PRACTICAL WIRELESS MESH NETWORKS AND THEIR APPLICATIONS

by
Raluca Musăloiu-Elefteri

A dissertation submitted to the Johns Hopkins University in conformity with the requirements for the degree of Doctor of Philosophy.

Baltimore, Maryland
January, 2010

Abstract

Low cost wireless routers are revolutionizing the way people connect to the Internet. The ease of deployment on the one hand, and the freedom in the ability to connect on the other hand, have made these wireless routers ubiquitous. *Wireless mesh networks* extend the connectivity area of mobile devices beyond the limited range of a single access point. These networks can be easily deployed inside a building, campus, on a large geographical area or at a disaster site without requiring every access point to be physically connected to the Internet. They are also very affordable when implemented with off-the-shelf low cost wireless routers.

This thesis is an effort of several years towards making mesh networks a reality. The first part of the thesis introduces the architecture of the first high-throughput 802.11 wireless mesh network that provides seamless connectivity to mobile users using off-the-shelf low cost routers. The second part of the thesis explores the realm of newly enabled mesh networks applications, presenting the architecture and protocols of the first robust Push-To-Talk service for wireless mesh networks.

This work is part of the SMesh wireless mesh network developed in the Distributed Systems and Networks Lab at the Johns Hopkins University. SMesh is available as open-source at `http://smesh.org`.

Dr. Yair Amir *Advisor*	Professor Department of Computer Science The Johns Hopkins University
Dr. Andreas Terzis *Primary reader*	Assistant Professor Department of Computer Science The Johns Hopkins University
Dr. Claudiu Danilov *Secondary reader*	Senior Researcher Boeing Research and Technology

To my grandmother, Maria Elefteri

Acknowledgments

In addition to its academic contributions, this dissertation concludes a wonderful chapter of my life. It's been a great journey for me and I want to take this chance and thank all the amazing people that I had the honor to meet and collaborate with. Without their help, this work would not have been possible.

I thank my advisor, *Yair Amir*, who accepted me to the Ph.D. program at the Johns Hopkins University. Building SMesh was the most fun I ever had as an engineer and a scientist, and I am very grateful for this opportunity. Yair is an excellent teacher, from whom I learned not only how to do research, but also that embracing challenges and always striving for the best are among the most important things in life.

I am grateful to many faculty members and colleagues from JHU. I want to thank *Andreas Terzis* for working with me on one of my qualifying projects and for his guidance and comments related to this dissertation. I thank *Claudiu Danilov* for building an excellent messaging system that enabled my work and for his guidance over the years. I thank *Bruce Barnett*, *Cristina Nita-Rotaru*, and *Avi Rubin* for participating in my Graduate Board Oral exam and for their insightful comments. I thank *Fabian Monrose* for his awesome courses that taught me not only Network Security but how to conduct quality research.

I thank my collaborators, colleagues, and friends from the Distributed Systems and Networks Lab: *Jonathan Kirsch*, *John Lane*, *Michael Kaplan*, *Michael Hilsdale*. Especially, I want to thank *Nilo Rivera* with whom I worked together for many years. His endless pa-

tience and attention to details were much appreciated.

I also want to thank all the people who have volunteered to host our routers over the years. Their kind support in giving us access to their offices was of great help during our "rescue" operations.

I thank my friends from the interNetworking Research Group, who, together with the PhD Comics strip [5], made the graduate school a fun place to be: *Chieh-Jan Mike Liang*, *Jayant Gupchup*, *Jong Hyun Lim*, *JeongGil Ko*, *Moheeb Abu Rajab*.

I also want to thank Prof. *Siau Cheng Khoo* who supervised my internship at the National University of Singapore, and *Sreedhar Mukkamalla* for hosting my awesome summer internship at Google, in the Mountain View office.

My professors from Politehnica University of Bucharest hold the pillars of my engineering education: *Nicolae Tăpuş*, *Irina Athanasiu*, *Cristian Giumale*, *Mircea Petrescu*, *Adrian Petrescu*, *Valentin Cristea*, *Lorina Negreanu*, *Adina Florea*, *Eugenia Kalisz*, *Ştefan Trăuşan-Matu*. Many thanks to *Răzvan Rughiniş* and *Octavian Purdilă* for their enthusiasm and efforts to bring innovation to the rigid CS courses.

Mihaela Vişinoiu, my high school Informatics teacher, introduced me to the world of programming, algorithms, and Computer Science. Without her, my career could have taken a totally different direction. I am very grateful for this.

I especially want to express my thanks to the people who always have been around me and whose support I took for granted. I thank my twin brother, *Răzvan Musăloiu-Elefteri*, who has been my best friend even before we were born. I thank my parents, *Ion Musăloiu* and *Maria Elefteri*, for their never ending encouragements. I thank my grandparents, and most of all, my grandmother, *Maria Elefteri*, to whom this dissertation is dedicated.

In the end I would like to thank all the wonderful people that I was fortunate to encounter in my life.

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Chapter 1

Introduction

"And so it begins..."

—Kosh, Babylon 5

Most wireless network installations today involve a set of *access points* with overlapping coverage zones, each access point being connected to a wired network tap. *Mesh networks* are a paradigm shift. They remove the wired connectivity requirement by having only a few of the access points connected to a wired network, and allowing the others to forward packets over multiple wireless hops. This thesis is in the area of wireless mesh networking.

Low cost wireless routers are changing the way people connect to the Internet. The ease of deployment at home or office on one hand, and the freedom in the ability to connect that they provide on the other hand, have made these wireless routers ubiquitous. Implementing mesh networks using off-the-shelf low cost wireless routers makes these installations affordable and very appealing. These networks can be easily deployed inside a building, campus, on a large geographical area, or at a disaster site without requiring every access point to be physically connected to the Internet.

A great deal of research have been conducted on wireless mesh networks. Channel assignment [64] [60] [56], network capacity analysis [52] [63], mobility protocols [61] [74],

handoff [62], security, audio and video streaming [73]—all these are frequent topics in mesh networking conferences and journals. Some of them got extensive attention. However, turning research ideas and protocols into practical systems is not an easy task. Many times, the difficulty of running real-world experiments limit the evaluation of wireless systems architectures and protocols to simulations. We tried to bridge this gap between theory and practice by taking the challenge of building a real mesh system.

Typically, the systems that we currently see in academic world and in industry are either experimental testbeds (tailored to evaluate special kind of protocols), they use expensive hardware for mesh nodes, or have limited (or none) support for mobility. The model we chose for a mesh network is the following. The network is comprised of mesh routers (*mesh nodes*), which are stationary, and mesh clients, which can be mobile. Few of the mesh nodes are connected to the Internet (*Internet gateways*), while the rest of the nodes rely on multi-hop wireless paths to reach Internet connected nodes. In a practical setting, a mesh network needs to:

i) Provide seamless access to its users.

ii) Maintain users connections and handoff them quickly from one access point to another when users roam in the coverage area of the mesh.

iii) Be easy to deploy.

iv) Be robust and continue to operate even if part of the network is not available.

v) Be cost-effective, i.e., it must perform well using off-the-shelf low cost wireless routers.

These are the challenges that motivated this work. The SMesh system is the outcome of several years of research. It went through several stages of development [65] and is available as open source software [7]. We deployed the system in a 18-nodes testbed throughout three

buildings at Johns Hopkins University.

1.1 Highlights & Contributions

Off-the-shelf wireless routers provide good performance when functioning as regular access points, however, their limited CPU capacity is a performance bottleneck when these routers are part of a mesh infrastructure, as will be shown in Chapter 5. The reason is that a mesh network requires routing services that are not natively supported by current operating systems. This lack of support limits the routing mechanisms that can be used in such networks to *user-level* implementations. Routing the entire traffic through user space is very convenient but becomes problematic for routers with limited processing power. It is widely known that forwarding packets through user space results in higher CPU utilization when compared to kernel space. The overhead can be attributed to two primary factors: memory copies and context switches. Each routing node must copy the packets from kernel space to user space in order to determine the next hop. After a routing decision is made, the packet must be returned to kernel space where it is sent on the network. That is, the user-kernel boundary must be crossed a minimum of two times per hop.

This thesis presents the architecture of the first high-throughput wireless mesh network that provides seamless connectivity to mobile clients using off-the-shelf low cost wireless routers. The design captures the flexibility of user-level based systems without the performance degradation that is normally associated with using such systems on resource limited devices. Specifically, the mesh packet routing is controlled from user space by an overlay system, while the actual packet forwarding is done at the kernel level. To accomplish this separation while preserving seamless mobility, we introduce a novel redundant multipath routing mechanism. Our approach requires minimal additions to the kernel (essentially a loadable kernel module), preserving portability, a very much desired property of overlay sys-

tems.

Internet access is the most common usage of wireless mesh networks today. However, by taking advantage of the mesh infrastructure, mesh networks open the door to applications beyond the ones typically used in wired and wireless LANs. We explore the realm of these applications by looking at Push-To-Talk, a half-duplex communication service between multiple participants. While Push-To-Talk systems usually rely on centralized architectures, implementing such a system in a distributed manner makes it very appealing for emergency response workers. Many times, first responders need a rapid way to deploy a network infrastructure and an efficient way to communicate. Wireless mesh networks are an excellent way to establish an instant infrastructure.

This thesis presents the architecture and protocols of the first robust distributed Push-To-Talk service for wireless mesh networks. Collectively, the mesh nodes provide the illusion of a single third party call controller (an entity that manages communication between two or more parties in a telephone call), enabling clients to participate via any reachable mesh node. Mesh users participate using a SIP-based VoIP phones (an actual device, or a soft-phone). In addition, we allow cell phone users to join a Push-To-Talk group established in the mesh by connecting to a SIP gateway.

In our approach, each Push-To-Talk group (also referred to as a PTT session) instantiates its own logical floor control manager that arbitrates the order in which participants can speak, based on their requests. Any of the mesh nodes in the network can play the controlling role for a session. To maintain high availability, each controller node is continuously monitored by every mesh node with a participating PTT client and is quickly replaced if it becomes unavailable due to a crash or network partition. The controller relinquishes its role to another mesh node upon determining that this node is better situated (network-wise) to control the PTT session, based on the current locations of the clients participating in the

session.

A lesson learned from building such a Push-To-Talk system is that the distributed system support provided by the mesh infrastructure plays an important role in building appealing applications for wireless mesh networks.

1.2 Organization of the Dissertation

This thesis is organized as follows. Chapter 2 presents related work and overviews the current efforts in making mesh networks a reality. Chapter 3 specifies the mesh network model considered in this thesis and the main challenges of building a practical wireless mesh system. Chapter 4 overviews the SMesh system, followed by an in-depth description of its routing architecture (Chapter 5). Finally, Chapter 6 presents the design, implementation and evaluation of the Push-To-Talk service.

Chapter 2

Related Work

This chapter presents several existing mesh networking systems and projects, coming both from academic and industry worlds. Then it overviews a few ways of extending the routing functionality of the current operating systems. In the end, it includes a brief background on the Push-To-Talk application.

2.1 Wireless Mesh Networks

MIT Roofnet. This is a experimental multi-hop 802.11b network consisting of 50 nodes deployed in volunteers' apartments in Cambridge, MA [30] [25]. The nodes are off-the-shelf desktop computers equipped with 8 dBi omni-directional antennas mounted outside. The network cards are used in "Pseudo-IBSS" mode, a modification of the standard ad-hoc mode, such that it does not suffer from the IBSS partioning problem. Few of the nodes are Internet gateways. The testbed is used for providing Internet access, as a community network, but also to carry on experiments that evaluate various routing protocols [26] and link-layer measurements [10].

Microsoft Mesh Connectivity Layer (MCL). This is a loadable Windows driver developed by Microsoft Research Lab that creates a virtual network adaptor used to form an ad-hoc network between Windows computers. The routing system is based on the DSR (Dynamic

Source Routing) protocol, which was extended to support link quality metrics. The system was deployed in office buildings and a local apartment complex [1].

UCSB MeshNet. This is a 25 nodes experimental testbed deployed on the campus of University of California, Santa Barbara. Each mesh node consists of two Linksys WRT54G wireless routers strapped together. One of them is used to run the AODV (Ad-Hoc On-Demand Distance Vector) routing protocol and the second one for out-of-band management. A gateway node is a small desktop computer which runs Linux.

Stony Brook iMesh. This is a infrastructure-mode wireless mesh network designed to provide seamless networking services to mobile users, both for last-mile access and for peer-to-peer access [74]. The mesh nodes communicate via WDS (Wireless Distribution System) links established using a neighbor discovery protocol. For routing iMesh uses OLSR (Optimized Link State Routing) protocol. The system is evaluated on a testbed of six Soekris Engineering net4521 processor boards.

Metricom Ricochet. In mid 1990s, Ricochet Networks, owned by Metricom, was one of the first wireless mesh networks in the United States deployed for generic public [14]. Its goal was to provide wireless Internet access, emerging as an "always-on" replacement for the popular, at that time, 28.8 Kbps telephone modems. Users' traffic was forwarded by repeaters, running in the 900 Mhz ISM band of the RF spectrum, to a wired access point. The system was shut down in 2001 when Metricom filed for bankruptcy.

The Champaign-Urbana Community Wireless Project (CUWiN). This is an effort of independent developers to build community-owned, not-for-profit mesh networks. CUWiN developed the Hazy-Sighted Link State (HSLS) routing protocol, a link-state protocol which uses both proactive and reactive techniques to disseminate the link-state updates. The software, called CUWiNware, is open source and supports radio chipset such as Intersil Prism, Atheros and Hermes.

IEEE 802.11s. In the light of emerging mesh networking systems, IEEE formed a Task Group whose goal is to develop a standard that allows inter-operability between proprietary mesh systems. The aspects that are being discussed include path selection, security, and enhancements of the MAC (Medium Access Control) protocols. The existence of such a standard is a good thing and could stimulate a large-scale adoption of mesh systems.

Although is still under development, with many aspects yet to be addressed, IEEE 802.11s draft has already started to be adopted by industry. One of the most notable examples is the One Laptop per Child project [3], which supports 802.11s draft on their OLPC XO computer. Also, since version 2.6.26, there is a reference implementation in the wireless stack of the Linux kernel.

2.2 Routing Infrastructure

Using inexpensive wireless access points as nodes in a mesh network requires additional routing services. These services, such as anycast or redundant multipath routing, are necessary for achieving efficient and robust routing in the mesh, in the presence of mobile clients.

Chapter 5 evaluates various methods for routing packets for mobile users in wireless mesh networks, compares the performance of user-level and kernel-level forwarding, and introduces a new routing architecture based on a redundant multipath mechanism. To be supported, this mechanism requires only a few additions to the current Unix-based kernels. As such, our work relates to previous work on extending the routing capabilities beyond what the basic functionality offered by existing operating systems, as well as the use of redundant multipath for mobility.

The routing process involves computing routes to a destination, usually taking into account the distributed and dynamic nature of the underlying network, and the actual forwarding of the packets. Packet *routing* is commonly performed in user space, allowing different

protocols to be easily deployable and upgradable (OSPF [58], RIP [43]). Packet *forwarding*, on the other hand, typically resides in the kernel to forward packets as fast as possible. Thus, routing relies on the forwarding capabilities provided by the operating system. This approach allows operating systems to be both flexible and efficient.

Software routers, such as the Click Modular Router [49] and Router Plugins [36], have received much attention because they extend routing capabilities allowing the development of a rich set of network protocols, routing platforms (e.g., XORP - eXtensible Open Router Platform [42]), or wireless protocols (e.g., DIRAC system [76]). These kind of systems are powerful and very appealing for networking research. However, they usually require complex architectures (XORP codebase consists of more than half a million lines of C++ code) as compared to those of more general forwarding solutions provided by regular Unix-like operating systems. Our focus is the on systems that can run on off-the-shelf wireless routers, where the least changes of the operating system, the better. Changing the operating system at the kernel level adds a lot of freedom in routing approaches, but it diminishes the portability of the system because it needs to be kept up to date with the new kernel releases.

Another approach to extend routing services is to build user-level routers that forward packets at the application level. RON [19] uses this approach to route packets through an overlay network, increasing the reliability of the end-to-end path compared to using the underlying direct path. End-System-Multicast [45] and Spines [15] systems also route through an application router, providing services like multicast (in the overlay) without infrastructure support. Another interesting system is X-Bone [72], however, its focus is on building and managing overlay networks over the existing IP infrastructure and not on providing modularity and flexibility in extending the existing routing services.

Other work has looked into operating systems support for wireless ad-hoc routing protocols. Allard et al. [11] describe a user level router that supports ad-hoc routing protocols.

Complementary to our work, Chakeres and Belding showed in [29] an in-kernel design and implementation of the ad-hoc AODV protocol using Netfilter modules, and showed performance improvement compared to user-level ad-hoc protocols. Kawadia et al. [47] proposed a complete architecture to support ad-hoc protocols in-kernel and a generic ad-hoc support library for user-level programs to control different ad-hoc protocols.

Redundant multipath, or the ability to send packets through multiple paths simultaneously, is a necessary component in wireless infrastructures that provide seamless, lossless and fast handoff (Chapter 5). In these networks, redundant multipath can be achieved by multicasting packets to the access points handling the client during handoff. Several handoff protocols that use multicast and/or signaling to control path redundancy have been proposed. Seshan, Balakrishnan, and Katz [69] showed in the Daedalus project how low data loss can be achieved during handoff on cellular wireless networks by using multicast to nearby base stations. Helmy, Jaseemuddin, and Bhaskara [44] also show how fast handoff can be achieved in wireless networks by requiring mobile clients to explicitly join a multicast group to which packets are multicast-tunneled through the infrastructure.

Some routing protocols are specific and should be handled by application level routers. Redundant multipath is general in context and can assist a number of routing protocols. Mobility is only one of several uses of redundant multipath. Other applications will be discussed in Section 5.5.

2.3 Applications for Wireless Mesh Networks

2.3.1 Push-To-Talk

PTT allows half-duplex communication between multiple participants which request to speak by pressing a button. On a PTT group only one user is granted permission to speak at a time,

while all the other users listen. DaSilva et al. [34] provide a good survey about PTT technologies. Floor control (floor arbitration), an integral part of PTT, has been studied extensively over the years [37, 50, 54]. Some approaches to decentralized floor control are presented in [24]. A basic level of fault tolerance is built into some of these protocols to enable crash recovery.

PTT is commonly used by law enforcement and public safety communities to efficiently communicate between multiple users. Public safety agencies usually rely on trunked networks, known as Land Mobile Radio (LMR) systems, for voice and data communication [70]. The two major LMR systems are Project-25 [6], which is deployed over North America, and Terrestrial Trunked Radio (TETRA), which is deployed over Europe. Stringent guidelines for PTT, such as 500 ms one-way delay for voice packets to all listeners of a group, ensure that the system operates with acceptable performance.

Cell phone users also benefit from PTT type services that are now offered by telecommunication companies. A common standard, known as Push-To-Talk over Cellular (PoC) [12], allows PTT from different cellular network carriers to inter-operate with one another. PoC uses VoIP protocols (SIP, RTP, etc) between clients and the PoC server. A floor control mechanism, referred to as Talk Burst Control Protocol, arbitrates communication in each group. The performance requirements of PoC are less demanding than those in LMR systems. For example, the standard specifies that end-to-end delay should typically be no more than 1.6 seconds and that the turnaround time from the time a user releases the floor until it hears another user speak should be no longer than 4 seconds. An initial evaluation on a GPRS cellular network is shown in [23].

Balachandran et al. show a unifying system for bridging LMR and commercial wireless access technologies [22]. Both LMR and commercial PTT solutions (PoC) rely on a central point of arbitration and send a separate unicast voice stream to each member of the PTT

group. On these networks, the inherent inefficiency of using multiple unicast streams is not that costly over the wired backbone medium. Such a design would yield a multi-hop wireless mesh network useless with just a few users, and therefore is not a good fit in our case.

A decentralized approach with a full-mesh conferencing model is presented by Lennox and Schulzrinne in [53]. Florian Maurer [55] shows a decentralized scheme for PTT. Both approaches rely on all-to-all communication of control and voice packets between users. While adequate for small conferences or PTT sessions, this approach does not scale well and does not provide the robustness necessary to support node crashes and network partitions and merges that may occur in a wireless environment.

Complementary to our work, some research has looked at optimizing routes for PTT data traffic in wireless mesh networks. Kado et al. [46] propose a centralized tree-based routing protocol that enables a root node to compute and arbitrate routes in the network. While we also optimize routes by using multicast dissemination trees from each mesh node to each PTT group in the system, our focus is on the fault tolerance and availability aspects for providing a highly robust PTT system.

Chapter 3

Practical Wireless Mesh Networks

This chapter describes the model used when we refer to a wireless mesh network in this thesis. Different models are adopted in wireless research. We consider what we think is the most simple and realistic way to look at a mesh network. The second part of the chapter outlines the challenges one faces in designing and building such a wireless mesh system in practice.

3.1 Model

802.11 single radio. While part of the research covered by this thesis is generic and it can be applied to other kind of networks, we assume a mesh node is equipped with an 802.11 b/g radio and omnidirectional antennas. Unless otherwise noted, each mesh node has a single radio.

Some nodes are Internet connected. In a typical wireless LAN each access point is wired either directly to the Internet (for very small networks) or to a WLAN hardware controller. This makes such installations costly and difficult to extend in buildings and open places where wired infrastructure in not readily available. Wireless mesh networks changes this paradigm: only a small number of mesh nodes are connected to the Internet, sharing their connection over the wireless links with the rest of the mesh nodes. We refer to the Internet

connected nodes as Internet gateways.

Multi-hop networks. There is no single-hop connectivity between all the mesh nodes. As only some of mesh nodes are wired, traffic from one non-Internet connected node is forwarded over multiple hops to reach an Internet gateway. If we consider user-to-user communication, a multi-hop path is essentially necessary between any two nodes in the mesh.

Multi-homed networks. A wireless mesh network can span a large geographical area. This makes Internet gateways likely to reside in different network domains, effectively creating a multi-homed wireless mesh network.

Mesh nodes are stationary. This is one of the main characteristics that sets mesh networks apart from ad-hoc networks. While nodes in the ad-hoc networks are in general mobile, in mesh networks, even though topology can change over time (nodes are added or removed), they are stationary.

Mesh users are mobile. In a mesh network users are not part of the infrastructure. They can connect and access the network from any location in the mesh. Once connected they can roam in the network, moving from the coverage area of an access point to another. While typically most of user traffic is with the Internet, peer-to-peer communication should also be possible.

3.2 Challenges

Seamless access for mesh users. In a practical setting, a mesh network cannot assume special software installed on user's side. Neither can assume network driver modifications. Any unmodified 802.11 device (laptop, PDA, smartphone, etc.) needs to be able to connect and access the network transparently. In contrast, mesh nodes do not have such a challenge, i.e., they can run any software as long as transparency with the clients is maintained.

Fast handoff for users that roam within the mesh. Mesh users must be able to freely roam

within the area covered by the wireless mesh nodes. Their existing connections must be maintained at all time while roaming, and users should not perceive any interruption in their traffic. This is particularly important for interactive VoIP applications. This requirement may seem obvious, as it is what a user expects to have in a cell phone network, however, 802.11 standard does not specify a mechanism for roaming. Cell networks achieve a smooth handoff using signaling in their low-level protocols and sharing information between the towers.

Rapid deployment. The ideal case for deploying a wireless network is to perform a site survey and to carefully place access points in the most appropriate locations. This is also what usually happens in planned academic testbeds. However, in a practical setting, mesh networks are unplanned, nodes are placed in convenient places for the people who host the routers, and not where the connectivity is the best. This poses a challenge on the mesh network. It needs to dynamically self-organize, and establish mesh connectivity between all the nodes in the mesh network. In addition, self-configuration of the nodes is important, as it lowers network administration time and it makes really easy new nodes deployment.

Robustness. Due to the inherent instability of the wireless environment, network's connectivity changes over time. In a well-established network this may rarely happen, however, a mesh network, especially one rapidly deployed on an emergency site, will often experience such problems. Topology changes, like network partitions and merges, should not impede the overall functionality of the network. If an Internet gateway crashes, no user should be left out of service. Users traffic must be redirected to another Internet gateway.

Low cost. A mesh network is attractive in practice not only if it is easy to operate but also if it is cost-effective. Using a full-fledged computer as a mesh node makes things easier but also increases the deployment's cost. Our goal is to build a mesh network using off-the-shelf wireless routers. While these low cost routers provide good performance when functioning

as regular access points, their limited CPU capacity can be a bottleneck when they are part of a mesh infrastructure. This requires routing services that are not natively supported by current operating systems, limiting the routing mechanisms that can be used to user-level implementations, which can greatly degrade performance.

Security. Wireless networks' security, and mesh networks in particular, is challenging for multiple reasons. First, the way typically a WLAN provides authentication, authorization and accounting for its users is using a centralized server, such as RADIUS[1]. This requires planning, it does not scale well, and it is not suitable for mesh networks that are rapidly established in an emergency situation. Secondly, mesh nodes cannot be physically secured, and they can be compromised. In a multi-hop network this is very problematic as it may disrupt the functionality of the entire network. Fortunately, a great deal of research has been done on secure routing protocols. However, using such protocols in practice, and in general, securing a mesh network in the proper way, is a challenging and interesting problem by itself. For this reason, in this thesis we adopt an open access model, which is suitable and commonly used in community wireless networks.

Applications support. Internet access is the most common usage of wireless mesh networks today. However, by taking advantage of the mesh infrastructure, mesh networks enables applications beyond the ones typically used in wired and wireless LANs, in areas such as emergency response, remote monitoring and control, security surveillance. As an example of such applications, Chapter 6 presents Push-To-Talk, a half-duplex communication service between multiple participants. Implemented in a mesh network, this system is beneficial as a robust communication service for first responders.

[1]Remote Authentication Dial-in User Service.

Chapter 4

The SMesh Wireless Mesh Network

The SMesh system is the outcome of several years of research. It went through several stages of development. This chapter provides a brief overview of the system, setting the stage for Chapter 5, which describes in detail system's routing architecture. An in-depth view of the mobility protocols in SMesh is provided by [18].

4.1 System Overview

The entire handoff and routing logic in SMesh is provided solely by the access points, and therefore connectivity is attainable for any 802.11 mobile device that supports DHCP, regardless of its vendor or architecture. In order to achieve this complete transparency to mobile clients, our approach uses only standard MAC and IP protocols. The entire mesh network is seen by the mobile clients as a single, omnipresent access point, giving the mobile clients the illusion that they are stationary.

4.2 Architecture

The core of the SMesh system consists of two components (Figure 4.1): the *Interface with Mobile Client* and the *Communication Infrastructure*. Together, these components cover the routing logic, mobility protocols, and client management sub-system. There is also the Push-

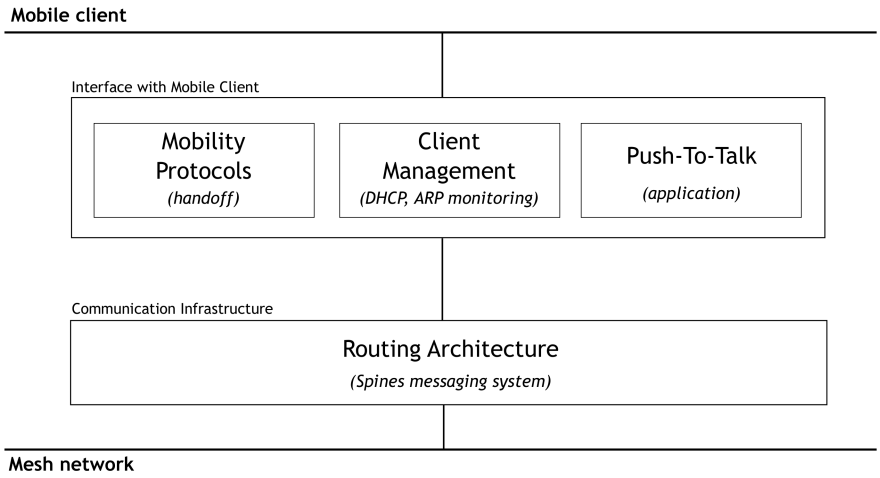


Figure 4.1: High-level view of the SMesh architecture.

To-Talk sub-system, which provides functionality for a half-duplex communication service between multiple mesh users.

4.2.1 Communication Infrastructure

To allow any node to directly communicate with all the nodes within its range, SMesh uses 802.11 in IBSS mode (ad-hoc). Communication between the mesh nodes is necessary in order to forward packets over multiple hops, to an Internet gateway.

Multi-hop communication in SMesh is achieved using the Spines messaging system [15] [8]. This is a convenient tool for unicast, multicast, and anycast communication in an overlay network. In SMesh we instantiate a Spines daemon in each node in the network and create an overlay topology that maps to the mesh topology. The Spines daemon discovers nearby nodes using periodic hello messages, and creates links between nodes when connectivity in both directions is above a certain threshold. Virtual links are created between the nodes connected on the wired network.

Spines uses a link-state protocol to disseminate link state updates in the entire network

(flooding). To minimize the medium usage, these updates are incremental and are sent over reliable links to a node's direct neighbors. As the mesh nodes are stationary and mesh users, which can be highly mobile, are not part of the mesh topology, most of time the protocol will exchange very little link-state information.

Of particular interest to our system is Spines ability to provide overlay multicast and anycast communication (multicast and anycast IP addresses are defined in the Spines virtual address space, not in the actual IP address space of the network). We leverage this for building mobility protocols that provide a seamless handoff for mobiles clients. Traffic on multicast groups is routed according to multicast trees that are computed in a way similar to MOSPF [57]. Whenever a node joins or leaves a multicast group the local Spines daemon updates the rest of the network with a reliable flood. In experiments ran on regular desktop machines Spines could handle several hundred thousand group members, being limited only by the available memory to maintain its data structures. [33].

Topology formation. SMesh forms its topology in the following way. Each access point periodically broadcasts hello messages, allowing the nodes in its coverage area to establish direct links. All the nodes that are Internet gateways are connected with virtual links over the wired infrastructure in a fully connected graph. This is achieved using an overlay multicast group, called *Internet Gateway Multicast Group* (IGMG). An Internet gateway joins this group and broadcasts its IP address, allowing other Internet gateways to connect.

Routing metric. SMesh topology is hybrid, including both wired and wireless links. Each link has an actual cost (which can be latency for wired links or ETX [32] for wireless links) that is adjusted in order to give preference to wired links when computing the cost of a path. We do this in order to reduce the usage of wireless medium, and also because wired links are much more reliable and have a higher bandwidth than wireless links. Therefore, even within the mesh, packets between two source and destination nodes might be routed via hybrid

paths, by short-cutting long wireless paths with Internet gateways. [18] explains in detail how this metric is computed.

4.2.2 Interface with Mobile Client

In a practical mesh network a user must connect and access it without any special software or network driver installed on the device. This constrain, combined with our aim of providing a seamless and fast handoff between access points, led us to the following approach of interfacing with the client. To begin with, we rely on standard MAC and IP protocols, available on any user's networking stack. Then, SMesh provides the illusion of a single distributed access point to mobile clients. This is achieved by providing connectivity information to clients through DHCP [38] and by always giving the same information (IP address, netmask, and default gateway) to the mobile client.

When a new client is requesting connection information, a special DHCP server running on the mesh nodes provides an IP address by computing a hash function on the user's device hardware address. This is private a IP in the `10.0.0.0/8` address space (mesh nodes' IP address are assigned in the same address space). In addition, mesh nodes advertise a virtual gateway IP address to the client in their DHCP offers and acknowledgments. The mobile client sets its default gateway to this virtual IP address regardless of which access point he is connected to. There is no node in SMesh with this IP address. Instead, SMesh makes the client "believe" that this address is reachable by associating it to a *mesh node* hardware address. This association between the virtual IP address and a hardware address is done via gratuitous ARP packets sent by the mesh nodes. We detail this in [16]. This way, mobile clients get the illusion of being connected to a single access point that follows them as they move.

4.3 Client Management

In SMesh, two multicast groups are associated with *each* mobile client.

Client Control Group. An access point in the vicinity of a client joins this multicast group to coordinate with other mesh nodes in the client's vicinity. This coordination is necessary for sharing information required by the handoff protocols. Specifically, each node tracks its own connection with the client, computes a link metric (using ARP packets sent every 4 seconds and also packet RSSI[1] and retransmit flag, if frame's full 802.11 header is available), and advertises the information periodically on this multicast group.

Client Data Group. This multicast group is used to deliver actual data packets to the client. A mesh node joins the client Data Group if it believes it has the best connectivity with the client based on the link quality metrics it receives from other nodes.

4.4 Mobility Support

The Client Control Group and Client Data Group provide the basic mechanisms for achieving a fast **intra-domain** handoff. In contrast to the roaming mechanisms employed by 802.11 devices[2], in SMesh the handoff is controlled from the mesh infrastructure and it relies on sending data through multiple paths to the mobile client while it transitions from one access point to another. The access points continuously monitor the connectivity quality of any client in their vicinity and efficiently share this information with other access points in the vicinity of that client to coordinate which of them should serve the client. During a handoff, multiple access points may believe they have the best connectivity with the mobile client, and data packets to the client will be duplicated by the system in the client's vicinity. Using multiple access points during the handoff minimizes the packet loss, allowing real-time applications

[1] Received Signal Strength Indicator.

[2] IEEE 802.11 standard does not specify a roaming mechanism, which led to various (and proprietary) techniques being employed by the manufactures.

such as VoIP. During stable connectivity times the mobile clients are handled by a single access point. Our protocol guarantees that, at all times, there is at least one member in the Data Group of each client, such that the client will be served by at least one mesh node [18].

A fast **inter-domain** handoff is achieved by using multicast groups through the wired network to coordinate decisions and seamlessly transfer connections between Internet gateways as mobile clients move between access points. The idea is to make new connections always use the closest Internet gateway at the time of their creation, while existing connections are forwarded through the wired infrastructure to the Internet gateway where they were originally initiated. In this way the data is routed optimally to the closest Internet gateway, without breaking existing connections (as clients are in a private address space, a NAT operation is performed at each Internet gateway). We treat UDP and TCP traffic separately. As opposed to the intra-domain handoff protocol, the coordination now is between the Internet connected nodes, and is performed over the wired links. [17] explains in detail how is the routing agreement between Internet gateways reached and how are the connections transferred to the appropriate nodes.

While duplicating packets and tightly coordinating access points in a client's vicinity may seem to incur high overhead, we quantified the overhead and demonstrated it is negligible compared to data traffic [18].

4.5 SMesh Testbed

Because of the difficulty of conducting experiments on real-world wireless networks, very often wireless research relies on simulations to evaluate various protocols. While simulations play a very important role in testing network protocols, most of these simulations are based on simplifying assumptions. Kotz et al. summarize in [51] the most important "mistaken axioms" that make wireless simulations unrealistic: *"The world is flat."*, *"A radio's transmission*

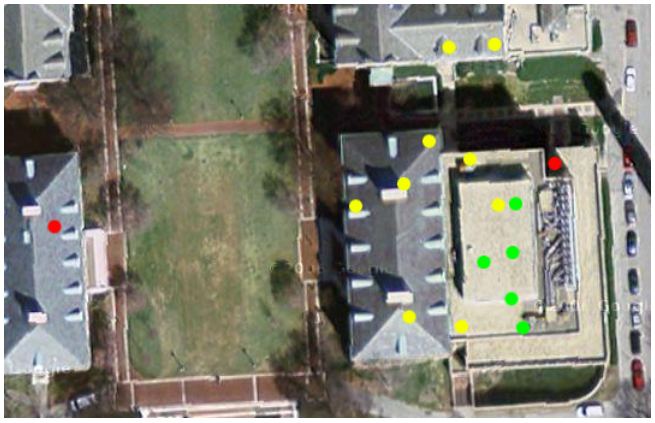


Figure 4.2: View of the SMesh testbed, with the approximative locations of the nodes. A node's color indicates the floor on which the node is located. Three of the nodes are Internet gateways.

area is circular.", *"All radios have equal range."*, *"If I can hear you, you can hear me (symmetry)."*, *"If I can hear you at all, I can hear you perfectly."*, *"Signal strength is a simple function of distance"*. While efforts have been made to build more realistic simulation tools (e.g., *ns-2*, *OpNet*), real radios and wireless environments are very complex and remain hard to model.

From the very beginning we aimed at building a real system, and understand the practical problems that such a system experiences. This resulted in a 18-node testbed deployed over three buildings at Johns Hopkins University, Homewood campus.

In terms of hardware we used off-the-shelf Linksys WRT54G wireless routers. They are equipped with 802.11b/g Broadcom BCM947XX radios, omnidirectional antennas, 16 MB RAM, 4 MB flash memory, 200 Mhz CPU speed. Our choice of the router was motivated by its low cost and because it provides a simple Linux environment. We re-flashed each router with the open source OpenWrt firmware [4], and used a MIPS cross-compiler to build our system, written in C.

Nodes' locations slightly changed over the years. Figure 4.2 shows a map with their approximative locations, in the most current configuration. Three of these nodes are Internet

gateways.

Our efforts of deploying and maintaining a real-world testbed payed off, as we gained a lot of insight on the problems that wireless mesh networks encounter in practice.

Chapter 5

Routing Architecture

Low cost wireless routers are revolutionizing the way we connect to the Internet, becoming very popular both because of the easiness to deploy, and because of the freedom in the ability to connect from everywhere they provide. These routers are a revolution from another, less known perspective: They are very cheap, albeit quite limited, Linux boxes (around $50 a piece). These attributes make them very attractive and convenient for developers to implement their own applications.

In a mesh network, as opposed to a network of independent access points, a wireless router must participate in a hybrid wireless-wired, multi-hop, routing mechanism to allow Internet access from any point in the mesh. In addition, special mechanisms are required to allow users to seamlessly roam in the network. These kind of routing services must be built using the native routing capabilities offered by router's existing operating systems. To extend routing capabilities without requiring special operating system support, developers often resort to user-level overlay routing systems, such as the ones overviewed in Chapter 2. The limitation of resources (in terms of processing power) of a mesh node that uses such user-level systems impacts the performance, in terms of throughput, of the routing mechanisms employed in the mesh.

SMesh routing architecture captures the flexibility of user-level overlay systems without

the performance degradation associated with using such systems on resource-constraint devices. This chapter first describes our high-throughput routing architecture, followed by the key elements that facilitated the implementation of this architecture in Linux, and it ends with the evaluation of the system using 17 nodes from our testbed.

5.1 Design

Redundant multipath routing (i.e., the ability to simultaneously send the same packet over multiple routes) is an essential service for increasing the reliability of wireless mesh networks. As mobile clients (laptops, smartphones) roam throughout the area covered by the mesh network, their access point must change to avoid loss of connectivity. Redundant multipath can help achieve uninterrupted connectivity during handoff by:

i) Sending packets through multiple access points to the mobile client, to deal with unexpected client movements, until the access point with the best connectivity is chosen.

ii) Avoiding loss while route changes take place in the wireless mesh.

Related work has looked into these benefits in wireless environments ([69], [44], [16]). Other applications can also benefit from redundant multipath routing (Section 5.5). However, redundant multipath is not a routing service provided by current operating systems. There are several different approaches we could take in designing our routing architecture, without changing the operating system's networking stack:

(1) Use unicast forwarding to route packets to one of the access points handling the mobile client during handoff. This approach benefits from kernel-routing performance, but loses messages during handoff due to the usage of a single path.

(2) Use IP multicast to achieve redundancy by reaching several destinations with one transmission. It efficiently transmits packets but suffers from lower reliability due to lack of

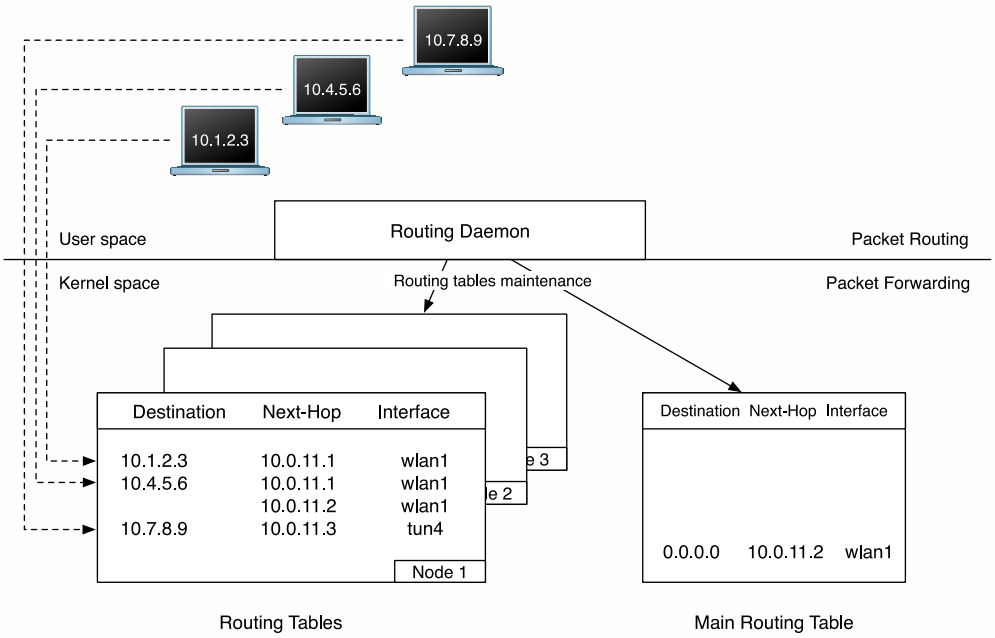


Figure 5.1: SMesh routing architecture.

802.11 link-layer retransmissions.

(3) Use IP multicast with unicast tunnels on each hop to take advantage of wireless link-layer retransmission. However, this approach incurs additional space and processing overhead on each node.

(4) Use user-level overlay routing to provide redundant multipath by routing packets through user space, at the expense of higher CPU utilization.

Each of these alternatives is described in Section 5.4 and a discussion of the tradeoffs between them is included in Section 5.4.1.

We adopted a hybrid design, where packet routing is managed in user space but packet forwarding is efficiently performed at the kernel level. Figure 5.1 gives a high level view of our routing architecture. There are two main types of communication in mesh networks: between the client and the Internet, and between clients connected to the mesh (peer-to-peer).

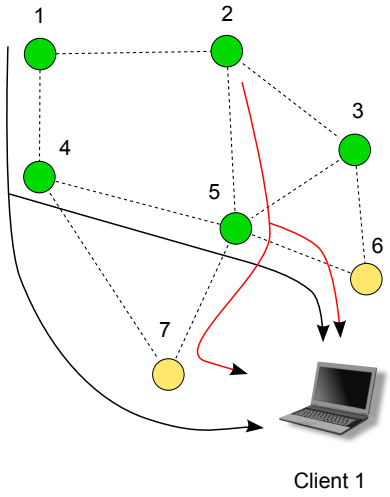
We direct the packets destined to the Internet to the closest Internet gateway. This is accomplished by routing packets using an *anycast* group in which all the Internet gateways join. The membership of this group is conveniently maintained in user-space and is translated into a unicast routing table at the kernel level.

On the other hand, traffic directed *to* the mobile client requires multipath communication, which cannot be translated directly into a unicast routing table in the kernel.

Entry point based routing. Traffic to a client may originate from different source nodes in the mesh network. This is because, first, in any non-trivial network there will be more than one Internet gateway (a multi-homed wireless mesh network [17]), any of which may need to forward packets to the client. And second, since clients may communicate with other clients in the mesh network, virtually every access point could possibly be a source of packets in the mesh. To provide optimal redundant multipath routing in these networks, each node must consider the mesh source, in addition to the destination of each packet[1] in order to determine the appropriate forwarding rule for that packet. We refer to this node as the packet's *entry point*.

This becomes more evident if we analyze the scenario from Figure 5.2. The mobile client is experiencing a handoff and the traffic from Internet must be directed to both nodes (access points) 6 and 7. We show how packets are routed from two different sources: node 1 and node 2. Note that these mesh nodes are not the actual sources of the packets. Rather, they are the entry points of the packets in the mesh network, either from Internet, or from clients directly connected to these nodes, in case of peer-to-peer communication. Router 5 must forward the packets differently depending on the entry point (either to node 6, if the source is node 1 or to both node 6 and 7, if the source is node 2).

[1]Traditional routing is destination based, that is, the destination address will determine which outgoing link is used by a router when forwarding that packet.



Node 5 routing rules

Source	Destination	Next−Hop(s)
Node 1	Client 1	6
	...	...
Node 2	Client 1	6, 7
	....	...

Figure 5.2: The routes to a mobile client when redundant multipath routing is used for mobility support. Client 1 is experiencing a handoff between node 6 and 7, thus, the traffic towards him must be forwarded to both nodes. Router 5 forwards packets differently depending on the entry point.

Therefore, during the routing process, a mesh node must decide what are the next hops for a packet based on the mesh *entry point* as well as the *destination address* of the packet. However, the entry point cannot be determined by just looking at the packet destined to the client (the source address in the IP header is not the address of the mesh entry point, but the actual address of the sender, which can be another mesh client or an Internet address). One solution to keep track of the entry point is to tunnel each packet from the entry point in the mesh to the mobile client. However, to maintain client access transparency, we need to instruct the kernel to remove the tunnel in the last hop, right before sending the packet to the client. That requires new kernel functionality. Otherwise, the mobile client may discard these packets. Another, less obvious approach, is to encode the mesh entry point in some of the existing space in the IP header of the packet. Specifically, we encode the IP address of the entry point into the identification field from the packet's IP header (also referred to as IPID). This is a 16-bit field used to identify the fragments of the IP datagrams. Together with the offset field, it is used by the IP layer to reassemble the fragmented datagrams. We discuss

the implications of using this field, in the presence of fragmentation, in Section 5.3.2.

Multiple routing tables. For a given multicast group associated with a client (referred to as Client's Data Group in Chapter 4), there is one multicast tree for each possible entry point in the network (Figure 5.2). That is, in the most general case, we need to maintain N multicast routing trees for each client, where N is the number of mesh nodes in the network. This led as to the following approach. We define a routing table in the kernel for each access point in the mesh and include in each table an entry for each existing multicast group (Figure 5.1). When a packet comes in, we first choose what routing table to use, and then we forward it according to the entry that has the client address as the destination.

For the example presented in the Figure 5.2, router 5 will use forwarding table 1 if the packets come from entry 1 and table 2 when the packets come from entry 2.

In addition to these routing tables the routing daemon space needs to maintain one additional routing table that corresponds to the anycast group used to route traffic from the clients to the closest Internet gateway.

Multiple next-hops. The last key element of our design refers to the *routing rules* that populate the aforementioned routing tables. These rules differ from the ones we see in traditional unicast routing. Instead of a single next-hop, a rule may have multiple next-hops. This is because our rules reflect a routing *tree* and not a single path.

In a nutshell, our routing architecture can be summarized as follows:

i) Maintain multiple routing tables, one for each node in the mesh network (or at least for each Internet gateways is peer-to-peer communication is not considered).

ii) In each table add a route entry for each possible destination, i.e., for each client. This entry may include multiple next-hops, depending on the multicast trees determined by the routing daemon.

iii) Encode the entry point in each packet's IP header when the packet is first seen in the mesh network. Every node along the path uses this information to select what routing table to use in routing that packet.

iv) Finally, forward the packet at the kernel level, according to the entry that has the client address as the destination in the previously selected routing table.

5.2 Implementation

We implemented this routing architecture in Linux, using Netfilter [2] modules, and deployed it on Linksys WRT54G routers running the open source OpenWrt firmware [4]. We present here the main elements that enabled this implementation in a Linux-based system. While the architecture is generic in nature, the implementation is operating system specific, thus, other operating systems may require slightly different mechanisms.

5.2.1 Packet Marking

As we have seen, as a packet travels along the path to the mobile client, it is being routed based on the entry point in the mesh, which is encoded in packet's IP header. Specifically, at the entry node in the mesh, if the destination of a packet is a client, then the identification field (IPID) is set to be the last byte from the IP address of the corresponding mesh entry node.

To alter the IPID field of the IP header, we wrote a Netfilter target module (`ipt_IPID`). However, only the entry point of the mesh network modifies this field—intermediate routers must leave it unchanged. To make a distinction between the original sender and the rest of the routers along the path we use a bit from the *Differentiated Services Code Point* (DSCP) field to indicate if the IPID field was already overwritten at the entry point, or not. Specifi-

cally, we use the *second* bit from the DSCP field. If the bit is unset then a router will update both IPID and DSCP fields, otherwise it will not.

Going into more details, altering the packet header is done with the following *iptables* rule:

```
# iptables -A PREROUTING -t mangle
           -d $MESH˙NET/8
           -m tos --tos 0
           -j IPID --set-ipid $MESH“˙NODE“˙ID
```

(The rule can be translated to: *“If the destination address of the packet is inside the mesh network mask (i.e., a client), and the TOS (the old name for DSCP field) is not set, then alter the IPID field to be the current node’s identifier”*.)

At the kernel level, the selection itself of which routing table to use, given the IP encoded in IPID, is done using *fwmark*, a tag carried by the kernel as the packet travels through the kernel networking stack. This tag is exposed by the packet filtering mechanism from the Linux kernel. We set this tag to reflect the IPID value from the IP header. The following *iptables* rule tags a packet whose IPID is 35 with *fwmark* 35.

```
# iptables -A PREROUTING -t mangle
           -m u32 --u32 ”2&0xFFFF=35”
           -j MARK --set-mark 35
```

(*u32* is an existing Netfilter module that facilitates checking any bytes from the packet against certain values. In translation: *“Grab 4 bytes starting with byte 2, apply a mask of 0xFFFF and check the result against value* 35*. If it matches, set fwmark to value 35”*.)

Note that these Netfilter rules that set *fwmark* are added/deleted at run-time since all possible entry-points are not known in advance.

5.2.2 Policy Routing

One of the key elements of our design is the use of multiple tables in the packet forwarding process. Fortunately, since version 2.2, the Linux kernel permits defining multiple routing tables and has support for policy routing (a.k.a. rule based routing), which allows selecting different routing tables based on criteria other than the packet destination address. One of these criteria is the tag carried by the packet in the kernel, *fwmark*.

The following command defines a rule that routes all packets marked with value 35 (i.e., packets with IPID set to 35) via routing table number 35:

```
# ip rule add fwmark 35 table 35
```

As before, these rules are added at run-time, as all possible entry points are not known in advance.

5.2.3 MULTIHOP Kernel Module

Normally, a routing table specifies a single forwarding action to be taken in a deterministic manner for a given packet. The `CONFIG_IP_ROUTE_MULTIPATH` option in the kernel configuration permits specifying several alternative paths to a destination. If no weight if given, the kernel considers all these paths to be of equal cost and chooses in a non-deterministic way which *one* to use when a packet arrives. Instead, we would like to send the packet to multiple nodes *simultaneously*, if the multicast tree indicates that.

We wrote a Netfilter target module (`ipt_MULTIHOP`), which sends a copy of the packet to each next-hop found in the routing rule for a given destination. Consider the following rule to the mobile client:

```
# ip route add 10.233.59.169/32 table 35
              nexthop via 10.0.11.32 dev eth1
              nexthop via 10.0.11.33 dev eth1
```

(Which can be read as: *"Using routing table number 35, set mesh nodes 10.0.11.32 and 10.0.11.33 as the next-hops for the packets whose final destination is client 10.233.59.169."*)

In this case the `MULTIHOP` module duplicates the packet and sends it to the next-hops not chosen by the kernel in the routing process. That is, the native kernel routing process sends the packet to one of the next-hops, while the `MULTIHOP` sends to all additional ones. The general steps performed by the module for each next-hop found in the routing table, are the following:

(1) If the next-hop is the one chosen by the kernel, then stop and go to the next next-hop.

(2) Create a copy of the packet.

(3) Search for the next-hop IP address in the neighbor cache maintained by the Address Resolution Protocol (ARP) to get the hardware address (MAC) of the next-hop.

(4) If found: copy to the packet buffer and send the packet.

(5) If not found: send an ARP request and queue the packet in the device queue, unless the next-hop is a tunnel, in which case the packet is sent directly using the device send function.

In order to use the module, one needs to recompile the kernel to export a function required to access the routing table (*fib_lookup*). Other than this, no changes are required in the kernel. The module is available for download at `http://smesh.org`.

5.2.4 Path of a Packet Through the Linux Kernel

Figure 5.3 shows the path of a packet through the Linux kernel and the places where it interacts with our scheme. Immediately after the packet enters the network, the entry-point of the mesh must change the IPID field and set the DSCP bit. Both the entry-point and

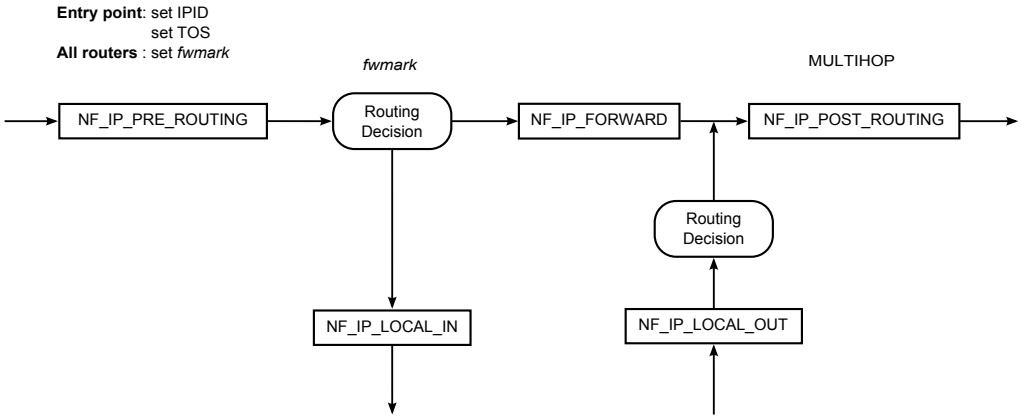


Figure 5.3: The actions performed on a packet while it travels through the Linux networking stack. The entry point alters the IPID and TOS fields. The rest of the routers tag the packet using the encoded entry point and use this tag to select the appropriate routing table. Before leaving the network interface, the packet is processed by the MULTIHOP module, which creates additional copies if necessary.

any intermediate router set the *fwmark* when processing a packet for routing We use the NF_IP_PRE_ROUTING Netfilter hook to do these modifications. The packet is then passed back to the kernel networking stack, where it goes into the routing mechanism in which the *fwmark* is used to choose the appropriate routing table. After the routing decision is taken, but before leaving the interface, the packet reaches the MULTIHOP module. Additional copies of the packet are created if there is more than one next-hop in the route. The module will simply exit if there is only one next-hop. We register the MULTIHOP module at the NF_IP_POST_ROUTING Netfilter hook, such that there will be no routing decisions afterwards.

Note that if the destination of the packet is in the network's private space address[2] (i.e., a mobile client), the packet is not routed as above; instead it follows the default route from the main routing table, as described in Section 5.1.

[2]We consider here the packet as it is after the network address translation (NAT) is performed by the Internet gateway. A P2P packet does not need address translation, as the destination is inside the same network.

5.3 Other Considerations

5.3.1 Multiple Gateways

In large wireless multi-hop networks, having multiple Internet connected nodes is a necessity. To minimize the usage of wireless links but also to reduce latency and increase routing stability, the wired links are utilized whenever possible. We describe here how the existence of multiple Internet gateways impacts the peer-to-peer communication between mesh clients in our system.

Figure 5.4 shows the path of a packet from client `10.A.B.C` to client `10.X.Y.Z` (private addresses in our mesh network). The packet is generated by the first client, having an IPID assigned by its own networking stack, and the TOS bit unset. As soon as the packet acquired by the access point, the IPID is changed to the value of the router ID and the TOS bit set. This will prevent any intermediate router to alter these values. Eventually, the packet arrives at Internet Gateway 1. In order to maintain the destination address of the IP header, we set up an IP tunnel between gateways, so that an additional IP header is used for the packet to be transported to Gateway 2. When the packet arrives at Internet Gateway 2, the tunnel header is removed and the packet continues its trip to the final destination (as specified in the original IP header), using the multicast tree indicated by its IPID. Therefore, the modifications done on the IP header only affects packets inside the mesh, without interfering with the Internet traffic.

The additional overhead caused by IP tunneling only affects the wired part of the network, which is likely to have higher capacity than the wireless mesh.

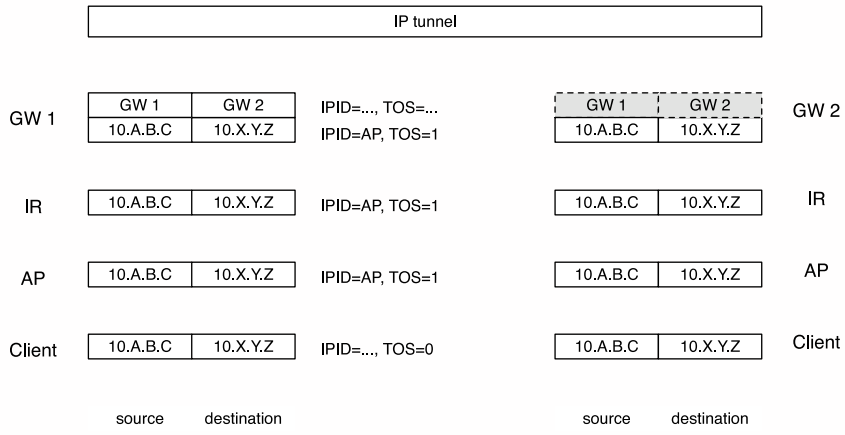


Figure 5.4: Changes in the header of a packet that travels between two clients in the mesh, when two Internet gateways are used to shortcut a wireless path. GW=Gateway, IR=Intermediate Router, AP=Access Point.

5.3.2 Fragmentation

Even if very convenient, modifying the IPID of the packets in order to encode the entry point in the network interferes with fragmented traffic. However, current studies show that IP packet fragmentation is not commonly used today, and it amounts to 1% to 2% [27] of the overall traffic. While advocating for or against the use of fragmentation [48] [31] is outside the scope of this work, we choose to ignore the mesh entry-point when the packet is fragmented, and forward it through a single path.

Other solutions, that allows both fragmentation and the of mesh-entry points, are possible. For example, a practical workaround is the following. Instead of using all 16 bits from the IP identification field, we could utilize only the most significant 8 bits (or less) to encode the entry point. The remaining bits will continue to represent the fragment identifier of the packet, specifically the 8 least significant bits. This will allow the correct reassembly of the packets, since the same operation is performed on all the fragments of the packet. On the down side, the identifier wraps around every 256 packets instead of the regular $65,536$. How-

ever, considering the small amount of fragmented traffic, we believe this is a practical way to support it alongside our system.

5.3.3 Limitations

A restriction one needs to be aware of is that the Linux kernel currently allows specifying up to 255 routing tables when policy routing is used. This limits the number of mesh entry points to 254 (one routing table is reserved for driving the traffic from the clients towards the Internet). In large-scale mesh networks, a practical approach to overcome this limitation is to maintain routing tables only for the Internet connected mesh nodes. This makes peer-to-peer communication between two clients sub-optimal, because the clients will communicate with each other via their closest Internet gateways instead of a multi-hop wireless path. On the up side, if the clients are not very close to each other, then the optimal path is actually via the Internet gateways.

The maximum number of clients supported by our architecture is limited only by the internal memory of the routers. We maintain one entry in each routing table per client, which requires a total of 32 x N bytes in kernel memory, where N is the number of nodes in the mesh network. As an example, the Linksys WRT54G wireless router has 16 MB of RAM, and assuming only 5 MB is available to be used for the routing, we could theoretically support at least 9,000 mobile clients. In reality, this number is much greater because an entry is added in a routing table only if the router is on the path towards that client. As the size of the mesh network increases, more routing tables need to be maintained; however, as the clients are likely to be spread evenly throughout the network, the number of entries maintained by each router does not significantly grow.

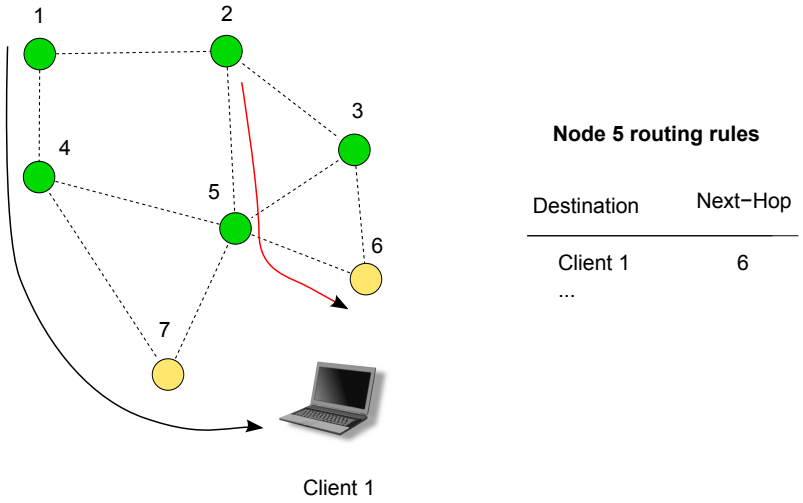


Figure 5.5: The routes to a mobile client when unicast forwarding is used. Client 1 is experiencing a handoff between node 6 and 7, however, packets are delivered only to one of them.

5.4 Alternative Approaches Using Current Operating System Support

Kernel-level unicast routing. Due to the limitation of using only one outgoing link, unicast forwarding cannot, in fact, route through multiple redundant paths. That is, with unicast forwarding we can forward packets only to one of the access points serving the mobile client.

Figure 5.5 examines the same scenario as in Figure 5.2 and shows the unicast routes that could be used to forward the packets flowing from node 1 and node 2 to the mobile client. For both streams sonly one of the access points nearby the client (6 or 7) will receive the packets, and not both, as intended. In addition, router 5 always forwards the packets to router 6, regardless of the entry point. For efficiency, we forward packets to the closest AP handling the client. Other ways to compute the routes are possible, but the routing protocol must carefully consider the race conditions that may arise.

Although unicast forwarding benefits from any performance improvement that kernel

Link #	Multicast	Multicast with background traffic	Unicast	Unicast with background traffic
1	52	108	0	0
2	134	249	0	1
3	114	678	0	77
4	16	94	0	0
5	158	246	0	1

Table 5.1: Number of lost packets on different links when transmitting unicast and multicast packets on 802.11 radios, with and without background traffic. The background traffic consisted of a 1 Mbps stream covering the complete mesh.

routing may provide, it can only send to one of several destinations. This limitation can result in loss during handoff for two reasons. First, route disruptions will occur while the network is converging to a new route and in-transit packets will likely loop and be dropped by the network. Second, the signal quality of the mobile client may drop off rapidly from one of the APs used during handoff. Any packets forwarded from this AP to the mobile client will likely be lost.

Kernel-level IP multicast routing. IP multicast is a communication primitive commonly supported by Unix-like operating systems that allows a sender to efficiently send a packet to several destinations simultaneously. Redundant multipath routing can be provided with native IP multicast by making the access point(s) handling a client members of an multicast group (not only in the overlay, but also at the kernel level multicast routing table).

Natively, IP multicast reaches all destinations that reside in the same LAN or are within wireless range with a single transmission. In 802.11 networks, however, multicast packets do not benefit from link-layer retransmission mechanism. Due to the lossy nature of the wireless medium, especially in a mesh network for mobile clients where collisions are more likely to occur, the number of lost multicast packets can be much greater than the number of lost unicast packets.

The impracticability of multicast in wireless networks can be seen in our testbed. Table

5.1 shows the number of lost packets at different links when using unicast and IP multicast to transmit packets. We sent 1,000 packets on different links at different times, with and without background traffic in the network. The wireless routers had default retransmit limit of 4 for unicast packets. As we can see, the loss rate for multicast streams varies between 1% and 67% while unicast streams experience at most 7% loss.

Kernel-level IP multicast routing with unicast tunnels. Some IP multicast protocols are capable of using unicast tunnels on each link. These protocols, overviewed in [13], were mainly developed to support multicast on wide area networks. Achieving redundant multipath with these protocols requires two levels of indirection per packet. First, since the destination of the packet is the IP of the actual mobile client, we must create a mesh end-to-end tunnel with the mesh-node that first sends the packet as the source and the multicast address that is used to manage the client as the destination. Then, another unicast tunnel, commonly known as virtual interface, must be created for each link in the mesh. Both tunnel headers must be stripped out before sending the original packet to the mobile client, thus allowing seamless end-to-end communication. Managing these tunnels adds unwanted complexity to the system, requires modifications of the packet at each hop (replacement of one tunnel header with another one), and also increases the packet size with two tunnel headers.

User-level overlay routing. User-level overlay routing allows users to implement any protocol without requiring any special support from the kernel. In early stages of development, SMesh routing architecture used an user-level overlay system. While very convenient, routing the entire traffic through user space is challenging for resource-constraint devices. In Section 5.6 we evaluate the performance of user-level routing using off-the-shelf Linksys WRT54G routers, and show that even on a single hop, using only a wired connection, the maximum forwarding throughput achieved is about 2.1 Mbps due to CPU saturation.

5.4.1 Discussion of Tradeoffs

Table 5.2 summarizes the tradeoffs of the routing mechanisms we presented in previous section. We compare them in terms of performance, packet overhead, and amount of kernel modifications required to implement them. Since we focus on wireless mesh networks, we also consider wireless reliability (the ability to take advantage of the 802.11 link-layer retransmissions) and path redundancy for mobility among the metrics. As we show in Section 5.5, generic redundant multipath support may be useful in other kinds of applications, therefore, we include this feature in the comparison, too.

Among the kernel-based routing mechanisms, unicast routing (first alternative) seems to be the easiest approach but it gives up the redundancy in favor of using kernel unicast forwarding to achieve high throughput. It does not add any overhead per packet and it can provide good performance depending on mesh topology and client movements.

The first IP multicast-based approach, while providing redundancy, cannot be a viable option for wireless environments where the loss rate, even between two adjacent access points, is high. In addition, this method requires one tunnel header in which the entry point in the mesh network is encoded. For generic applications such as those presented in Section 5.5, this method cannot provide path redundancy because a tree is not a good abstraction for these situations[3]. This drawback is also shared by the second type of multicast-based protocols, those in which we can use an additional hop-by-hop tunnel to trigger 802.11 link-layer retransmissions. This tunnel header must be replaced at each hop along the path of a packet.

The user-space overlay routing, while adding an overlay header per packet, is advantageous when processing power of the routers is not an issue. As it is based on unicast hop-by-hop forwarding, it benefits from 802.11 link-layer retransmissions, thus providing wireless

[3]It is however for mobility in wireless mesh networks, where the access points can join a *multicast* group to receive the packets.

reliability. Fully implemented in user-space, this approach does not require any special kernel support and can run over a large set of platforms. However, when low-cost routers come into play, the performance hit is quite significant. In our tests, one-hop throughput degraded from about 10 Mbps to 2 Mbps.

Our redundant multipath approach aims to address the drawbacks associated with previous solutions by combining the benefits of both user-space and kernel-space. The approach does not add overhead per packet and achieves high performance by taking advantage of kernel routing. However, unlike all previous options, it does require minimal kernel modifications. Specifically, one function from the routing subsystem needs to be exported, and two new kernel modules need to be installed. The rest of the implementation is in the routing daemon (user space). While some minimal processing overhead is associated with this approach, the difference in performance over native unicast kernel routing is not significant, as experiments will show. Similar to the overlay approach, it provides a generic redundant multipath support for various applications.

Routing Mechanism	Performance	Redundant Multipath for Mobility	Generic Redundant Multipath	Wireless Reliability	Packet Overhead	Kernel Modifications
Unicast (Shortest Path)	high, kernel level	no	no	yes	no	no
Multicast 1	high, kernel level	yes	no	no	one tunnel header	no
Multicast 2	high, kernel level	yes	no	yes	two tunnel headers	no
User-Space Overlay	limited, user level	yes	yes	yes	overlay header	no
Redundant Multipath	high, kernel level	yes	yes	yes	no	very little

Table 5.2: Tradeoffs between several routing mechanisms using existing operating system support.

5.5 Additional Applications for Redundant Multipath Routing

Traditionally, redundancy is equivalent to resiliency, allowing the entire system to function as a whole when some of its components fail. In the case of routing, protocols are in general

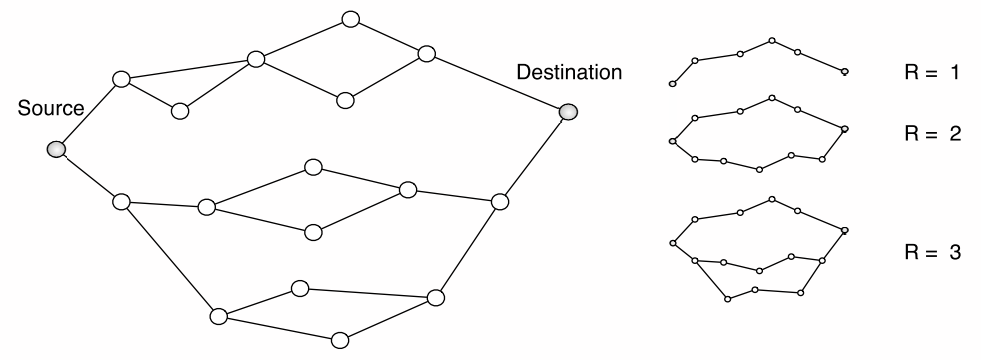


Figure 5.6: Generic redundant multipath. The packets between the source and the destination nodes can be routed with different levels of resilience (R). The resilience level is encoded in the header of the packet.

self-healing, in the sense that, upon detection of a route failure, they will search and/or establish a new routing path if one exists. The challenge lies in the process of detecting the failure, which usually involves a timer to expire after a relatively large amount of time. While reducing these timeouts is not desirable in routing protocols, as they may generate dangerous oscillations, we believe that using redundant multipath forwarding in case of uncertainty offers a viable solution for resilient routing.

Using the principles of our routing architecture, wireless networks could route packets with different levels of resilience, on a per-packet basis. Consider the network shown in Figure 5.6. The source can reach the destination via multiple paths. These paths may be disjoint and the number of paths may depend on the current state of the network and the levels of resilience needed. The graph shows the paths for three possible levels of redundancy. Similar to our architecture, different routing tables can have entries that correspond to different levels of redundancy. Then, each packet can carry its desired reliability level in its IPID so that each router is able to forward the packet using the appropriate forwarding table.

Redundant multipath can also be used to send time-sensitive information in hostile environments where nodes may be compromised by an adversary. While existing Byzantine

routing protocols [21] try to detect bad links and avoid routing through compromised nodes, this detection will take some time. With redundant multipath, the redundancy level of packets can be based on the importance of the data as well as on the current threat level in the working environment.

Wide area networks can benefit from generic support of path redundancy in several ways. Andersen et al. [20] evaluated the benefits of redundant multipath on the RON overlay network, and concluded that 40% of the losses observed were avoidable. Zhao et al. introduced constrained multicast [77] which sends packets through multiple outgoing links when entering a lossy region on a peer-to-peer overlay network. Redundant multipath is also used by Castro et al. [28] in developing a secure routing protocol for structured peer-to-peer overlay networks.

Path redundancy has been shown beneficial in various wireless protocols. Pan et al. [59] proposed an end-to-end smooth handoff scheme for streaming media which sends packets through multiple paths while adapting the streaming data rate to the available bandwidth of the new connection. Gabrielyan and Hersch [39] showed how Forward Error Correction benefits from redundant multipath to increase reliability of real-time streams where there is a constraint on the buffer size of the receiver. Path redundancy has also been used in wireless sensor networks to disseminate critical information with the desired reliability [35].

5.6 Experimental Results

5.6.1 Setup

We evaluate our kernel redundant multipath scheme using low-cost Linksys WRT54G wireless routers running with a third-party OpenWrt firmware [4]. As this firmware was initially built using Linux kernel 2.4, we implemented our kernel modules for this version of Linux.

Some of the experiments were performed using only wired connections, showing the CPU limitation in routers' performance, while other were performed using 17 nodes from our SMesh testbed. In these experiments, the transmit power is set to 50 mW, the short retransmission limit to 7 and long retransmission limit to 4.

In the first set of experiments we show the CPU limitation of both overlay and kernel routing methods, and how it affects the loss rate of the packets as the sending rate increases. Then, we evaluate the *maximum* TCP throughput that can be achieved by both methods in a multi-hop network, as the number of hops increases. Finally, using our deployment, we show throughput and packet latency that can be achieved in our mesh network deployment, in both kernel and overlay schemes.

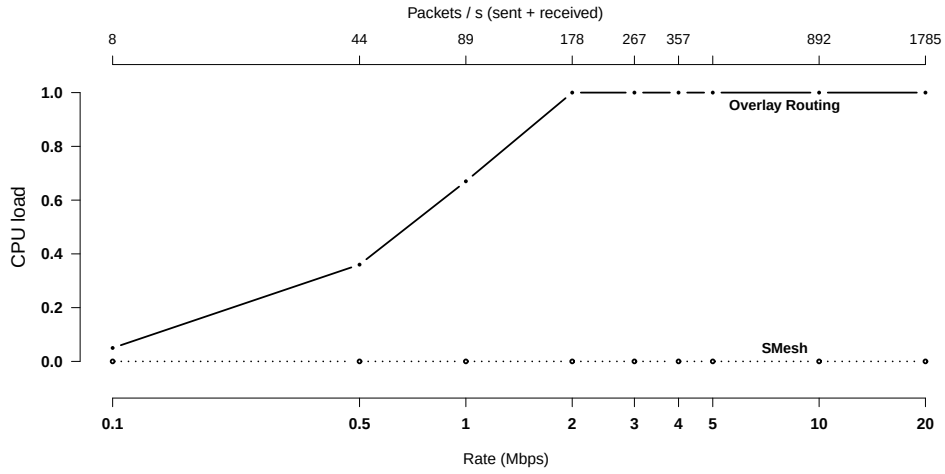
5.6.2 Measurements

Overlay vs Kernel CPU Test

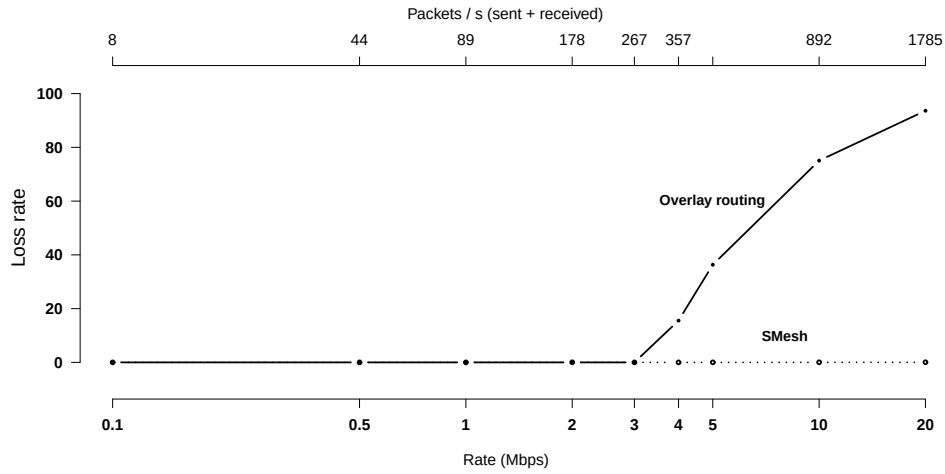
This experiment demonstrates the CPU limitation and its effect on the packet loss rate, for both overlay and kernel methods, as the transmission rate increases.

A client computer was connected to a router that was setup as an Internet gateway. Packets were routed to and from the client computer. To avoid losses caused by the wireless link-layer contention, we performed this test connecting the client to the router with a network cable.

In the first part of the experiment, we sent 1,400-byte UDP packets from the Internet to the client with various transmission rates: 100 Kbps, 500 Kbps, 1, 2, 3, 4, 5, 10 and 20 Mbps. In the second part, we sent 160-byte UDP packets, emulating VoIP streams, at rates corresponding to an increasing number of full-duplex VoIP streams between the client and the Internet. Each (one-way) stream has a rate of 64 Kbps. In both cases we monitored

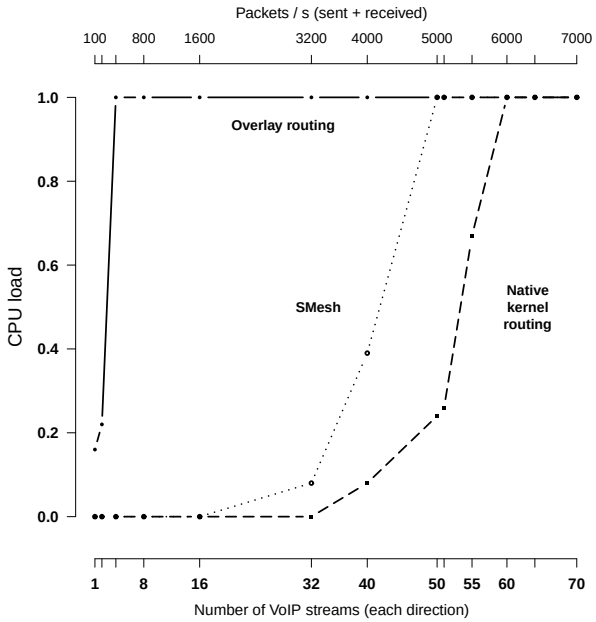


(a) CPU load.

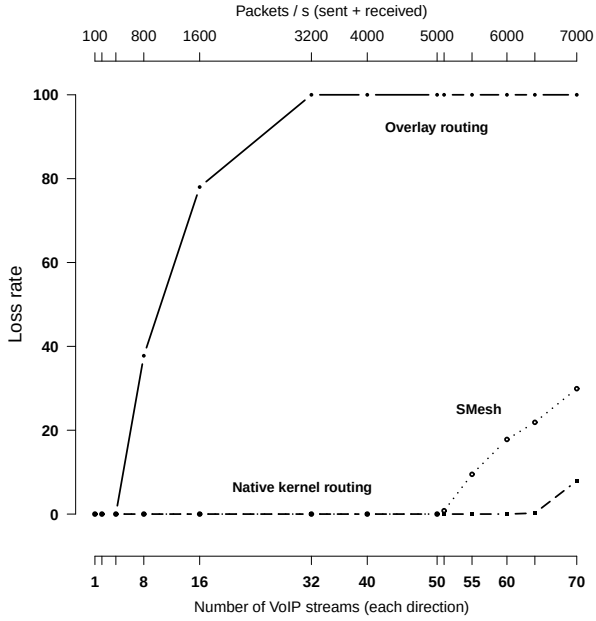


(b) Loss rate.

Figure 5.7: CPU load and loss rates while sending a stream of 1400-byte UDP packets with transmission rates varying from 100 Kbps to 20 Mbps. The top x-axis shows the corresponding number of packets/sec.



(a) CPU load.



(b) Loss rate.

Figure 5.8: CPU load and loss rates while sending an increasing number of full-duplex streams of 160-byte UDP packets. The top x-axis shows the corresponding number of packets/sec.

the average CPU load (Figures 5.7(a) and 5.8(a)) and the loss rate (Figures 5.7(b) and 5.8(b)). The top x-axis shows the corresponding number of packets/sec.

In the $1,400$-byte test, overlay routing consumes about 67% of the CPU at 1 Mbps and it goes up to 100% at 2 Mbps. However, the packets continue to be delivered at 3 Mbps, while at 4 Mbps (about 350 packets/s) the loss rate is already 16% and continues to grow quickly after that. In kernel multicast implementation, we do not see a noticeable increase in CPU load.

In contrast, in the 160-byte test, kernel routing shows an increase in CPU load and becomes saturated when the number of VoIP streams sent by one side is 50, while with overlay routing this happens at only 4 VoIP streams.

To understand better the overhead of our kernel approach, we included an additional scenario: kernel routing without the overhead of iptables rules required by our scheme. We can see that with the overlay implementation, the CPU starts to be saturated at 400 pkts/s (4 one-way VOIP streams, or 512 Kbps), in our kernel implementation at 5,000 pkts/s (50 streams or aprox 6.4 Mbps) while in kernel "native kernel routing" implementation at 6,000 pkts/s (60 streams or aprox 7.6 Mbps). In each of these three scenarios, after a while, the loss rate starts to be non-zero: less than 8 streams (1 Mbps) for overlay, 51 streams (11 Mbps) for kernel and 64 streams (13 Mbps) for kernel "native kernel routing" routing[4].

Overlay vs Kernel Throughput Test

This experiment evaluates the *maximum* throughput that can be achieved in a multi-hop wireless network. We connected 5 Linksys WRT54G routers in a simple "line" topology, and measured the TCP throughput while sending traffic from Internet to the client. Note that this continues to be a very controlled test. We only use one client, and we do not use background

[4]When we refer to the number of VoIP streams, it is the number of streams in one direction, but we send the same amount of traffic in both directions, and the number of packets/second and the throughput is presented as a sum of those.

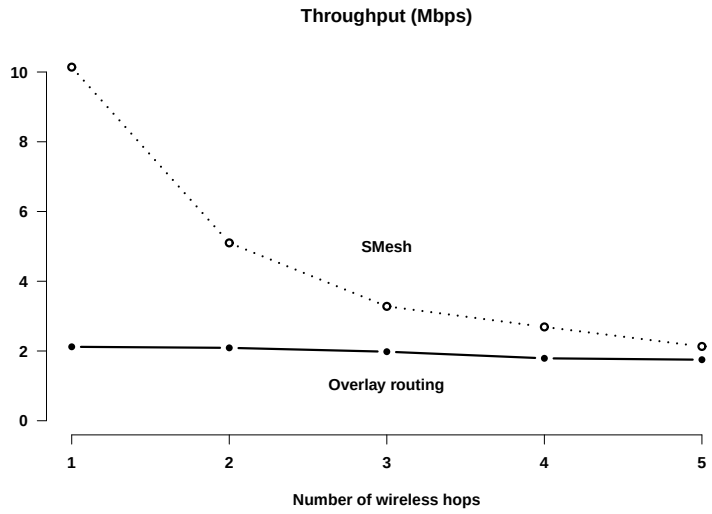


Figure 5.9: The average TCP throughput between the Internet and a client situated at different hops away from the Internet gateway. The routers are in a simple "line" topology.

traffic to influence our results, as our goal is to obtain the throughput upper bound. In the experiment we did not use the RTS/CTS mechanism for collision avoidance, as previous studies show that it does not provide a higher TCP throughput [25] [75].

We performed tests with the client placed 1, 2, 3, 4 and 5 hops away from the Internet gateway. The throughput results are presented in Figure 5.9. We also measured the round trip time (RTT) for both, overlay and kernel routing (Figure 5.10).

With the overlay implementation, the maximum throughput was about 2.1 Mbps for 1 hop and it slowly decreased to 1.7 for 5 hops. In the kernel scheme we obtained a throughput of 10.1 Mbps for 1 hop, which decreased by half, to about 5.1 Mbps for 2 hops. We believe this is due to the influence of inter-hop interference. We notice that even if our scheme yields quite bit of improvement when the number of hops is low, at 5 hops away from the Internet gateway, the difference between overlay and kernel methods is relatively small (2.1 Mbps for kernel versus 1.75 Mbps for overlay). Placing multiple Internet gateways in an wireless mesh network will prevent having a very low throughput by decreasing the number of wireless hops

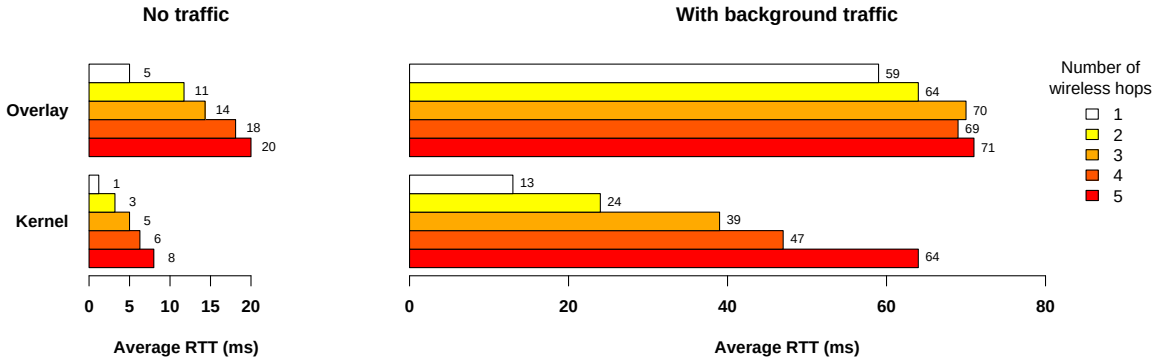


Figure 5.10: The average RTT between the Internet and a client situated at different hops away from the Internet gateway. The routers are in a simple "line" topology.

to the clients.

The round-trip times are averaged over the duration of each test (Figure 5.10). In the overlay implementation, the RTT increases from 59 ms for 1 hop to 71 ms for 5 hops, while in the kernel implementation, it increases from 13 ms to 64 ms for 5 hops. In both situations, we also show the RTT when no traffic is present. We notice again the influence of link-layer collisions, which cause packet loss, trigger 802.11 retransmissions, and increase the packet latency. The round-trip latency from overlay is more than 3 times the one from kernel for 1 hop, and even at 4 hops it is much above the kernel implementation.

Overlay vs Kernel in deployed testbed

This experiment evaluates the TCP throughput and packet latency for a mobile client walking throughout our deployed testbed of 17nodes.

TCP throughput. Figures 5.11 and 5.12 present the TCP throughput achieved over time in both overlay and kernel modes. In order to see how far we are from the Internet gateway, we plot with a dotted line the access point that currently services the client. The horizontal lines mark when the number of hops increases by one. To simplify the graphs, we included only

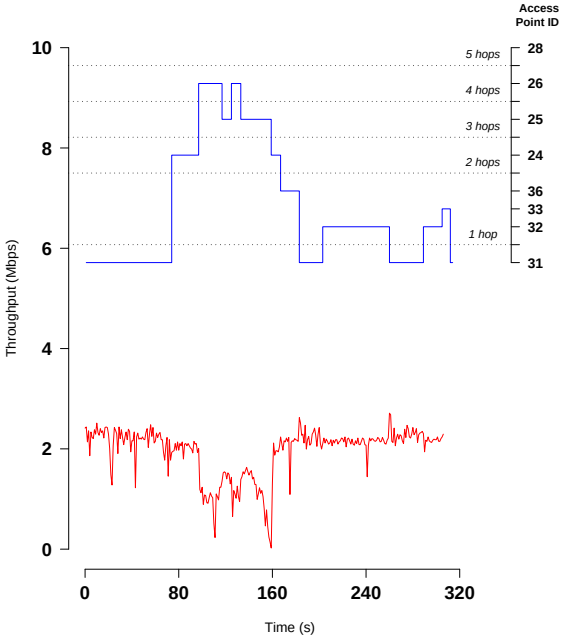


Figure 5.11: TCP throughput of a client moving in the network when user space overlay routing is used. The top line tracks the access point that currently serves the client. The horizontal lines mark when the number of hops increases by one.

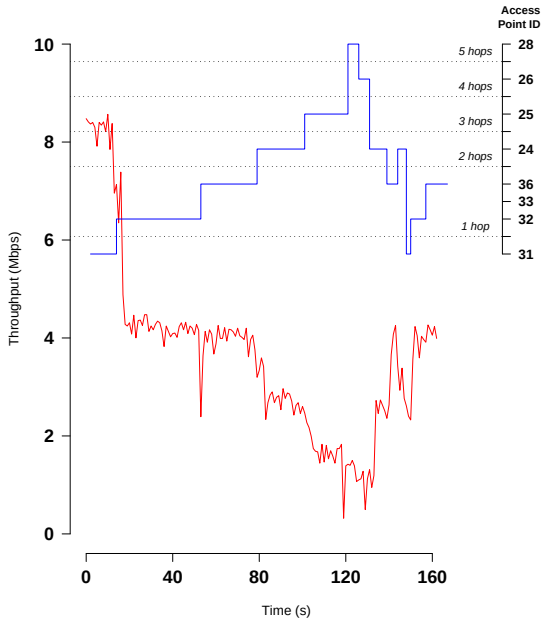


Figure 5.12: TCP throughput of a client moving in the network when our proposed routing architecture is used. The top line tracks the access point that currently serves the client. The horizontal lines mark when the number of hops increases by one.

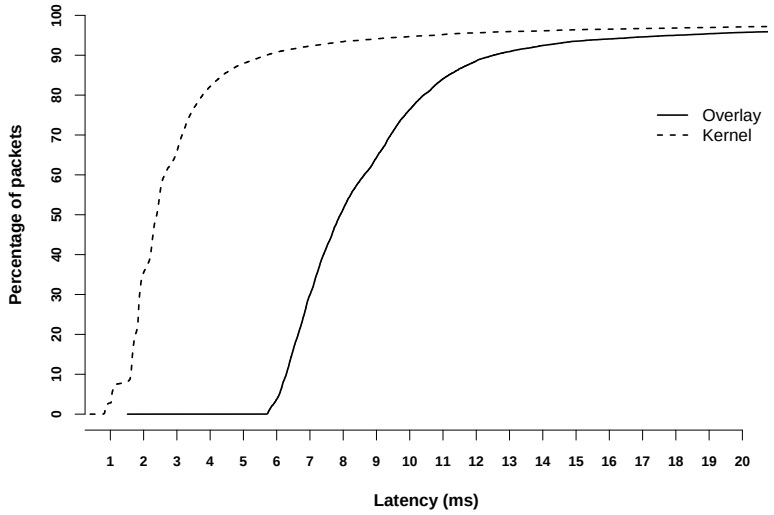


Figure 5.13: CDF of the one-way latency of the packets delivered to a client moving throughout the mesh. The traffic is a full-duplex 64 Kbps UDP stream.

the routers that were involved in handling the client.

With the overlay routing, the throughput is just above 2 Mbps if the client is 1 or 2 hops away from the Internet gateway (routers 31 and 32, 33 and 36). This is consistent with the throughput reported in the previous test. As the number of hops increases, the throughput drops to 1 Mbps and even lower. In the kernel routing, the throughput was about 8.5 Mbps for 1 hop access points, 4.3 Mbps for 2 hops and it drops to 1 Mbps when the client is 6 hops away from the gateway (router 28). In both cases, the wireless link losses prevented us from achieving the throughput obtained in the controlled test (Figure 5.9).

UDP latency. We compare the improvement of packets latencies when routing in overlay and in kernel modes. We use a full-duplex UDP traffic, consisting in 160-byte packets sent every 20 ms at a rate of 64 Kbps, for 5 minutes. We focused on a VoIP-like traffic as a representative application that poses severe latency requirements. As opposed to the RTT presented in the controlled test (Figure 5.10), we measure the one-way latencies. Figure 6.12 shows a CDF

of the packets latency. We can see that about 80% of the packets were delivered in under 10.5 ms using the overlay while in the kernel implementation they arrived within 3.7 ms.

Chapter 6

Application: Push-To-Talk Service for First Responders

Push-To-Talk (PTT) is a well known service in the law enforcement and public safety communities, where coordination and spectral efficiency are key for efficient communication. Some cell phone companies offer a similar service in the commercial world. However, core differences in motivation drive these two sectors. Cellular phone systems are designed for the busiest hour, as outages impact revenue, while public safety systems are designed for worst case scenarios, as outages impact lives.

Unfortunately, first responders cannot always rely on pre-existing ground communication infrastructure. For example, the White House report on hurricane Katrina [9] states that 1,477 cell towers were incapacitated, leaving millions unable to communicate. The report concludes that "The complete devastation of the communications infrastructure left emergency responders and citizens without a reliable network across which they could coordinate."

Wireless mesh networks have emerged as a viable technology that allows for rapid deployment of instant infrastructure [71]. In these networks, mobile clients can roam throughout the area covered by the mesh and seamlessly handoff between access points while utilizing real-time applications such as VoIP [16, 40]. These attributes make wireless mesh networks an appealing technology for first responders. While centralized solutions for providing PTT

service exist (e.g., POC [12]), there are currently no solutions for a robust and efficient PTT service that can be applied in much more dynamic environments such as wireless mesh networks.

Because it relies on half-duplex communication, a PTT system requires an arbitration mechanism (also known as *floor control*), which determines the order in which participants speak. All participants that wish to communicate with each other form a PTT group. As the name suggests, they request to talk by pressing a button. In contrast to peer-to-peer VoIP systems, data must be disseminated from the speaker to *all* the participants in a given PTT group.

Building a robust and practical Push-To-Talk system for the wireless mesh environment is challenging for several reasons. First, it requires the ability to coordinate communication between users even when part of the infrastructure is unavailable (mesh node crashes) or when there is intermittent connectivity between nodes (network partitions and merges). This rules out traditional approaches such as POC, where arbitration is assured by a centralized point. Second, it must operate correctly when users join and leave the network, when they are partitioned away, lose their connectivity, or move from one access point to another. Third, it must use the wireless medium efficiently and should provide low transfer times between users' requests. Last but not least, an important property for first responders is the ability to integrate regular PSTN (Public Switched Telephone Network) and cellular phone users, allowing them to seamlessly participate in the PTT sessions conducted by the wireless mesh PTT service at a disaster site.

We present here the first architecture and protocol of a robust distributed PTT service for wireless mesh networks. Collectively, the mesh nodes provide the illusion of a single third party call controller (3pcc), enabling clients to participate via any reachable mesh node. Mesh users with SIP-based VoIP phones participate by connecting to an IP address that

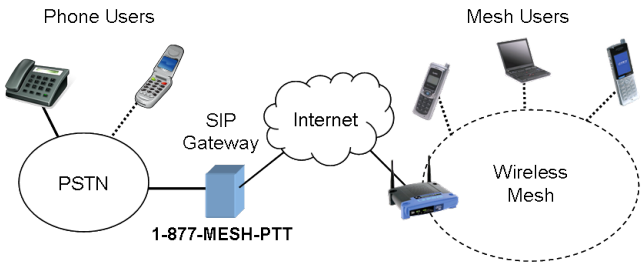


Figure 6.1: Overview of the Push-To-Talk system. Mesh users connect to the system using a SIP-based VoIP application. Phone users connect via the PSTN network to a SIP gateway that routes their calls to the mesh network.

corresponds to the virtual 3pcc server address. In addition, regular phone and cell phone users dial a phone number that connects to the mesh network through a SIP gateway that routes the call to the mesh (Figure 6.1).

In our approach, each PTT group (also referred to as a PTT session) instantiates its own logical floor control manager that is responsible for keeping track of the floor requests of the participants and for issuing *Permission-To-Speak* when a participant releases the floor. Any of the mesh nodes in the network can play the controlling role for a session. To maintain high availability, each controller node is continuously monitored by every mesh node with a participating PTT client and is quickly replaced if it becomes unavailable due to a crash or network partition. The controller relinquishes its role to another mesh node upon determining that this node is better situated (network-wise) to control the PTT session, based on the current locations of the clients participating in the session. In addition to improved performance, this migration increases the availability of the service in the face of network partitions because it keeps the controller in the "center of gravity" of the clients in the PTT session.

The main contributions of this work are:

i) The first robust Push-To-Talk service for wireless mesh networks that can withstand connectivity changes such as node crashes, network partitions, and network merges.

ii) Novel use of multicast for localized access points coordination to share PTT client state, such that the entire network appear to the client as a single call controller.

iii) Novel decentralized floor control protocol that maintains a different logical controller for each PTT session and adaptively migrates it to the most suitable node in the network.

iv) The first architecture that allows regular PSTN phones users (e.g., cell phone users) and unmodified VoIP SIP phones to seamlessly participate in PTT sessions.

We implemented this Push-To-Talk architecture and protocol within the SMesh system and evaluated using a set of 14 nodes from our testbed. In our tests, users experienced less than 150 ms interruption while the system switches between speakers. We show how the system scales to tens of clients, with an overhead of under 1 Kbps per client with 42 clients in the mesh. Then, we show that in our testbed, the system scales to 18 simultaneous PTT groups when dual-radio and packet aggregation are used. Lastly, an elaborate scenario with 40 clients divided among 10 different PTT sessions demonstrates that the system remains highly available during mesh network connectivity changes.

6.1 Push-To-Talk System Architecture

The mesh topology changes when wireless connectivity between the mesh nodes changes, when mesh nodes crash or recover, or when additional mesh nodes are added to expand the wireless coverage. These changes may create network partitions and merges in the wireless mesh.

Push-To-Talk users are regular mesh clients, that is, unmodified 802.11 devices. We do not assume any specific drivers or hardware capabilities present on the clients. Clients connect to the mesh by associating with the wireless-mesh 802.11 SSID. A client should be able to participate with any compliant VoIP application. Therefore, *any* regular unmodified mobile

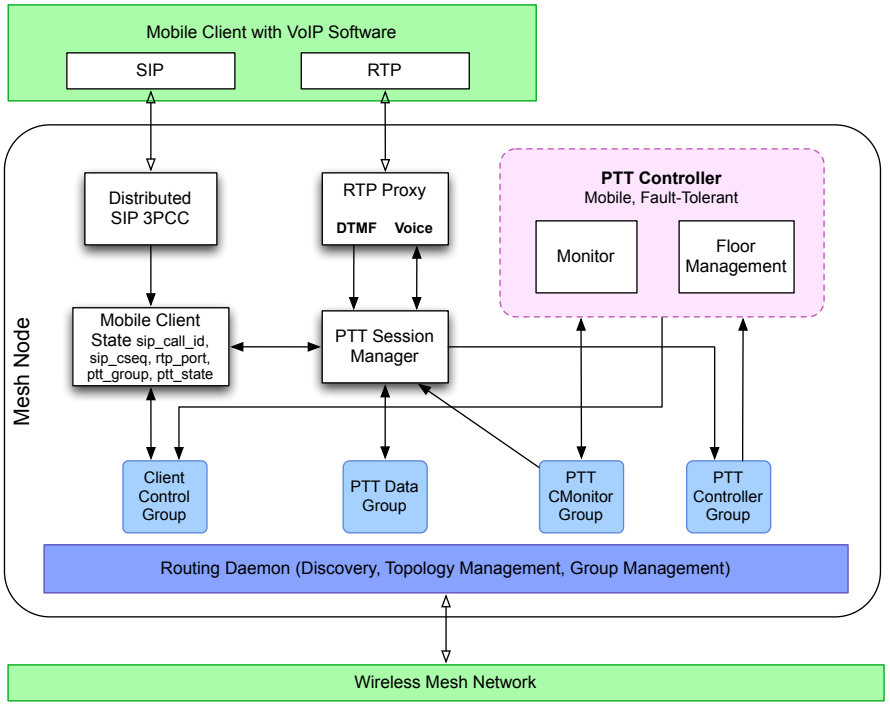


Figure 6.2: Push-To-Talk system architecture.

device should be able to connect to the mesh and use our PTT service transparently.

Regular phones from the Public Switched Telephone Network (PSTN) such as home phones, and cell phones, connect to the mesh by dialing a regular phone number, in our case 1-877-MESH-PTT. The call is routed by the PSTN to a SIP gateway that is connected to the Internet (Figure 6.1). Normally, a regular VoIP client registers with the SIP gateway in order to receive incoming calls. In our architecture, the mesh Internet gateway registers as an end-client with the SIP gateway and routes messages between the mesh and the phones in the PSTN. We do not make any changes to SIP, therefore our protocol integrates with already deployed SIP gateways without any changes.[1]

Figure 6.2 illustrates the software architecture of our PTT system. It includes the inter-

[1]For the SIP gateway we used a service provided by Vitelity (http://vitelity.com), which redirects the packets from the telephone network to our mesh gateway.

face with the mobile client, the mesh PTT session manager for the mobile client, and the mesh PTT controller for each PTT session in the wireless mesh network. Various multicast groups, over which communication takes place, are shown. We benefit from the underlying routing infrastructure, which provides us with overlay group management to effectively communicate on a group-based abstraction.

Each of these components is described in detail in the next sections.

6.2 Interface with Mobile Clients

A mobile client should be oblivious to the heavyweight protocols employed in the mesh network. Further, we want to allow any 802.11 client, as well as PSTN clients, to use the PTT service without changing any of the standards. To do so, our architecture interacts with clients by using well established VoIP protocols.

VoIP applications use the Session Initiation Protocol (SIP [67]), to establish, modify, and terminate a VoIP session. During the SIP session establishment, the Session Description Protocol (SDP [41]) is used to describe the content of the session (i.e., voice), the underlying transport protocol (i.e., RTP[2]), the media format, and how to send the data to the client (address, port, etc). Data is then sent using the designated transport protocol between the parties.

A third party call control (3pcc) server is normally used to inter-connect multiple parties together through a rendezvous point. Conference call managers are one type of 3pcc. Good practices for SIP-based VoIP 3pcc servers are specified in RFC 3725 [66]. In essence, from an end-client point of view, the 3pcc server looks exactly the same as another end-client.

In our architecture, all mesh nodes act as a single 3pcc server and share the state of the SIP connection with every other mesh node in the vicinity of the client (between mesh

[2]Real-time Transport Protocol (RTP) is a standard protocol (RFC 3550) that provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video.

nodes that can hear the client). This is key for the system to scale as it efficiently shares information only between nodes that can potentially need the state of the SIP connection as the client moves throughout the mesh, or in case the client's mesh node crashes.

To participate in the mesh PTT session, the user specifies in its VoIP application the IP address of our virtual SIP server (i.e., *"sip:ptt@192.168.1.10"*). This IP is the same throughout the mesh. Every mesh node intercepts packets sent to this address and follows the SIP protocol to connect the client to the mesh. Therefore, the mesh network provides the illusion of a single 3pcc to the client.

Once a SIP connection is established, the user can start using the mesh PTT service by simply dialing the PTT group that it wishes to join. Each dialed key generates a Dial-Tone Multi-Frequency (DTMF [68]) signal that is sent over the RTP channel (by default, this signal is repeatedly sent over multiple RTP packets to ensure that the end-node receives it). In our approach, we intercept DTMF signals for control purposes between the end-client and the mesh. For example, a client dials *"#12#"* to join PTT group 12. In the same way, every time a user wishes to speak, pressing *"5"* or any pre-defined key combination will be interpreted as a *"Request-To-Speak"* control message. Once the system determines that it is the user's turn, it sends an audio signal (*"beep-beep"*) to let the user know that it can start to speak. While other means for signaling control information are possible, DTMF is supported by most communication networks such as PSTN, allowing us to seamlessly support users from these networks.

RTP data is then sent from the client to the 3pcc virtual IP address through the client's access point (mesh node), which forwards the packets to every mesh node that has a PTT client on that group using a source-based multicast tree. Finally, each receiving mesh node forwards the packets to its corresponding end-clients.

6.3 Push-To-Talk Protocol

Providing a robust and scalable way to coordinate client communication is the essence of the Push-To-Talk protocol. There are several ways to approach it. One possibility is to have a unique point of management in the network that every mesh node needs to contact in order to register a request and get permission to speak. Such a protocol is easy to design and implement and is appropriate for deployment in certain environments. However, this approach is not a good choice for networks that require high availability. For example, if a partition occurs in the mesh, all the clients connected to nodes that cannot reach the arbitration point will be left out of service. At the opposite extreme is the approach of total decentralization in which there is no unique entity that arbitrates the communication. Instead, the nodes in the mesh must coordinate and collectively decide on the order of serving the clients. While more complex, such a protocol is very resilient to infrastructure failures, at the expense of a continuous communication overhead in order to maintain a consistent view between the mesh nodes in the network.

Our PTT architecture uses a hybrid protocol that shares characteristics with both approaches. As in the centralized approach, each PTT session is managed by a controller node which is responsible for keeping track of floor requests and for issuing *Permission-To-Speak* after a participant releases the floor. However, each PTT session has is own controller node and any of the mesh nodes in the network can play the controlling role for any session. The controller node is continuously monitored by other nodes and rotated when a more suitable node (i.e., a node with a better geographical position in the network) becomes available.

In addition, we completely separate floor control from data dissemination. While the arbitration is left to the best node to be the controller, the data is routed optimally to all participants through source-based multicast trees. This allows the system to be efficient and

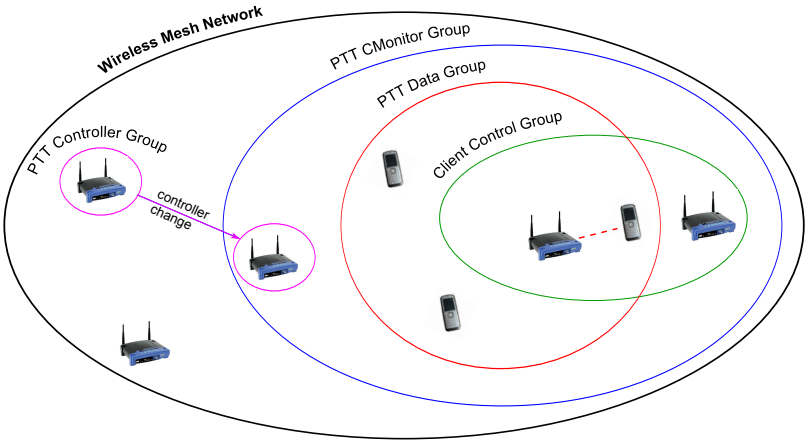


Figure 6.3: The multicast groups maintained by the system to manage a client (Control) and its PTT session (CMonitor, Data, Controller).

scalable.

The details of our protocol are presented below. We start by describing how a mobile client is managed by a mesh node and how a PTT session is managed by a controller. We then present the floor arbitration mechanism. Finally, we show how the protocol withstands node crashes and connectivity changes such as network partitions and merges.

6.3.1 Client Management

For a PTT client, the entire mesh network behaves as a *single* 3pcc server. This is achieved by maintaining the state of the client on the mesh nodes in the vicinity of that client, such that any node that becomes the client's access point (the client is mobile) has the appropriate SIP and PTT information. A virtual IP is assigned to the 3pcc server, and is used by the client VoIP application to connect to.

Specifically, in order to service a client, the system requires information such as the SIP call identifier, SIP sequence number, RTP port, PTT group, PTT state (e.g., the client requests permission to speak, or has permission to speak). We chose to maintain client's state locally

for several reasons. First, there is no single node responsible for the state. Instead, any node that can hear the client maintains a state for it. Thus, the state is preserved even when the client's access point crashes. Second, as the state is maintained in the vicinity of the client, the overhead is localized in the part of the network where the client is located. Finally, the client state is decoupled from the controller node, allowing the clients' requests to be recovered when the controller node crashes (or is partitioned away), as we discuss below.

Client Control Group. To share the client state between mesh nodes that can reach a client, we associate with each client an overlay multicast group. Specifically, any node that can hear the client (that is, not only its current access point) joins and periodically advertises the client state on the Client Control Group (Figure 6.3). In our experiments, we share this information every four seconds. Note that the system is not synchronized and different nodes may see different states for a client at a given time. We use a combination of client timestamps (available in the SIP and RTP packets) and controller logical timestamps to correctly identify the most recent state of a client. This multicast group is an extension of the Client Control Group introduced in Section 4.3.

Using a localized multicast group per client has another benefit: The client is mobile and it can freely move from one access point to another. When the client is granted permission to speak, the controller node uses this group to reach and notify the client, without actually having to keep track of its current access point (a node can send a message to a multicast group without being a member of the group).

6.3.2 Session Management

A client joins a PTT session by initiating a VoIP conversation with the virtual 3pcc server as described in Section 6.2, independently of its network location. In our protocol a PTT session

is coordinated by a controller node, whose presence is continuously monitored by other nodes. The controller relinquishes its role to another mesh node upon determining that this node is better situated (network-wise) to control the PTT session, based on the current location of the clients participating in the session. Three multicast groups are used to manage a PTT session in a distributed manner.

PTT Controller Group (PTT_CONTROLLER). For each PTT session, there is a single mesh node, the *controller*, responsible for managing the floor at a given moment in time. It receives and arbitrates requests and grants the right to speak. In our architecture, when a node becomes the controller for a PTT session, it joins an overlay multicast group associated with that session. Maintaining an overlay multicast group with the controller as the only member allows any mesh node in the network to reach the controller node without actually knowing its identity. Unicast communication with the controller is used by the protocol, however, only in response to a message previously received from the controller. All client floor requests are sent by their access points (mesh nodes) to this group and are stored by the controller in a FIFO queue. Periodically, the controller checks whether another mesh node is more appropriate to manage the PTT session. In such a case, it initiates a procedure to migrate the control to that node.

PTT Controller Monitoring Group (PTT_CMONITOR). This overlay multicast group is used to monitor the controller node. A mesh node joins the monitoring group of a PTT session if it is the access point of a client that participates in that session. In addition, the controller joins this group to detect the presence of another controller during a network merge. A ping message is periodically sent by the controller to this group, allowing its members to monitor controller's presence and take action if the controller is no longer available.

PTT Data Group (PTT_DATA). This multicast group is used to deliver the actual voice data to the clients. A mesh node joins the PTT Data Group of a session if it is the access point

of a client in that session. Thus, we completely separate floor arbitration, coordinated by a single controller node, from data dissemination. This allows us to optimally route data from the sender node to all the participants in a PTT session.

To simplify the management of names for these three multicast groups, we generate their IP multicast addresses using a hash function of the PTT session identifier, such that any mesh node in the network knows which groups are associated with each PTT session without coordination. Similarly, the Client Control Group is generated as a hash of the client IP address.

6.3.3 Floor Control

Requests handling

Figure 6.4 describes how the controller arbitrates the floor such that only one user speaks at a time, while all other users listen.

When a PTT client requests the floor, a `REQUEST_FLOOR` message is sent by its access point to the `PTT_CONTROLLER` group. The controller queues the request and sends back an acknowledgment. Since the messages are not reliable, the access point will retransmit the request until it receives an acknowledgment from the controller, or until the client cancels its request. Note that the acknowledgment is sent on unicast, to the access point of the client.

Release floor requests are sent to the controller in a similar manner. When a `RELEASE_FLOOR` is received, the controller node grants the right to speak to the next client in the queue by sending a `PTS` (*Permission-To-Speak*) message. This message is sent to the client via the Client Control group. We do this because during the life of a PTT session, the client location can change, and it may no longer be associated with the access point that initially issued the request to speak. If the client is no longer available, a simple timeout

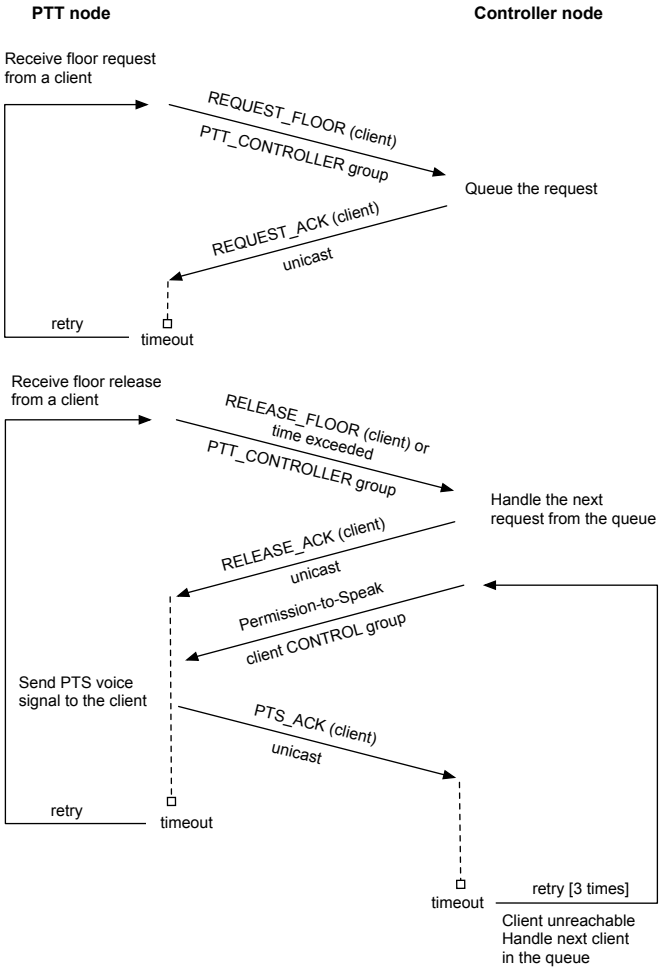


Figure 6.4: The sequences of steps performed by the mesh node and the controller node in order to service user requests. The first part shows how an user requests to speak, while second part shows how the user is notified when its permission to speak is granted.

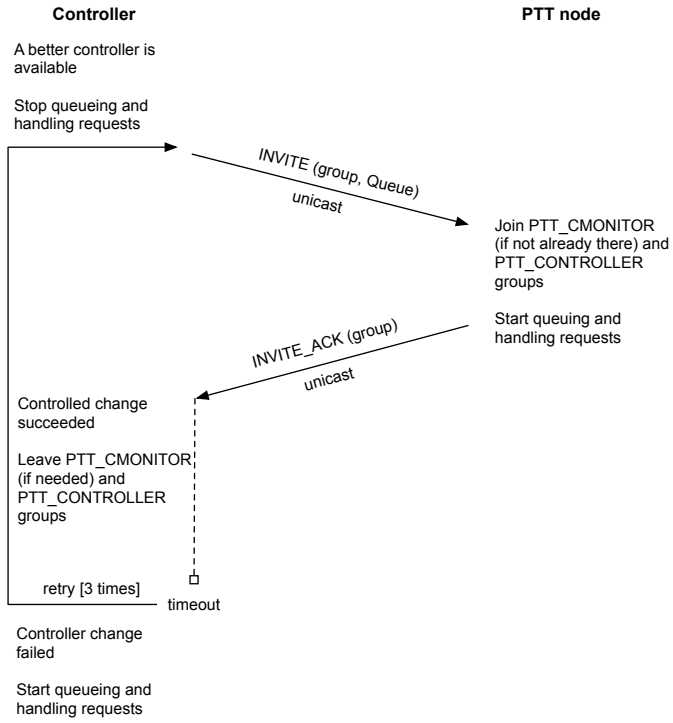


Figure 6.5: The sequence of steps and actions performed to migrate the controller. The migration process is initiated by the current controller, based on existing network conditions.

mechanism allows the controller to move to the next request in its queue.

Migrating the controller

While there is a single controller node for a PTT session at a given time, the system may change the controller over time, depending on participants' placement in the network. The idea is to avoid situations such as when a majority of the clients in a PTT session are localized in some part of the network while the controller node is in another. Placing the controller closer to where most participants are reduces the latency and the amount of control traffic in the network. In addition to improved performance, this migration increases the availability of the service in the face of network partitions because it keeps the controller in the "center of gravity" of the clients in the PTT session. Specifically, the system computes the cost that

each node would incur if it was the controller as the sum of the costs to reach each member of PTT_DATA group. In our experiments we computed this cost every minute. By cost we refer to a wireless metric that may incorporate latency or the number of hops, for example[3]. Note that any node in the mesh network can be chosen to be a controller, regardless if it services PTT clients.

The sequence of steps performed for migrating the controller are as follows: First, the current controller enters a block state, in which it does not respond to any floor requests or releases and does not grant the right to speak to any client. Next, the controller sends an INVITE message to the selected node—the one with the lowest cost to be a controller—which includes the queue of the pending requests. Upon receiving such a message, the invited node joins the PTT_CMONITOR group—in case it was not already a member—and also joins the PTT_CONTROLLER group. It now has the queue of requests and can safely begin controlling the session, queuing new requests and issuing PTS. An acknowledgment is sent back to the initial controller so that it can leave the PTT_CONTROLLER group. In case of a timeout during this process, the original controller unblocks and continues to manage the PTT session.

6.3.4 Protocol Robustness

Due to the inherent instability of the wireless environment, it is possible for the network to partition and merge. While in a well-established mesh network this should rarely happen, a rapidly deployed network during an emergency is likely to experience such problems. In addition, any node in such a network can crash. The PTT service must continue to operate even if it experiences a few seconds of interruption. Below we explain how the protocol is resilient to these conditions.

[3]Additional functionality from was added to retrieve topology and membership information from the link-state and group-state updates, which in turn allows a controller to compute the Euclidean distance from every node to a given PTT group.

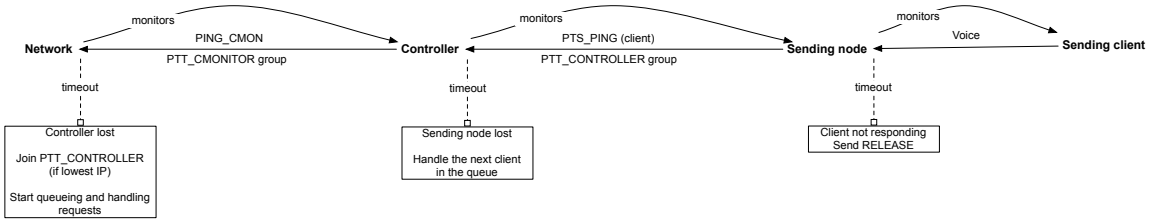


Figure 6.6: Mechanisms that ensure the protocol robustness. The critical components (the controller nodes and sending nodes) are continuously monitored and re-instantiated if the connectivity is lost because of a node crashes or the network partitions.

To operate correctly, there must be a controller and a sending node (that is, a node with a client with permission to speak) for each PTT session. If one of these is missing, either there is nobody to arbitrate the floor or nobody is currently speaking as the system waits for a node which is no longer available. Thus, we introduce the following mechanisms to monitor the operation of each of these two nodes (Figure 6.6). Note that asymmetric links are eliminated by the routing protocol.

Controller node monitoring

The controller node periodically sends a keep-alive message (`PING_CMON`) to the `PTT_CMONITOR` group, allowing other nodes that service PTT clients for that session to monitor its presence. When the controller crashes or is partitioned away, the node with the lowest IP address on the `PTT_CMONITOR` group volunteers to be the controller by joining the `PTT_CONTROLLER` group. However, its queue of requests is empty. We use a special flag in the subsequent `PING_CMON` messages to notify everybody that a new controller was instantiated. All the nodes with pending PTT requests must re-send their requests as if they were new. This is possible because the user requests are part of the user state which is maintained and shared via the Client Control Group. Thus, the controller's queue is reconstructed in a best-effort way, with the requests from the current partition. Note, however, that the order of the requests in the

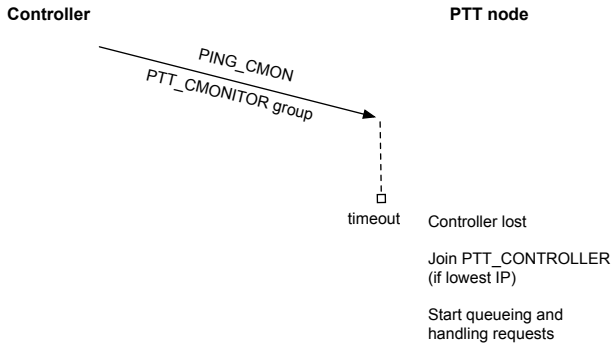


Figure 6.7: The controller is considered lost when its presence messages sent on the monitoring group timeout. The node with the lowest IP address becomes the new controller of the session.

new queue may be different than the one from the original controller. With minimal changes, the protocol can be adapted to recover part of the original order established by the previous controller.

Another situation from which we have to recover is when there are multiple controllers in the network. This occurs after a network merge but also when the controller is lost and multiple nodes decide to control the session (unlikely but possible, as the nodes can temporary have a different view of the network's topology). Figure 6.8 shows the mechanism to recover from this situation. Since the controller node is the only one sending keep-alive messages on the `PTT_CMONITOR` group, receiving a keep-alive that is not its own indicates to the controller that there is at least one additional controller in the network. Once this situation is detected, the node with the lowest IP address remains the controller, while the other(s) must leave the controller's group. A redundant controller sends a `LEAVE_REQUEST` message to the `PTT_CONTROLLER` group with the content of its queue as it leaves the group. Upon receiving such a message, the controller with the lowest IP appends the queue to its own, removing duplicate requests if necessary, and acknowledges the leave.

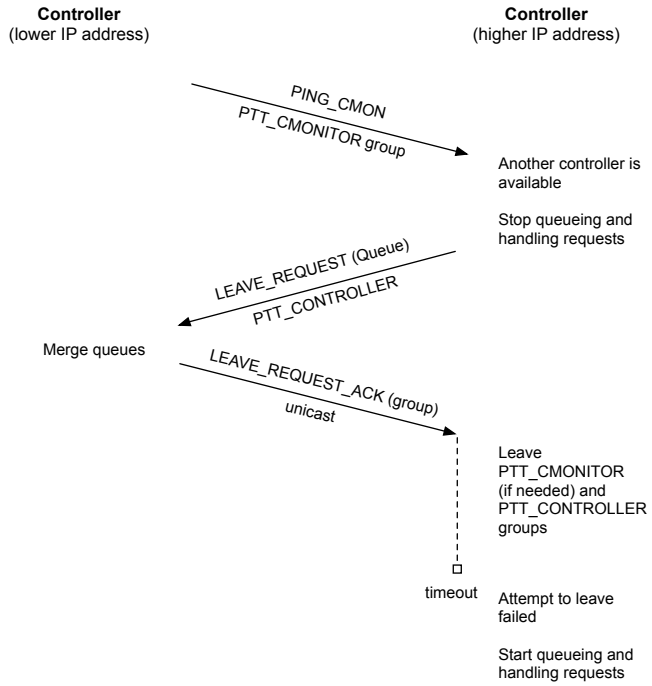


Figure 6.8: Sequence of steps and actions taken to detect and recover from the situation when there are multiple controllers in the network. This happens whenever two partitions of the network merge, or because the nodes may temporary have a different view on the network's topology.

Sending node monitoring

While the members of PTT_CMONITOR group monitor the controller, the controller in turn is responsible for monitoring the sending node (Figure 6.6). The sending node periodically issues a keep-alive message (PTS_PING - *Permission-To-Speak Ping*) on the PTT_CONTROLLER group. This allows the controller to quickly move to the next client in the queue in case of a timeout. An alternative to this approach would be to simply wait for the maximum allotted time per speaker to expire; however, system responsiveness is important in emergency situations, ruling this option out.

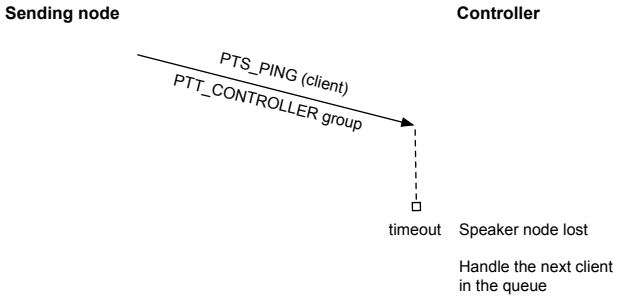


Figure 6.9: The sending node is considered lost when its presence messages sent on the PTT_CONTROLLER group timeout. In this situation the controller will immediately move to the next request in the pending queue.

When two or more network partitions merge, there will be multiple controllers, but also multiple sending nodes in the network. The controller of the newly established network withdraws the right to speak to all additional clients by sending a REVOKE message to their access points (mesh nodes), which in turn notify their associated clients.

Table 6.1 presents a summary of the messages handled by the controller node, including the frequency of the PING_CMON and PTS_PING messages.

Type	Sent by	Sent to	When
REQUEST_FLOOR	mesh node	PTT_CONTROLLER group	client requests floor
RELEASE_FLOOR	mesh node	PTT_CONTROLLER group	client releases floor
REQ_REL_ACK	controller	mesh node	
PTS	controller	CLIENT_CONTROL group	
PTS_ACK	node	controller	
INVITE	controller	mesh node	controller changes
INVITE_ACK	node	controller	
LEAVE_REQUEST	controller	controller	multiple controllers
LEAVE_REQUEST_ACK	controller	controller	multiple controllers
REVOKE	controller	node	multiple speakers
PING_CMON	controller	PTT_CMONITOR group	every second
PTS_PING	node	PTT_CONTROLLER group	every second when client has PTS

Table 6.1: Types of messages sent and received by the controller node.

6.4 Experimental results

6.4.1 Setup

We implemented the Push-To-Talk protocol within the open source SMesh wireless mesh system and evaluated using 14 nodes from the SMesh testbed, consisting of Linksys WRT54G wireless routers deployed across several floors in three buildings at Johns Hopkins University.

Each of the mesh nodes is equipped with one radio configured in ad-hoc mode. The data rate was set to 18 Mbps, the transmission power to 50 mW, and the 802.11 link-layer retransmission limit to 7. Unless specified, the topology of the mesh, depicted in Figure 6.10, was stable.

In all experiments, when a client is granted permission to speak it transmits a 64 Kbps VoIP stream as 160 bytes UDP packets every 20 ms.

Some experiments require a large number of simultaneous clients. To support such experiments, we implemented a client emulator that generated the appropriate control and data traffic associated with the emulated client. From the 802.11 network and from the PTT

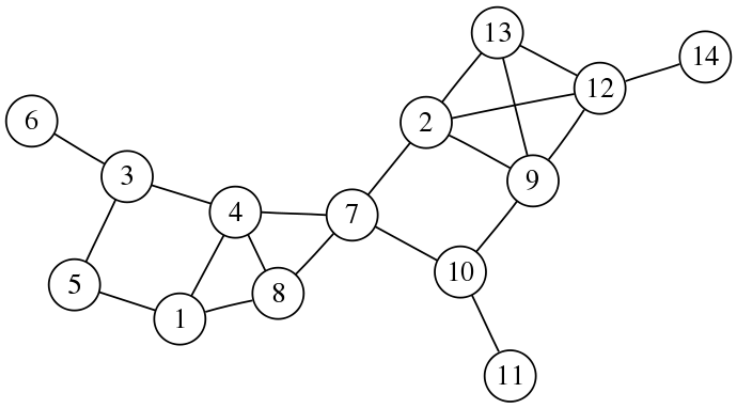


Figure 6.10: The wireless mesh network testbed used in the Push-To-Talk experiments.

system perspective, there was no difference between an emulated client and a real client in terms of control and data traffic. Each client is instantiated on a mesh node, and packets between the emulated client and its access point are always transmitted over the wireless medium. In spite of generating the appropriate amount of traffic in the network, our metrics (such as latency and loss rate) are reported from the mesh nodes perspective. A real client should perceive slightly higher values. Despite this shortcoming, we resort to this method in order to evaluate our system in a real testbed, with a large number of users. In our opinion, though it requires more work and complete implementation, this method is significantly more realistic than simulations, e.g., using *ns-2* or *OpNet*.

6.4.2 Measurements

We present four types of experiments. First, we demonstrate the system's normal operation with a small number of clients. Second, we demonstrate the ability of the system to scale with the number of clients in a PTT group. Third, we demonstrate the ability of the system to scale with the number of PTT groups. Last, we demonstrate the robustness of the system through its ability to handle network partitions and merges correctly while PTT sessions are

in progress.

Normal Operation

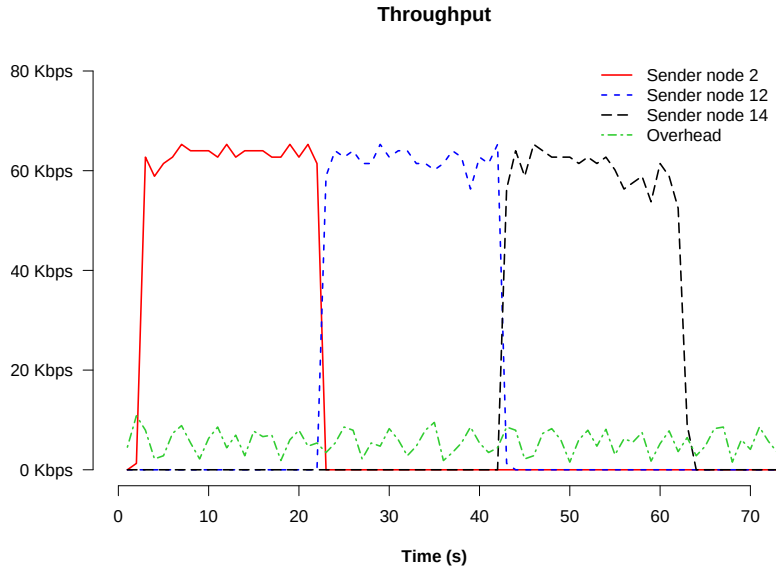
This experiment involves four mobile clients, each of them connected to a different mesh node in the network (nodes 1, 2, 12, and 14 in Figure 6.10). All four clients join a PTT session and continuously request to talk. When a client is granted the floor, it immediately speaks for 20 seconds, releases the floor, and then renews its request. Thus, the PTT session's queue of requests is never empty.

Figure 6.11 depicts the VoIP data throughput and our protocol overhead, as seen by node 1. The overhead includes the control traffic of the PTT protocol as well as the SMesh traffic associated with maintaining the mesh and the multicast groups. This overhead ranges between 1.5 Kbps and 5.8 Kbps, with an average of 3.4 Kbps, which is reasonable considering that each VoIP session is 64 Kbps.

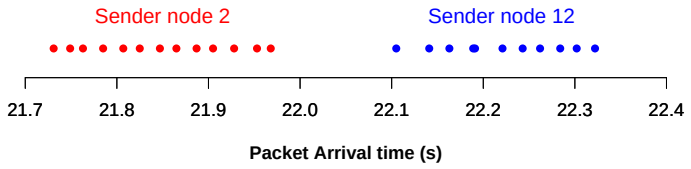
Figure 6.11 also includes a view with the arrival time of each VoIP packet. The figure shows a turnaround time of 137 ms from the moment the last packet from a client is received to the moment the first packet from the next client's voice stream arrives. This demonstrates that only a small part of the time is consumed for synchronizing the PTT clients.

Scaling with the number of clients in a PTT session

To test the scalability of our system, we gradually increase the number of clients participating in a single PTT session. Each client connects to one of the 14 mesh nodes in the network according to a round-robin order of their identifiers (i.e., the first client connects to node 1, the second to node 2, ..., the 14^{th} to node 14, the 15^{th} to node 1, etc.) and requests to speak. Upon acquiring the floor, each client speaks for 10 seconds, releases the floor, waits



(a) Data as seen by node 1. Every 20 seconds the permission to speak is granted to a different user.



(b) Packets received by node 1 while the system transitions between users connected to nodes 2 and 12. The time between the last packet from node 2 and the first from node 12 was 137 ms.

Figure 6.11: Experiment showing the normal operation of the system. Four users are connected to four random nodes in the 14-node testbed.

for another 10 seconds, and requests to speak again. Therefore, at any point, some client is authorized to speak.

We considered three scenarios:

1) Each mesh node is equipped with a single radio.

2) Each mesh node is equipped with dual radios. The first radio is used for communication with other mesh nodes, while the second is used for communication with the users. In this way the mesh traffic does not interfere with the mesh-to-client traffic, as the radios can be set on non-interfering channels. We emulated this dual-radio scenario in our single-radio environment by generating the client's messages locally on the corresponding mesh nodes and by avoiding sending data packets from the mesh nodes towards the clients.

3) Each mesh node is equipped with a single radio and the mobile clients have no PTT support. In our system, one can simply participate in a PTT session using a standard VoIP phone or application. If using G.711, a well supported VoIP codec, a duplex voice stream is transmitted regardless of the data content. By PTT support we refer to the ability of the user's device to control the flow of outgoing packets, e.g, a button to suppress sending unnecessary data when the user is not allowed to speak in the session. A silence suppression technique employed by the codec, on the user's device, would also be a good alternative to reduce unnecessary data.

For the case where clients have no PTT support, there are two main disadvantages that considerably affect performance. First, such clients continuously send VoIP packets, even when not having the floor. These packets are dropped by the mesh node serving the client, except for the durations when the client has acquired the floor. This case incurs considerable overhead as clients send unnecessary packets in their vicinity. The second disadvantage is that a node needs to send individual packets to all the clients directly connected to it, even

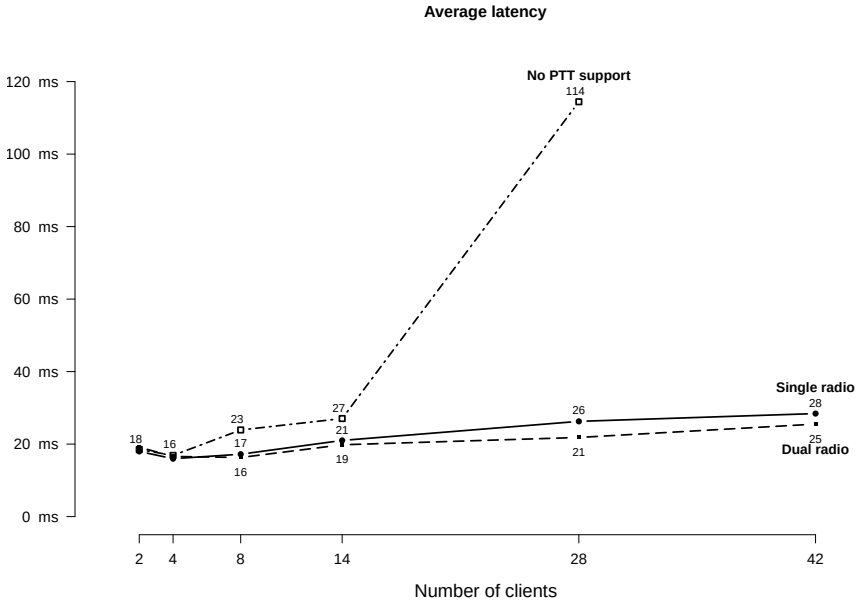


Figure 6.12: Average latency of the packets received by the nodes when the number of clients in a PTT group increases from 2 to 42 (3 clients connected to each of the 14 access points.)

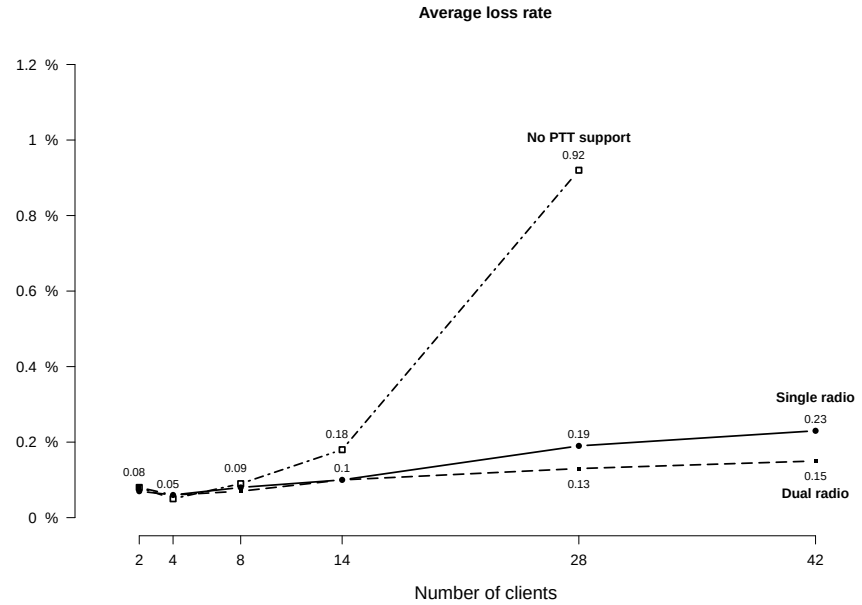


Figure 6.13: Average loss rate of the packets received by the nodes when the number of clients in a PTT group increases from 2 to 42 (3 clients connected to each of the 14 access points.)

if they are on the same PTT session. In contrast, with PTT support clients can use the same multicast address and local port, allowing a single stream of multicast packets to be sent by the mesh node to all of them.

The quality of the PTT service is affected by several factors, the most important being latency, jitter (which influences the responsiveness of the system), and the loss rate of the packets. While the half-duplex nature of the PTT communication makes it much more latency tolerant compared to an interactive VoIP conversation, the loss rate has a direct impact on the voice quality. It needs to be maintained at low levels. [citation needed].

To evaluate the quality of the voice data we use the one-way latency of the packets, and the loss rate. As the clients used in our experiments are emulated on the mesh nodes, both metrics are measured from the mesh nodes perspective.

Figures 6.12 and 6.13 summarize this entire experiment, showing the average latency and loss rate of the VoIP packets received by the mesh nodes, in each of the above three scenarios, averaged over 10-minute tests. In the single radio setup, the average latency of the nodes increases from 15.97 ms for 4 clients to 28.42 ms for 42 (three clients connected to each mesh node). In dual radio setup the latency is slightly lower, varying from 16.59 ms for 4 clients to 25.52 ms for 42. With single radio and no PTT support setup, the latency was significantly higher, because of the large number of packets competing for the medium. With only 28 clients the average latency was 114 ms. Increasing the number of clients in the system beyond this point resulted in a high loss rate (above 1%).

Figures 6.14 and 6.15 illustrate the variation of these average latencies and loss rates of each node. For an easier visualization we used different ranges for the Y-axis, in each of the three setups. For each experiment we show a boxplot, i.e., the 1^{st} and 3^{rd} quartile, the median, the lowest and the highest value. The variation of the latency in general increases with the number of clients in the system, the maximum difference between the highest and

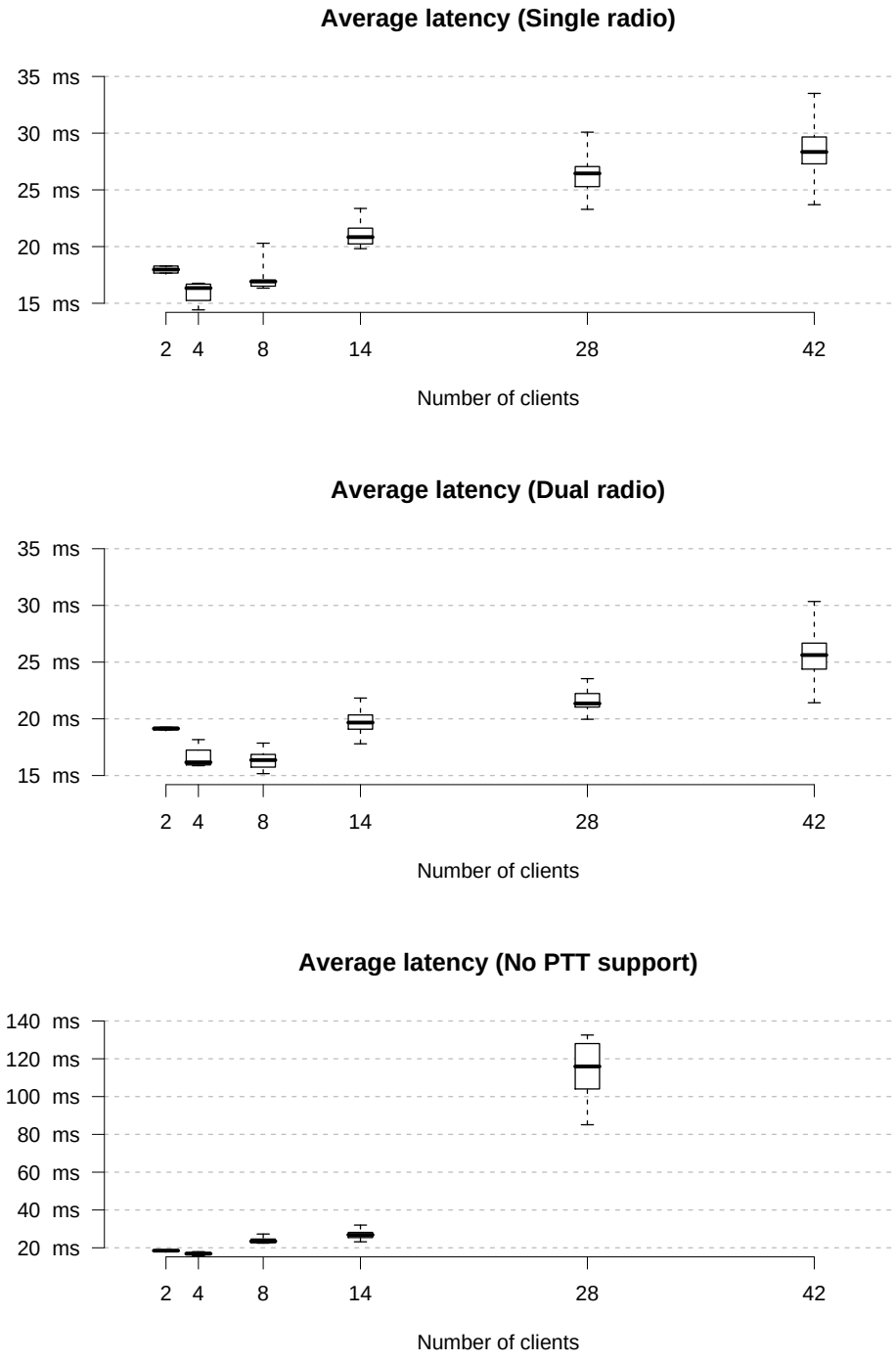


Figure 6.14: Boxplots with the average latency of the packets received by the nodes when the number of clients in a PTT session increases. Note that the Y-axis has different ranges in all three scenarios.

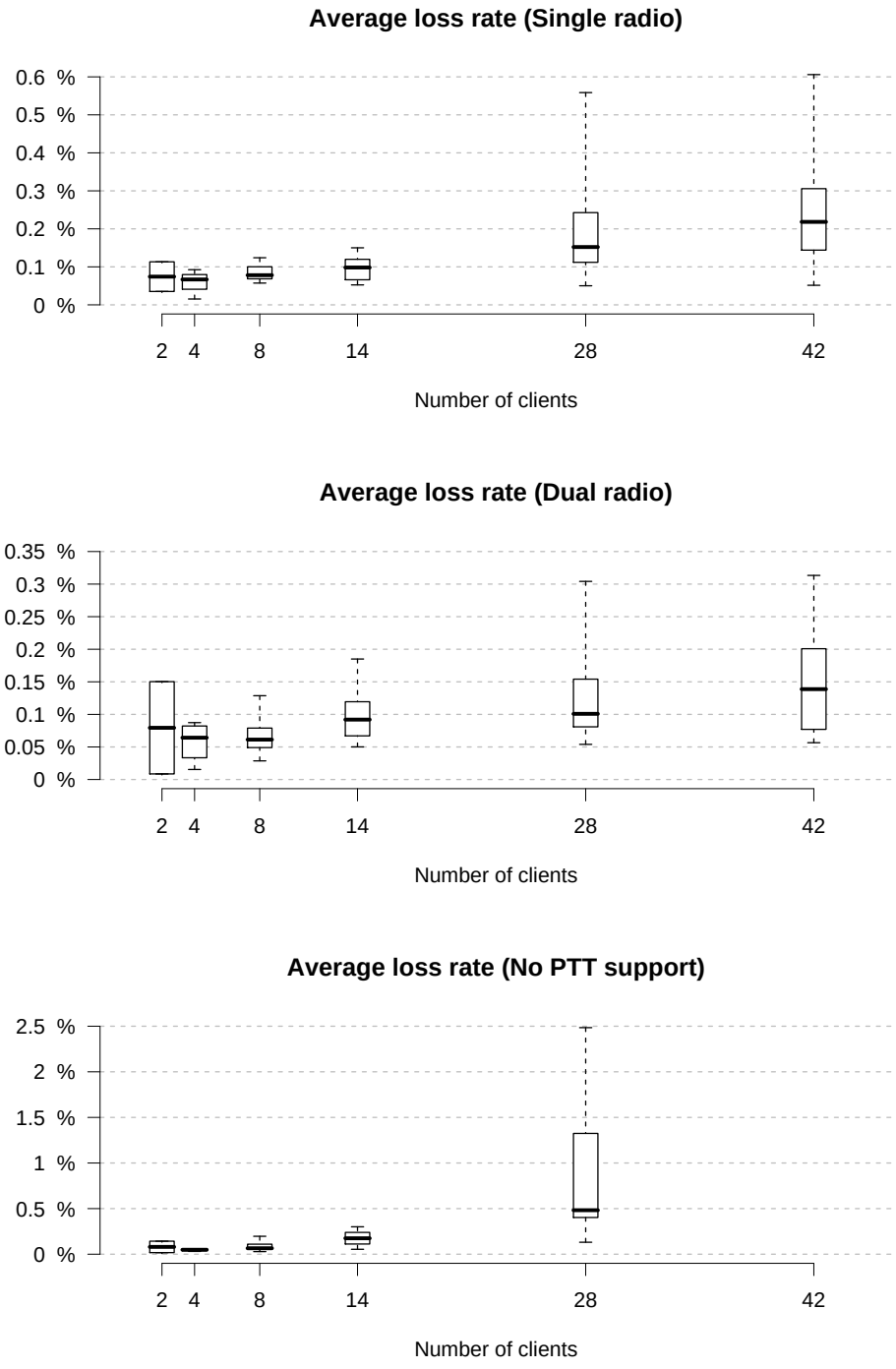


Figure 6.15: Boxplots with the average loss rate of the packets received by the nodes when the number of clients in the PTT session increases. Note that the Y-axis has different ranges in all three scenarios, and its maximum value is 100%.

lowest average latency being less 10 ms. For loss rates (Figure 6.15), the deviation from the average is higher. With 28 clients in the single radio setup, the maximum average loss rate of node was 0.5%, while the average was only 0.19%. The same wide range of loss rates is visible in all three setups.

The latencies we used so far are the average latency of all the packets received by a node, regardless of the sending node. It is expected that in an unplanned mesh network the connectivity between nodes may vary, both in terms of latency and loss rate. Many times the links between nodes are not symmetric. Moreover, the length of the multi-hop paths between the sending and the receiving node has a direct impact on the packet latency. We provide now a more detailed view of this experiment by looking at the pairwise latencies between the nodes.

For every node we compute the average latency of the packets received from each other node (Figures 6.16 and 6.17). If the number of clients is less than the number of nodes in the network, we show only the nodes that actually handle at least one PTT client. The graph is plotted as a heatmap, that is, the lighter colors correspond to a lower latency, while the darkest color corresponds to a latency of 50 ms or above. The diagonal line is not defined (the sender is the receiving node), and we used by convention value 0. By analyzing the single and dual radio setups we can see that for a given receiving node the latency per sending node is far from being uniform. For example, in the experiment with 28 clients in the single radio setup, the average latency for node 14 is 16 ms when the sending node is 2, and 42 ms when the sending node is 5. This is expected as the paths from the sending to the receiver node vary. Vice-versa, for a given sender, the packet latency perceived by the receiver nodes spans over a wide range. When node 14 was the sender, the minimum latency was 24.57 ms latency for node 2 and 56.39 ms for node 5. We can also see that, based on the location in the network, some sending nodes are better than others. For instance, the average latencies

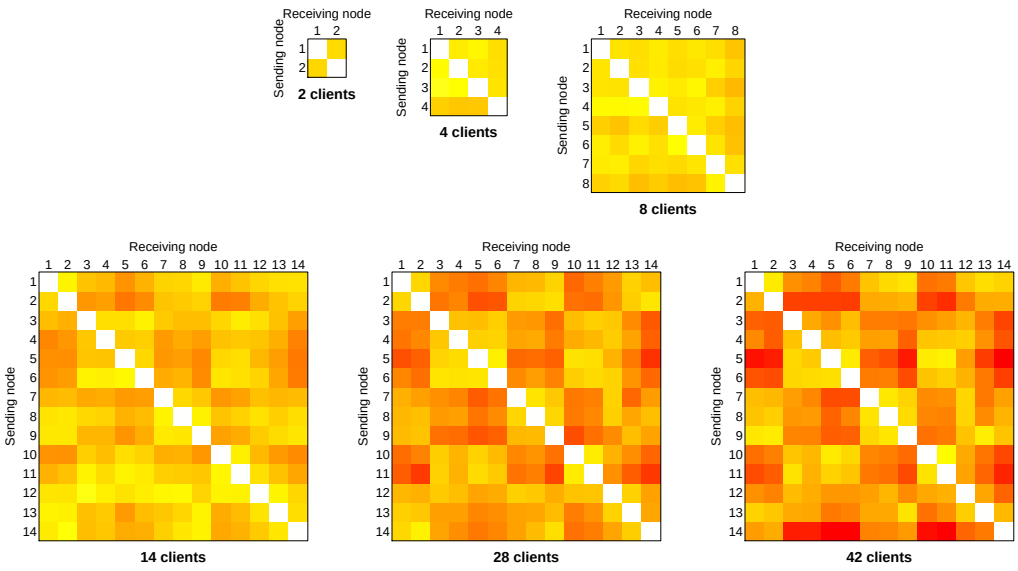


Figure 6.16: Average latency of the received packets for all sending nodes, in the single radio setup. The lighter colors correspond to a lower latency, while the darkest color corresponds to a latency of 50 ms or above. The diagonal line is by convention zero.

for all receiving nodes are lower when node 13 is sending data, compare to when node 14 is sending data, in the experiment with 42 clients. Therefore, it is likely that a PTT user will experience a change in the quality of the voice stream both when he roams from one access point to another, or when a different user acquires the floor.

In terms of the symmetry of the links, they are not perfectly symmetric, but the values are fairly close. The maximum difference in the latency between a pair of nodes was 10.9 ms for single radio and 12.38 ms for the dual radio setup.

Similar patterns can be observed in the pairwise loss rates (Figures 6.18 and 6.19). Here there is an even more variation between the receiving nodes. Nodes 3, 4, 5 and 6 experienced a lower loss rate for the same senders (nodes 3, 4, 5, 6, 10, and 11) in the single radio setup with 28 clients. Also, node 7's loss rate was very high regardless of the sender, in the same setup. As for the symmetry of the links' loss rates, the maximum difference was 0.97% for single radio and 0.4% for dual radio.

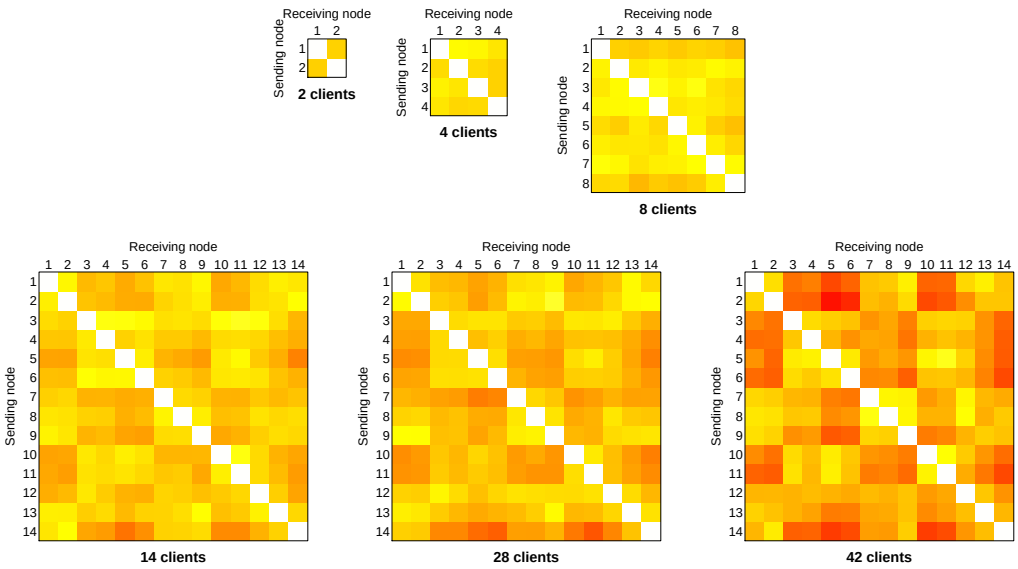


Figure 6.17: Average latency of the received packets for all sending nodes, in the dual radio setup. The lighter colors correspond to a lower latency, while the darkest color corresponds to a latency of 50 ms or above. The diagonal line is by convention zero.

Figure 6.20 presents the Cumulative Distribution Function (CDF) of the latency for each mesh node in the experiment with 14 clients, in a single radio scenario. We can see that 95% of the packets are received within 50 ms. Note that the numbers for end-to-end client communication will be somewhat higher as each client is one wireless hop away from a mesh node. The experiment shows that all nodes received most packets within 50 ms. Note that for PTT applications, latencies as high as 400 ms are considered acceptable in PTT systems built for first responders [6].

Figure 6.21 presents the overhead traffic, as seen by a single mesh node (node 1 in Figure 6.10). This overhead depends on the distribution and the density of the clients in the network. For better analysis, we separate the overhead into three distinct components: **(1)** mesh control traffic (i.e., link state updates generated by topology changes and control traffic for managing multicast groups). The amount of this traffic is very small, less than 1 Kbps. **(2)** Control traffic generated by our PTT protocol (e.g., requests, releases, ping messages, ac-

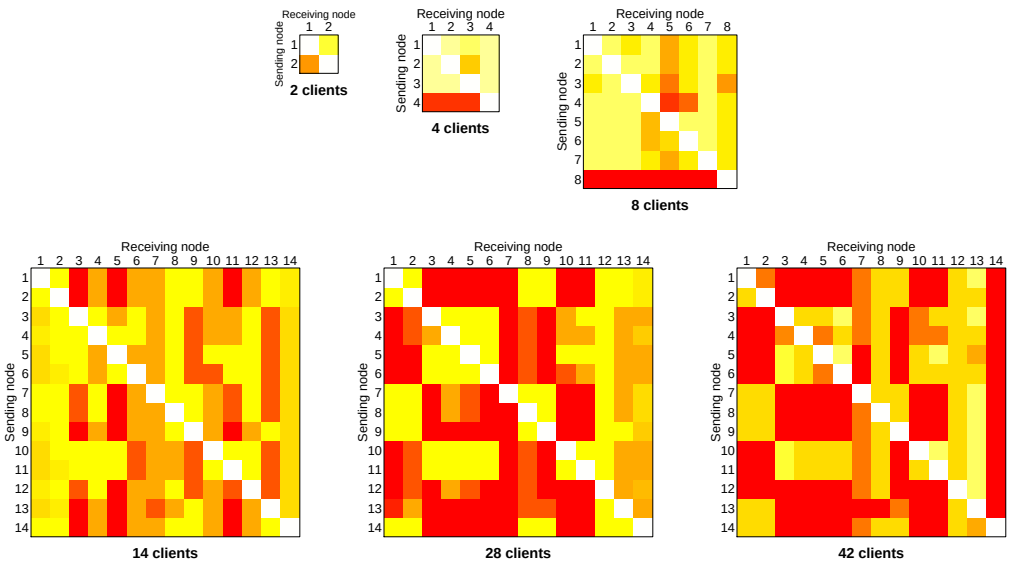


Figure 6.18: Average loss rate of the received packets for all sending nodes, in the single radio setup. The lighter colors correspond to a lower loss rate, while the darkest color corresponds to a loss rate of 0.2% or above. The diagonal line is by convention zero.

knowledgments, etc.) Since the size of these messages is very small, this overhead is also low, less than 1 Kbps on average in our experiments. **(3)** Traffic required to locally share clients' PTT state (i.e., traffic on the Client Control groups). This represents the majority of the overhead, increasing from 1.3 Kbps for 2 clients to 27 Kbps for 42. This overhead depends on the density of the mesh (how many mesh nodes can hear a client) and the number of clients. The experiment shows that the overhead of the system as the number of clients grows is minimal, below 1 Kbps per client.

To sum up, this experiment shows how the system scales and demonstrates that when utilizing a PTT enabled phone or a dual-radio configuration, the system can scale to at least 42 clients in the mesh network with minimal impact on latency and loss rate.

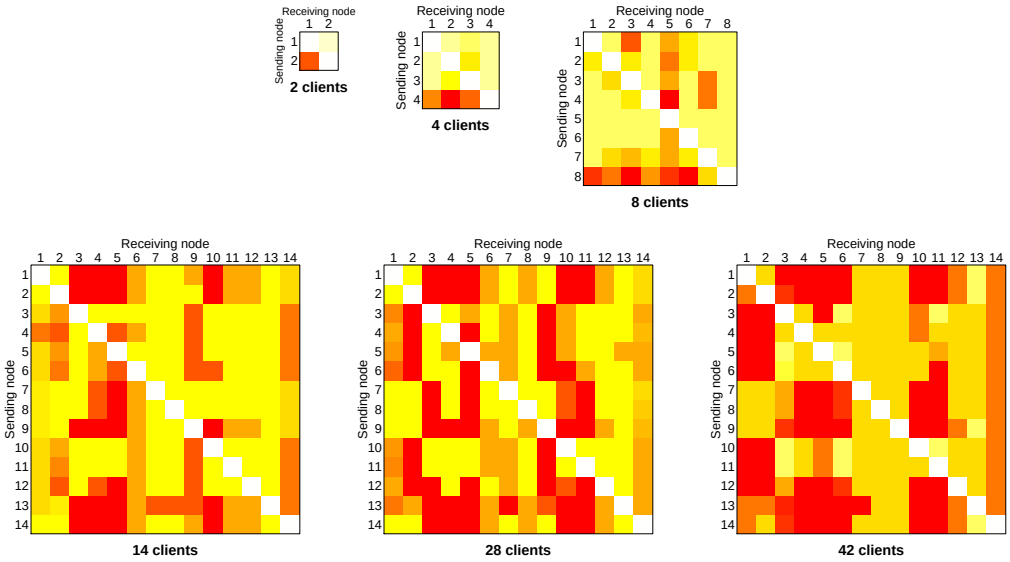


Figure 6.19: Average loss rate of the received packets for all sending nodes, in the dual radio setup. The lighter colors correspond to a lower loss rate, while the darkest color corresponds to a loss rate of 0.2% or above. The diagonal line is by convention zero.

Scaling with the number of PTT sessions

To test the scalability of our system in another dimension, we gradually increase the number of simultaneous PTT sessions in the system. Each PTT session includes four clients connected to random mesh nodes in the network. Each PTT session contributes a single VoIP stream (50 packets per second, total of 64 Kbps).

Figures 6.22 and 6.23 show the latency and loss rate as the number of PTT groups increases. We first compute the latency of the packets received by all the nodes that handle PTT users in a given group. The average of these values represent the group average latency. Then we average these values for all the groups and obtain the average latency for the experiment.

With a single radio, the system scales to 6 PTT groups, while with dual-radio, the system scales to 8 PTT groups. Noting that the scalability of the system was impaired by the

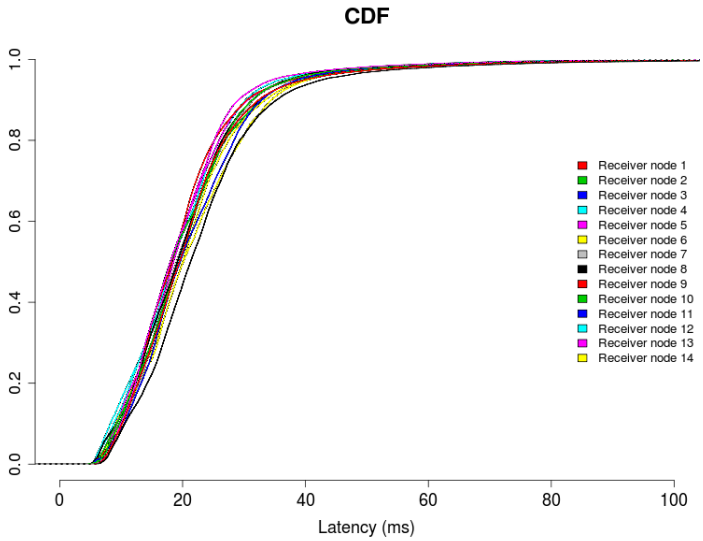


Figure 6.20: CDF of the latency of the packets received by each of the mesh nodes, in the experiment with 14 clients, in a single radio scenario.

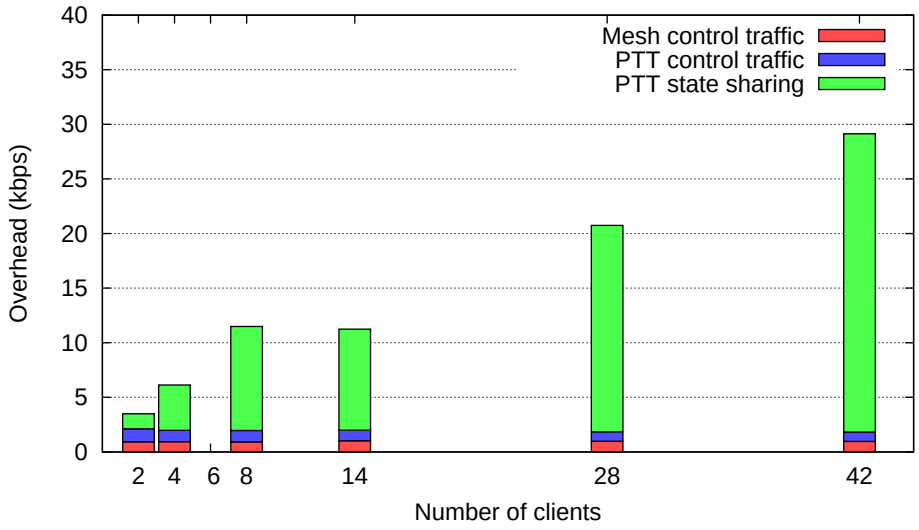


Figure 6.21: Overhead traffic as seen by node 1 when the number of clients in a PTT group increases from 2 to 42 (3 clients connected to each of the 14 access points.)

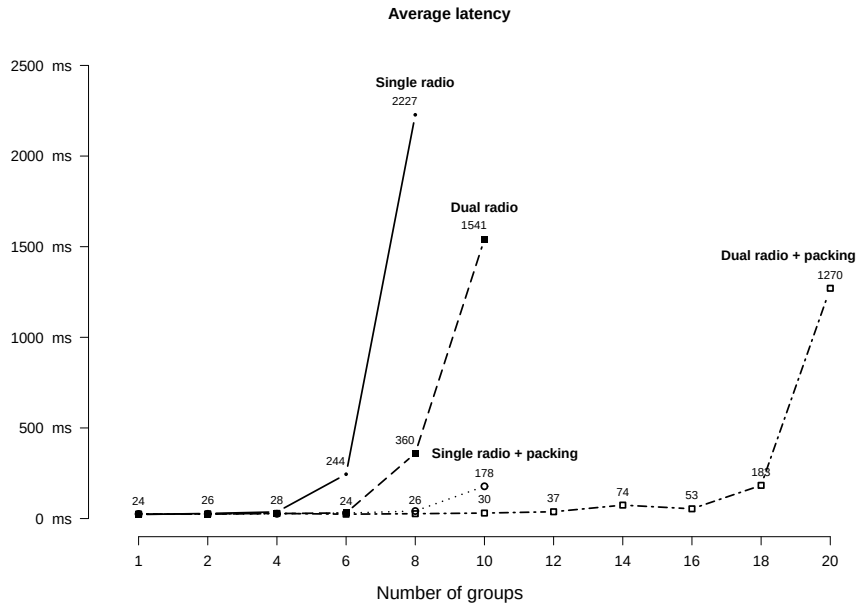


Figure 6.22: Average latency of the packets received by the nodes when the number of PTT groups increases from 1 to 20. There are 4 clients in each PTT group.

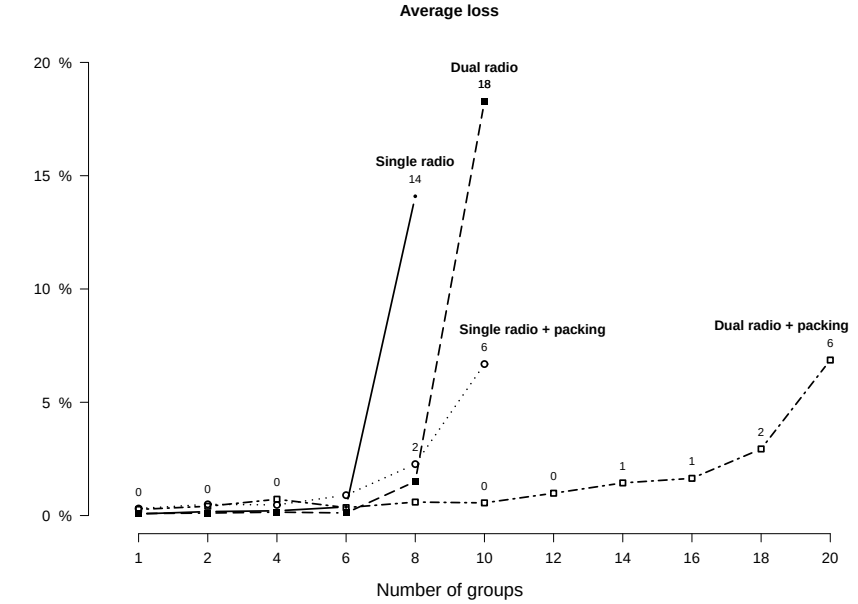


Figure 6.23: Average loss rate of the packets received by the nodes when the number of PTT groups increases from 1 to 20. There are 4 clients in each PTT group.

high overhead associated with sending small packets in 802.11 networks, we tested the same two scenarios with packing 160 ms of VoIP packets into one network packet at the mesh node. This approach allows us to trade some latency (20 ms x 7 packets = 140 ms) for an 8 fold reduction in the number of packets in the mesh. Note that PTT systems used by first-responders [6] employ a slightly higher packing scheme of 180 ms. Packet aggregation allowed us to support up to 18 PTT sessions before the latency jumped to over 500 ms.

To better understand the quality of the service perceived by the users, we now analyze the variation of the latency and loss rate on each group. Figures 6.24 and 6.25 show a boxplot for each experiment, as we increase the number of PTT groups in the system, for each setup (single radio, single radio with packing, dual radio, dual radio + packing). As before, for each setup we use different ranges for the Y-axis. Notice that the variation of the latency in the setups without packing is much higher than if packing is used: 65-493 ms (6 groups, single radio), 107-784 ms (8 groups, dual radio), compared to 31-47 ms (8 groups, single radio with packing), and 68-305 ms (18 groups, dual radio with packing). The corresponding ranges for loss rates are: $0.19\% - 0.6\%$ (6 groups, single radio), $0.6\% - 4.19\%$ (8 groups, dual radio), $0.88\% - 4.59\%$ (8 groups, single radio with packing), and $1 - 5\%$ (18 groups, dual radio with packing). Latencies for each individual PTT group are shown in Figure 6.26. Groups are sorted in the increasing order of their average latencies. This is still a high level view on the system, as a group latency is computed as an average over the latencies of the four users communication on that group. Therefore, depending on their position in the network, users experience very different latencies.

To sum up, this experiment shows that it pays to trade some latency with scalability in order to support a larger number of PTT groups.

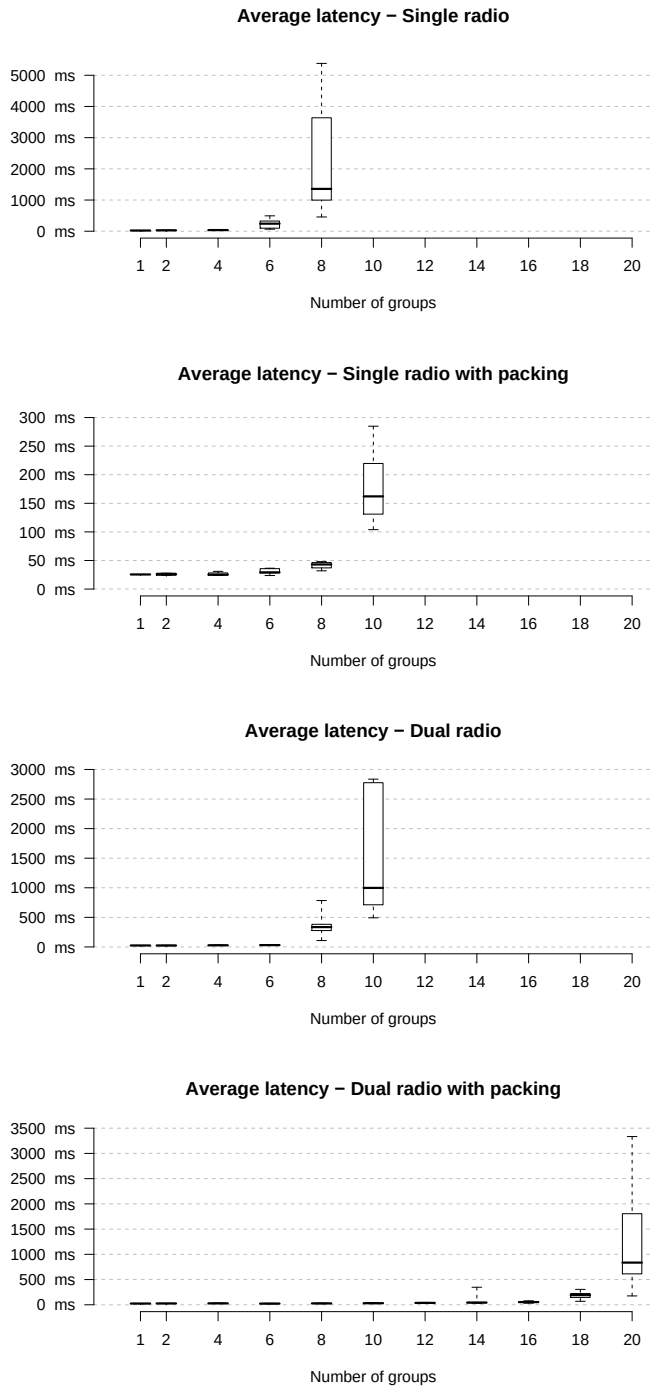


Figure 6.24: Boxplots with the average latency of the packets received on each group when the number of PTT groups increases from 1 to 20. There are 4 clients in each PTT group. Note that the Y-axis has different ranges in all four scenarios.

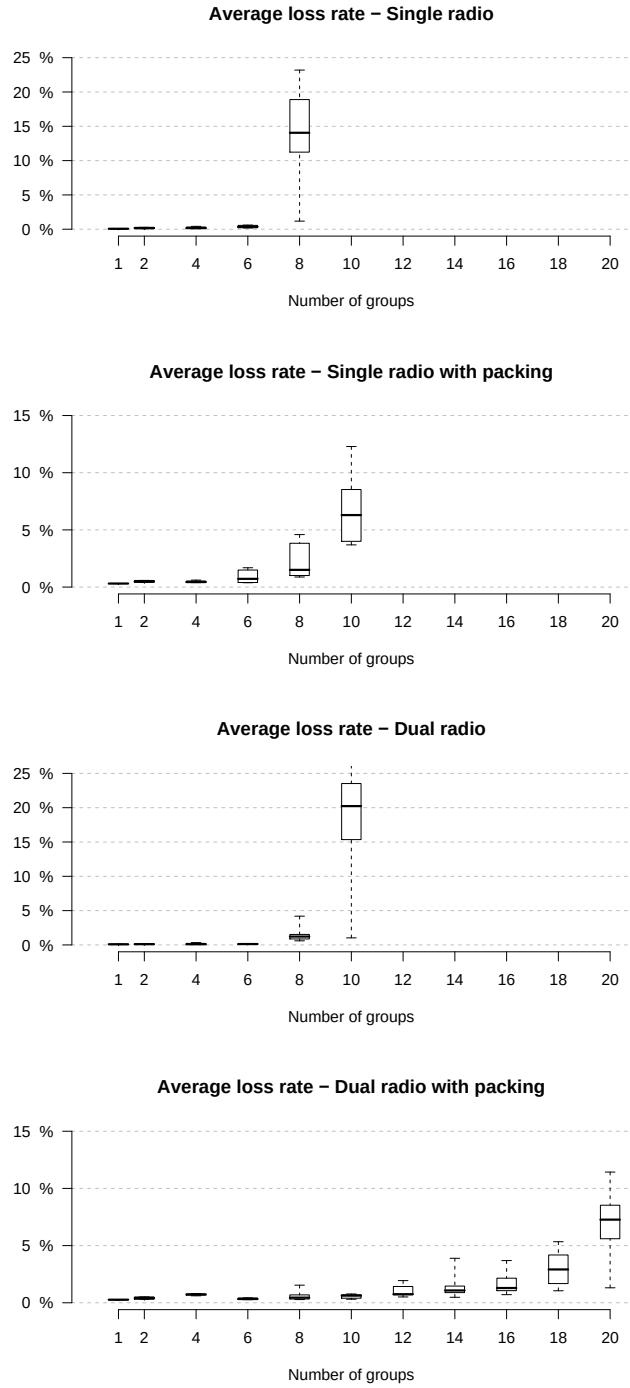


Figure 6.25: Boxplots with the average loss rate of the packets received on each group when the number of PTT groups increases from 1 to 20. There are 4 clients in each PTT group. Note that the Y-axis has different ranges in all four scenarios.

Single radio

	1	2	3	4	5	6	7	8
1 group	22							
2 groups	26	29						
4 groups	32	35	37	41				
6 groups	65	97	237	246	326	493		
8 groups	457	951	1043	1274	1438	3412	3862	5380

Single radio + packing

	1	2	3	4	5	6	7	8	9	10
1 group	25									
2 groups	24	27								
4 groups	23	24	25	31						
6 groups	23	28	28	29	35	36				
8 groups	31	36	37	41	44	45	46	47		
10 groups	103	127	131	146	157	166	197	219	244	284

Dual radio

	1	2	3	4	5	6	7	8	9	10
1 group	24									
2 groups	22	26								
4 groups	25	26	27	31						
6 groups	24	28	29	31	35	36				
8 groups	107	245	307	320	350	379	385	784		
10 groups	492	644	712	769	830	1164	2378	2776	2811	2836

Dual radio + packing

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
1 group	24																			
2 groups	24	27																		
4 groups	25	26	27	34																
6 groups	21	22	22	23	27	27														
8 groups	21	23	25	26	27	28	30	31												
10 groups	23	26	28	29	30	30	31	33	34	35										
12 groups	29	30	33	34	36	36	37	38	40	41	45	47								
14 groups	31	32	35	40	40	41	42	43	43	43	49	58	190	347						
16 groups	31	38	38	48	48	50	51	52	53	54	57	60	63	66	67	79				
18 groups	68	94	102	115	145	155	160	178	186	197	199	206	219	222	239	241	256	305		
20 groups	175	424	503	578	589	635	647	763	814	828	843	858	1216	1267	1556	2053	2462	2767	3089	3334

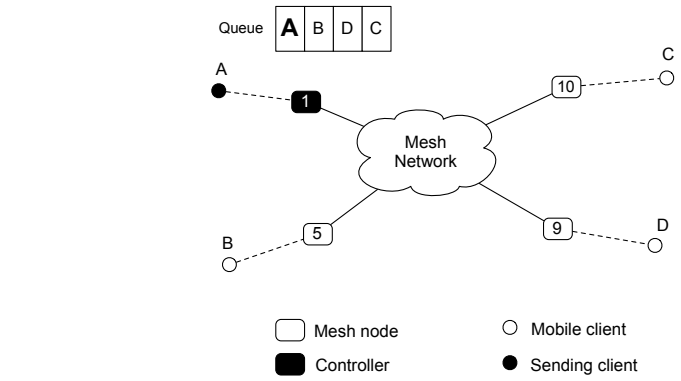
Figure 6.26: Average latency of the packets on each PTT group. Groups are sorted in the increasing order of their average latencies.

Robustness test

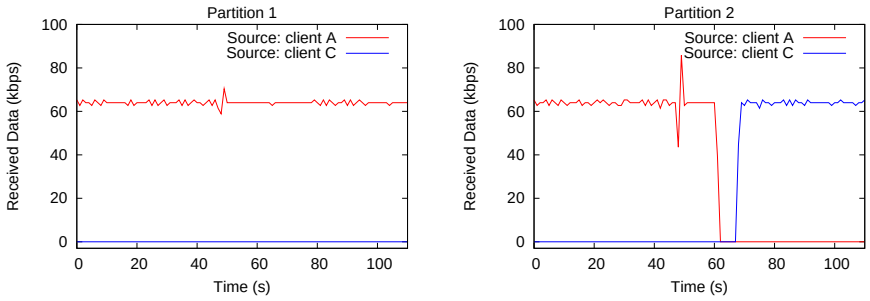
This experiment demonstrates the system's behavior when there is a partition and a merge in the wireless mesh network.

We first present a small-scale scenario with 4 clients (A, B, C and D) joining the network in 4 different places (node 1, node 5, node 9 and node 10), with A and B in one "side" of network, and C and D in the other side, as illustrated by Figure 6.27. In the beginning, the controller of the PTT session is node 1 and client A is granted permission to speak. We then partitioned the network, such that node 9 and node 10 became unreachable from node 1 and node 5's side of the network. Figure 6.27 shows the voice traffic as received by client B in the first partition and by client D in the second one. We can see that in the first partition the data packets are generated by client A, and this does not change, as expected, even after the partition occurs (around second 60). However, the right side of the partition lost the controller. After approximately 7 seconds, a new controller is generated (node 9), the requests are recovered, and client C is granted permission to speak, as shown by the second partition's view in Figure 6.27. This demonstrates that the system gracefully handles network partitions.

Next, we started with the network partitioned, with node 1 and node 5 in one partition and node 9 and node 10 in the other, as shown in Figure 6.28. Each of the partitions has its own controller, node 1 and node 9, respectively. We then merged the network by connecting a mesh node that is the only connection between the two sides of the network. We analyze the voice data received by clients B and C (Figure 6.28, partition views). We can see that before the merge the data is sent by client A in the first partition and by client D in the second. As the network begins to merge, both B and C start getting voice traffic from both senders. Shortly after the network routes became stable, the controllers detected each other and client A's right to speak is revoked by node 9, the newly established controller. Thus,



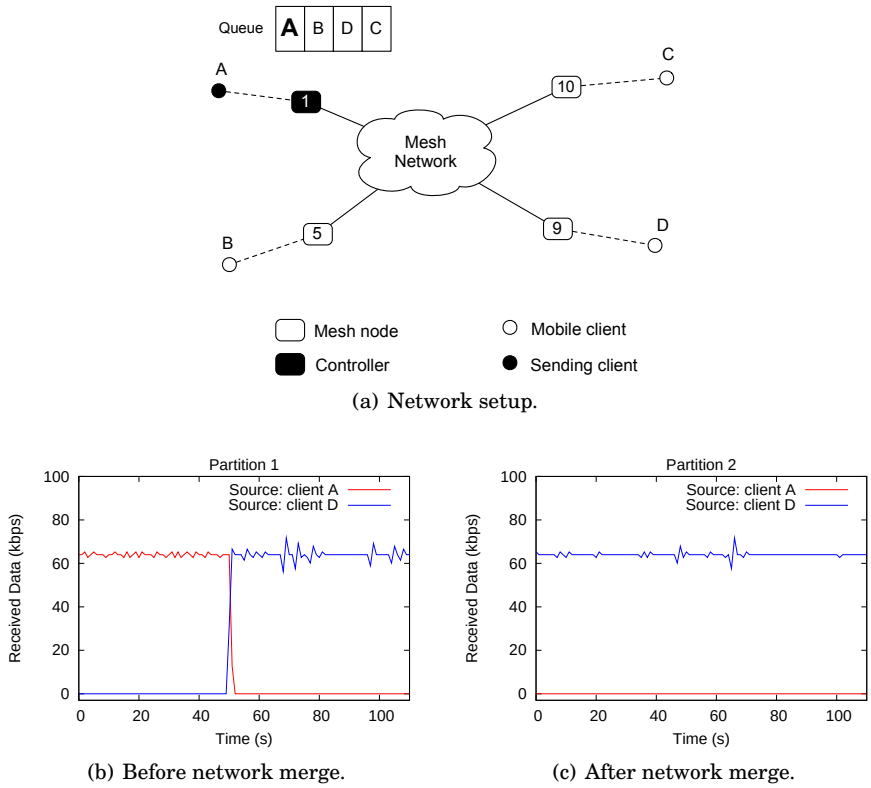
(a) Network setup.



(b) Before network partition.

(c) After network partition.

Figure 6.27: Traffic before and after a network partition, as seen by client B in the first partition and by client D in the second partition. A new controller is generated in the second partition.



(a) Network setup.

(b) Before network merge.

(c) After network merge.

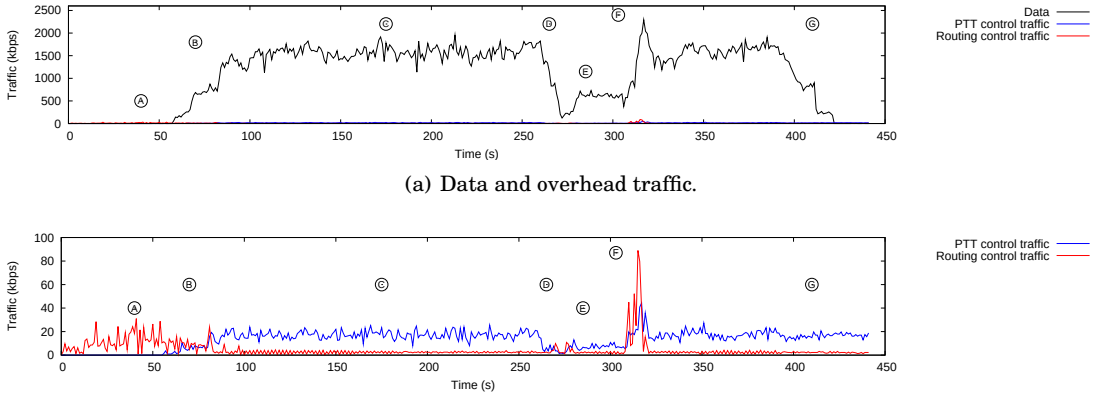
Figure 6.28: Traffic before and after a network merge, as seen by client B in the first partition and by client C in the second partition. For 686 ms the client's voice traffic is corrupted, due to multiple voice streams. After the merge there is only one controller and one sending node.

the one-to-many communication is reinforced. Clients' voice traffic was corrupted—due to multiple voice streams—for about 686 ms (35 packets). This demonstrates that the system gracefully handles network merges, quickly eliminating redundant controllers.

Large-scale scenario

Finally, we benchmark the system in a large scale partition and merge scenario, with 40 participants in 10 simultaneous PTT sessions. Partition was performed by disconnecting one of the routers. Similar to the scalability experiment, the sending client in each group changes every 10 seconds. Figure 6.29 shows the overall traffic in the vicinity of node 1 (as observed by setting node 1 in promiscuous mode and counting all the packets in its vicinity). To better understand the system's behavior, we present in Figure 6.29(a) both the data and overhead traffic, and separately, in Figure 6.29(b), two components of the overhead traffic: routing control traffic (link state, multicast group management) and PTT protocol control traffic. For clarity, we do not show the overhead traffic associated with sharing the state of a client within his vicinity, as it was already shown in Figure 6.21.

Following the overhead traffic, we can see the route updates that are generated when clients join the network (point A), as well as the overhead related to clients joining a PTT group and asking for permission to speak (point B). The system operates normally until second 265 (point D), when the network partitions. Then, many of the sessions in node 1's partition lose their speaker or their ability to route to some PTT members. When the connectivity stabilizes, new speakers are granted permission to speak (point E). Note that the amount of VoIP traffic is smaller, as some PTT sessions no longer have members in the current partition, or do not have to route through node 1's vicinity PTT session messages. Around second 310 (point F), the network merges, causing a spike in both data and overhead traffic. Shortly after that, network routes stabilize and normal operation is resumed. Lastly, around second



(a) Data and overhead traffic.

(b) Overhead traffic.

Figure 6.29: Network partition and merge in a large-scale experiment with 40 clients on 10 PTT groups. (A) clients join, (B) clients request to speak, (C) regular operation, (D) network partitions, (E) network stabilizes after the partition, (F) network merges, (G) clients stop speaking. The marks indicate approximately the middle of each stage.

380 (point G), all the clients stop speaking and the data rate drops to zero. Since clients did not leave their PTT groups, the overhead associated with maintaining PTT sessions remains constant through the end of the experiment. This elaborate scenario demonstrates the robustness of the system to network connectivity changes while supporting a large number of distinct PTT sessions.

Chapter 7
Conclusions

In an effort to make them a reality, this dissertation looks at the practical aspects of the wireless mesh networks and introduces the first high-throughput 802.11 wireless mesh system that provides seamless connectivity to mobile users using off-the-shelf low cost routers.

Our design is both efficient and flexible by using an overlay approach that maintains the control of the mesh in user space but forwards the data at the kernel-level. Redundant multipath routing, the mechanism that enabled this separation while preserving seamless mobility, is a general concept that could be beneficial in other types of networks.

While Internet access is the most dominant usage of mesh networks today, the distributed support that mesh networks can provide enables new applications. This dissertation looks at Push-To-Talk as an example of such an application. It includes the architecture and protocols that allow the entire mesh network to act a single distributed point of coordination between PTT users. The system is robust in the face of connectivity faults, such as network partitions and merges, and node crashes. The architecture seamlessly integrates regular and cell phone users, allowing them to connect via a gateway to the mesh network, and participate in locally established PTT sessions.

I enthusiastically look forward to see appealing applications and a wider adoption of wireless mesh networks in the years to come.

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Vita

Raluca Musăloiu-Elefteri received a B.S. and M.S. in Computer Science from Politehnica University of Bucharest, Romania, in 2003 and 2004. She enrolled in the Computer Science Ph.D. program at the Johns Hopkins University in 2004 joining the Distributed Systems and Networks Lab. Her research focuses on wireless mesh networking. More generally, she enjoys building systems that make things easier.

Starting in January 2010, Raluca will join Meraki, Inc. in San Francisco.