop Wireless Mesh Networks 67

1 Introduction 69

2 Predicting Packet Size for Aggregation through Link Quality 71
   2.1 Determining packet size ...................................... 72
   2.2 Adaptive Packet Aggregation ................................. 75

3 Performance Evaluation 76

4 Conclusion 80

5 Acknowledgment 80

Paper IV: Performance Evaluation of Structured P2P over Wireless Multi-hop Networks 83

1 Introduction 85

2 Structured P2P Overlay Networks in MANETs 87
   2.1 Structured Overlay Networks ................................. 87
   2.2 Challenges of Structured P2P in Multi-hop Environment .... 88
   2.3 Bamboo DHT ................................................ 89
CONTENTS

3 Performance Evaluation 91
4 Related Work 95
5 Conclusion 95
6 Acknowledgment 96
Introductory Summary
1 Introduction

Wireless devices such as cellular phones, laptops, personal digital assistants (PDAs), etc. have become indispensable. The prevalence of wireless devices can be attributed to the mobility they provide. With the development of new wireless standards and new wireless technologies, the usage of wireless devices tends to increase even more in the future.

A wireless network is a network of nodes or devices that have wireless communication capabilities. Based on the communication model, wireless networks can be classified as cellular networks and multi-hop networks. In a cellular network, a set of devices communicate with a static central device, called the base stations. The cellular networks have centralized communication architecture with the base station coordinating the communication activity of other devices that are usually mobile.

Wireless multi-hop networks usually don’t have a dedicated infrastructure and rely on multi-hop communication. Nodes in a wireless multi-hop network cooperatively forward other nodes’ data. These networks have a distributed communication architecture, where nodes make individual decisions on routing and medium access. Since wireless multi-hop networks can be deployed rapidly and flexibly, it is attractive to numerous potential applications, ranging from multi-hop wireless broadband Internet access to multimedia services such as Voice over IP, and P2P applications.

In this introductory summary, we give the big picture introduction to the issues that arise when internet service provisioning is built on top of wireless multi-hop networks. Thus, in Section 2.1, we give an introduction to the general characteristics of wireless multi-hop communications, focusing mainly on well-known examples such as mobile ad-hoc networks (MANETs) and wireless mesh networks. Those general characteristics are necessary to understand the issues brought forward while discussing multimedia-based service and P2P communication in wireless multi-hop networking, respectively in Sections 2.2 and 2.3. In those sections we also give an overview of the related work in the area. Section 3 addresses the research questions, while Section 4 presents the research methodology used. The main contributions of this work are described in Section 5. Section 6 gives short summaries of the included papers, whilst Section 7 presents our conclusions with discussions of future works.
2 Internet Service Provisioning over Wireless Multi-hop networks

The interest in wireless communications has grown constantly for the past decades, leading to an enormous number of applications and services embraced by billions of users. Ubiquitous Internet service provisioning suitable for wireless systems has been one of the engines that pushed the research and industry societies to innovation and growth.

The dissemination of wireless data networks has been increasing in an astonishing rate since the first release of the IEEE 802.11 standard in late 1999. In order to meet the increasing demand for high bandwidth network services, high data rate radio networks have recently been proposed to replace wired networks in many applications. New families of technologies, such as WiFi [1] and WiMAX [2], have been conceived to provide high speed wireless communications to a large number of users. A new generation of standard-based devices has been developed to offer a mobile and quickly deployable alternative to the current cabled networks. Consequently, the provision of scalable services for future wireless networks is becoming an increasingly critical aspect in networking.

The spread of wireless link technologies has diversified the number of ways that computing devices can interact and exchange data. It also allows an unprecedented level of mobility. As these technologies evolve, they become smaller and less expensive, allowing them to be integrated in very small devices, such as sensors and radio frequency identification’s (RFID) tags.

However, with the broad spectrum of technologies and device capabilities also come many challenges. The Internet protocols struggle in mobile and wireless environments. They were not initially designed to operate over loss wireless links, with large variances in delay. Thus, the challenge of Internet service provisioning over mobile and wireless environments, in special over wireless multi-hop networks, has spawned lots of research in how to improve current Internet protocols.

2.1 General Characteristics of Wireless Multi-hop Communication

Most traditional wireless networks operate using a central coordinator, called a base station or an access point. The base station is part of a wireless infrastructure, which is usually deployed by a network operator, e.g., a cellular provider, or as part of a company, university, or home network. This infrastruc-
Internet Service Provisioning over Wireless Multi-hop networks

ture typically provides wireless edge access to client hosts that want to access an internal Local Area Network (LAN), or the Internet. The clients (nodes) thus access the Internet over a wireless link, which connects them with the base station. With WiFi technology, these networks are often called Wireless LANs (WLANs). The WiFi technology, through the IEEE 802.11 standard, also allows direct peer-to-peer communication between wireless nodes by operating in so called ad-hoc mode, without involving base stations or any type of infrastructure. Such communication can be extended over several hops, known as a wireless multi-hop communication.

The wireless multi-hop communication has many use cases, both in standalone deployments, but also to extend the reach of infrastructure, e.g. hotspots, in areas in which there is little or no communication infrastructure or the existing infrastructure is expensive or inconvenient to use. Examples of networks that apply wireless multi-hop communication are so called mobile ad-hoc networks [3] and wireless mesh networks [4]. Mobile ad-hoc networks are the most general wireless network formed dynamically by an autonomous system of nodes that are connected via wireless links without using the existing network infrastructure or central administration. The interconnection of MANETs to the internet (e.g. a fixed infrastructure based IP networks) is a very important characteristic in order to provide the ubiquitous user internet access anywhere at any time [5, 6]. In such scenarios, also known as "internet connected MANET", or "hybrid MANET", the user within an ad-hoc network will get access to the public internet by using the packet forwarding capabilities of intermediate nodes towards the access routers or gateways. Several gateway discovery protocols have been proposed [7–9], and important research issues including load-balancing techniques [10], self-configuration [11] and mobility [12].

Wireless mesh networks is an area that has been receiving a lot of attention within the last few years. Figure 1 shows a wireless mesh network architecture, where dash and solid lines indicate wireless and wired links, respectively. They can be considered as a quasi-stationary ad-hoc networks that very much resemble the early multi-hop packet radio networks. A common wireless mesh network architecture includes mesh routers forming an infrastructure/backbone for clients that connect to them. The mesh routers form a mesh of self-configuring, self-healing links among themselves. With gateway functionality, mesh routers can be connected to the Internet. In such scenario, the wireless mesh backbone permits the integration of existing wireless networks through gateway/bridge functionalities in mesh routers. In contrast to MANETs, stationary mesh routers can self-optimize to a degree not possible in mobile scenarios. By using commodity hardware and unplanned deployment of routers, anyone should be able to integrate, e.g., their home router
into the mesh network. However, the unplanned structure of the mesh networks can still lead to severe interference. Several experimental mesh networks have been created for research purposes (e.g. MIT’s Roofnet [13], Berlin’s Roofnet [14] and KAUMesh [15]), and eminent research efforts include wireless link quality [16, 17], multi-hop effects [18], channel assignment strategies [19, 20] and cross-layer mechanisms [21].

As the communication is extended through multiple hops, several wireless issues come into play in such scenarios. For instance, the different ranges of wireless signal propagation cause a number of adverse effects when wireless nodes simultaneously try to access the medium. Carrier Sense Multiple Access - Collision Detection (CSMA/CD) attempts to prevent a node from transmitting simultaneously with other nodes within its transmitting range by requiring each node to listen to the channel before transmitting. Unfortunately, hidden terminal problems [22] degrade the performance of CSMA substantially, because carrier sensing cannot prevent collisions in that case, thus increased packet loss and reduced throughput occurs. Exposed terminal problem is com-
2. Internet Service Provisioning over Wireless Multi-hop networks

Complementary to hidden terminals, and it occurs when a node is prevented from sending packets to other node due to a neighboring transmitter within the same range but out of range of the receptor. The IEEE 802.11 uses request-to-send / clear-to-send (RTS/CTS) acknowledgment and handshake packets to partly overcome those problems. However, RTS/CTS is not a complete solution and may decrease throughput even further since in many cases the reservation of the channel is less efficient than just dealing with the collision through retransmissions [23], or network coding [24].

Therefore, in multi-hop communication, collision and interference become more complex and depend on many factors such as radio environment, modulation schemes, transmission power, or sensing ranges. As a result, adjacent links and even links further separated, affect each other during transmission and they might have to share the wireless channel. In single channel networks, a two-hop configuration hence effectively halves the available bandwidth [25]. Other links still within interference range also might affect links further down a multi-hop path, reducing the link bandwidth even further. Such behavior has many subtle performance implications to higher layers such as TCP [26, 27], which are not visible in single hop networks.

Introducing mobility into the network also introduces new challenges to wireless multi-hop communication. As nodes can move in and out of each other’s range, the network topology changes frequently. Such changes must be communicated across the network to update routes accordingly. To maximize wireless communication channels’ bandwidth, communications about topology changes must be minimized. Therefore, frequent topology changes and limited bandwidth place significant requirements on routing protocols. Routing protocols supporting high levels of mobility have mainly been developed within the wireless multi-hop area, by adopting two main goals: 1) supporting dynamic and mobile environments, and 2) reducing the overhead of routing updates. The approaches to achieve this are generally classified as proactive (e.g. OLSR [28] and DSDV [29]), reactive (e.g. AODV [30] and DSR [31]) or hybrid routing (e.g. HWMP [32]).

2.2 Multimedia-based service in WMNs

The advancement of wireless multi-hop networks enables the delivery of ubiquitous and pervasive computing scenarios that supports a range of mobile services in addition to conventional mobile internet access [33]. The further success of these networks derives from their ability to provide users with cost-effective services that have the potential to run anywhere, anytime, and on any device without (or with little) user attention.
In this context, demands for multimedia-based services are emerging. Voice over IP (VoIP), also referred to as IP Telephony, constitutes one of the most flourishing applications. It has emerged as an important application over Internet with the tremendous popularity of Skype [34]. The cost savings and the easy deployment benefits achieved by VoIP using existing network infrastructures are the main factors driving the steady growth of VoIP.

VoIP holds a considerable appeal both from users’ and service providers’ viewpoint, the most important one being the edge in cost savings over Public Switched Telephone Network (PSTN). The interest in VoIP dates back to the end of the previous decade where users and service providers were praised by the countless benefits and new business opportunities. On the one hand, VoIP opens up exciting possibilities for users, such as flexibility and monetary savings. On the other hand, VoIP promises new revenue sources to service providers, given them also an easy and cost-efficient way to compete with incumbent operators.

Pervasive commercial deployment of VoIP over wired networks, and the mobility, flexibility and the scalability provided by WiFi technology have attracted great research effort recently in the area of wireless VoIP [35–39]. In addition, many researchers advocate Session Initiation Protocol (SIP) as a feasible and important enabler for VoIP applications, because it is simpler and more efficient than H.323 [40]. SIP [41] 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. SIP has been selected as the call control protocol for the third generation (3G) IP-based mobile networks [42].

In SIP, before an end-user can start a VoIP session, the session setup needs to be performed in order to negotiate session and media parameters. The time interval to perform the session setup is called the session setup time. The SIP session setup phase may involve 1 end-users (also called user agents, that acts as an agent on user behalf), registration server (also called registrar, which keeps and made available SIP contact information to other SIP servers), and proxy server2 (which act on behalf of the end-users in forwarding or responding to the SIP requests).

As compared to the wired networks, communication over wireless channels inherently involves dealing with time-varying and stochastic channel conditions and scarcity of resources. Therefore, to deploy SIP in wireless multi-hop environments, we must deal with many technical challenges that have never been faced in wired networks. These new challenges are raised by the inher-

1The SIP architecture may also involves other entities, such as redirect server, which are not directly related to this thesis.

2The registration server and the proxy server can be deployed in the same machine.
ent combination of decentralized wireless infrastructure which impose limited applicability to the standard SIP architecture as registrar and proxy servers are static and centralized entities. Therefore, the mechanism applied by SIP in fixed IP networks to locate the end-users and to map user names to destination IP addresses involving those centralized servers (typically owned by the network operator and located in the access network) may not exist in a wireless multi-hop environment. Hence, the SIP protocol cannot be deployed as is in MANETs or mesh networks.

The session setup time has a direct impact on the users’ satisfaction, who expects to experience the same waiting time even if the technology is different. When SIP is deployed over wireless multi-hop networks, the end-users located in the wireless network can reach other parties located in the Internet (and thus also SIP proxies and registrars) through gateway nodes. But whenever two end-users inside the wireless network need to communicate via SIP, all the signaling must traverse the gateway, that could be located several hops always, in order to reach the SIP servers. Therefore, in such scenario, three factors have a major impact on the performance of session setup time; namely, the multi-hop communication (several hops will lead to higher wireless medium contention, thus leading to higher network delay and loss), the physical channel characteristics (low quality channels should increase the number of frame retransmissions, thus also leading to higher network delay and loss), and the underlying protocols used by SIP. [43] considers SIP as the signalling protocol enabling VoIP and investigates the performance of SIP session setup delay by using an adaptive retransmission timer adjustable to SIP transaction over wireless link communications. In addition to this work, [44] shows that SIP session setup delay depends not only on the average frame error rate (FER), but also on the amount of burstiness in the wireless channel, where the use of UDP instead of TCP can make the session setup shorter in case of higher FER.

According to [45], numerous scenarios require multiple networked devices to be able to communicate with each other without a single point of failure, and it argues that decentralized architectures is often very useful for robustness. Thus, several researches [37–39, 46, 47] have proposed decentralized approaches, by using SIP as a decentralized protocol to establish direct signalling and media session between users. In [39], a framework for conference signalling in ad-hoc network as an extension of SIP is proposed. The objective of this framework is to allow users of ad-hoc networks to communicate with each other and exchange instant messages without the SIP centralized entities. Thereby, it unifies the network layer routing protocol and the application layer SIP, by using SIP register messages to update AODV routes, reducing then the number of transmitted messages in the ad-hoc network and therefore improving bandwidth use and decreasing collision probability.
Complementary solutions, such as SIP multicast, service discovery and peer-to-peer SIP (P2PSIP), are evaluated in [38], [46], [48], respectively. In [38] a middleware framework that works between the application layer and the MANET routing layer is proposed. As exemplified by [46], the support of a service discovery framework is also useful in wireless multi-hop networks to give users the possibility to discover people, services, or devices in the network. Thus by using the service location protocol (SLP) [49], the support for user discovery in decentralized SIP is achieved either by finding out the bindings of users within reach in the ad-hoc network or to discover the IP address of a user by SIP address of record (AOR). In P2PSIP, a SIP system uses a P2P overlay network for management of distributed functions such as user location [48].

Even if the SIP session has been established between two SIP end-users, media communication needs to be transmitted among them. Therefore, the capacity of multi-hop network needs to be known [50]. Packet losses and an increased delay due to interference in a multiple hop network can significantly degrade the end-to-end VoIP call performance. High traffic leads to high medium contention which increases packet loss rates compared to single hop deployments. The existence of potential hidden nodes further intensifies this problem. Moreover, when using VoIP in wireless multi-hop networks the overhead induced by the IEEE 802.11 physical and medium access control layer and the IP/UDP/RTP protocol stack accounts for large portion of the channel utilization time, while the actual 20 byte payload 3 only uses small amounts of it. As a consequence, the voice over IP capacity is very low [53].

To increase the channel utilization efficiency and the capacity several IP packets can be aggregated in one large packet and transmitted at once. The enhancement of the VoIP capacity in multi-hop networks by aggregating packets is studied in [53–57]. While trying to reduce the IEEE 802.11 MAC overhead, different techniques were applied, such as end-to-end, hop-by-hop, and hybrid aggregation schemes. As an example, the proposed accretion (hybrid) aggregation algorithm in [53] proved to increase the number of supported calls with the given quality measured over single-radio single-channel multi-hop networks. In such a scheme, the aggregation is done at the ingress node for all flows routed to a common destination. The medium access queuing delay of intermediate nodes is used for a further aggregation without imposing an extra delay to the packets. In addition, header compression schemes such as robust header compression (ROHC) are presented in [53] and [58] as a complementary technique to aggregation, while increasing VoIP capacity over wireless multi-hop networks. However, the size of the aggregation packets is a very important performance factor, since too small packets yield poor aggregation efficiency and too large packets are likely to get dropped when the channel quality is

3Using G.729 codec [51, 52]
2.3 P2P computing in WMNs

Recently, applications based on the peer-to-peer (P2P) communication paradigm are increasing in popularity. Examples are popular file-sharing applications (e.g., Kazaa [59], Gnutella [60]), upcoming P2PSIP solutions for Voice over IP, or P2P video streaming that use P2P techniques to form an overlay on top of existing networks. P2P computing refers to technology that enables two or more peers to collaborate spontaneously in a network of equals (peers) by using appropriate information and communication systems without the necessity for central coordination. In that sense, P2P networks are overlay networks typically operated on infrastructure (wired) networks, such as the Internet. However, the P2P overlay network is dynamic, where peers come and go (i.e., leave and join the group) for sharing files and data through direct exchange.

There are numerous P2P overlay networks proposed with very different architectures and protocols. The architectures for P2P overlays can be categorized into two main classes: unstructured P2P overlays and structured P2P overlays. Unstructured overlays do not impose a rigid relation between the overlay topology and where resources or their indices are stored. This has a number of advantages like; easy implementation and simplicity, supporting dynamic environments and keyword search (instead of exact match queries). But the major drawback of such overlay is scalability problem. Search operation for a resource may take a long time and consume network resources extensively, since most of the time there is no relation between the name of resources and their locations. Well-known examples of unstructured P2P are BitTorrent [61] and Gnutella [60].

To overcome the scalability issues of the unstructured approach, structured P2P networks have been proposed, where the P2P overlay network topology is tightly controlled. Content is placed not at random peers but rather at specified locations that will make subsequent queries more efficient. Very popular representatives of structured P2P networks are realized through the so called Distributed Hash Tables (DHTs), such as CAN, Chord, Pastry, and Bamboo [62–65]. At the core of each DHT lies the ability to route a packet based on a key, towards the node in the network that is currently responsible for the packet’s key. This process is referred to as indirect or key-based routing. This structure enables DHTs to introduce an upper bound on the number of overlay hops towards the node currently responsible for the packet’s key. This upper bound is commonly $O(\log n)$, with $n$ being the number of nodes in the network. This bound is achieved through routing strategies employed by the respective DHTs. Those strategies include reducing the Euclidean distance in the over-
lay ID space to the destination in each overlay routing step (e.g., CAN [62]), halving the numerical distance to the destination in each routing step (e.g., Chord [63]), or increasing the length of the matching prefix/suffix between the current node’s overlay ID and the key in each overlay routing step (e.g., Pastry [64] and Bamboo [65]).

The P2P communication paradigm will be very important in wireless multi-hop networks as centralized servers might not be available or located in the Internet. Therefore, P2P will be an interesting alternative for decentralizing services or making its own local resources available in the wireless multi-hop network to serve local user communities. Indeed, not only do mobile nodes require content delivery but they also act as content providers. Therefore, mobile users are expected to offer data services in an effective manner, despite the scarcity of bandwidth and the intermittent connectivity due to the highly-dynamic nature of wireless multi-hop networks.

2.3.1 Challenges Deploying P2P Services in WMNs

The P2P overlays designed for the wired Internet rely on the IP routing infrastructure, which is resource rich especially in terms of bandwidth availability. Therefore, high P2P management traffic, e.g. as it is used currently in structured overlay networks to guarantee consistency, will lead to scalability problems when legacy P2P services are used as-is in multi-hop environments. One of the main issues is therefore how to efficiently provide the same kind of P2P services implemented in legacy wired networks in wireless multi-hop networks, and how to enable efficient overlay services and applications on the resource constrained wireless environment. Several approaches, such as [66–70], try to overcome the bandwidth constraints challenge by integrating, or applying cross-layering techniques between the P2P and the WMNs’ routing layer.

While reactions to changes in the routing layer operate on very small timescale, reactions to changes in structured overlay are not so fast. To avoid invalid and inconsistent routes in the overlay, DHT protocols employ maintenance mechanisms to keep the routing tables up to date [63]. Typically, nodes probe their neighboring nodes via periodic ping request and response messages to learn whether they are still available or not. In wireless multi-hop networks, such maintenance traffic can further contribute to congestion and collisions. Moreover, as nodes mobility might lead to topology changes, there might be potential for misrouted messages if the overlay and the routing layer have inconsistent topology information. Also, triggering such maintenance traffic during network rerouting further contributes to network instability. To this end, approaches such as CrossROAD [70], SSR [69] and VRR [68] exploit network routing messages and cache information in order to maintain the P2P overlay consistency.
When DHT protocols are used in a wireless multi-hop environment, resilience is also a very important issue. The resilience of a DHT determines how much time may pass before expensive recovery mechanisms have to be evoked. As the quality of connections in wireless multi-hop networks is highly dependent on the environment and on the nodes mobility, nodes may often become temporarily inaccessible. If the recovery process is started too early, an avoidable overhead is caused if the node becomes accessible again. However, if the topological structure allows the DHT protocol to delay recovery mechanisms without losing routing capability, these costly recovery measures can be avoided and the maintenance costs of a DHT can be significantly reduced. Therefore a compromise between the overlay maintenance and network congestion must be reached [71].

Unlike the P2P overlay in the Internet, where the neighbor is directly reachable using an underlying routing protocol, in the P2P overlay in WMNs, contacting the neighbor may require going through multiple (wireless) hops. When routing to a destination via DHTs, the node resorts to simple greedy routing by selecting the overlay’s neighbor that makes the most progress in the ID space, and then forwarding the packet along the hop-by-hop route. The ratio between the cost of selected route using the overlay’s neighbor to the optimal shortest path routing through the WMNs is defined as the routing stretch metric. Small routing stretch means that the selected route is efficient compared to the shortest path route. This is a key quantitative measure of route quality used by the P2P overlay, and affects global resource consumption, delay, and reliability. Thus, minimizing routing stretch is also a critical issue for a multi-hop environment as both delay and packet loss increase significantly with the growth of the number of hops in the physical path. In order to reduce the routing stretch, [69] proposes to use the source routes in the node’s routing cache in order to prune unnecessarily long routes, e.g. routes containing cycles. In [72], routing stretch is reduced as routing entries are augmented by overhearing data packets and by applying proximity awareness using random landmarking.

3 Research Questions

The research questions for this thesis are threefold as described below.

1. What are the impacts of deploying SIP-based architecture in wireless multi-hop networks, such as internet connected MANETs?

As discussed in section 2.2, the architecture of the session initiation protocol (SIP) is based on centralized entities and therefore imposes sev-
eral challenges when deployed in decentralized environments such as MANETs or mesh networks. In Paper I we present simulation performance which exploit the challenges of establishing SIP-based sessions in internet connected MANETs, and propose alternative approaches. In Paper II, we investigate the alternative approach where the SIP proxy is co-located at the MANET's gateways, and also present results on multiple gateways strategy. We conclude from this question that by applying alternative approaches SIP session establishment can be improved in internet connected MANETs.

2. How to increase VoIP scalability in a wireless multi-hop environment?

This question is investigated in Paper II and Paper III. In Paper II, we start investigating the influence of intra and inter-flow interference in channel utilization by accessing VoIP quality metrics, such as end-to-end delay, jitter and packet loss, in a multiple gateway scenario. We conclude that inter-flow interference combined with the overhead of small VoIP packets leads to higher channel utilization and also an increased number of retransmissions due to MAC data frame collisions resulting in lower VoIP capacity. Therefore, we proposed in Paper III a novel aggregation scheme that increase VoIP capacity by combining several small VoIP packets into larger aggregated ones. However, as larger aggregated packets can lead to higher packet loss for a low signal quality wireless link, our adaptive hop-by-hop aggregation scheme calculates the target aggregation packet size based on the wireless link characteristics. We conclude from this question that the VoIP capacity is an important issue in wireless multi-hop scenario, and the packet aggregation scheme can provide an important capacity improvement in wireless multi-hop environments.

3. How to cope structured P2P and wireless multi-hop networks?

Due to various reasons, such as characteristics of wireless links, multi-hop forwarding operation, and mobility of nodes, performance of traditional P2P applications is rather low in wireless multi-hop networks. Therefore, to answer this question demands the definition of an acceptable trade-off between P2P service performance and management overhead in such networks. In Paper IV we provide various simulation results characterizing the overhead of management and control traffic and give recommendations for performance improvement. We conclude from this question that a balance can be reached between the P2P management and control traffic in the overlay and network congestion in the wireless multi-hop network.
4 Research Method Used

The scientific research method that led to this thesis included four main (re-current) steps. First, the literature review used to get an overview of the field. Second, the theory building composed by problem statement and hypothesis formulation. Third, the theory testing used to verify the claims of our theory. Finally, the reflection on the experiments and results through conclusions [73].

The theory testing can take many forms. Commonly used theory testing in computer science are the analytical and experimental methods. In analytical method, problems are modelled with mathematics and results are derived by formal symbol manipulations. The modelled system’s performance can be predicted under a range of conditions by varying the input parameters of the model. Analytical models generally provide better insight into the effects of various parameters and their interactions. However, it requires many simplifications and assumptions.

In experimental method, problems are modelled by simulation, emulation, and real measurement [74]. Experimental methods are often used when the model is too complex to allow analytical methods. In the same manner as analytical method, a simulation uses an abstract representation of the system. The abstraction is created by a computer program called the simulation tool. Compared to analytical method, it is easier to incorporate more details in the simulation, and, thus, simulations often produce more realistic results. Despite the advantages of simulation, simulators may require high computational complexity which leads to longer simulation time. The aforementioned complexity can be reduced by simplifying assumptions or heuristics into the simulator engine, increasing the lack of accuracy of the simulation results.

During emulation, measurements are performed on a real implementation of a system running on real hardware. However, some aspects of the system are abstracted through an emulation tool. Emulation combines advantages with the simulation (controlled and reproducible environment) and real measurements (more realistic test environment). Emulation can also make use of host virtualization in order to increase scalability [75].

In a real measurement an operational system is studied. One obvious advantage is that since real code is being tested in a real environment, eventual doubts whether the modelled system represents the real systems are prevented, since in this case they are the same. However, when complex systems are tested it is generally hard to produce controlled and reproducible experiments.

For this thesis we mainly used simulation as research method. We used basically the network simulator ns-2 [76]. By using simulation, we were able to
test complex protocols which were difficult using analytical methods. It also allowed comparison under a wider variety of workloads and environments. Real measurements is planned to be carried in the KAUMesh testbed [15] in order to validate the results from our performance evaluation.

5 Contributions

This section summarizes the main contributions of this thesis. They are directly related to the research questions presented in Section 3. The contributions are divided in research and implementation results:

Research Results

- We evaluate the challenges of deploying SIP-based service in internet connected MANETs, and suggest some approaches to it. In particular, we:
  - evaluate the performance of SIP service when centralized proxies/registrars located in the access network are used by MANET's nodes. Four alternative approaches to provide SIP services in such environment are discussed and compared against each other.
  - describe and evaluate the approach where the SIP proxy is co-located at the MANET's gateways. And also present results of the influence of multiple gateways strategy.

- The performance evaluation of VoIP intra and inter-flow interference in internet connected MANETs using multiple gateways, based on the quality of service metrics; end-to-end delay, jitter and packet loss probability.

- The design and performance evaluation of an adaptive hop-by-hop aggregation scheme for VoIP packets in wireless multi-hop networks based on wireless link characteristics.

- The identification of trade-offs when deploying a structured P2P overlay solution over wireless multi-hop networks, the conduction of simulation results characterizing the overhead of P2P management and control traffic in such environment and the proposed recommendation to balance P2P lookup efficiency and management traffic overhead in wireless multi-hop networks.

Implementation Results

- We adapted the ns-2 SIP module developed by [77] to cope with the CMU's Monarch mobility extension of ns-2 (wireless node).
6. Summary of Papers

This section contains short summaries of the papers included in this thesis.

Paper I – SIP based Service Provisioning for hybrid MANETs

The interconnection of Mobile Ad-Hoc Networks (MANETs) to fixed infrastructure based IP networks is very important in order to provide the ubiquitous user internet access anywhere at any time [5, 6]. As a partner institution in EU Daidalos project [80], we were working through the development of protocols and mechanism for connecting MANETs to the public internet. In such scenarios, also known as "internet connected MANET", or "hybrid MANET", the user within an ad-hoc network gets access to the public internet by using the packet forwarding capabilities of intermediate ad-hoc network nodes towards the Access Router (AR) or gateway to the Internet. This gives users the opportunity to gain access to different services through a diversity of interconnected networks (fixed, wireless access and ad-hoc networks). Among the advanced services specified in the project, multimedia communication for hybrid MANETs in terms of Voice over IP (VoIP) service was mainly required.

For this purpose, we started to analyze the applicability of Session Initiation Protocol (SIP) [41] as a way to enable VoIP service in hybrid MANETs. The SIP protocol was chosen as a signaling protocol for establishing and controlling multimedia sessions, by enabling VoIP communication between SIP user agents. Since SIP is based on client/server architecture, typically owned by the network operator, the use of standard SIP architecture in hybrid MANETs presented severe performance limitations due to the SIP centralized entities (e.g. SIP proxies and registrar). In the scenario studied, all SIP signaling exchanged between MANET nodes (or a MANET node and an external node in the internet) need to pass through gateways that connect the MANET to the
The first contribution of this paper is the identification of the performance
limitations of SIP in hybrid MANETs. Even not considering a QoS provisioning
architecture in our study, we deliberately used VoIP quality metrics in terms
of the SIP call setup delay and the call blocking probability. Through our sim-
ulation results we have shown that those metrics are severely degraded as the
background traffic and number of hops to the gateway increase. In face of
avoiding such limitations, the second contribution of this paper is the proposal
of alternative approaches to provide SIP services in hybrid MANETs. The first,
and straight forward, approach was to enable MANET’s gateways with SIP
proxy/registrar functionalities. This approach does not require any modifica-
tion to the SIP standard, however still recall to single point-of-failure of SIP
centralized entities. The other three approaches exploited the decentralized
techniques; distributed SIP, SIP service discovery, and peer-to-peer SIP. The
third contribution was the analysis of impacts of the proposed approaches on
SIP architecture and functionalities. For each approach, we also described the
necessary modifications to MANET’s node and gateway architectures.

If SIP is used over TCP or SCTP, no retransmission mechanism is used at
the application layer, however the total session setup delay may increase as
it is the addition of the setup time of the TCP or SCTP session and the suc-
cessful transmission time of all the SIP messages necessary to open a session.
According to [44], the session setup delay for TCP tends to overpass the one for
UDP thus in our study the UDP was used as the SIP underlying protocol. The
use of a simplified two ray ground propagation model introduced limitations to
our scenarios, as the additional impact of the physical channel characteristics,
such as low quality channels, in the performance of session setup time is not
analyzed. With respect to the topology chosen, our study aimed at identifying
the SIP performance limitations in a wireless multi-hop network with a varied
number of hops among source and destination. For example, in the experiment
with 32 voice background traffic, where source and destination are 2 hops away
from each other, we observed that only 25% of new SIP session setups are com-
pleted within 5 seconds, as the high channel contention provokes SIP timeout
and many SIP messages will be dropped. Despite the simulation results pre-
sented, SIP retransmission strategies are not considered in this work.

Our work did not aim to specify in detail architectural changes required to
optimize SIP-based service provisioning in hybrid MANETs. Rather, it gave
a rough overview on what would need to be changed in order to integrate the
proposed research alternative into the EU Daidalos architecture. Also more
analysis is required with respect to architectural issues in relation to full IMS
(IP Multimedia Subsystem) capabilities, such as the IMS registration and ses-
Paper II – Challenges of SIP in internet connected MANETs

This paper is a continuation of the work carried out in Paper I. Given the performance limitations of standard SIP in hybrid MANETs, we verified the necessity to compare it against the alternative approaches. As an initial step, we selected the first alternative approach, where the gateway implements SIP proxy/registrar functionalities, not requiring SIP architecture modification and guarantees SIP interoperability. Thus, in this scenario, by using a modified gateway discovery mechanism, the MANET's nodes are not just informed about internet connectivity but also the SIP service capability of the gateway.

In this paper we extended the scenario analyzed in Paper I by introducing a new set of SIP traffic patterns. Here, we compared the VoIP quality metrics against different background flows, while now the SIP session traffics could be directed to nodes inside the MANET, in the internet, or in both networks with different probabilities. In addition, we also verified the influence of having multiple gateways in the hybrid MANET scenario. Therefore, the first contribution of this paper is the performance evaluation of the proposed approach against the standard SIP approach in a hybrid MANET. The results show that by having gateways with SIP proxy/registrar functionalities, the extra delay imposed to SIP messages to reach the SIP servers in the access network is avoided. Thus a reduction of SIP call setup delay can be achieved by, for example, 42.0% in the case where all SIP session flows are directed to the internet. The second contribution is a performance analysis of network capacity in terms of average end-to-end delay, jitter and packet loss for the VoIP flows in the hybrid MANET scenario studied. The results show the impact of intra and inter-flow interference as the number of VoIP flows increases. We identified that MAC layer collisions (due to high network contention), packet dropped at routing layer (due to no route available) and packet dropped at node's queue (due to no queue space) are the three main reasons for packet loss. The results also show the influence of multiple gateways in the SIP call setup delay and the packet loss rate.

As in Paper I, the use of a simplified two ray ground propagation model introduced limitations to our scenarios because the additional impact of channel characteristics on session setup time was not analyzed. By using the SIP proxy/registrar capability at MANET gateways, mobile nodes have additional freedom to choose a proper gateway. In our work, the shortest path to the gateway is the metric used by MANET's nodes during the gateway selection process. Other metrics such as gateway's load or path's expected transmission time could further enrich the gateway selection process. The integration of the
proposed approach into 4G networks architecture is left as future work.

**Paper III – VoIP Packet Aggregation on Link Quality Metric for Multihop Wireless Mesh Networks**

During the last years VoIP over wireless networks is gaining momentum due to the popularity of applications such as Skype and higher WLAN availability. However, the network capacity of wireless multi-hop networks is still a challenging topic. From the results obtained in Paper I and II, it became obvious that the 802.11 MAC layer access method DCF, commonly deployed in wireless multi-hop networks, suffers highly from intra and inter-flow interference in multi-hop packet forwarding scheme. While a multi-radio multi-channel solution could minimize such problem, we noted that the MAC and physical layer overhead of 802.11 networks is still high for the transmission of small packets such as packetized voice samples.

In order to avoid such overhead, we proposed in this paper a new adaptive hop-by-hop packet aggregation scheme. It seemed straightforward that by aggregating small packets the overall number of packets in the network is reduced while minimizing multi-hop contention and packet loss due to collisions. However, such larger aggregated packets lead to higher packet loss for a link that operates at low signal quality. We noted that for such links, aggregating fewer packets could be beneficial. While such aggregation mechanism have been proposed for a single-hop infrastructure WLAN, designing an aggregation strategy for wireless multi-hop networks is a hard problem, since the sender node usually does not have complete knowledge of all link characteristics. Therefore, our proposed aggregation scheme calculates the target packet aggregation size per each hop based on wireless link characteristics information exchanged among neighboring nodes.

In our proposed scheme, every node participating in the wireless multi-hop network has capability to aggregate or de-aggregate packets. The wireless link characteristic was given by the mapping of signal-to-noise ratio to frame loss probability for different frame sizes. For example, with such mapping, we could obtain the optimum packet size for a given signal-to-noise ratio under a certain packet loss rate. Signal-to-noise ratio was constantly measured for each link or neighbor by the receiver node, and its moving average was informed to the sender node via beacon messages (e.g. hello packets). Different parameters were proposed in order to control the trade-off between loss due to frame error plus medium contention and the forced delay imposed by the aggregation. The results showed that by using adaptive aggregation led to increase in VoIP capacity in terms of additional VoIP flows accepted within a QoS boundaries. For the topology studied, the adaptive aggregation led to the increase of capacity
6. Summary of Papers

in order of 200% compared to no aggregation.

The algorithm we have developed adopts a simulation approach and VoIP application was the only service considered. It is true that different services tolerate different packet loss rate. Therefore different application requirements and network topology should be used to determine the correct packet loss suitable for a given application. The signal-to-noise ratio (SNR) is a function of peak signal strength to noise. In the simulation, we augmented signal-to-noise ratio estimation by considering SNR of corrupted packets. It is also important to note that our proposed schemes are not dependent on any specific routing protocol.

**Paper IV – Performance Evaluation of Structured P2P over Wireless Multi-hop Networks**

In this paper we raised the question: How is the performance of peer-to-peer applications in wireless multi-hop networks? Our starting point is the observation that many peer-to-peer based applications over the internet are increasing in popularity and their applicability in wireless multi-hop environments will be necessary. Thus, in our scenario we evaluated a resource lookup application through a structured P2P overlay in wireless multi-hop networks. As an example, the resource lookup application can be used to leverage P2PSIP where the P2P overlay manages the SIP user location.

The structured P2P overlay network was realized through a distributed hash table (DHT), where the topology is tightly controlled and an upper bound on the number of overlay hops towards the node responsible for the requested information is guaranteed. To avoid invalid and inconsistent routes in the overlay, the DHT protocol commonly employs maintenance traffic to keep routing tables up to date. As we have seen through simulation, high maintenance traffic led to scalability problems and congestion in the wireless multi-hop network.

The first contribution of this paper is the characterization of the P2P management overhead and control traffic of DHTs in wireless multi-hop networks. We evaluated different overlay configurations for the DHT while verifying important metrics such as success lookup probability, lookup delay distribution, and overhead introduced by the P2P management and lookup requests. The results showed that the lookup efficiency, translated as the ability to find destination nodes which are responsible for the specific key information, can be severely degraded as the P2P management overhead increases network contention. As verified through the results, too frequent management traffic led to high overhead in the multi-hop environment and thus network congestion. On the other hand, no management led to low lookup efficiency as node's effective
connectivity forced it to choose suboptimal routes though the overlay. Therefore, the second contribution of this paper is the trade-off suggested between the P2P lookup efficiency and management traffic overhead while deploying structured P2P over wireless multi-hop networks.

In a peer-to-peer network, the process of nodes leaving and joining the overlay network is known as churn process. In our evaluation, the churn process is not realized through join/leave node’s process but through up/down link connectivity, as the wireless links are subjected to higher contention as management traffic and network topology increases. Thus, in our scenarios, a temporary loss of a routing neighbor weakens the correctness and performance guarantees of the DHT. Node mobility can also augment the churn process, however it is not considered in our evaluation. Since this paper is our first try through the realization of P2P applications over wireless multi-hop networks, we have not explored here cross-layer possibilities such as proximity neighbor selection or P2P overlay and routing layer interactions.

7 Conclusions and Future Work

Multi-hop communication for wireless systems has been attracting significant attention in the research community. Unlike centralized point-to-point communication, multi-hop communication offers tremendous advantages such as allowing users to share resources through distributed transmission and processing. Since wireless multi-hop networks can be deployed rapidly and flexibly, it is attractive to numerous potential internet-based applications, ranging from multi-hop wireless broadband internet access to multimedia services such as Voice over IP, and P2P applications. In this thesis we investigated how two different applications can be influenced by the characteristics of wireless multi-hop networks and how the applications and the network can be adapted to provide scalable communication.

In the first part of the thesis we considered the issues introduced by wireless multi-hop characteristics while deploying voice over IP communication. We have shown that VoIP quality metrics, such as SIP call setup delay and call blocking probability, are severely degraded as the background traffic and number of hops to the gateway increase. To avoid such limitations, we have proposed four alternative approaches. Since network capacity is also a key issue to provide scalable services, in the second part we have focused on the VoIP capacity over wireless multi-hop networks. Our performance evaluation has shown that the number of gateways and traffic flows have important impact over the VoIP capacity. Therefore, in order to avoid the small VoIP packet overhead, we have proposed a new adaptive hop-by-hop packet aggregation scheme.
based on wireless link characteristics. Our performance evaluation has shown
that while considering wireless link quality the VoIP capacity over wireless
multi-hop networks can be increased by a two-fold gain for the scenario studied. Finally, in the third part we investigated the applicability of structured
peer-to-peer overlay in wireless multi-hop networks by analyzing the trade-off
between the P2P lookup efficiency and management traffic overhead.

In future work, the short term activity is a testbed evaluation of VoIP ca-
pacity through a multi-radio multi-channel wireless multi-hop network. Here,
the idea is to verify the impact of different channel assignment and scheduling
techniques to improve VoIP capacity. We also noted the importance of analyz-
ing the influence of packet aggregation in such scenario. Another short term
research activity is the optimization of the mapping between wireless multi-
hop network topology and the DHT overlay. In order to integrate WMNs and
DHT overlay in a system with higher performance, the goal of these activities
are to better understand the behavior of DHTs in WMNs and to model the in-
fuence of important parameters (i.e., cache size, cache lifetime, and number of
distinct shortcuts in the logical space) of those systems.

In the long term, we plan to audaciously investigate the integration of
WMNs and P2P overlay through a P2P traffic aware channel assignment al-
gorithm that ensures channel assignment according to P2P traffic demands.
The basis for a P2P traffic aware channel assignment scheme is the use of the
P2P overlay information and continuous monitoring of certain node specific pa-
rameters at the mesh routers (such as channel utilization, traffic demands) to
derive the best channel assignment strategy. Based on those information data,
channels can be re-assigned dynamically to balance P2P traffic demand across
the channels and thus utilize all available spectrum more efficiently.

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SIP based Service Provisioning for hybrid MANETs

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SIP based Service Provisioning for hybrid MANETs

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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.

1 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 2 we present the problems related to the deployment of standard SIP approach in internet connected MANETs, where performance simulation is presented in Section 3. Section 4 describes alternative approaches, exploring advantages and disadvantages of each one. Section 5 concludes this paper and discusses future work.

2 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 $T_r(1)$ seconds if no response is received, and the timer $T_r(1)$ doubles after each retransmission following an exponential backoff behavior. $T_r(1)$ 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.

3 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 [5] 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 [6]. 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 [8]. 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.

Figure 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 $T_r(1)$ expires, it will be retransmitted 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.

As can be seen from Figure 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 Figure 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
3. Performance Evaluation of Standard SIP Architecture in Internet connected MANETs

32 background flows have been established, resulting in a blocking probability of 75%, which is high for voice communication systems [10].

In order to evaluate the reasons for the high SIP call setup delay, we plotted in Figure 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). Figure 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.

4 SIP based Service Provisioning for Internet Connected MANETs

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

4.1 SIP Proxy/Registrar co-located at Gateways

As discussed in Section 2, 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 ad-
dress 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 [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 con-
tinuity [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 gate-
way 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 gate-
way 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-

Figure 3: Invitation Attempts versus network load for 1 gateway and hops
GW_INFO message format extension proposed for this approach does not impose more overhead to 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. Figure 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 Figure 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 Figure 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 Figure 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].
4.2 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.

4.3 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 4.2). 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].

4.4 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 located.
tacted. 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).

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.

### 4.5 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 4.1, 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 convey this information to the SIP stack.

Distributed SIP and Integration with routing protocols (Section 4.2), Integration of SIP with Service Discovery Frameworks (Section 4.3) and Peer to Peer SIP (Section 4.4) 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
REFERENCES

SIP.

5 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.

6 Acknowledgment

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References


<table>
<thead>
<tr>
<th>Parameters</th>
<th>Proxy-based</th>
<th>Distributed SIP</th>
<th>Service Discovery</th>
<th>P2P SIP with DHT</th>
</tr>
</thead>
<tbody>
<tr>
<td>MN SIP Stack</td>
<td>Standard SIP Stack is sufficient</td>
<td>A SIP Proxy/Registrar module needs to be installed in some MNs</td>
<td>A Service Discovery module and SIP Proxy/Registrar module needs to be installed in MNs</td>
<td>DHT module needs to be installed in MNs. Extension to SIP REGISTER messages are required to transport DHT</td>
</tr>
<tr>
<td>GW Discovery</td>
<td>Extension needed (insert bit &quot;P&quot; in GW_INFO message)</td>
<td>Standard GW Discovery is sufficient</td>
<td>Standard GW Discovery is sufficient</td>
<td>Standard GW Discovery is sufficient</td>
</tr>
<tr>
<td>Additional GW Functionalities</td>
<td>GW must implement SIP Proxy/Registrar functionality</td>
<td>GW must implement SIP Proxy/Registrar functionality</td>
<td>GW must implement SIP Proxy/Registrar functionality</td>
<td>GW must implement Peer-to-Peer SIP Proxy/Registrar</td>
</tr>
<tr>
<td>Additional Interfaces Required</td>
<td>Interface between GW discovery module and SIP client module in each MN, Interface between GW discovery module and SIP Proxy/Registrar module at GWs</td>
<td>Interface between MANET Routing Protocol module and SIP Proxy/Registrar module at MNs and GWs</td>
<td>Interface between Service Discovery module and SIP Proxy/Registrar module at MNs and GWs</td>
<td>Interface between DHT module and SIP client module at MNs, Interface between DHT module and SIP Proxy/Registrar module at GWs</td>
</tr>
</tbody>
</table>

Table 1: Comparison of proposed approaches
Challenges of SIP in internet connected MANETs

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Challenges of SIP in internet connected MANETs

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Abstract
Mobile Ad Hoc networking has been considered as one of the most important technologies to support future Ubiquitous and Pervasive Computing Scenarios and internet connected MANETs will be an integral part of future wireless networks. For providing multimedia services and 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 study the implications of using standard SIP architecture in internet connected MANETs. We analyze performance limitations of SIP service scalability when centralized proxies/registrars located in the access network are used by MANET nodes. We also present and compare an alternative approach to provide SIP services in internet connected MANETs in order to minimize the impact of such performance limitations.

1 Introduction

Two major trends in technology can be observed over the last decade: pervasive communication and the Internet. Pervasive communication is slowly but surely becoming an integral part of our daily life. The Internet has also become a central aspect for almost everybody, both in business, at home or during education. It is thus essential for us to bridge those two technologies, where future wireless communication systems are expected to provide a broad range of multimedia services. MANETs are reaching a stage where they can support these services in order to provide an infrastructure useful for the common user. Thus, the interconnection of MANETs to fixed infrastructure based IP networks is very important in order to provide the ubiquitous user internet access anywhere at any time [1, 2]. In such scenarios, also known as "internet connected MANET", or "hybrid MANET", the user within an ad hoc network will get access to the public internet by using the packet forwarding capabilities of intermediate ad hoc network nodes towards the Access Router (AR) or
gateway to the Internet, providing the user with access to different services through a diversity of interconnected networks (fixed, wireless and ad hoc networks). To offer advanced services, such as multimedia communication for hybrid MANETs, the support of Voice over IP (VoIP) is required.

One important part of VoIP technology is the Session Initiation Protocol (SIP) [3], which is a signaling protocol for establishing and controlling multimedia sessions. SIP is a client/server based protocol, and user agent server (UAS) needs to be reachable on the network so that SIP systems work properly. As MANETs are dynamic networks formed by peer nodes, standard SIP architecture has its problems as registrars and proxies are fixed, static and centralized, normally located in the access network.

This paper addresses scalability limitations for VoIP services in internet connected MANETs. We present and analyze performance of the approach where the SIP proxy is located in the access network. We compare this result with the alternative approach where the proxy is co-located at the MANET gateways, which significantly reduces call setup latency for successful calls. Also, the paper presents results on multiple gateway strategy and capacity limitations of voice traffic in such environment. The paper is structured as follows. In Section 2 we present a brief introduction to SIP and its standard operation in internet connected MANETs. In Section 3 we describe our alternative approach and compare it with standard SIP approach. Section 4 presents simulation results, and Section 5 concludes this paper and discusses future work.

2 SIP Service in Internet Connected MANETs

SIP architecture [3] is based on centralized entities: registrar and proxy servers. With registrars, SIP users register their contact information once they connect to the SIP network. In the registration scenario, a SIP user agent 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 IP addresses where the user is reachable. Proxy servers, that can be co-located in the same element as registrar server 4, are also needed because SIP users do not know, in most cases, the complete contact information of the callee but only its AOR. Therefore, SIP proxy server acts on behalf of user agents forwarding or responding to the request received from another user agent or proxy.

Figure 1 shows a typical SIP message exchange between UAs and proxy

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4Throughout this paper we consider SIP Proxy server and SIP Registrar servers located at the same device, as shown in Figure 2.
server, in order to locate the appropriate service or the endpoint for media exchange. User A calls user B using user B's SIP URI. As shown in Figure 1, INVITE (message 1-2) is used to request establishing a session between user A and user B. The node which receives an INVITE returns a provisional response 100 Trying (message 3-4) immediately indicating receipt of the INVITE and call progress. When user B receives an INVITE, it checks and confirms parameters such as codecs. If user B decides that the parameters are appropriate, it returns a response 180 Ringing (message 5-6). When user B picks up the phone, it sends a response 200 OK (message 7-8). Finally user A receives the 200 OK and returns an ACK (message 9) to user B. Then the session is established and 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 10-11) and 200 OK (messages 12-13).

Therefore, the mechanism applied in fixed IP networks to locate the SIP endpoint and to map SIP user name to destination IP address involves centralized servers [3] that may not exist in a MANET. Hence, 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 connecting the MANET with the public internet.

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 \( T_r(1) \) seconds if no response is received, and the timer \( T_r(1) \) doubles after each retransmission following an exponential backoff behavior. It is an estimate of the round-trip time with default value of 500ms [3]. 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.

3 Optimizing SIP Service Provisioning in Internet Connected MANETs

Using standard SIP architecture where registrar and proxies are located in the access network, 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 (see Figure 2). Instead of using concepts of peer-to-peer SIP [4, 5] or service discovery frameworks [6] where SIP architecture modifications are necessary and interoperability with standard SIP is not truly guaranteed, we propose an alternative approach where MANET gateways have limited SIP proxy/registrar capabilities, offering SIP services to MANET nodes [7]. Our approach optimizes SIP services in internet connected MANETs without any modification on standard SIP architecture which guarantees SIP service interoperability. Gateways with SIP proxy/registrar functionalities can act on behalf of MANET nodes.

Figure 2 illustrates our approach, where gateway (AR1) operates as a SIP proxy/registrar server for the internet connected MANET. All control traffic (1. + 3. REGISTER, 2. + 4. INVITE, etc.) is exchanged between MANET nodes (circles) and gateways (in the example AR1) and media traffic (green arrow) is exchanged directly between MANET nodes (or between a MANET node and a node in the internet).

Instead of using pre-configured SIP outbound proxy server IP address in every MANET node or DHCP to discover it, we propose the use of gateway discovery mechanisms [7] to inform SIP user agent clients inside the MANET about the existence of gateways with SIP proxy/registrar functionalities. 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 suggested by
Jelger [8], where the MANET is virtually divided into as many subnets as there are gateways, allows to deploy proxy/registrar functionalities co-located with each gateway. In order to inform MANET nodes about gateways with SIP proxy/registrar functionalities, we propose the modification of GW_INFO message (as shown in Figure 3) through the insertion of the bit "P" in the reserved field which indicates gateway capability to act as a SIP proxy/registrar. This does not increase overhead but mobile nodes have now additional freedom to choose a proper gateway based on its SIP proxy capabilities. 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 5.

Service differentiation can be applied to prioritize SIP signaling and voice packets in hybrid MANET when voice traffic coexists with data traffic. The IEEE 802.11e standard allows service differentiation using multiple queues within the mobile stations. Prioritized channel access is provided for flows with different priorities finding the right balance between prioritizing SIP signaling, voice packets and routing traffic which is not addressed in this paper.

5http://www.ist-daidalos.org/
4 Performance Results of SIP in Internet Connected MANETs

4.1 Simulation Description

In order to analyze performance of the proposed approach we show in this section results based on the usage of SIP proxy/registrar located at the gateway, and compare it with the standard approach. 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 10x10 nodes are placed on a grid at a distance of 200 m with 250m of transmission range and 500m of carrier sense range using two ray ground as radio propagation model. The parameters used in the simulation are summarized in Table 1.

AODV-UU routing protocol [9] was adopted, enabling the use of multiple gateways in our scenario (see section 4.3). In AODV-UU, nodes discover routes on demand. If the destination is located in the internet, gateways respond with a special proxy RREP. Mobile nodes then use RREQ/RREP phase create tunnels towards the gateway for all traffic destined to the Internet. MANET gateways are connected to the fixed network and offer wireless ad hoc connectivity and SIP services to MANET nodes, through the support of SIP proxy/registrar capability in the gateway. MANET users try to establish calls to other MANET users or to users in the Internet using SIP messages. We measure average SIP call setup delay and SIP call blocking probability under different background traffic and different number of gateways. SIP call setup delay was measured as the time between user agent sends an INVITE request (message 1 in Figure 1) until the time it gets the 200 OK response (message 8 in Figure 1). SIP call blocking probability indicates the percentage of SIP sessions that were not established within 5 seconds, as this is the maximum time permitted to interwork with ISDN [10]. Voice calls are also used to model the network load. We use G.729 voice codec and model it as bi-directional exponential distributed
Table 1: Simulations parameters used in ns-2

<table>
<thead>
<tr>
<th>Parameters</th>
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<tr>
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<td>Grid Space</td>
<td>2000x2000 m</td>
</tr>
<tr>
<td>Routing Protocol</td>
<td>AODV-UU [9]</td>
</tr>
<tr>
<td>Hello Interval</td>
<td>1 s</td>
</tr>
<tr>
<td>Number of gateways scenarios</td>
<td>3 (1, 2, 4 gateways)</td>
</tr>
<tr>
<td>Number of Wireless Nodes</td>
<td>100</td>
</tr>
<tr>
<td>Number of Wired Nodes</td>
<td>40</td>
</tr>
<tr>
<td>Wired Links</td>
<td>5Mbps and 2ms</td>
</tr>
<tr>
<td>Traffic Model</td>
<td>Exponential ON-OFF (350ms-650ms) G.729</td>
</tr>
<tr>
<td>Application</td>
<td>SIP [12]</td>
</tr>
<tr>
<td>Simulation time</td>
<td>180 s</td>
</tr>
</tbody>
</table>

traffic with mean values for talk/silent time of 350ms/650ms [11].

The total simulation time was 185 seconds, where background traffic duration was 180 seconds. We assume that 100% of voice sources are inside the MANET. For voice sinks, 75% are located inside the MANET and 25% in the Internet. Source and destination was picked manually and the distance between two nodes (or a node and the gateway) was 4 hops. As an example, for 24 background voice calls, 24 sources are MANET nodes, 6 sinks are outside the MANET and 18 sinks are within the MANET.

Results in terms of average end-to-end delay, average jitter, and packet loss probability for background voice calls are also presented in order to estimate voice capacity in hybrid MANETs.

4.2 SIP Proxy co-located at GWs versus SIP Proxy at ANs

In this simulation, we picked source and destination for SIP call setup attempts manually with 2 hop distance between them and the gateway. To calculate SIP
call setup delay, we assume that all callers are located inside the MANET and callees could be inside the MANET (Out-In: 0%-100%), outside the MANET (Out-In: 100%-0%), or in both networks with two different probabilities (Out-In:25%-75% and Out-In:75%-25%).

Figure 4 shows the average SIP call setup delay as a function of background voice flows. Increasing the number of background voice calls increase the SIP call setup delay of new call attempts, and the worst case is where all nodes (callers and callees) are inside the MANET (Out-In: 0%-100%), because every SIP message has to traverse the gateway twice to reach the proxy which means that the delay between the gateway and the proxy influences the SIP signaling delay. Here, the MANET becomes more and more congested as the number of background voice calls increases. However, as all SIP signaling must pass through the gateway, the average time to establish a new SIP call can reach more than 25 seconds under 48 background voice calls.

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 a timer 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 overhead. Also, SIP messages compete at the MANET gateway with background voice traffic for buffer space and might be dropped. But as the SIP proxy is located at the gateway, the delay imposed to SIP messages to reach the proxy in the access network is avoided. Finally,
the SIP processing delay at the proxy also contributes to the overall SIP setup delay.

Figure 5 shows the SIP call blocking probability. As the number of background voice calls increases, the call blocking probability of the new SIP calls also increases. Even if we choose the best case (Out-In:100%-0%) where all callees are outside the MANET, only 55% of the new SIP calls are completed within 5 seconds when 48 background voice calls have been established, resulting in a blocking probability of 45%, which is high for voice communication systems [13].

Figure 6 compares the gain of the proposed alternative where SIP proxy is located at gateway (GW) with the standard approach where SIP proxy is located in the access network (AN). We plot the SIP call setup delay for the best (Out-In: 100%-0%) and worst case (Out-In: 0%-100%) and we are only interested in successfully established calls, which means all calls established in less than 5 seconds. For the standard approach, we use a delay of 150ms between gateway and SIP proxy. Our proposed approach improves SIP call setup delay significantly in comparison to standard approach for all number of background voice calls. Taking 8 background voice flows as an example, in average SIP calls are established with 32.8% less time for the worst and 42% less for the best case.

4.3 Multiple gateways

SIP signaling messages compete with background traffic in the MANET, thus contributing to the SIP call setup delay. To analyze this factor we expanded our
set of simulations increasing the number of gateways in the hybrid MANET to 2 and 4 gateways, also varying the number of hops between source and sink for the background traffic. MANET gateways (red circles in Figure 7) were distributed uniformly along the grid, where the green circles represent the wireless nodes and the blue ones the wired nodes. All the lines represent the wired connections between the nodes.

Figure 8 shows the cumulative distribution function (CDF) for SIP call setup delay for 24 background flows and different number of gateways (as presented in Figure 7). The contention at access layer can be decreased through
4. Performance Results of SIP in Internet Connected MANETs

Figure 8: SIP call setup time for each attempt for 24 background voice calls

the use of additional gateways. Hence, increasing the density of gateways in hybrid MANETs also influences the overall SIP call setup delay. Analyzing Figure 8, we note that less than 40% of all SIP call attempts generated are established in 5 seconds for 1 gateway, and using 2 and 4 gateways in the same scenario we increase the number of SIP call attempts been established within the same time to 70% and 95%, respectively. Also, when using only 1 GW more than 70% of call attempts require 20 s or more to complete, where if we use 2 GWs only 5% of calls need so long to be established.

Figure 9 shows results for SIP call setup delay when the number of hops between source and sink increases from 2 to 7 for different number of gateways in 24 background traffic scenarios. In this case, the increase of number of hops for background traffic leads to the increase in channel utilization and contention for the wireless medium, and the increase in the number of retransmission due to Medium Access Control (MAC) data frame collisions which in turn also increases channel utilization.

4.4 VoIP capacity

In order to analyze the performance of the background traffic and thus the network capacity for VoIP services, Figure 10 shows the average end-to-end delay and average jitter, while Figure 11 presents the packet loss probability for 1 gateway. On this scenario, all background traffic traverses 4 hops from source
62 Challenges of SIP in internet connected MANETs

Figure 9: SIP call setup delay versus number of hops for 24 background calls

to sink or to the gateway. Inter-flow interference increases which results in increased channel utilization and also an increased number of retransmissions due to MAC data frame collisions, as the number of background voice traffic increases. Therefore, idling time of the nodes waiting for the other nodes to finish their transmission and the time spent in exponential backoffs increases leading to a reduction in transmission efficiency. Retransmission also adds to the delay for packet transmission and since channel resources are used for retransmission, packets have to wait longer in the network queues, also contributing to increased jitter. As illustrated in Figure 10, the delay of ongoing calls seems to be prohibitive for 48 background voice calls, where end-to-end delay reaches 1 second in average, accompanied also by the average jitter, which in this case reaches 1.46 seconds.

We identified three main reasons for packet loss, (1) MAC layer contention leads to high number of collisions (2) packets dropped at routing layer due to no route available (3) packet dropped due to no more space in queue. The reason (2) is a signal of increased congestion as our nodes were static and routed were always available. However, if MAC layer retransmission is exceeded route error message is triggered even if there is a route available. When the packet delays become too large, packets will be also dropped at the network queues, causing degradation of voice calls and respectively dropping the voice capacity. As shown in Figure 11, the results for 24 and 48 background voice calls appear to be impractical for real voice conversation, as packet loss probability reaches 26% and 70%, respectively. For the given voice codec G.729 it is recommended that the total packet loss is below 8% at a maximum delay of 125 ms [14] in
order to achieve a R-score of more than 60% for medium voice quality. For the given network topology, we can thus support not more than 8 calls with one gateway. With 8 parallel calls, the average delay is 17ms at a jitter of 98ms and packet loss probability of 7.93%. However, at 8 background calls, the SIP call setup delay is between 520 ms for the best (Out-In:100%-0%) and 1.2 second for the worst case (Out-In: 0%-100%) as can be seen from Figure 4, if the SIP proxy at the gateway is used.

Figure 12 presents the influence of multiple gateways on packet loss rate for scenario with 8 background calls. It also shows the influence of number of hops between source and sink for the background traffics in PLR. For all number of hops, the strategies of multiple gateways reduce network contention hence less packets are dropped due to network congestion. As can be seen from Figure 9, the increase of number of hops between source and sink increases channel utilization and network contention leading to the increase of packet loss rate.

5 Conclusion

As wireless communication technologies become mature, providing multimedia services to mobile nodes is crucial. Also, the widespread use of VoIP services is likely to play a key role in successful deployment of IP-based convergence of wireless networks. In this paper, we focus on the limitation of standard SIP architecture in order to provide SIP services to internet connected MANETs. We presented performance results of two approaches for VoIP systems in internet connected MANETs where SIP proxy/registrar servers are located in the access network or co-located at the MANET gateway. The enhancements
presented show that for MANET nodes to enable SIP communication with internet nodes, MANET gateways with SIP proxy functionality are beneficial, reducing around 40% SIP latency when network capacity is reached. Our study demonstrates that in static hybrid MANETs, VoIP services can be provided but capacity is limited around 8 calls over 4 hops if one gateway is deployed.

Our future work will concentrate on evaluating the limiting factors in detail, such as reasons for high packet loss and delay for the background voice traffic, impact of deploying multiple gateways on network capacity and providing support for mobility of users. Also, we will look into other factors that determine the capacity such as the impact of routing metrics and gateway selection strategies.

6 Acknowledgment

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Figure 12: Packet loss rate versus number of hops in a multiple gateway scenario

References


VoIP Packet Aggregation on Link Quality Metric for Multihop Wireless Mesh Networks

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VoIP Packet Aggregation based on Link Quality Metric for Multihop Wireless Mesh Networks

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Abstract

Providing Voice over IP in WLAN based wireless meshed networks is an important service for the future wireless internet. However, the transmission of small (voice) packets imposes high overhead which leads to low capacity for VoIP over 802.11 based multihop meshed networks. By combining several small packets into larger aggregated ones, it is expected that overhead can be significantly reduced thus incapacity for VoIP services. When mesh relay nodes aggregate small packets, there is an inherent trade-off regarding packet size. Aggregating more packets leads to larger packets, reduces the overall number of packets in the mesh and thus reduces multi-hop contention and packet loss due to collisions. However, such larger aggregated packets can lead to higher packet loss for a link that operates at low signal quality. For such links, aggregating fewer packets can be beneficial. Classical aggregation strategies typically operate end-to-end without considering those aspects and are therefore suboptimal. In this paper, we propose a new adaptive hop-by-hop aggregation scheme that calculates target aggregation sizes per each hop based on wireless link characteristics. Simulation results show that our approach is very effective, outperforms static aggregation schemes in various performance parameters and increases capacity for VoIP in meshed networks significantly.

1 Introduction

Voice over IP (VoIP) over Wireless is gaining momentum due to the popularity of applications such as Skype and higher WLAN availability. Increasing the popularity of such wireless VoIP requires, among other things, ubiquitous WLAN coverage. To this extent, 802.11s based multihop WLAN meshed networks (WMNs) are a viable and inexpensive solution. In WMNs, wireless access points communicate with each other wirelessly forming a true wireless,
mesh based access network of mesh relay nodes (MRN). Mesh gateways (MG) provide internet connectivity and standard mobile clients attach to MRNs, which forward packets via other MRNs to other meshed clients or through MGs to the internet. Therefore, the wireless backbone comprised of MRNs and MGs is similar to static, internet connected Ad Hoc networks. WLAN meshes are likely to be part of 4G networks and several projects such as the EU-funded IST-DAIDALOS (www.ist-daidalos.org) are currently investigating how to seamlessly integrate such heterogeneous network technologies.

A major problem is scalability of WMNs. Currently, 802.11s WMNs are based on 802.11 basic MAC layer access method DCF, and they suffer from the same intra- and interflow interference of multihop forwarding as in standard MANET. While multi-radio, multi-channel solutions could minimize that problem, MAC and PHY layer overhead for the transmission of small packets such as packetized voice samples is still high. For example, when G.729a voice codec is used, voice payload is 20 bytes but additional 40 bytes are required for RTP/UDP/IP headers per packet, every 20 ms. In addition, MAC layer needs to spend significant time in contention phase (in average around \(310\mu s\)) depending on competing traffic and additionally contributes to the overhead using SIFS, DIFS and ACK. Including PHY overhead, sending a 20 byte voice packet (15\(\mu s\)) requires thus around \(800\mu s\) at 11 mbps [1]. The large number of small packets common with VoIP services is the primary reason for contention and congestion in multi-hop WMN as all intermediate relay nodes have to process and forward a significant number of small packets. Therefore, an important suggestion is to keep the transmission overhead per packet small. There are several mechanisms how overhead can be reduced. For example, header compression schemes such as robust header compression (ROHC) [2] can bring down the overhead of 40 byte RTP/UDP/IP headers to 2 byte connection ID when a single flow and one hop is used. However, payload size is still small compared to MAC and PHY overhead and WMNs are inherently multi-hop in nature. Therefore, other mechanisms to reduce overhead are strongly desired.

Capacity of WMNs can be increased significantly by aggregating (combining) several smaller packets into larger ones. The overall number of packets is thus reduced, average packet size increased and contention will take place only once for the larger packets. A relay node will then process fewer large packets instead of many small ones. While such aggregation mechanisms have been proposed for single-hop infrastructure WLAN, designing an aggregation strategy for multihop WMNs is a hard problem. In infrastructure WLAN, the sender has complete knowledge about link characteristics of one hop neighbors and can thus calculate an optimal packet size for aggregation [3]. In a multihop environment, signal quality and congestion for each link is different. When mesh relay nodes aggregate small packets, there is an inherent trade-off
2. Predicting Packet Size for Aggregation through Link Quality

Regarding packet size. Aggregating more packets leads to larger packets, reduces the overall number of packets in the mesh and thus reduces multi-hop contention and packet loss due to collisions. However, such larger aggregated packets can lead to higher packet loss for a link that operates at low signal quality. For such links, aggregating fewer packets can be beneficial.

Packet aggregation can be classified as end-to-end or hop-by-hop. In end-to-end aggregation, all packets towards a common destination are aggregated at the ingress node only. In hop-by-hop aggregation (de-)aggregation is done at every node, which leads to higher complexity and potentially higher delay but yields better aggregation possibilities. In a realistic WMN deployment, link characteristic and load will be different for each hop. Therefore, a hop-by-hop aggregation scheme will allow to optimize the packet size used for aggregation for each hop separately. This allows then to trade-off packet loss due to contention and bit errors potentially outperforming end-to-end aggregation strategies. However, current approaches on packet aggregation such as [1, 4–6] do not consider the influence of link quality on packet size or consider only end-to-end path quality [6] due to the use of routing metrics such as WCETT. Here, packet size for aggregated packets is calculated based on the route quality because a ”good” route can potentially carry larger aggregated packets. The problem with such approach is that metrics such as WCETT reflect path characteristics and are suitable for end-to-end aggregation and thus achieve suboptimal performance. If there is a bottleneck link such as one characterized through low signal quality, this will lead to a small packet size for end-to-end aggregation and the benefit of aggregating will be small.

The key idea of our paper is to use hop-by-hop aggregation and for each hop derive an optimal packet size for the aggregated packets. Therefore, the overall aggregation along a path will not be constrained by the weakest link leading thus to significant performance improvement, as our evaluation section will demonstrate. The rest of this paper is structured as follows: In Section 2 we will derive a mapping to an optimal packet size for aggregation based on link characteristics. Based on this packet size estimation, an adaptive aggregation algorithm was developed. Section 3 presents experimental results gained through simulations that demonstrate the significant capacity increase of our adaptive scheme. Finally, we draw conclusions of this work in Section 4.

2 Predicting Packet Size for Aggregation through Link Quality

This section presents the strategies adopted to find the optimum packet size based on current link characteristics (sub-section 2.1). It also presents a de-
2.1 Determining packet size

End-to-end delay, jitter and packet loss ratio are the most important QoS parameters determining VoIP quality. To achieve a good voice quality all three parameters need to stay below certain thresholds. For example, using G.729a voice codec requires smaller than 150 ms end-to-end delay and less than 2% packet loss in order to provide acceptable quality [7]. Due to the retransmission scheme of the IEEE 802.11 MAC layer, a reduction of packet loss ratio has also beneficial effects on jitter and delay. Therefore, any aggregation scheme should aim to increase the number of VoIP calls which can be supported at an acceptable quality by reducing the overall packet loss ratio while keeping end-to-end delay low.

The reasons for packet loss in WMNs are manifold. If the signal-to-noise ratio (SNR) is too low, the receiver cannot decode a frame correctly and drops it. If a node tries to send or forward packets at a higher rate than the MAC layer can service, the packet queue of the node fills up and eventually packets in the queue have to be dropped. As a consequence of the hidden-terminal problem, two nodes in the vicinity can start to transmit at the same time, resulting in collisions and packet loss. Finally, in multi-hop networks a node might not have a valid route to a destination so packets are queued until a route becomes available. If this takes too long, the queue might overflow.

Packet size is an important factor for controlling the network load and the packet loss ratio. As stated in section 1, due to the MAC layer overhead, small packets have poor efficiency and increase multihop contention. Large packets have better efficiency, but are more likely to be dropped due to frame errors (FER) than small packets for a given bit error rate. In theory, there is a relationship between the SNR and the bit-error rate (BER). Depending on the coding scheme and the card sensitivity, a bit error probability can be found for a given SNR value [8]. Furthermore, for a given BER the frame error rate can be approximated as 1-(1-BER)^n, where n is the frame length in bits. Consequently, the SNR can be used to predict the loss probabilities of frames with different lengths. Although some papers [9, 10] argue that it is hard to obtain such mapping, [11, 12] show that the SNR has the potential to significantly improve wireless link quality prediction and hence packet loss estimation. Therefore our scheme uses the SNR for determining the optimum frame size used for aggregated packets.

To obtain the mapping between SNR, packet size and packet loss ratio, a series of simulations using ns-2 with the shadowing model and the channel
2. Predicting Packet Size for Aggregation through Link Quality

model proposed by [8] was conducted. UDP packets were sent between two static wireless nodes at a rate of 64 Kbit/s. The packet size and node distance were gradually increased from 128 to 1528 bytes and from 10 to 105 m. For each packet size and node distance the average SNR and the packet loss ratio was calculated. SNR values have been calculated using unicast packets with standard 802.11 link layer retransmission enabled. SNR values of erroneous packets have also been taken into consideration, different from [9] which measures SNR using fixed broadcast packet size (1500 bytes) and accounts only for successfully received packets. Figure 1 shows packet loss ratio curves for different packet sizes and channel qualities. Above 11 dB SNR there is no packet loss due to FER. Between 11 and 5 dB a sharp increasing packet loss ratio can be observed. Here, for a given SNR, smaller packets cause less packet loss than larger packets. For example, with an SNR of 8.5 dB 128 byte packets virtually cause no packet loss, while the packet loss ratio for 1528 byte packets is about 48%. The curves in Figure 1 are specific for the network topology and settings described in Section 3.

The values in Figure 1 were used to find the optimum packet size, which yields 0.2, 0.5 and 1% packet loss ratio due to frame errors for different chan-

![Figure 1: Packet loss rate as function of SNR and packet size](image)
kernel qualities per link. For each packet size $p$, the point $[x/0.2\%]$ was determined. The obtained $x$-values and the corresponding packet sizes $p$ are plotted in Figure 2. These values can be approximated with the exponential functions, which were found through curve fitting. For example, the packet size $SIZE_{\text{max}}$ which yields 0.2% packet loss due to frame errors can be calculated according to Equation 1.

$$SIZE_{\text{max}} = 0.0035 \times e^{1.2255 \times SNR} \quad (1)$$

Different services tolerate different packet loss rates. Therefore, application requirements and network topology determine the curve fitting which yields the packet loss suitable for a given application. For example, if application is less sensitive to packet loss, the curve with target packet loss 1% could be used.
2.2 Adaptive Packet Aggregation

Our packet aggregation scheme consists of two components - a protocol for determining and exchanging the optimum packet size on a link and a packet aggregation algorithm.

The SNR of a link is a function of signal strength and noise, which are not necessarily the same at the sender and the receiver of a given packet. Therefore, the receiver has to propose a given packet size \( SIZEmax \) based on the SNR and the target acceptable loss ratio and inform the sender about it. We propose that the receiver constantly measures the SNR of all received packets and stores a moving average of the SNR for each link or neighbor. Using the Equation 1, it is possible to calculate the packet size which yields e.g. 0.2% packet loss ratio. If all nodes apply the same policy, less than 1% packet loss due to FER is caused on a five hop route. The receiver of a given packet then periodically informs its neighbours about that packet size \( SIZEmax \). In our implementation, the packet size information is piggy-packed on AODV-HELLO messages, which are sent to keep routes alive. When a node receives an AODV-HELLO message, it reads the packet sizes for its neighbouring links and stores it in its routing table. Although our implementation utilizes AODV protocol messages to promote packet size information, other options are possible. For example the packet size information can be promoted by modified MAC ACK frames, neighbour discovery protocol messages, or special probe packets.

The second part of our aggregation scheme is a hop-by-hop aggregation algorithm for IP packets, which is based on the end-to-end aggregation algorithm proposed in [13]. The aim of the algorithm is not to add extra delay or overhead unless necessary while providing a good aggregation ratio and to adapt the packet size to the link quality. These properties are controlled by four parameters \( SIZEmin \), which represents the minimum size of an aggregation packet, \( SIZEmax \), which specifies the maximum packet size obtained as described before, \( SIZEfactor \) for controlling the trade-off between loss due to FER and contention and \( MAXdelay \) which denotes the maximum forced delay expected.

The algorithm works as follows: at every hop, received packets are marked with a timestamp and stored in a queue located between the routing module and the MAC layer. As soon as the MAC layer becomes idle, the aggregation algorithm tries to create an aggregation packet which is passed down to the MAC layer. All packets in the queue with the same next hop can potentially be aggregated (up to a maximum size of \( SIZEmax \times SIZEfactor \)). If the cumulative size of those potential packets with same next hop is greater than \( SIZEmin \) the packets are aggregated and passed to the MAC layer. If the size is below \( SIZEEmax \), only packets which are older than \( MAXdelay \) are aggregated. If none
is older, the queue stays idle and nothing is sent. If exactly one packet is older, the queue sends the packet as it is, without adding an additional header. If at least two are older, they are aggregated and passed to the MAC layer.

Packets are aggregated by adding a new IP-header in front of the concatenated packets. The additional IP-header causes an overhead of 20 bytes per aggregation packet. Using a specific value in the protocol field of the IP-header, a node can distinguish between aggregation and non-aggregation packets.

SIZEmax only considers loss due to FER and the calculations imply that all packets are exactly of size SIZEmax. In fact, the aggregation algorithm cannot always produce such large packets, because the traffic rate might be too low or packets might be received in bursts instead of a constant flow. In such a situation MAXdelay can expire before enough packets are available for creating an aggregation packet of size SIZEmax. Consequently, increasing the maximum allowed packet size by a factor of SIZEfactor does not result in too high packet loss due to FER and reduces loss due to contention by increasing MAC efficiency. In future work we plan to make the SIZEfactor adaptive to network traffic and topology.

SIZEmin determines a lower bound for the size of an aggregation packet. SIZEmin should be chosen such that the ratio of overhead and payload (the aggregated packets) is small. However, a too large SIZEmin (i.e. a small quotient) results in many delay operations or many packets which are sent without aggregation. SIZEmin needs to be large enough to require at least two packets for aggregation.

MAXdelay denotes the maximum forced delay for a single packet. When the network traffic is low, this parameter induces artificial delay, increases the number of packets in the queue and thereby increases the aggregation ratio. However, since VoIP traffic is time critical, the value of MAXdelay should not be too high. In higher loaded networks, MAXdelay has minimal impact as the queue fills up due to natural MAC layer service delay.

3 Performance Evaluation

We implemented the adaptive hop-by-hop aggregation in ns-2 simulator version 2.26. For comparison we also implemented a static hop-by-hop aggregation, which is explained later in this section. The 802.11 physical layer in ns-2.26 is overly simplified. Nodes receive packets only when the signal from the sender is greater than the receive threshold. However, the impact of wireless transmission error due to bad channel quality is completely ignored, meaning that BER is not considered in order to determine whether one frame is trans-
3. Performance Evaluation

mitted correctly. We replace this part of 802.11 function with code developed in [8], so that SNR is used to determine BER and FER by the empirical curves measured for Intersil HFA3861B card [14], and determine whether the frame is received correctly. We use shadowing radio propagation model with path loss exponent of 2.5 and shadowing deviation of 1.1dB, and the receiving and carrier sense threshold of 83.50dB and 90.81dB, respectively. We use 1Mbps basic rate and 11Mbps channel rate based on IEEE 802.11b without RTS/CTS. Each simulation runs for 120 second. The topology is depicted in Figure 3, which is comprised of wireless and wired nodes. Node 0 represents a server, e.g. a PSTN-gateway, connected with a Fast-Ethernet link (100 Mbit/s) to the router Node 1, which itself has a wired Ethernet connection to Node 2. Node 2 is a Mesh Gateway interconnecting the wired and wireless network. Node 3 is a Mesh Relay Node, which only forwards traffic inside the mesh network. Node 4 and 5 are Mesh Relay Node where clients are connected.

We used AODV-UU [15] as the base routing protocol to provide internet connectivity for mesh relay nodes. AODV-UU is used in half-tunneling mode, which adds an encapsulation header for all packets towards the internet. However, our packet aggregation mechanism can be deployed with any routing protocol more suitable to a multi-channel, multi-radio mesh scenario.

We compare our adaptive algorithm with a static aggregation algorithm that does not take link quality information into account for packet aggregation. It is also known as forced delay aggregation [5]. Here, each incoming packet is given an explicit timestamp; when that packet has been delayed for a period longer than a statically configured maximum aggregation delay (MAXdelay), it is marked as available for transmission. When sufficient packets arrive with the same next routing step such that their combined size meets the MTU of the outgoing link, they are combined into an encapsulation packet and placed in an explicit transmission queue. The size of each internal queue can not exceed that of the MTU for the outgoing link; any available packet are aggregated and marked for transmission when an incoming packet would increase the sum of the packet sizes to be larger than MTU. In our experiments, maximum aggregation delay was fixed to 5ms for both aggregation methods. In the static aggregation, the MTU of 1500 bytes was used. A SIZEmin of 101 bytes and SIZEfactor of 2 was used in the adaptive aggregation.

VoIP traffic is exchanged between Nodes 5 and 0, Nodes 4 and 0 and the respective reverse direction. In order to estimate gains achieved by aggregation, we need a more realistic traffic load than constant bit rate (CBR). The VoIP traffic is modeled using ITU G.729a voice codec, where 40 byte UDP packets are generated at a rate of 50 packets per second using an exponential distribution with mean values for talk/silent time of 350ms/650ms [7]. Flows are equally distributed between Node 5 and 0 and Node 4 and 0. No background
Figure 3: Simulation Topology

and signaling traffic was used.

All MRNs are capable of aggregating and de-aggregating traffic. During the simulation, MRNs are stationary and two distances were used; $d_0$ (distance between Nodes 3 and Node 4, and Node 3 and Node 5 - see Figure 3) and $d_1$ (distance between Node 2 and Node 3). The values of $d_0$ and $d_1$ have been set to 55 and 56 m respectively and have been varied in some simulation runs to analyze the effects of SNR versus distance between nodes.

To provide good voice quality we need to consider packet loss rate (PLR), delay and jitter of VoIP packets. As described in Section 2.1, for a satisfactory call quality using G.729a codec, voice packet should not experience more than 150ms of end-to-end delay (including jitter) and maximum 2% of packet loss ratio. In order to verify such boundaries we present in Figure 4 average values of packet loss rate, delay and jitter in terms of number of injected VoIP flows for adaptive aggregation, static aggregation and "no aggregation". The results shown are macro averaged across all flows. The diagram shows that up to a certain threshold the number of injected flows can be increased, while the QoS parameters stay within acceptable boundaries. Above this threshold, packet loss, delay and jitter increase immediately. With "no aggregation" this threshold is about 40 flows, while for static aggregation it is about 60 flows. For adaptive aggregation it is about 120 flows, leading to capacity increase of 200% and 50% if compared to "no aggregation" and static aggregation, respectively. Figure 4 also shows that both static and adaptive aggregation can be unfavorable in low traffic scenarios (up to 20 injected flows), leading to higher delay and jitter. With low traffic, some packets wait until MAXdelay in order to get aggregated, and jitter increases. But if MAXdelay is very low, an aggregation will perform similar as "no aggregation".
Channel quality is an important factor while performing packet aggregation. As stated before, a good link quality can potentially carry longer packets for a given bit error ratio than a bad link quality. Therefore, more packets can be aggregated on a good link, while the aggregation ratio might be lower on a bad link. To verify this assumption, the distance parameter $d_0$ and $d_1$ were varied and the number of supported VoIP flows using static and adaptive aggregation was analyzed in Table 1. The improvement factor listed gives an idea of capacity increase between static and adaptive aggregation for the different topologies (1-5). Topologies 1-4 show that a higher difference in the channel quality (i.e. a higher difference between $d_0$ and $d_1$) yields a higher improvement on number of supported calls using adaptive aggregation. In topology 4, the performance of adaptive aggregation is more than twice the performance of static aggregation. This is due to the fact that the adaptive aggregation uses current link channel quality information in order to determine the optimum
Table 1: VoIP Capacity with different channel quality

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<td>56 m</td>
<td>0 m</td>
<td>130</td>
<td>112</td>
<td>0.86</td>
</tr>
<tr>
<td>2</td>
<td>56 m</td>
<td>55 m</td>
<td>1 m</td>
<td>68</td>
<td>115</td>
<td>1.69</td>
</tr>
<tr>
<td>3</td>
<td>60 m</td>
<td>55 m</td>
<td>5 m</td>
<td>66</td>
<td>108</td>
<td>1.93</td>
</tr>
<tr>
<td>4</td>
<td>65 m</td>
<td>55 m</td>
<td>10 m</td>
<td>32</td>
<td>70</td>
<td>2.18</td>
</tr>
<tr>
<td>5</td>
<td>65 m</td>
<td>50 m</td>
<td>15 m</td>
<td>26</td>
<td>32</td>
<td>1.23</td>
</tr>
</tbody>
</table>

packet size for the link. With equal channel conditions the static aggregation is even slightly better (improvement factor = 0.86) because larger packets are created. However, in topology 5 the high distance between nodes causes a high BER resulting in a high PLR. Therefore, the number of supported flows is reduced for both aggregation methods. As expected, in this simulation the VoIP capacity is extremely low, and even with the use of current link quality information does not benefit the aggregation algorithm.

4 Conclusion

In this paper, we have looked at the relation between link quality and packet size for packet aggregation in multi-hop wireless mesh networks. The proposed adaptive hop-by-hop aggregation algorithm shows around 200% and 50% achieved capacity improvement on an arrow wireless mesh topology compared to “no aggregation” and static aggregation, respectively. Future work will include the evaluation of the proposed aggregation with more realistic topologies and different traffic models, and its implementation in a real environment.

5 Acknowledgment

The work described in this paper is based on results of IST FP6 Integrated Project DAIDALOS, which receives research funding from the European Community’s Sixth Framework Program. Apart from this, the European Commission has no responsibility for the content of this paper. The information in this document is provided as is and no guarantee or warranty is given that the information is fit for any particular purpose. The user thereof uses the information at its sole risk and liability.
References


Performance Evaluation of Structured P2P over Wireless Multi-hop Networks

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Performance Evaluation of Structured P2P over Wireless Multi-hop Networks

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Abstract

Internet connected wireless multi-hop networks are an interesting alternative for providing broadband wireless access. In order for the network to be transparent, the same services need to be available as in standard infrastructure wireless deployments. However, there is a significant challenge in providing services such as authentication, name resolution, VoIP over multi-hop mesh networks as dedicated servers implementing those services might be not available. Therefore, deploying overlay networks in the mesh to decentralize those services and move towards a Peer-to-Peer paradigm is an interesting approach. However, the multi-hop nature of wireless mesh networks and the restrictions in resource availability might cause problems when deploying overlay networks on top of such environments. In this paper, we investigate the overhead and trade-offs when deploying a structured overlay solution such as the Bamboo DHT over wireless multi-hop mesh networks. We provide various simulation results characterizing the overhead of management and control traffic and give recommendations for performance improvement.

1 Introduction

Over the past several years, the area of peer-to-peer (P2P) overlay networks has attracted much attention from the research community due to their new communication paradigm. P2P overlay networks are distributed systems in nature, without any hierarchical organization or centralized control. Peers form self-organizing networks that are overlaid on top of Internet Protocol (IP) based networks, enabling the deployment of services such as robust wide-area routing, efficient search of data items, selection of nearby peers, redundant storage, naming, trust and authentication, anonymity, massive scalability and fault tolerance.
Due to their distributed nature, P2P networks can and have been used for a wide variety of Internet-based applications. First-generation P2P networks have been largely used for popular file-sharing applications (e.g., Gnutella [1]). These systems are also referred to as unstructured P2P networks since they usually do not impose any structure on their topology. Instead, nodes connect to each other largely at random. The network uses flooding as the mechanism to send queries across the overlay. While flooding-based techniques are effective for locating highly replicated items and are resilient to peers joining and leaving the system, they are poorly suited for locating rare items, and are not scalable as the load on each peer grows linearly with the total number of queries and the system size.

To overcome the scalability issues of the unstructured approach, structured P2P networks have been proposed, where the P2P overlay network topology is tightly controlled. Very popular representatives of structured P2P networks are realized through so-called Distributed Hash Tables (DHTs) (e.g., Pastry [2], and Bamboo [3]), in which data object location information is placed deterministically, using unique keys of the data objects corresponding to the peers identifiers. DHT-based systems have the property that uniform random NodeIDs are assigned to the set of peers in an identifier space. Data objects are assigned to unique identifiers called keys, chosen from the same identifier space. Keys are mapped by the overlay management protocol to a unique live peer in the overlay network. As a result, P2P overlay networks support scalable storage and retrieval of key,value pairs on the overlay network but might suffer from control traffic overhead necessary for ensuring consistent structure.

P2P overlay networks in the Internet and wireless multi-hop networks share many key characteristics such as self-organization and decentralization due to the common nature of their distributed components: both consist of a dynamically changing set of nodes, and consequently, share the characteristic of frequently changing topology (join and leave of the nodes in the overlay and additionally terminal mobility of the nodes in MANETs), and both use hop-by-hop connection establishment. The common characteristics shared also dictate that both networks are faced with the same fundamental challenge, that is, to provide connectivity in a decentralized and dynamic environment.

The key contributions of this paper are an evaluation of the behavior of a well-known DHT based lookup protocol (BAMBOO) in a static multi-hop environment common to wireless mesh networks using the network simulator ns-2 [4]. We evaluate different configurations for management traffic to find a balance between management traffic in the overlay and network congestion. We investigate the challenges when deploying structured overlay networks in multi-hop environments using standard routing protocols such as AODV. While discussing those challenges, we note that such integration is not straightfor-
ward because the overlay network is a virtual network running in the application layer and the underlying network is transparent to the nodes in the overlay, while nodes in a multi-hop environment should participate in the actual routing in network or link layer.

The remainder of this paper is structured as follows: In the next section, we provide a short overview of structured overlay networks, identify the main challenges of deploying such technology in multi-hop environments, and detail about Bamboo DHT. Our performance evaluation of Bamboo in a multi-hop mesh environment is presented and discussed in section 3. Section 4 presents related work and we conclude our findings in section 5.

### 2 Structured P2P Overlay Networks in MANETs

In this section we give an overview of structured overlay networks. We first describe the basics of structured overlay concepts, challenges of deploying structured P2P in wireless multi-hop environments, and then provide information on the design of a popular candidate - the Bamboo DHT.

#### 2.1 Structured Overlay Networks

A common feature provided by overlay networks is a lookup service handling flat identifiers with an ordinary query-response semantic. Such a service is often implemented using DHTs (Distributed Hash Tables) [2, 3, 5, 6]. A DHT allows a node to insert values connected to keys much like ordinary hash tables. At the core of each DHT lies the ability to route a packet based on a key, towards the node in the network that is currently responsible for the packet's key. This process is referred to as indirect or key-based routing. To do this, DHTs form a virtual overlay identifier space (e.g. 160-bit identifiers in Bamboo [3]) from which each peer assigns itself a random overlay ID - e.g. by hashing its IP or MAC address into the overlay ID space. Each object is also hashed into the same overlay ID space. A node is now responsible for an object if the node's own overlay ID is closest to the object's ID among all live nodes in the network. Closeness in this aspect depends on the nature of the overlay ID space and the characteristics of the respective DHT, but common examples are numerical distance or Euclidean distance.

Unlike unstructured P2P networks with their random topology, DHTs impose a structure on the overlay topology by no longer choosing routing table entries arbitrarily. Instead, routing table entries have to satisfy certain criteria depending on the respective DHTs. This structure enables DHTs to introduce
an upper bound on the number of overlay hops towards the node currently responsible for the packet’s key. This upper bound is commonly $O(\log N)$, with $N$ being the number of nodes in the network. This bound is achieved through the routing strategies employed by the respective DHTs. Those strategies include reducing the Euclidean distance in the overlay ID space to the destination in each overlay routing step (e.g. CAN [5]), halving the numerical distance to the destination in each routing step (e.g. Chord [6]), or increasing the length of the matching prefix/suffix between the current node’s overlay ID and the key in each overlay routing step (e.g. Pastry [2], Bamboo [3]).

Although DHTs can route packets very efficiently in comparison to unstructured P2P networks, they usually induce high overhead due to the maintenance of their overlay routing tables. However, as argued by [7], DHT approaches outperform unstructured approaches when the number of nodes, the number of objects, or the query rate increases, since they do not introduce flooding in the network.

### 2.2 Challenges of Structured P2P in Multi-hop Environment

In a structured P2P overlay of $N$ nodes, each node maintains direct links to $O(\log N)$ other nodes. The reduction of the routing state to $O(\log N)$ is obtained by sending a packet, destined to a destination node, to one of the neighbor nodes contained in the routing table. This process is repeated until the packet reaches the final destination. Furthermore, if the structured overlay is proximity-aware (also known as topology-dependent structuring), the $O(\log N)$ nodes information stored at each node are among the closest candidates in the network proximity space (e.g. Geographical Hash Table (GHT) and landmark routing) [8].

However, in topology-dependent structuring approaches, mobility of nodes requires the transfer of large number of objects and/or large objects: as nodes move, their coordinates change and they may no longer be the nodes with coordinates closest to the keys of some of the objects they store. This can require a large amount of objects to be transferred to other nodes, which could potentially cause scalability problems. With topology-independent structuring, objects only need to be transferred between nodes when nodes fail or new nodes arrive.

In addition, P2P overlays in the Internet operate in a resource rich environment, especially in terms of bandwidth availability. Multi-hop networks are however rather limited in bandwidth. Therefore, high maintenance traffic, as it is used currently in structured overlay networks leads to scalability
problems and congestion due to excessive maintenance traffic of the overlay when legacy P2P services are used without modification in wireless multi-hop environments. Path length is also a critical issue for such multi-hop environment as both delay and packet loss increases much more significantly with the number of hops in the physical path compared to fixed networks. This is due to the shared nature of the transmission medium where more hops will lead to more contention and thus higher packet loss ratio, something not expected from standard internet environment.

Instead of using overlays, other mechanisms (e.g. lower-layer broadcast) might be a more effective and/or less costly alternative compared to using overlays for small-scale multi-hop environments [7]. However, there are also arguments supporting visions of large-scale MANET/Sensor/Mesh networks based on P2P overlays, such as: SSR [9], and VRR [10]. As a consequence, we believe that overlays will be necessary for medium to large scale deployments of wireless multi-hop networks covering large rural or urban environments.

2.3 Bamboo DHT

Bamboo [3] is referred to as a third generation DHT, which improves previous systems by taking into account problems such as management traffic congestion in Pastry. While Bamboo is based on the routing logic of Pastry, management of overlay structure is different in order to be more scalable in a dynamic environment.

To maintain the network structure, Bamboo maintains two sets of neighbor information in each node, leafset and routing table information. The leafset consists of successors and predecessors that are the numerically closest in key space. While two nodes may be neighbors (in the leafset) in the overlay of a given node, they may be physically far away. When routing a query, it is ultimately forwarded to a node which has the key in its leafset to ensure correct lookups. However, if only the leafset is used when doing lookups, a lookup complexity of $O(\log(N))$ is achieved. To reduce the lookup performance, a routing table is used, which is populated with nodes that share a common prefix. Routing table lookups are then ordinary longest prefix matches.

When data is stored in the DHT using the put command, the data is routed through the DHT to the node primarily responsible for storing the data. When the responsible node gets the data, it caches it within its leafset neighbors in each direction according to the number of desired replicas. For certain applications, the number of desired replicas can cause large demands for storage space (if each node has equal amounts of keys to store, every node needs to store "desired replicas" times that amount). Therefore, for data storage up
dates, a node periodically picks a random node in its leafset and synchronizes the stored keys with it. The correspondent node calculates the set among its stored keys that should also be stored at the sender node, sending those keys to the sender, including the hash values of the data.

The major difference between Pastry and Bamboo is the way they handle management traffic. In Pastry, management is initiated when a network change is detected, while in Bamboo management traffic is periodic regardless of network status. While reactions to changes in the routing layer operate on very small timescale, reactions to changes in overlay structure are not so fast. However, the approach to use periodic updates has been shown to be beneficial during churn [3], since it does not cause management traffic bursts during congestion. Such traffic bursts can further increase packet loss probability; can lead to management messages being dropped and other overlay network disturbances.

In order for Bamboo to be able to serve requests and maintain a consistent network view among its nodes, it needs to perform overlay maintenance message exchanges between nodes. Periodic management traffic occurs in all layers of the Bamboo system. Neighbor ping is generated by every node in order to make sure that the node can still reach its one-hop neighbors in the overlay, and it is also used to maintain a RTT estimate used for retransmission timeout calculations. Such timer values are used to derive, if e.g. members left the overlay or messages need to be re-transmitted. However, re-transmitting too early will lead to too high number of packets. An accurate timeout value is crucial in order to predict if a packet is lost and needs to be resent along a different path in the overlay. Nodes also perform leafset update by periodically choosing a random node from its leafset, and execute a leafset push followed by a leafset pull. Both messages involve sending the complete leafset to the synchronizing node where the information is incorporated.

Bamboo considers that two nodes share the same level when one node contains the other node in its routing table. Therefore, the local routing table update is used to exchange the node information in that level. If a node gets information about other nodes that fit into the routing table, it probes the nodes to test reachability and to get a RTT estimate. If a node is reachable and fits into an empty field in the routing table, it is added. If the matching routing table entry is occupied, the node with the lowest latency is chosen. In standard configuration, Bamboo optimizes latency, although other schemes could be available, such as optimizing for uptime. It is important to note that an optimized routing table does not influence lookup correctness, but only lookup latency [11].

The Bamboo system has been evaluated through simulation and using testbeds such as the PlanetLab [12]. However, to the best of our knowledge, the evalu-
3. Performance Evaluation

To analyze the performance of Bamboo DHT [3] over multi-hop networks we show in this section simulation results with different overlay management settings. As described before, Bamboo uses proactive management traffic in order to maintain the network structure. Table 1 presents different setting for Bamboo management timeout periods used for different simulation scenarios.

In order to analyze the impact of Bamboo periodic management traffic over multi-hop networks, we compare three different settings; "no management", "standard management" used by [3], and "custom management" which tries to find a balance between lookup efficiency and management traffic overhead. Too frequent management traffic will lead to high overhead in multi-hop environments and thus lead to network congestion. No management, on the other hand, will lead to low lookup efficiency.

Simulations were performed using ns-2 in a static wireless multi-hop topology common with mesh backhaul networks for urban deployment with different number of nodes. The nodes are positioned on a grid at a distance of 200 meters, with 250m of transmission range and 500m of carrier sense range using two ray ground as radio propagation model. The transmission rate of the nodes was set to 11Mbps, and basic rate to 1Mbps.

AODV-UU routing protocol was adopted using default settings proposed by [13]. Simulations were performed for 60 seconds without bootstrapping period. During the experiments, every 2 seconds, each node generates 500 bytes put message with a random key to store data in the overlay. All nodes also try to acquire random selected keys that are located on other nodes generating 32 bytes get message every 2 seconds.

We defined the total management traffic as the aggregation of the overlay

Table 1: Bamboo Management Periods (secs)

<table>
<thead>
<tr>
<th></th>
<th>NO</th>
<th>Standard</th>
<th>Custom</th>
</tr>
</thead>
<tbody>
<tr>
<td>Leafset update</td>
<td>-</td>
<td>1</td>
<td>5</td>
</tr>
<tr>
<td>Local routing table</td>
<td>-</td>
<td>5</td>
<td>10</td>
</tr>
<tr>
<td>Global routing table</td>
<td>-</td>
<td>10</td>
<td>20</td>
</tr>
<tr>
<td>Data storage update</td>
<td>-</td>
<td>2</td>
<td>6</td>
</tr>
<tr>
<td>Neighbor Ping</td>
<td>0.5</td>
<td>0.5</td>
<td>0.5</td>
</tr>
</tbody>
</table>

Simulations have not taken multi-hop scenarios where wireless links and standard routing protocols, such as AODV-UU [13], are used.
management traffic (e.g. neighbor ping, leafset update, routing table update, and data storage update) during the whole simulation time. It is important to note that the routing traffic generated by AODV-UU is not considered in the total management traffic.

The total management traffic per number of nodes is shown in Figure 1, for the three different scenarios. As can be seen from the figure, the overhead introduced by Bamboo increases with number of nodes, and is much higher for standard timeout settings compared to 'no', and 'custom' management. This is mainly due to the aggressiveness of periodic updates required by Bamboo to monitor the status of other nodes in the overlay and update the overlay data structures. On the other hand, in the case of no management, each node does not generate periodic updates, but neighbor ping is still performed in order to maintain the leafset table status.

Figure 2 presents the percentage of management overhead compared to the total user traffic (gets, getreplies and puts messages) generated over simulation time. It can be seen that the periodic management traffic used in 'standard' and 'custom' management imposes a high overhead for the multi-hop network, and it increases as larger topologies are used. However, 'custom' management has a smaller impact compared to the 'standard' management. For 'no' management, the overhead remains constant for different number of nodes, as every node generates gets and puts and also neighbor pings.

Success ratio represents the percentage of lookups that are eventually de-
livered to the correct responsible node in the overlay. When number of nodes increase, network load increases (Figure 2) and success ratio decreases according to Figure 3. In the 36 nodes grid, success ratio is 61%, 41% and 19%, respectively for ‘no’, ‘custom’ and ‘standard’ management. This low success ratio for higher number of nodes can be explained by the higher percentage of management and routing overhead in order to maintain the overlay structure, as shown in Figure 2 and Table 2. The ability to find the destination nodes which are responsible for the specific keys degrades as management overhead increases network contention. This results in resent and dropped packets due to network congestion and consequently problems in the routing layer. This can be seen from Table 3, where we analyze the reasons for packet drop in the 36 nodes topology. In ‘standard’ management case, most of the dropped packets are due to failure on route establishment (TOUT, TTL and NRTE) which consequently leads to queue overflow (IFQ).

In order to analyze the performance of the management strategies in relation to lookup efficiency, we present in Figure 4 the cumulative distribution function of success lookup delay for the 36 nodes topology. The lookup delay is defined as the time elapsed between nodes sending a get request into the overlay until the receipt of the corresponding getreply message. The CDF shows that ‘custom’ and ‘standard’ management strategies maintain lower lookup delay performance for successful gets if compared to ‘no’ management. The figure also shows that some lookups take longer to complete for ‘custom’ than for ‘standard’ management. This is due to the aggressiveness of ‘standard’ man-
Performance Evaluation of Structured P2P over WMNs

Figure 3: Request success ratio

agagement using frequent periodic exchange of leafset table, routing table and data storage information. This results in fewer hops on average in the overlay. In contrast, for ‘no’ management scenario where the number of nodes performing forwarding operations is higher as no leafset and routing table information are exchanged among nodes, resulting in more hops on average (for example 3.70, 3.85, and 4.85 hops on average, respectively for ‘standard’, ‘custom’, and ‘no’ management in the 36 nodes topology).

Table 2: Avg. routing traffic received per node in the 36 nodes topology

<table>
<thead>
<tr>
<th></th>
<th>NO</th>
<th>Standard</th>
<th>Custom</th>
</tr>
</thead>
<tbody>
<tr>
<td>Routing Traffic (bps)</td>
<td>7875.67</td>
<td>6981.16</td>
<td>6322.87</td>
</tr>
</tbody>
</table>

Table 3: Number of dropped packets and correspondent reason in the 36 nodes topology

<table>
<thead>
<tr>
<th></th>
<th>NO</th>
<th>Standard</th>
<th>Custom</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Timeout (TOUT)</td>
<td>32</td>
<td>32</td>
<td>32</td>
</tr>
<tr>
<td>Route TTL (TTL)</td>
<td>186</td>
<td>651</td>
<td>3526</td>
</tr>
<tr>
<td>No Route (NRTE)</td>
<td>421</td>
<td>1190</td>
<td>2699</td>
</tr>
<tr>
<td>Node Queue Full (IFQ)</td>
<td>49</td>
<td>264</td>
<td>694</td>
</tr>
<tr>
<td>Total Dropped</td>
<td>688</td>
<td>2037</td>
<td>7251</td>
</tr>
</tbody>
</table>
5. Conclusion

4 Related Work

A number of publications have been devoted to P2P overlay protocols in the Internet ([1–3, 5, 6, 11]). Some are related to P2P in mobile ad-hoc networks. However, many of the publications are mainly conceptual, presenting architecture proposals but not evaluating them ([7, 8]).

It is common to use simulations to evaluate the performance of DHTs in MANETs [9, 14], although many evaluations are done using simplification of the underlay networks. One example is the evaluation presented in [14], where the wireless network is modeled as a geometric random graph, not considering direct effects resulting from wireless node such as link breaks and protocol overhead. Evaluations of DHTs, such as Bamboo, are also done often using emulation [3]. By using emulation, studying very large networks is difficult, but modeling queue drops is possible. Testbeds, such as PlanetLab, are also used, but as it occurs with emulation, large network size studies are difficult to execute.

5 Conclusion

Peer-to-Peer networking is an interesting communication paradigm for multi-hop networks, such as Wireless Mesh Networks and Mobile Ad-hoc Networks,
where it is not feasible to use centralized servers. As peer-to-peer networks share key characteristics with multi-hop networks such as resilience to dynamic network topologies, lack of any central infrastructure, self-organization, the convergence of mobile multi-hop networks and peer-to-peer overlay networks appears to be a very promising way to build distributed applications in such environments.

Using simulations, we have studied the performance of Bamboo DHT over different number of nodes, comparing results through different overlay management strategies. The simulation results have shown that the aggressiveness of standard Bamboo overlay management together with the multi-hop nature and the restrictions in resource availability of wireless multi-hop networks, cause problems when deploying structured overlay networks on top of such environments. The proactive overlay management used by Bamboo increases the network overhead, especially if aggressive management updates are performed (e.g. 'standard' management). On the other hand, the Bamboo management maintains the correctness of the network structure providing efficient queries along the overlay. Therefore, we believe that it can be advantageous to find a balance between lookup efficiency and management traffic overhead in order to deploy Bamboo DHT over multi-hop networks.

6 Acknowledgment

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References


Enabling Multimedia Services over Wireless Multi-Hop Networks

With the constant development of wireless technologies, the usage of wireless devices tends to increase even more in the future. Wireless multi-hop networks (WMNs) have emerged as a key technology to numerous potential scenarios, ranging from disaster recovery to wireless broadband internet access. The distributed architecture of WMNs enables nodes to cooperatively relay other node’s packets. Because of their advantages over other wireless networks, WMNs are undergoing rapid progress and inspiring numerous applications. However, many technical issues still exist in this field.

In this thesis we investigate how Voice over IP (VoIP) and peer-to-peer (P2P) application are influenced by wireless multi-hop network characteristics and how to optimize them in order to provide scalable communication. We first consider the deployment of VoIP service in wireless multi-hop networks, by using the Session Initiation Protocol (SIP) architecture. Our investigation shows that the centralized SIP architecture imposes several challenges when deployed in the decentralized wireless multi-hop environment. We find that VoIP quality metrics are severely degraded as the traffic and number of multiple hops to the gateway increase. In the context of scalability, we further propose four alternative approaches which avoid current limitations. In the second part of this thesis we tackle the network capacity problem while providing scalable VoIP service over wireless multi-hop networks. The performance evaluation shows the influence of intra and inter-flow interference in channel utilization, which direct impacts the VoIP capacity. In order to avoid the small VoIP packet overhead, we propose a new adaptive hop-by-hop packet aggregation scheme based on wireless link characteristics. Our performance evaluation shows that the proposed scheme can increase the VoIP capacity by a two-fold gain. The study of peer-to-peer applicability over wireless multi-hop networks is another important contribution. A resource lookup application is realized through structured P2P overlay. We show that due to several reasons, such as characteristics of wireless links, multi-hop forwarding operation, and structured P2P management traffic aggressiveness the performance of traditional P2P applications is rather low in wireless multi-hop environments. Therefore, we suggested that a trade-off between the P2P lookup efficiency and the P2P management traffic overhead can be achieved while maintaining the overlay network consistency in wireless multi-hop networks.