CHASING EME: ARGUMENTS FOR AN END-MIDDLE-END INTERNET

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Saikat Guha

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Saikat Guha, Ph.D.

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Connection establishment in the Internet has remained unchanged from its original design in the 1970s: first, the path between the communicating endpoints is assumed to always be open. It is assumed that an endpoint can reach any other endpoint by simply sending a packet addressed to the destination. This assumption is no longer borne out in practice: Network Address Translators (NATs) prevent all hosts from being addressed, firewalls prevent all packets from being delivered, and middleboxes transparently intercept packets without endpoint knowledge. Second, the Internet strives to deliver all packets addressed to a destination regardless of whether the packet is ultimately desired by the destination or not. Denial of Service (DoS) attacks are therefore commonplace, and the Internet remains vulnerable to flash worms.

This thesis presents the End-Middle-End (EME) requirements for connection establishment that the modern Internet should satisfy, and explores the design space of a signaling-based architecture that meets these requirements with minimal changes to the existing Internet. In so doing, this thesis proposes solutions to three real-world problems. First, it focuses on the problem of TCP NAT Traversal, where endpoints behind their respective NATs today cannot establish a direct TCP connection with each other due to default NAT behavior. It presents a set of techniques, called STUNT, that solves this problem without any changes to NATs or to existing operating systems. In STUNT, the communicating endpoints use signaling to coordinate the creation of NAT state that

then enables a direct TCP connection. The second problem this thesis focuses on is that of mitigating unwanted traffic on the Internet, such as DoS attacks and worms, originating from botnets. It presents a simple architecture, called ShutUp, that mitigates unwanted traffic in a completely End-to-End (E2E) manner without requiring any changes to the network. Trusted code near the source of unwanted traffic, for instance in the virtualization layer, network card, or nearby router, responds to signals from the destination by taking corrective action. Finally, this thesis focuses on the broader problem of establishing connections that adhere to all applicable network policy, including access control, multihomed route control, and middlebox usage — all open problems in today's Internet. This thesis presents the NUTSS architecture which takes into account policy set by all stakeholders, including both the endpoints and the middle networks. NUTSS uses name-based signaling to negotiate high-level policy before connection establishment, and couples it to address-based signaling for efficient enforcement during the connection lifetime. NUTSS does not change the protocol stack and can be deployed incrementally.

Solving each of the aforementioned problems requires a departure from the original Internet architecture. Yet in this thesis clean-slate solutions are expressly avoided in favor of evolutionary changes. The central argument of this thesis is that solving a wide range of architectural shortcomings of today's Internet, and incremental deployment are not mutually exclusive.

BIOGRAPHICAL SKETCH

Saikat Guha seeks nothing short of world domination through the blunt instrument of research in network systems. Following in the footsteps of his role models Dr. Evil, Dr. Doom, and Dr. Octopus (that may or may not be his nicknames for his committee members), he sought out his B.S., M.S., and Ph.D. in Computer Science from Cornell University in 2003, 2008, and 2009 respectively. Prior to escaping to Cornell, he spent the first year of his undergraduate incarceration at the Indian Institute of Technology (IIT) Delhi sentenced to study Chemical Engineering. Having established his base of operations at Cornell, Saikat reached out for venture capital: DoCoMo Labs introduced him to the evils of Peer-to-Peer (P2P) technology in the summer of 2005. This brought him to the attention of the evil empire, which offered him an internship at Microsoft Research Cambridge in summer 2006 where he worked on more P2P stuff. He finally graduated to the most evil of them all, Google Inc., where in the summer of 2007 he worked on projects too evil to put in print. Meanwhile at Cornell, he repeatedly broke the Internet architecture and put it back together, most notably having to do with NAT Traversal, which gained him notoriety in the IETF and the research community. With graduation imminent, Saikat evaluated the Max Planck Institute for Software Systems (MPI-SWS) in Germany as a potential new base of operations in the summer of 2008, and found it suitable for a temporary post-doctoral position. From there he plans to launch the privacy preserving advertising revolution, which would unseat the evil advertising overlord Google from its absolute monarchy over cloud computing. He then plans to ally with his enemy's enemy to re-imagine cloud computing so it protects user privacy, thereby winning over legions of delighted users who will naturally deliver him the world.

To Ma, Bapi, Didu, and Mashi.

ACKNOWLEDGEMENTS

Paul Francis, my advisor, taught me everything I know of architecting systems. He ingrained in me the principle that a good architecture is not one which has the most features, but one that meets the requirements with the fewest mechanisms. He constantly challenged me to reach higher, to solve a problem with fewer assumptions, or with more severe constraints. Above all, Paul taught me to be an independent researcher. He taught me to look for problems with real-world significance, and to solve them in a way that would create real-world impact. At the same time, Paul illustrated, by example, the role of an ideal mentor. His hands-off nature encouraged me to find my own way, while his endless patience, eye for detail, and brutal honesty made discussions with him my first recourse in times of doubt. I will especially treasure every hard-fought argument where I successfully defended my own for the thrill and sense of accomplishment it brought.

Andrew Myers cultivated the love of teaching in me. Emin Gün Sirer taught me much about the art of presenting one's research to an audience. Fred Schneider was a source of infallible wisdom and insight in analyzing and providing constructive criticism on research. Neil Daswani at DoCoMo Labs and Google, Pablo Rodriguez at Microsoft Research, and Nina Taft, Dina Papagiannaki, and Jaideep Chandrasekhar at Intel Research gave me the opportunity to expand my research interests and experience research from the industry perspective. My internship with Krishna Gummadi at MPI-SWS and the Glasnost team under him (Marcel Dischinger, Ratul Mahajan, Stefan Sariou, and Bryan Ford) was an enriching collaborative experience.

Research cannot be done in isolation, and I owe a big debt to my many cohabitants in the Cornell Systems Lab (Syslab). Under the stewardship of

Andrew Myers, Emin Gün Sirer, Ken Birman, and Johannes Gerkhe, the lab brought together Ph.D. students from different areas under one roof. The Syslab community was an invaluable resource in matters of research (and life in general). Permanent Syslab residents: Hitesh Ballani, Oliver Kennedy, Jed Liu, Alan Shieh, Krishnaprasad Vikram, Vivek Vishnumurthy, Dan Williams, and Bernard Wong, and visitors: Mahesh Balakrishnan, Tuan Cao, Lakshmi Ganesh, Tudor Marian, Patrick Reynolds, Yee Jiun Song, and Kevin Walsh were good company through the years. I will cherish the late hours spent discussing research on whiteboards, followed soon after by movies and takeout.

Finally, none of this would have been possible without the love and support of my family. From introducing me to computer programming at the age of six, to encouraging my obsession ever since, to supporting my decision to transfer from IIT Delhi to a university in another country, my parents, my (late) grandmother, and my aunt have been instrumental in enabling me to follow my every dream, and to them I dedicate this thesis.

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CHAPTER 1

INTRODUCTION

The Internet was designed to provide a small but critical set of transport services:

1. *User-friendly naming* of all Internet hosts (through DNS).
2. *Network-level identification* of all Internet hosts (through the IP address) and best-effort delivery of datagrams to identified hosts.
3. *Identification of the application* on the host that should receive a given packet (through the port number).

Implicit among these services was the idea that applications would individually take care of access control. The Internet[1] would deliver transmitted packets to the target application, and it was up to the application to decide whether to accept or reject the packet. A further implication of this approach is that there is no danger in asking an application to process an incoming packet. The application is assumed to be competent to look inside the packet and decide whether or not to accept it. Industry recognized in the early 90's that this approach was wrong: DoS attacks can overwhelm an application, and because of either bugs or just poor design, applications are incapable of securing themselves with certainty. The industry answer to this problem was the firewall, which effectively enunciated a fourth critical requirement for the Internet transport service:

[1]By "Internet", we mean the naming and transport services provided by IP addresses, ports, and DNS for today's "fixed" Internet (including wireless access to the wired Internet). Sensor networks and MANETs that perform their own naming and routing separate from the Internet are not included in this definition.

4. *Blocking of unwanted packets* before they reach the target application (through packet filters in firewalls).

Of course it is well-known that the Internet today is ill-equipped to satisfy these four core requirements. The IP address shortage prevents all hosts from being identifiable in the network. Port numbers do not adequately identify applications anywhere outside of the operating system (OS) that created the socket. As a result, firewalls cannot be certain what application is behind a given port number. Thus firewals tend to use costly deep packet inspection, and often err on the side of caution (preventing flows that might otherwise be acceptable).

The firewall compromised the End-to-End (E2E) [103] nature of the Internet architecture by placing a point of control outside of the end host. While this development was widely viewed as negative [49], we and others [127] believe that it is not only inevitable, but necessary and largely positive. A primary reason for this is the fact that there may be multiple legitimate stakeholders in a given packet flow—the end user, the corporate IT department, or the ISP—each with their own policies. The E2E nature of the Internet does not easily accommodate these policies. Another reason, however, is that sometimes it is simply economically expedient to deploy a function in the middle, even if it might ultimately be better done at the ends. Today there are often good reasons to want to route packets through middleboxes other than firewalls: for instance, virus scanners, web caches, traffic shapers, performance enhancing proxies and protocol translators (IPv4 to IPv6). These middleboxes sometimes interrupt E2E semantics. The legitimate rise of middleboxes leads to another requirement:

5. *Explicit negotiation of middlebox usage* between the endpoints and networks

in the middle, including the ability to steer packets through middleboxes not otherwise on the data-path between source and destination.

We refer to this set of five requirements as the *End-Middle-End* (EME) naming and addressing problem. Together they constitute what we consider to be the absolute minimum set of requirements that the modern Internet should satisfy. Put another way, a new standard sockets interface, and the networking infrastructure that supports it, should at a minimum satisfy the above requirements.

1.1 Clean Slate, Patches, or Evolution?

Before we discuss our proposed solution, it is worth taking a step back and exploring the design space of any Internet architecture that solves today's problems. There are three basic approaches: at one end of the spectrum is rearchitecting the Internet from scratch — the so called "Clean Slate" approach. At the other end are purely E2E patches that make use of existing (unmodified) network infrastructure and require changes only at the endpoints. The third option is a middle ground where E2E techniques are combined with incremental modifications to the existing Internet architecture. This allows maximal reuse of existing infrastructure while evolving the underlying architecture.

Clean slate redesign answers the question: "How would we have designed the Internet if we had had the hindsight of the last forty years of Internet evolution?" There has been phenomenal change since when the Internet was designed: the number of hosts once expected to be in the few thousands has exploded to several billions; the friendly collaboration between networks has

given way to legal and economic agreements that bind networks; and the once-open network, under constant siege by malicious parties with the ability to inflict widespread harm, is increasingly becoming less open. No doubt given full hindsight a clean slate redesign of the Internet architecture would address a majority of today's problems.

A clean slate redesign cannot, however, anticipate *future evolution* of the Internet. The evolution of the Internet has been marked by one disruptive technology after another — the World Wide Web (WWW), Peer-to-Peer (P2P), and Instant Messaging and Online Social Networks, to name a few. The role of the Internet itself has changed from remote access, to content delivery, to communication. A clean slate redesign before 1989 could not have accounted for the WWW, the resulting commercialization of the Internet, and the problem that would lead to (e.g. phishing). A redesign before 1999 would, arguably, have paid little attention to the P2P model, choosing instead to optimize for the client-server WWW. Today the Internet is undergoing yet another change; the emergence of online social networks is transforming the Internet from primarily a content-delivery platform to a platform for communication. It is therefore naïve to believe that a clean slate architecture based on the Internet of 2009 will be ideal even ten years hence. Nor is there reason to believe that a clean slate redesign is a one-shot solution; we are bound to seek clean slate solutions each time the Internet evolves in unpredictable ways, with each iteration coming at a deployment cost greater than the previous.

A pragmatic alternative is to solve problems within the confines of today's Internet architecture — in essence, hacks and patches to the Internet. Patching has two main benefits over a clean slate approach. First, patching capitalizes

on already deployed infrastructure and existing expertise. As a result, costs can be significantly lower than a clean slate redesign. Second, patching tends to be expedient. While a clean slate redesign affects components across the protocol stack, patching can target individual problems, resulting in a solution that can potentially be deployed more easily.

The downside to patching, however, is that it leads to point solutions with limited utility. At the same time, patching can be dangerous if it lacks foresight. The design of Network Address Translation (NAT) to solve the IP address space exhaustion problem is a prime example. NAT is largely transparent to the private-client public-server communication model that was dominant in the early '90s. As a result, NAT saw rapid deployment. Later when P2P applications were designed, in which any peer may communicate with any other peer, NAT proved to be a fundamental stumbling block. A lot of time and effort has since been expended on solving these problems; problems that, for the most part, will be rendered moot when IPv6 eventually displaces NAT altogether.

What is needed, then, is a middleground that solves a large class of problems, and yet makes maximal use of existing infrastructure. Such an approach can evolve the Internet architecture in an incrementally deployable manner. This is the approach we take in this thesis.

1.2 Using Signaling for Internet Connection Establishment

The EME problem fundamentally stems from the lack of coordination between the endpoints and the network. Ultimately, it is the endpoints that know whether a connection should be allowed or not allowed, while it is the network

that is in a position to make that connection feasible or infeasible. Yet, today, there is no direct communication between the two. This observation leads naturally to the solution we explore in this thesis: the design of a signaling primitive, which creates a dialog between the endpoints and the network, that can solve the End-Middle-End problem in an incrementally deployable manner.

The design space for a signaling primitive is defined by three key axes, which correspond to the answers to the questions: *who* participates, *where* they are located, and *how* they are contacted. The answer to *what* information is conveyed by signaling and *when* depends on the specific problem, but broadly it is understood to be metadata that endpoints and in-network elements can use to configure their behavior towards a connection when needed.

Who participates: Signaling can be performed between only the endpoints, or only between elements in the middle of the network, or between both endpoints and the middle. Endpoint-only signaling corresponds to the current Internet architecture with a "dumb" network and intelligent endpoints. Middle-only signaling corresponds to an intelligent network and dumb endpoints, much like the telephone network. Involving both endpoints and the middle in the signaling allows connection establishment intelligence to be shared.

Where are participants located: Signaling elements in the middle of the network can be located either *onpath* (i.e., along the physical path between the endpoints over which data will flow), or *offpath* (i.e., off on the side). Discovering onpath elements simply requires sending a packet to the destination, but for discovering offpath elements, the architecture must provide some discovery mechanism. Signaling, then, can be onpath-only, offpath-only, or a combination of the two. Onpath-only corresponds to the current Internet architecture that does

not have a separate control-plane. Offpath-only corresponds to an architecture with a control-plane that is tightly coupled to the data-plane. This is because, ultimately, elements on the data-plane must enforce control-plane decisions; since the data-plane elements do not participate in signaling, the control-plane must have full knowledge of the data-plane and must directly drive data-plane elements. A combination of offpath and onpath signaling allows for a loosely coupled control-plane. In this case, control-plane decisions can be more abstract, which the data-plane elements can then make concrete depending on the actual path taken by the data.

How are participants contacted: Signaling can be *implicit* or *explicit*. Implicit signaling is when the act of sending a data packet triggers elements in the network; the Internet architecture today is an example, where the first data packet establishes connection state in middleboxes. Explicit signaling is when the endhost engages in a separate signaling protocol to network elements before sending the first data packet. While implicit signaling is simpler, explicit signaling conveys richer information about the connection.

Broadly speaking, the more parties involved in signaling, and therefore the more information that is available during connection establishment, the more functionality that can be provided. At the same time, the farther away from the current Internet, which only has a data-plane, the more the deployment challenges. Figure 1.1 illustrates this design space and plots where the solutions to various End-Middle-End problems explored in this thesis lie. In the figure, the x-axis corresponds roughly to increasing functionality, and the y-axis corresponds roughly to increasing deployment challenges [2].

[2]Taking only technical issues into consideration. Issues of incentives and costs are discussed in later sections.

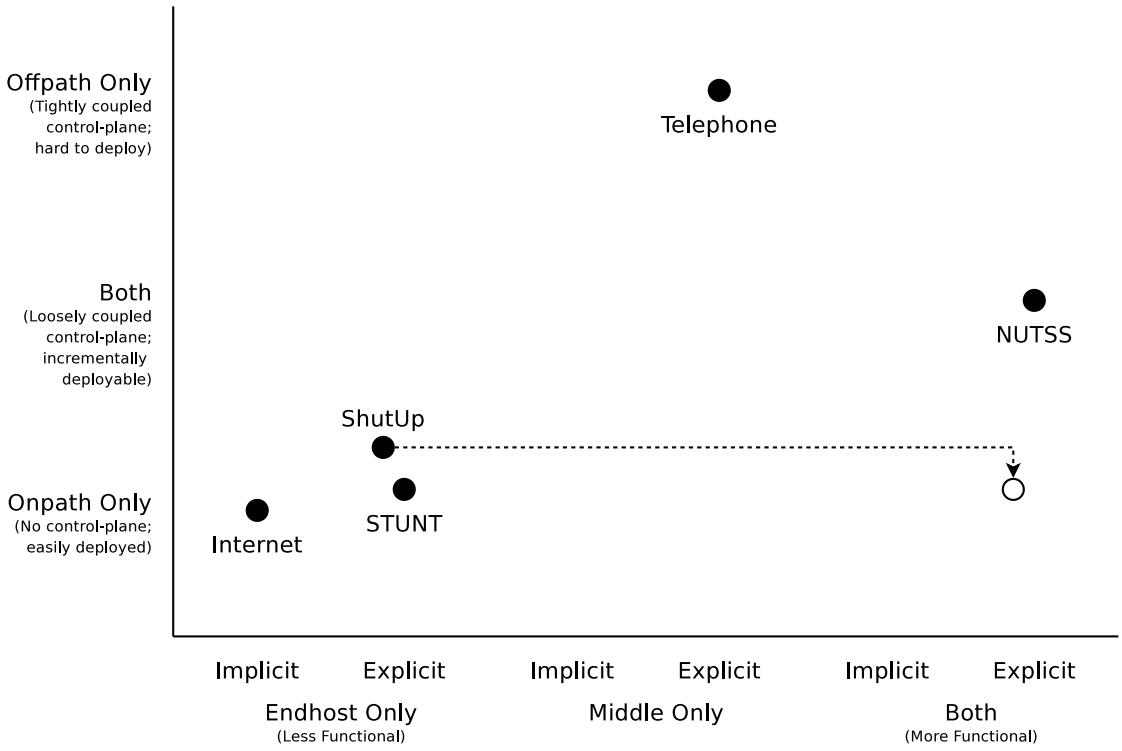


Figure 1.1: Design space for a signaling primitive, and where the systems presented in this thesis lie in comparison to existing networks.

As mentioned, the current Internet architecture corresponds to the endhost-only onpath-only implicit signaling design point in the Figure. In contrast, the telephone network uses complex signaling protocols (e.g., [99]) between gateways to negotiate such advanced functionality as mobility, billing, and steering (e.g., voice mail) before the direct voice circuit is created; this corresponds to the explicit middle-only offpath-only design point. The other three design points, namely STUNT, ShutUp, and NUTSS, represent our most practical solutions to the respective problems of NAT Traversal, unwanted traffic, and firewalls and middleboxes. We discuss these three systems in more detail below.

1.2.1 STUNT: NAT Traversal

First, we tackle the problem of NAT Traversal. The problem arises from the fundamental constraint of network address translation: multiple hosts behind a NAT share the same (external) IP address. When a host behind the NAT initiates a connection to an external host, the NAT creates state based on the connection 5-tuple (addresses, ports, protocol) that allows it to correctly route packets in the reverse direction. If, however, the external host initiates the connection, the NAT cannot disambiguate which internal host to route the connection to. When two hosts behind their respective NATs wish to communicate, neither host can successfully initiate the connection since each is external to the other's NAT.

In Chapter 2, we present a comprehensive set of solutions to the NAT traversal problem. In recent years, solutions have been developed for traversing NAT boxes using UDP (that is, establishing UDP flows between hosts behind NATs). The UDP solution relies on the connection-less nature of UDP to create NAT state. Unlike UDP, however, TCP requires connection initiation packets to be received and acknowledged by the destination before the connection is established [89]. Since NATs prevent delivery of these packets, TCP NAT traversal is more difficult. Indeed, TCP NAT traversal was considered impossible until recently when we proposed our solution.

STUNT establishes TCP connections between hosts behind NATs without any changes to existing NATs, or to existing endhost protocol stacks. We leverage the existing implicit signaling channel between the endhost and the NAT (that is, the act of initiating a connection) simultaneously on both ends to create the necessary state in both NATs. Explicit signaling between the two endpoints is needed to synchronize the connection establishment attempt.

Since our original proposal, other researchers have proposed additional TCP traversal approaches. The success of all these approaches depends on how NAT boxes in the real-world respond to various sequences of TCP (and ICMP) packets. To settle this question, we perform the first broad study of NAT behavior for a comprehensive set of TCP NAT traversal techniques over a wide range of commercial NAT products. We develop a publicly available software test suite for this purpose that measures the NAT responses both to a variety of isolated probes and to complete TCP connection establishments. We test sixteen NAT products in the lab, and 87 home NATs in the wild. Using these results, as well as market data for NAT products, we estimate the likelihood of successful NAT traversal for home networks. We find that NATs deployed on the Internet today can be traversed successfully 85%–90% of the time using the techniques presented in Chapter 2.

1.2.2 ShutUp: Reducing Unwanted Traffic

In Chapter 3, we take a fresh perspective on the problem of unwanted traffic — DoS attacks and Internet worms. The majority of existing defense proposals assume a purely in-network architecture, requiring changes to routers deployed deep in the Internet core or requiring new infrastructure to be deployed at specific points in the network. We pose, and answer in the affirmative, the question of whether a purely E2E architecture can solve DoS and worms. The challenge in doing so is to design the simplest set of primitives and mechanisms that address both classes of unwanted traffic within the same general framework.

We present the design of a "ShutUp Service", whereby the recipient of DoS

traffic explicitly signals the sender to slow down or stop. Tamper-proof endhost software, implemented for instance with trusted platforms and virtual machines, reacts to the signal by taking the appropriate action. The same basic approach is used to slow down the spread of flash worms: non-vulnerable endhosts that a worm attempts to infect explicitly signal the sender to block address or port scans performed by the worm.

As purely a deployment expedient, the design allows in-network elements to play the same role for a group of endhosts. ShutUp requires only minimal changes to the endhost, which can be achieved with buy-in from a small number of vendors, and requires no changes to Internet protocols or to the network. We present a detailed security analysis and show through experimentation that the service has little impact on legitimate traffic.

1.2.3 NUTSS: End-Middle-End Connection Establishment

In Chapter 4, we present the NUTSS architecture, protocol design, and implementation that satisfies the End-Middle-End requirements for connection establishment. NUTSS takes into account the combined policies of endpoints and network providers. Specifically, NUTSS solves a wide range of problems on the Internet, including access control, middlebox steering, multi-homing, mobility, and protocol negotiation.

While NUTSS borrows liberally from other proposals (URI-like naming, signaling to manage ephemeral IPv4 or IPv6 data flows), NUTSS is unique in that it uses explicit offpath and onpath signaling, and couples the two using lightweight mechanisms. As a result, NUTSS requires no changes to existing

network protocols, and combined with recent NAT traversal techniques, works with IPv4 and existing NATs and firewalls. Overall, NUTSS represents an argument that advanced connection establishment functionality, and incremental deployment are not mutually exclusive.

1.3 Evolving the Real-World Internet

Impact in the real world owes perhaps a small fraction to the research that went into it, and the majority to the engineering effort. A discussion about evolving the Internet without the engineering perspective is incomplete. In Chapter 5 we chronicle our efforts within the Internet Research Task Force (IRTF) and the Internet Engineering Task Force (IETF) to bridge the gap between research and practice.

We have succeeded in incorporating the lessons learned from STUNT into the specifications of designing NATs [7, 45, 110], and in incorporating the STUNT mechanism into the Session Traversal Utilities for NAT (STUN) and Interactive Connectivity Establishment (ICE) toolkits that application developers can use to traverse NATs [98, 96]. With NUTSS, we experienced a more mixed result within the End-Middle-End Research Group (EMERG) formed specifically to explore the architectural implications.

Finally, in Chapter 6, we conclude this thesis by reflecting on the challenges in bridging the gap from research to practice that go largely unnoticed by the research and engineering communities. Given the money at stake for companies maintaining and extending the Internet, we believe a purely technical perspective on evolving the Internet is no longer sufficient. As researchers, we must rise

to the challenge of tackling not only the technological issues, but doing so in a way that is aligned with the business interests of those who are in a position to bring the research into practice.

CHAPTER 2

STUNT: RESTORING GLOBAL CONNECTIVITY THROUGH NATS

The Internet architecture today is vastly different from that envisioned when TCP/IP was designed. Firewalls and Network address and port translators (NATs) often make it impossible to establish a connection even if it does not violate policy. NATs break the IP connectivity model by preventing hosts on the external side of the NAT from initiating a connection with a host behind the NAT since the external host cannot name the internal host using an IP address. If both endpoints are behind their respective NAT, ordinary TCP cannot be established since the end initiating the TCP is outside the other end's NAT. The problem is not specific to NATs; firewalls too have this problem, albeit because firewalls unilaterally block packets. Even if the connection would be allowed according to each end's firewall security policy, for instance, if the firewall policy is that internal hosts may initiate TCP connections and both hosts wish to initiate, still neither host's packet is delivered to the other host as each host is outside the other's firewall. In Section 2.3 of this chapter, we present our original set of workarounds that establish a TCP connection without the use of proxies or tunnels, and review more recent proposals [24, 11, 28]. These approaches set up the necessary connection state on the NAT or firewall[1] through a carefully crafted exchange of TCP packets. However, because NAT behavior is not standardized[2], not all NATs in the wild react the same way, causing these approaches to fail in various cases. Understanding such behavior in NATs and

[1] For the remainder of this chapter, the term NAT is understood to include firewalls.

[2] At the time this research was conducted (early 2005), there existed no Internet RFC specifying how NATs should behave. Indeed, the research presented in this chapter led directly to RFC 5382 [45], which lays out the NAT Behavioral Requirements for TCP.

measuring how much they detract from the original goal of universal connectivity in the Internet is crucial to integrating them cleanly into the architecture.

To illustrate the NAT problem, consider for instance Alice and Bob who wish to communicate with each other. Both Alice and Bob disallow unsolicited connections by hiding behind a NAT or by configuring their firewalls to drop inbound SYN packets. Yet when both Alice and Bob agree to establish a connection, there is no way to do so without reconfiguring their NAT since Alice's SYN is dropped by Bob's NAT and vice versa. Even so, NATs and firewalls have become a permanent part of the network infrastructure and will continue to remain so for a long time. Even if IPv6 is deployed globally, IPv4-IPv6 NATs will be needed during the lengthy transition, and IPv6 firewalls will be needed for security. As a result, mechanisms that enable two consenting hosts behind NATs to communicate with each other are needed.

This problem has been solved for UDP by STUN [100]. STUN leverages the basic NAT translation mechanism [109]: NATs maintain a mapping between the internal IP address and port, and the external IP address and port allocated for the first outbound packet of a flow; any packets to the external address and port are translated and routed to the internal address and port. In STUN, Alice sends a UDP packet to Bob. Although this packet is dropped by Bob's NAT, it causes Alice's NAT to create local state that allows Bob's response to be directed to Alice. A third party that both Alice and Bob are in contact with informs Bob that Alice attempted to contact him from her allocated external address and port. Bob then sends a UDP packet to Alice. Alice's NAT considers it part of the first packet's flow and routes it through, while Bob's NAT considers it a connection initiation and creates local state to route Alice's responses. This approach is

used by Skype, a popular VoIP application [9]. Unfortunately, establishing TCP is more complicated. Once Alice sends her SYN packet, her OS stack as well as her NAT expect to receive a SYNACK packet from Bob in response. However, since the SYN packet was dropped, Bob's stack doesn't generate the SYNACK. Proposed workarounds to the problem [24, 11, 28] are complicated, their interactions with NATs in the wild are poorly understood, and the extent to which they solve the problem is not known. Consequently, applications such as the file-transfer module in Skype, which require reliably in-order delivery that TCP is designed to provide, reinvent the wheel by building on top of UDP. While such approaches may work, we believe it is important that wherever possible, applications use the native OS TCP stack. This is in part to avoid increasingly complex protocol stacks, but more importantly because TCP stacks have, over the years, been carefully optimized for high performance and congestion friendliness.

In summary, this chapter describes five contributions. First, it identifies and describes the complete set of NAT characteristics important to TCP NAT traversal. Second, it reports on the prevalence both of these individual characteristics and of the success rate of peer-to-peer TCP connections for the various proposed approaches. Third, based on these measurements, it suggests modifications to the proposed approaches. Fourth, it provides insights for application developers into the implementation issues pertaining to NAT traversal. Additionally, it describes a public-domain software toolkit that can be used to measure NATs as they evolve, and can serve as the basis of TCP NAT traversal in P2P applications. Finally, the results presented in this chapter have been used to guide the standardization process of NATs and firewalls, making them more traversal friendly without circumventing security policies.

2.1 TCP NAT-Traversal

In this section we discuss the TCP NAT-traversal approaches that have been proposed in recent literature. All approaches share certain elements in common: First, in all the approaches, both ends initiate a TCP connection; this is necessary since NAT state for TCP connections can only be created by an outbound SYN packet. Second, each approach then reconciles the two TCP attempts into a single connection through different mechanisms. The reconciliation mechanism used triggers different behavior in different NATs causing the proposed approaches to fail in many instances. Third, each approach must predict the address and port the SYN will appear to come from so the other side can create the correct NAT mapping. This is performed through *port prediction*. Port prediction allows a host to guess the NAT mapping for a connection before sending the outbound SYN. Fourth, each approach also requires some coordination between the two hosts. This is accomplished over an out-of-band channel such as a connection proxied by a third party or a UDP/STUN session. Once the direct TCP connection is established, the out-of-band channel can be closed. Finally, it is possible for either endpoint to be behind multiple NATs[3]. In such cases the result of each approach depends on a composite of the behavior of all the NATs and firewalls in the path. For brevity we overload the term 'NAT' to mean the composite NAT/firewall.

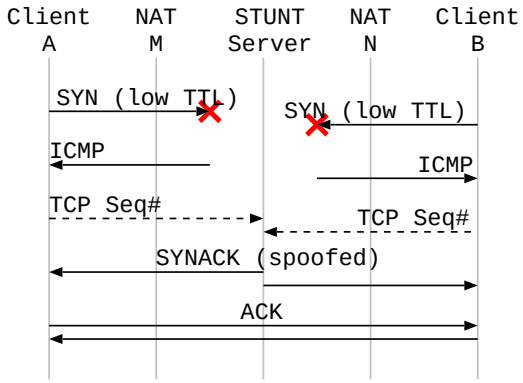


Figure 2.1: STUNT #1 Approach. Solid lines are TCP/IP and ICMP packets pertaining to the connection attempt while dotted lines are control messages sent over an out-of-band channel.

2.1.1 STUNT

In [48], we proposed two approaches for traversing NATs. In the first approach (STUNT #1), illustrated in 2.1, both endpoints send an initial SYN with a TTL[4] high enough to cross their own NATs, but small enough that the packets are dropped in the network (once the TTL expires). The endpoints learn the initial TCP sequence number used by their OS's stack by listening for the outbound SYN over PCAP or a RAW socket. Both endpoints inform a globally reachable STUNT server of their respective sequence numbers, following which the STUNT server spoofs a SYNACK to each host with the sequence numbers appropriately set. The ACK completing the TCP handshake goes through the network as usual. This approach has four potential problems. First, it requires the host to determine a TTL both large enough to cross its own NATs and small enough to not reach the other end's NAT. Such a TTL does not exist when the two outermost NATs share a common interface. Second, the ICMP TTL-exceeded error may be generated in response to the SYN packet and be inter-

[3]sometimes referred to as *dual* or *double NAT*
[4]IP time-to-live field

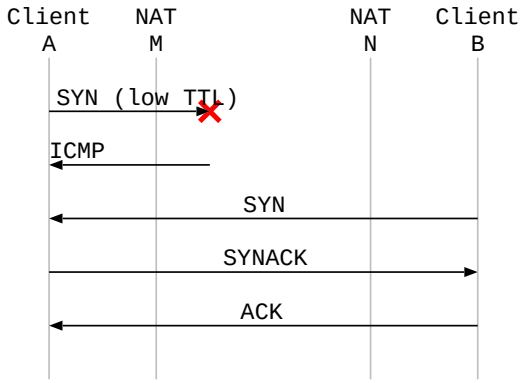


Figure 2.2: STUNT #2 Approach. Solid lines are TCP/IP and ICMP packets pertaining to the connection attempt while dotted lines are control messages sent over an out-of-band channel.

preted by the NAT as a fatal error. Third, the NAT may change the TCP sequence number of the initial SYN such that the spoofed SYNACK based on the original sequence number appears as an out-of-window packet when it arrives at the NAT. Fourth, it requires a third party to spoof a packet for an arbitrary address, which may be dropped by various ingress and egress filters in the network. These network and NAT issues are summarized in Table 2.1.4.

In the second approach (STUNT #2) proposed in [48], similar to the one proposed in [24], only one host sends out a low-TTL SYN packet. This sender then aborts the connection attempt and creates a passive TCP socket on the same address and port. The other endpoint then initiates a regular TCP connection, as illustrated in Figure 2.2. As with the first case, the host needs to pick an appropriate TTL value and the NAT must not consider the ICMP error a fatal error. It also requires that the NAT accept an inbound SYN following an outbound SYN — a sequence of packets not normally seen.

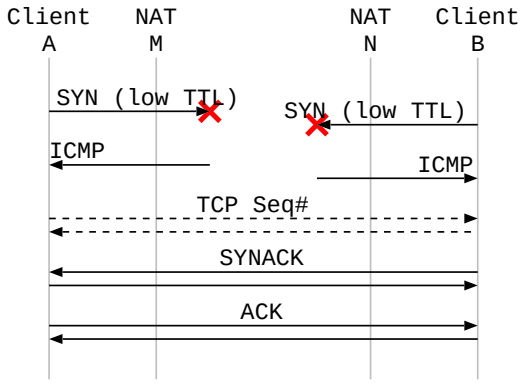


Figure 2.3: NATBlaster Approach. Solid lines are TCP/IP and ICMP packets pertaining to the connection attempt while dotted lines are control messages sent over an out-of-band channel.

2.1.2 NATBlaster

In [11], the authors propose an approach similar to the first STUNT approach but do away with the IP spoofing requirement (Figure 2.3). Each endpoint sends out a low-TTL SYN and notes the TCP sequence number used by the stack. As before, the SYN packet is dropped in the middle of the network. The two hosts exchange the sequence numbers and each crafts a SYNACK packet the other expects to receive. The crafted packet is injected into the network through a RAW socket; however, this does not constitute spoofing since the source address in the packet matches the address of the endpoint injecting the packet. Once the SYNACKs are received, ACKs are exchanged completing the connection setup. As with the first STUNT approach, this approach requires the endpoint to properly select the TTL value, requires the NAT to ignore the ICMP error and fails if the NAT changes the sequence number of the SYN packet. In addition, it requires that the NAT allow an outbound SYNACK immediately after an outbound SYN – another sequence of packets not normally seen.

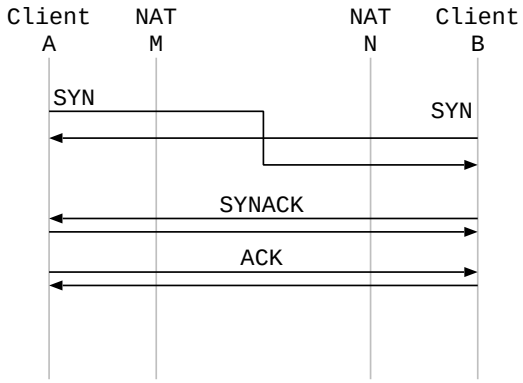


Figure 2.4: P2PNAT Approach. Solid lines are TCP/IP and ICMP packets pertaining to the connection attempt while dotted lines are control messages sent over an out-of-band channel.

2.1.3 Peer-to-Peer NAT (P2PNAT)

In [28], the authors take advantage of the simultaneous open scenario defined in the TCP specifications [89]. As illustrated in Figure 2.4, both endpoints initiate a connection by sending SYN packets. If the SYN packets cross in the network, both the endpoint stacks respond with SYNACK packets establishing the connection. If one end's SYN arrives at the other end's NAT and is dropped before that end's SYN leaves that NAT, the first endpoint's stack ends up following TCP simultaneous open while the other stack follows a regular open. In the latter case, the packets on the wire look like the STUNT #2 approach without the low TTL and associated ICMP. While the P2PNAT approach proposed in [28] does not use port-prediction, the approach can benefit from it when available. As with the STUNT # approach, P2PNAT requires that the NAT accept an inbound SYN after an outbound SYN. In addition, the approach requires the host to retry failed connection attempts in a tight loop until a timeout occurs. If instead of dropping the SYN packet a NAT responds to it with a TCP RST, this approach devolves into a packet flood until the timeout expires.

2.1.4 Implementation

We implemented STUNT #1 and #2, NATBlaster, and P2PNAT on both Linux and Windows. We also developed a Windows device driver that implements the functionality required by the approaches that are not natively supported by Windows. The STUNT # approach requires superuser privileges under both Windows and Linux to overhear the TCP SYN packet on the wire and learn its sequence number. In order to set the TTL on the first SYN packet, we use the `IP_TTL` socket option under Linux and our driver under Windows. We also implemented the STUNT server and host it behind an ISP that does not perform egress filtering in order to spoof arbitrary addresses. While the server was able to spoof most SYNACKs, it was not successful in spoofing SYNACKs where both the source and destination were in the same administrative domain and the domain used ingress filtering. When possible, an additional STUNT server is installed inside such domains. The STUNT #2 approach requires the driver to set the TTL under Windows. The NATBlaster approach requires superuser privileges to learn the sequence number of the SYN and to inject the crafted SYNACK through a RAW socket. Due to a restriction introduced in Windows XP SP2, the approach requires the driver to inject this packet. The P2PNAT approach requires the OS to support TCP simultaneous open; this is supported under Linux and Windows XP SP2 but not by Windows XP prior to SP2. On Windows XP SP1 and earlier, our driver adds support for this. These implementation issues are summarized in the top half of Table 2.1.4.

We found that setting the TTL is problematic under Windows; therefore, we consider the consequences of not using it. If the TTL is not reduced, the first SYN sent by one of the hosts reaches the other end's NAT before that end's SYN

Table 2.1: NAT and network issues encountered by various TCP NAT-traversal approaches as well as the implementation issues we encountered. Section 2.1.4 describes each issue in detail.

Approach	**NAT/Network Issues**	**Linux Issues**	**Windows Issues**
STUNT #1	• Determining TTL • ICMP error • TCP Seq# changes • Spoofing	• Superuser priv.	• Superuser priv. • Setting TTL
STUNT #2	• Determining TTL • ICMP error • SYN-out SYN-in		• Setting TTL
NATBlaster	• Determining TTL • ICMP error • TCP Seq# changes • SYN-out SYNACK-out	• Superuser priv.	• Superuser priv. • Setting TTL • RAW sockets (post WinXP SP2)
P2PNAT	• TCP simultaneous open • Packet flood		• TCP simultaneous open (pre WinXP SP2)
STUNT #1 default-TTL	• RST error • TCP Seq# changes • Spoofing	• Superuser priv.	• Superuser priv. • TCP simultaneous open (pre WinXP SP2)
STUNT #2 default-TTL	• RST error • SYN-out SYN-in		
NATBlaster default-TTL	• RST error • TCP Seq# changes • SYN-out SYNACK-out	• Superuser priv.	• Superuser priv. • RAW sockets (post WinXP SP2) • TCP simultaneous open (pre WinXP SP2)

exits the same NAT. The NAT can either silently drop the inbound packet, or respond with an ICMP unreachable error or a TCP RST/ACK. The response, if any, may trigger transitions in the sender's NAT and OS stack unaccounted for by the approach. If the TTL for the other end's SYN packet is not reduced either, the SYN may reach the intended destination triggering unforeseen transitions. The behavior may be favorable to the ultimate goal if, for instance, it triggers a TCP simultaneous-open, or it may be detrimental if it confuses the stack or NAT. To test the outcome of not lowering the TTL, we implement modified versions of the above approaches that use the default TTLs set by the operating system. Issues encountered are summarized in the bottom half of Table 2.1.4.

The astute reader will have noticed that the detailed description promised for each issue in Table 2.1.4 has been omitted from this section. This is intentional. We encourage the reader to contact the author for these details and a small reward.

2.2 Experiment Setup

We have defined the STUNT client-server protocol that both tests NAT/firewall behavior and assists in establishing TCP connections between NATed peers. A complete protocol description is available in [43]. The protocol is implemented by our test applications comprising of a client component and server component. As shown in Figure 2.5, the client is run on a host behind one or more NATs while the server is external to all of them. The STUNT test client detects the composite behavior of all the NATs and firewalls between the client and the server. While both the test client and server require superuser privileges to an-

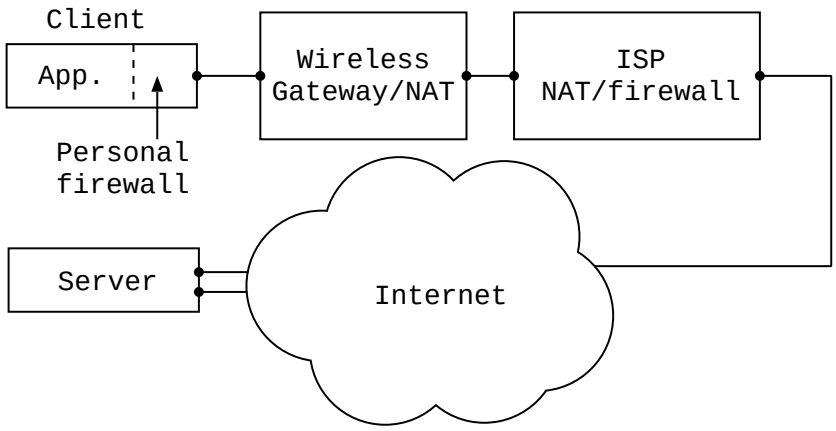


Figure 2.5: A possible experiment setup for STUNT. Client component is behind multiple NATs and the server component is outside all of them. The behavior determined by STUNT is the composite of the individual NAT behaviors.

alyze raw packets on the wire, the requirement can be dropped for the client in exchange for a small loss in functionality. The server, in addition, requires at least two network interfaces to properly differentiate between various NAT port allocation algorithms in use. The tests performed by the client and the NAT characteristics inferred from them are described later in Section 2.3.

We used the client to test a diverse set of sixteen NATs in the lab (Table 2.2). These include one of each brand of NAT that we could find in online stores in the United States. The NATs tested also include software NAT implementations in popular operating systems. In each lab test, the client host was the only host internal to the NAT. To measure the latency introduced by the approaches, the server was located on the same Ethernet segment as the external interface of that NAT. The client host was set to not generate any network traffic other than that caused by the test client.

In addition to these lab tests, we also tested NAT boxes in the wild. The main reason for this was to expose our test software to a wider range of scenarios than

Table 2.2: NATs tested in the lab, chosen to represent a variety of brands and implementations.

Brand	**Model**	**Firmware**
3Com	3C857	2.02
Allied Telesyn	AT-AR220E	R1.13
Belkin	F5D5231-4	1.0.0
Buffalo	WYR-G54	1.0 r31
Checkpoint	VPN-1/FireWall-1	(NGAI) release 55
DLink	DI-604	3.30
Linksys	BEFSR41	1.40.2
Linux	iptables	2.4.20
Netgear	RP614	5.13
Netopia	3386	8.0.10
Open BSD	pf	3.5
SMC	SMC7004VBR	R1.00
Trendnet	TW100-S4W1CA	1.02.000.267
USR	8003	1.04 08
VMWare	Workstation	4.5.2
Windows XP	Internet Connection Sharing	SP2

we could reproduce in the lab, thus improving its robustness and increasing our confidence in its operation. In addition, it provided a sample, albeit small, of what types of NAT we can expect to see in practice. Experimenting with NATs in the wild was vital to the impact created by this work: at the time this experiment was conducted, there were no defined specifications for NATs to follow and the behavior of deployed NATs with respect to non-common-case traffic (i.e. non outbound 3-way handshake) was poorly understood; this work lead directly to defining such a specification [45].

For testing against NATs in the wild, we requested home users to run the test client. This tested 87 home NATs (16 unique brands) being used by CS faculty and students at Cornell and other universities. Test traffic was in addition to typical network activity on the client host and other hosts behind the NAT and included web browsing, instant messaging, peer-to-peer file-sharing,

Table 2.3: Observed market share of NAT brands in our sample set and worldwide SOHO/Home WLAN market share of each brand in Q1 2005 according to Synergy Research Group

Brand	Sample	Market Survey
Linksys	24.1%	28.8%
D-Link	9.2%	20.2%
Netgear	6.9%	14.7%
Buffalo Technologies	1.1%	10.9%
Belkin	9.2%	4.6%
Other	49.4%	20.9%

email, etc. The resulting data draws from a mix of NAT brands with both new and old models and firmware; however, it admits a bias in the selection of NATs given the relatively small user base with most of them living in the north-eastern United States. This discrepancy is evident in Table 2.2, where the observed popularity of brands in our sample is listed under 'Sample' and the worldwide SOHO/Home WLAN market share of the brands in the first quarter of 2005 as per the Synergy Research Group [120] is listed under 'Market Survey'. In particular, Buffalo Technologies and Netgear were under-represented in our sample and the percentage of other brands was significantly higher. The full list of home NATs tested is available in [44].

2.3 NAT TCP Characteristics

In this section, we identify how different NATs affect TCP NAT-traversal approaches. We identify five classifications for NAT behavior; namely, NAT mapping, endpoint packet filtering, filtering response, TCP sequence number preserving and TCP timers. The classifications and the possible values that a NAT

Table 2.4: Important categories distinguishing various NATs.

Classification	**Values**
NAT Mapping	Independent Address$_\delta$ Port$_\delta$ Address and Port$_\delta$ Connection$_\delta$
Endpoint Filtering	Independent Address Port Address and Port
Response	Drop TCP RST ICMP
TCP Seq#	Preserved Not preserved
Timers	Conservative Aggressive

can receive in each class are listed in Table 2.3. We use the STUNT testing client and server described earlier in Section 2.2 to classify a collection of sixteen NATs in the lab and eighty-seven NATs in the wild. The full set of test results with a wider set of classifications is available in [44].

2.3.1 NAT Mapping

A NAT chooses an external mapping for each TCP connection based on the source and destination IP and port. Some NATs reuse existing mappings under some conditions while others allocate new mappings every time. The *NAT Mapping* classification captures these differences in mapping behavior. This knowledge is useful to hosts attempting to traverse NATs since it allows them to predict the mapped address and port of a connection based on previous connec-

Table 2.5: NAT Mapping test behavior observed. Nat1–5 show the 5 different mapping patterns that are observed in practice. Nat6 is a possible mapping pattern that has not been observed in our sample set.

#	From	To	Nat1	Nat2	Nat3	Nat4	Nat5	Nat6
1	a:p	B:Q	A:P	A:P	A:P	A:P_1	A:P	A:P
2	a:p	B:Q	A:P	A:P	A:P+1	A:P_2	A:P	A:P
3	a:p	B:Q	A:P	A:P	A:P+2	A:P_3	A:P	A:P
4	a:p	B:R	A:P	A:P+1	A:P+3	A:P_4	A:P+1	A:P
5	a:p	B:R	A:P	A:P+1	A:P+4	A:P_5	A:P+1	A:P
6	a:p	B:R	A:P	A:P+1	A:P+5	A:P_6	A:P+1	A:P
7	a:p	C:R	A:P	A:P+2	A:P+6	A:P_7	A:P+1	A:P+1
8	a:p	C:R	A:P	A:P+2	A:P+7	A:P_8	A:P+1	A:P+1
9	a:p	C:R	A:P	A:P+2	A:P+8	A:P_9	A:P+1	A:P+1
10	a:p	C:Q	A:P	A:P+3	A:P+9	A:P_{10}	A:P	A:P+1
11	a:p	C:Q	A:P	A:P+3	A:P+10	A:P_{11}	A:P	A:P+1
12	a:p	C:Q	A:P	A:P+3	A:P+11	A:P_{12}	A:P	A:P+1
13	a:s	B:Q	A:S	A:S	A:S	A:S_1	A:S	A:S
⋮								
Classification			NB: Independent	NB:Address and $Port_1$	NB: $Connection_1$	NB: $Connection_X$	NB: $Port_1$	NB: $Address_1$

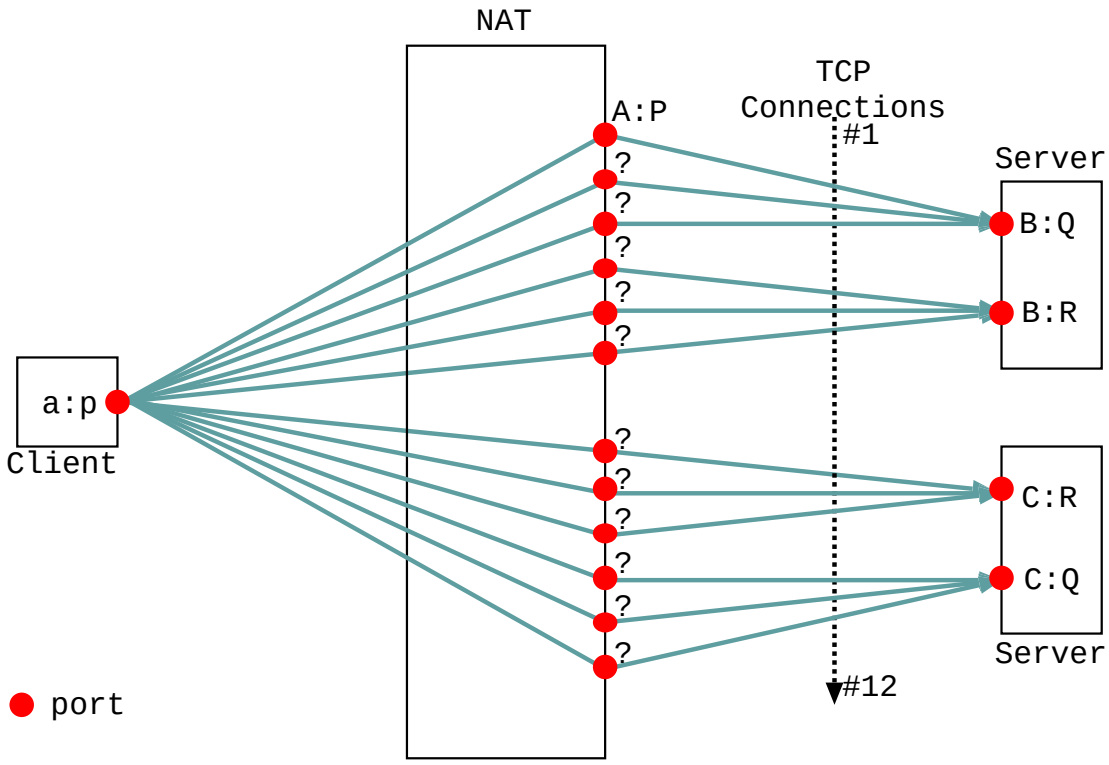


Figure 2.6: TCP connections established to find NAT Mapping classification. Client uses the same local IP address and port `a:p` to connect three times to two ports `Q` and `R` on two servers at IP addresses `B` and `C`. The pattern of mapped address and port for each connection determines the NAT Mapping classification.

tions. For UDP, it is known that some NATs assign a fixed address and port for all connections originating from a fixed source address and port [100]. We test for similar behavior for TCP in the STUNT client. The client establishes 12 connections in close succession from a fixed local address and port `a:p` to various server addresses and ports as shown in Figure 2.6 and tabulated in Table 2.3. Each connection is closed before the next is initiated. The server echoes back the mapped address and port it perceives to the client. For example, the first connection is initiated from local address and port `a:p` to the server at `B:Q`; the NAT allocates `A:P` for the connection, which the server echoes back. The con-

nection is closed, and a second connection is initiated from the same local port to the same server port; four out of the six types on NATs (Nat1–2,Nat5–6) reuse the `A:P` allocation, while Nat3 allocates a new external port `A:P+1` and Nat4 allocates a random other port (`A:`P_2). The third connection once again contacts the server at `B:Q`, the fourth connection contacts the server at a different port (`B:R`), and so on. The test is repeated multiple times for different choices of the local port (e.g. row 13).

We notice several distinct patterns among the mapped ports shown as Nat1–Nat6 in Table 2.3. Let the mapping allocated for the first connection be called `A:P` for each NAT. Nat1 reuses this mapping as long as the client source address and port of a new connection matches that of the first connection. We classify this behavior as *NB:Independent* since the mapping is determined only by the source address and port and is independent of the destination address and port. This is equivalent to *cone behavior* in [100] extended to include TCP. Nat2 reuses the mapping only if both the source and destination address and port for the new connection match the first connection. Such NATs are classified *NB:Address and Port*$_1$ since both the destination address and port affect the mapping. The subscript '$_1$' signifies that the difference between new mappings, denoted by δ, is 1. [113] shows that for UDP, δ is fixed for many NATs and is usually 1 or 2. We find that the same holds for TCP as well. All of the NATs we encountered, however, have $\delta = 1$. Nat3 allocates a new mapping for each new connection, however, each new mapping has port $\delta = 1$ higher than the previous port. We classify Nat3 as *NB:Connection*$_1$. Nat4, like Nat3, allocates a new mapping for each TCP connection but there is no discernable pattern between subsequent mappings. We classify such NATs *NB:Connection*$_\mathfrak{R}$ where the subscript '$_\mathfrak{R}$' indicates a random δ. Nat5 is a variation of Nat2 where the map-

Table 2.6: NAT mapping types observed in a set of 16 NATs in the lab, and that estimated for NATs in the wild based on our sampling of 81 home NATs and worldwide market shares.

NAT Mapping	**Lab**	**Wild**
NB:Independent	9	70.7%
NB:Address and Port_1	3	23.1%
NB:Connection_1	3	3.6%
NB:Port_1	0	2.2%
NB:Address_δ	0	0.0%
NB:$\text{Connection}_{\mathfrak{R}}$	1	0.5%

ping is reused if the destination port matches in addition to the source address and port. Nat6 is similar except the destination address needs to match instead of the port. Together Nat5 and Nat6 are classified *NB:Port*$_1$ and *NB:Address*$_1$ respectively. NATs 2–6 display *symmetric behavior* as per [100].

Table 2.3.1 shows the relative proportion of each type of NAT. Column 2 shows the number of NATs from our testbed of sixteen NATs that were classified as a particular type. A majority of them are NB:Independent. The only one that is NB:$\text{Connection}_{\mathfrak{R}}$ is the NAT implementation in OpenBSD's `pf` utility. We also noticed that our Netgear RP614 NAT with firmware 5.13 is NB:Connection_1, however, more recent Netgear NATs such as MR814V2 with firmware 5.3_05 are NB:Independent. Column 3 estimates the behavior of NATs in the wild. The estimates are computed by taking the proportion of each type and brand of NAT from eighty-one home NATs sampled and scaling them with a correction factor chosen to overcome the bias in our sample. The correction factor for each brand is the ratio between the surveyed and observed market shares presented in Table 2.2. The factor serves to increase the contribution of under-represented brands in the estimated result and decrease the contribution of over-represented brands. While our estimates are indicative of the general trend to the best of

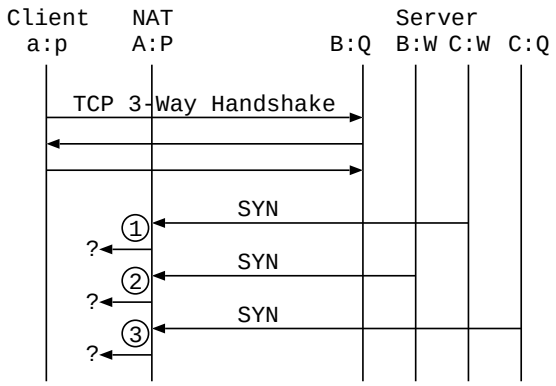


Figure 2.7: TCP packets exchanged for Endpoint Filtering test. Client establishes a connection to `B:Q`. Packet (1) is an inbound SYN from a different address and port (`C:W`), (2) from the same address but different port (`B:W`) and (3) from the same port but different address (`C:Q`). The response to each of these determines the Endpoint Filtering classification.

our knowledge, we note that in an industry changing at the rate of 47% per year [120] the accuracy of any results is short lived at best. Nevertheless, we estimate a majority of the NATs (70.7%) to be NB:Independent and almost none to be NB:Connection$_{\not R}$. A significant percent (29.3%) of NATs have *symmetric behavior* as defined in [100]. Consequently in a large fraction of cases, multiple connections from the same port will not be assigned the same mapping and applications must employ the more sophisticated port-prediction techniques described later.

2.3.2 Endpoint Filtering

Both NATs and firewalls may filter inbound packets addressed to a port unless certain conditions are met. If no NAT mapping exists at that port, a NAT is forced to filter the packet since it cannot forward it. If a mapping exists, how-

Table 2.7: NAT endpoint filtering behavior observed. Nat1′–Nat4′ show 4 different filtering behaviors that are observed for inbound SYN packets after an internal host establishes a connection from `a:p` to `B:Q` with allocated mapping `A:P`.

#	From	To	Nat1′	Nat2′	Nat3′	Nat4′
1	`C:W`	`A:P`	accepted	filtered	filtered	filtered
2	`B:W`	`A:P`	accepted	filtered	accepted	filtered
3	`C:Q`	`A:P`	accepted	filtered	filtered	accepted
Classification			EF:Independent	EF:Address and Port	EF:Address	EF:Port

ever, or if it the device is a firewall, then it may require that the source address and/or port of the inbound packet match the destination of a preceding outbound packet. These differences in conditions that trigger filtering are captured by the *Endpoint Filtering* classification. The STUNT test client determines this by first establishing NAT state by connecting to the server. It then requests the server to initiate connections to the mapped address and port from different addresses and ports as shown in Figure 2.7.

The different filtering behaviors observed for the test are tabulated in Table 2.3.2. Nat1′ accepts all three SYN packets. Such NATs allow inbound TCP connections independent of the source address and port as long as necessary state exists for routing the request. We classify such NATs as having the endpoint filtering behavior *EF:Independent*. Nat2′ filters all the packets thus requiring the source of the inbound TCP packet match both the address and port of the destination of the connection that created the mapping. The endpoint filtering of such NATs is classified *EF:Address and Port*. Nat3′ and Nat4′ allow inbound packets from the same address or port as the destination address of the connection but filter packets from a different address or port respectively. We classify the endpoint filtering behavior of such NATs as *EF:Address* and *EF:Port*

Table 2.8: NAT endpoint filtering types observed in a set of 16 NATs in the lab, and that estimated for NATs in the wild based on our sampling of 87 home NATs and worldwide market shares.

Endpoint Filtering	**Lab**	**Wild**
Address and Port	12	81.2%
Address	1	12.3%
Independent	3	6.5%

respectively. In general we find that endpoint filtering behavior of a NAT is independent of NAT mapping behavior. The subclassifications of cone NATs defined in [100] translate as follows: *full cone* is equivalent to NB:Independent and EF:Independent, *restricted cone* is NB:Independent and EF:Address and Port and *port restricted cone* is NB:Independent and EF:Port.

Table 2.3.2 shows the endpoint filtering classification for sixteen NATs in the lab and the estimated percentage of NATs in the wild. The estimates are computed based on the market survey as described earlier. 81.2% of the NATs are estimated to be EF:Address and Port while only 6.5% are EF:Independent. This implies that in most cases, to establish a connection between two NATed hosts an outbound SYN must be sent from each end before inbound packets are accepted.

TCP State Tracking

NATs implement a state machine to track the TCP stages at the endpoints and determine when connection state can be garbage-collected. While all NATs handle the TCP 3-way handshake correctly, not all of them implement the corner cases of the TCP state machine correctly, thereby prematurely expiring connection state. The STUNT client and server test how NAT/firewall implementa-

Table 2.9: Percentage of NATs not accepting various packet sequences. Inbound packets (-in) are in response to preceding outbound packets (-out). ICMP code used is TTL-exceeded (non-fatal error).

Sequence	**Filtered**
SYN-out SYNACK-in	0%
SYN-out SYN-in	13.4%
SYN-out ICMP-in SYNACK-in	7.0%
SYN-out ICMP-in SYN-in	22.7%
SYN-out RST-in SYNACK-in	19.8%
SYN-out RST-in SYN-in	27.8%

tions affect TCP NAT-traversal approaches by replaying the packet sequences observed for these approaches.

Table 2.3.2 lists some of the packet sequences tested. We estimate that 13.4% of NATs do not support TCP simultaneous open where an outbound SYN is followed by an inbound SYN. This affects the P2PNAT approach, which requires at least one end support simultaneous open as well as the second STUNT approach. 7.0% filter inbound SYNACK packets after a transient ICMP TTL-exceeded error. A similar number of NATs drop the inbound SYN packet after the ICMP but accept it in the absence of the error. This behavior affects all the approaches that set low TTLs on SYN packets. A fair number of NATs (27.8%) accept inbound SYN packets even after the SYN packet that created the connection state is met with a fatal TCP RST. This mitigates the issue of spurious RSTs that some approaches contend with. Not mentioned in the table is the sequence SYN-out SYNACK-out that is required for the NATBlaster approach. We did not test this case widely due to restrictions introduced by Windows XP SP2. In the lab, however, we found that the D-Link NAT (DI-604) does not support it.

Filtering Response

When an inbound packet is filtered by a NAT it can choose to either drop the packet silently or notify the sender. An estimated 91.7% of the NATs simply drop the packet without any notification. The remaining NATs signal an error by sending back a TCP RST acknowledgment for the offending packet.

2.3.3 Packet Mangling

NATs change the source address and port of outbound packets and the destination address and port of inbound packets. In addition, they need to translate the address and port of encapsulated packets inside ICMP payloads so hosts can match ICMPs to their respective transport sockets. All the NATs in our sample set either perform the ICMP translation correctly or filter the ICMP packets, which are not always generated in the first place. Some NATs change the TCP sequence numbers by adding a constant per-flow offset to the sequence number of outbound packets and subtracting the same from the acknowledgment number of inbound packets. We estimate that 8.8% of NATs change the TCP Sequence Number. Consequently in some cases, TCP NAT-traversal approaches that require the initial sequence number of the packet leaving the NAT cannot use the sequence number of the SYN at the end host in its stead.

2.3.4 TCP Timers

NATs and firewalls cannot indefinitely hold state since it makes them vulnerable to DoS attacks. Instead they expire idle connections and delete connection state

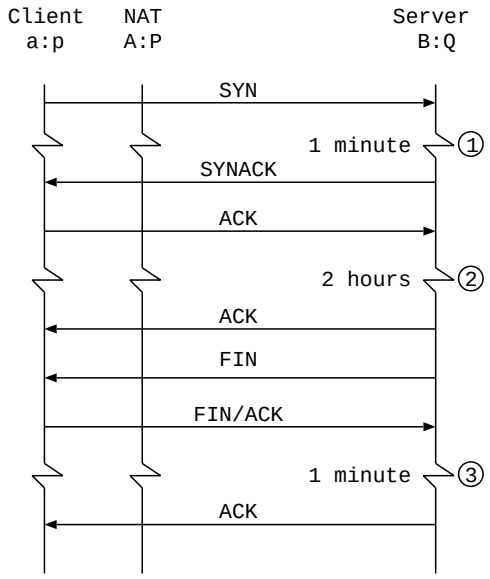


Figure 2.8: NAT timers in effect during a TCP connection. (1) Timer in SYN_SENT state, (2) Timer in established state, (3) Timer in TIME_WAIT.

for crashed or misbehaving endpoints. In addition, they monitor TCP flags and recover state from connections explicitly closed with a FIN/FINACK exchange or RST packets. They might allow some grace time to let in-flight packets and retransmissions be delivered. Once connection state has been de-allocated, any late-arriving packets for that connection are filtered. NATs typically use different timers for these cases, as illustrated in Figure 2.8. At location 1 and 3 in the figure, NATs use a short timer to expire connections not yet established or connections that have been closed respectively. RFC 1122 [13] requires that all Internet hosts wait for 4 minutes (2×`MSL`[5]) for in-flight packets to be delivered; however, most operating systems wait for about 1 minute instead. At location 2 in the figure, NATs use a longer timer for idle connections in the established state. RFC 1122 requires that TCP stacks should wait for at least 2 hours between sending `TCP-Keepalive` packets over idle connections.

[5]Maximum Segment Length

The STUNT test client checks the NAT timers for compliance with current practice and RFCs. It does so by performing three timed tests to check each case separately. In the first test, a TCP connection is initiated by the client, but the SYNACK from the server is delayed by a little under a minute. In the second test, the connection is established and left idle for a little under 2 hours, at which point a couple of bytes are sent across from the server. In the third test, the connection is established and then closed but the last ACK from the server is delayed by about a minute. In each case if the packet sent from the server after the introduced delay is delivered to the client then the corresponding timer is termed *conservative*, otherwise it is termed *aggressive*. We estimate only 22.6% of the NATs have conservative timers for all three cases while 31.3% have a conservative timer for the second case. 21.4% of the NATs have an extremely aggressive timer for the second case where they expire an established connection after less than 15 minutes of inactivity. This implies that applications should not rely on idle connections being held open for more than a few minutes.

2.4 Port Prediction

Port prediction allows a host to predict its own mapped address and port for a connection it is about to initiate. It therefore allows two hosts to initiate a connection with each other's mapped address and port even though the mapping is allocated by the NAT *after* the connection is initiated. Figure 2.9 shows a typical TCP NAT-traversal attempt using port-prediction information. In the figure, we assume that A has already determined the type of NAT it is using. When client A wishes to establish a connection with client B, A first establishes a TCP connection to the STUNT server and learns the mapping. Based on the NB:type of NAT

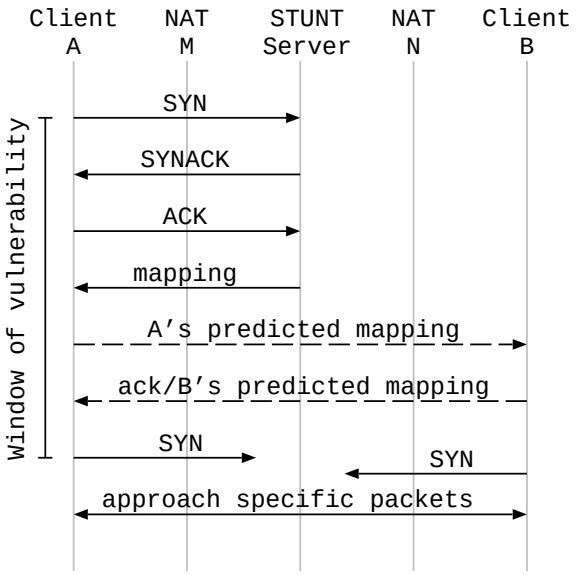


Figure 2.9: Port-prediction in TCP NAT-Traversal approaches.

M, A predicts the mapping for the next connection. B does the same and both A and B exchange their predictions over an out-of-bound channel. Each end then initiates a connection to the other's mapped address and port by sending a SYN packet. The remainder of the packets exchanged is designed to reconcile the two TCP attempts into one connection and vary from one NAT-Traversal approach to another as described in Section 2.1. The period between the first SYN and the SYN for the target connection may be a *window of vulnerability* depending on the type of NAT M. For some types of NAT, if another internal host A' behind NAT M initiates an outbound connection in this period, M will allocate the mapping predicted by A to the connection from A' instead.

Port-prediction depends on the NAT Mapping type explored earlier in Section 2.3.1. If the NAT is of type NB:Independent then the mapping for the connection to the STUNT server will be reused for any connection initiated soon afterward from the same source address and port. Since the reuse of the mapping is completely under the client's control, the window of vulnerability does

not exist in this case. However, this approach introduces a latency of 2×RTT[6] to the STUNT server before the mapping can be predicted. For a possible optimization, we noticed that a number of NATs usually allocate a mapped port equal to the source port used by the client. We term these NATs *port preserving*. Clients behind such a NAT can, with high probability, predict the mapped port without first establishing a connection to the STUNT server. If the NAT is not NB:Independent but has a fixed δ then a connection initiated immediately after the server connection will have a mapped port δ higher than the mapped port observed by the server. Since the mapping changes from connection to connection, a "rogue" connection attempt in the window of vulnerability can steal the mapping. In addition, this approach fails if the predicted mapping is already in use, causing the NATs allocation routine to skip over it.

We implemented port-prediction in the STUNT test client and predicted mappings for seventy-nine home users for an hour. Every minute, the test client initiates a connection to the STUNT server from a source address and port and learns the mapping allocated. Next it uses the same source address and port to initiate a connection to a remote host setup for the purpose of this experiment. The test client checks the mapping actually observed for the second connection against the one predicted based on the mapping for the first and type of NAT. Port-prediction is successful if and only if they match. The predictions are performed while users use their host and network normally. This includes web browsers, email readers, instant messaging and file-sharing applications running on the client host and other hosts behind the same NAT. 89.8% of the NB:Independent NATs are port preserving. This represents a big win for interactive applications that implement the optimization above. We find that in

[6]Round-trip time

82.3% of the cases the port was predicted correctly every time. This includes all but one of the NB:Independent NATs and 36.4% of the non-NB:Independent NATs. For the remaining 63.6% of the latter variety, at least one time out of the sixty another host or application stole the mapping the test client predicted for itself. In one particular case, the client host behind a NB:Connection$_1$ NAT was infected with a virus that generated several randomly-addressed SYN packets per second causing all predictions to fail! In another case, the user initiated a VPN connection midway through the test causing all subsequent requests to be sent over the VPN and thus through a different type of NAT. This suggests that long-running applications may cache the NAT mapping type for some time but must revalidate it from time to time. Overall, in 93.7% of the cases, more than three-fourths of the port-predictions were correct. Hence after a failed attempt if an application simply retries the connection, it is likely to succeed.

2.4.1 Effect of Port-Prediction Vulnerability Window

Outbound connections initiated by other applications or internal hosts during the window of vulnerability can foil a port-prediction attempt. This window of vulnerability lasts for about 3×RTT as illustrated in Figure 2.9. In [133], the authors measure RTTs for 16 million host-pairs and find that under normal network operation the maximum RTT is 1 second (250ms in the median case). Given this, we try to answer the question: what is the likelihood that a port-prediction attempt will fail due to another interfering connection attempt.

We first try to understand how typical hosts initiate connections. We traced TCP SYN packets from 641 hosts in the Cornell CS Department to hosts outside

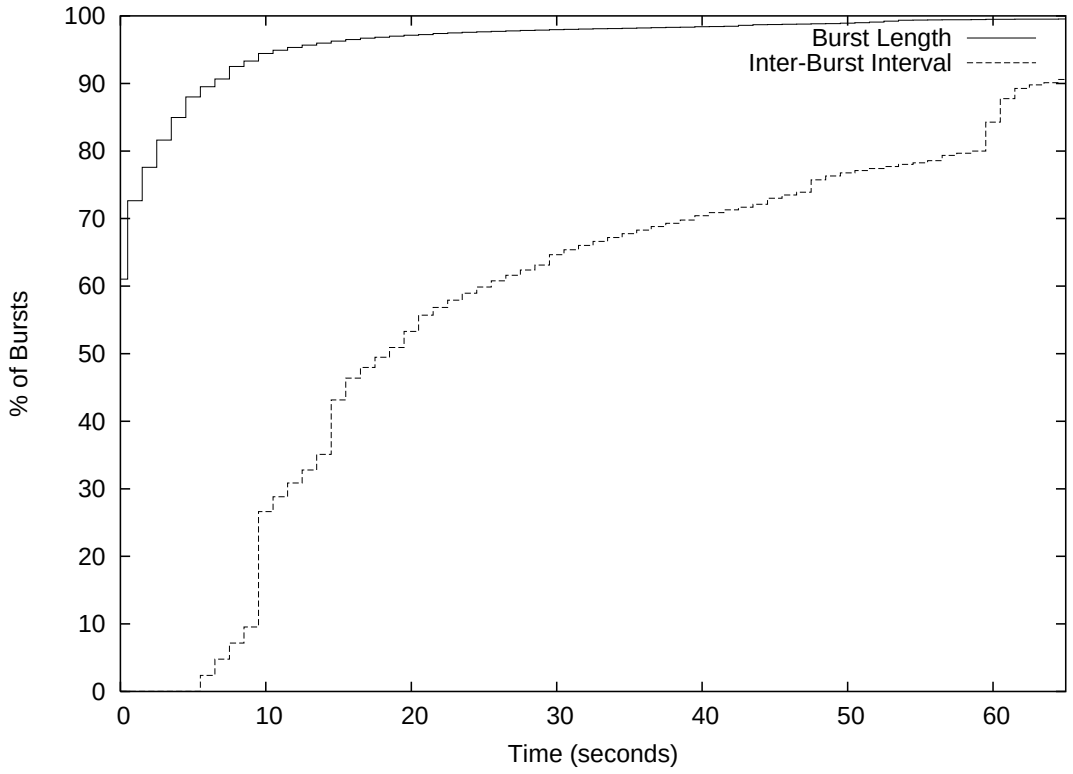


Figure 2.10: CDF of length and inter-arrival time of bursts.

for 33 days for a total of 13.6 million TCP connections. The hosts traced included hosts used by faculty, staff and students as their primary terminals, data acquisition and compute clusters but not webservers, mail servers and other hosts placed in the DMZ. The trace was collected at the department's firewall that only allows outbound connections. We find that the arrival of new connections is bursty in accordance with the observations made in [25]. Hosts established about 820 connections per day on average but 88.8% of the connections were initiated within 5 seconds of another connection from the same host. This is because a large fraction of the connections are HTTP requests where the browsers download a website and then initiate additional connections to same or other servers for content embedded in the webpage.

We clustered these connections into *bursts* where a sequence of connections from a host with at most 5 seconds between connections is clustered together. This resulted in 1.52 million bursts. For each burst, we define the *length* of the burst as the time between the first and last connection in that burst. The *inter-burst interval* of two sequential bursts for the same host is defined as the time between the last connection of the first burst and the first connection of the second burst. Figure 2.10 shows the CDF of burst lengths and inter-burst intervals observed. 61.0% of the bursts have a length less than 1 second while 95% of bursts have length less than 14 seconds. The median inter-burst interval, on the other hand, is 26 seconds. Together this suggests that if a port-prediction attempt is made, it is more likely to fall in the gap of inactivity between two bursts and therefore succeed.

To quantify the how often port-prediction attempts will be interrupted by rogue connection attempts, multiple active hosts behind the same NAT need to be modeled. The type of the NAT is irrelevant as we only need to determine the likelihood of seeing a SYN packet in the window of vulnerability. This likelihood increases with the number of hosts in the internal network thereby decreasing the chances of a successful port-prediction. We simulate multiple internal hosts by picking a group of k hosts from our trace. Their traced connection requests are replayed while a simulated client application running on one of the hosts attempts port-prediction. The window of vulnerability is set to 3 seconds corresponding to 3 times the maximum RTT observed. We compute the probability of success by computing the fraction of time a port-prediction attempt can be initiated without it being interrupted by a rogue connection attempt in its window of vulnerability. This simulation is repeated for all groups of size k in our trace and the resulting CDF is shown in the semi-log plot in Fig-

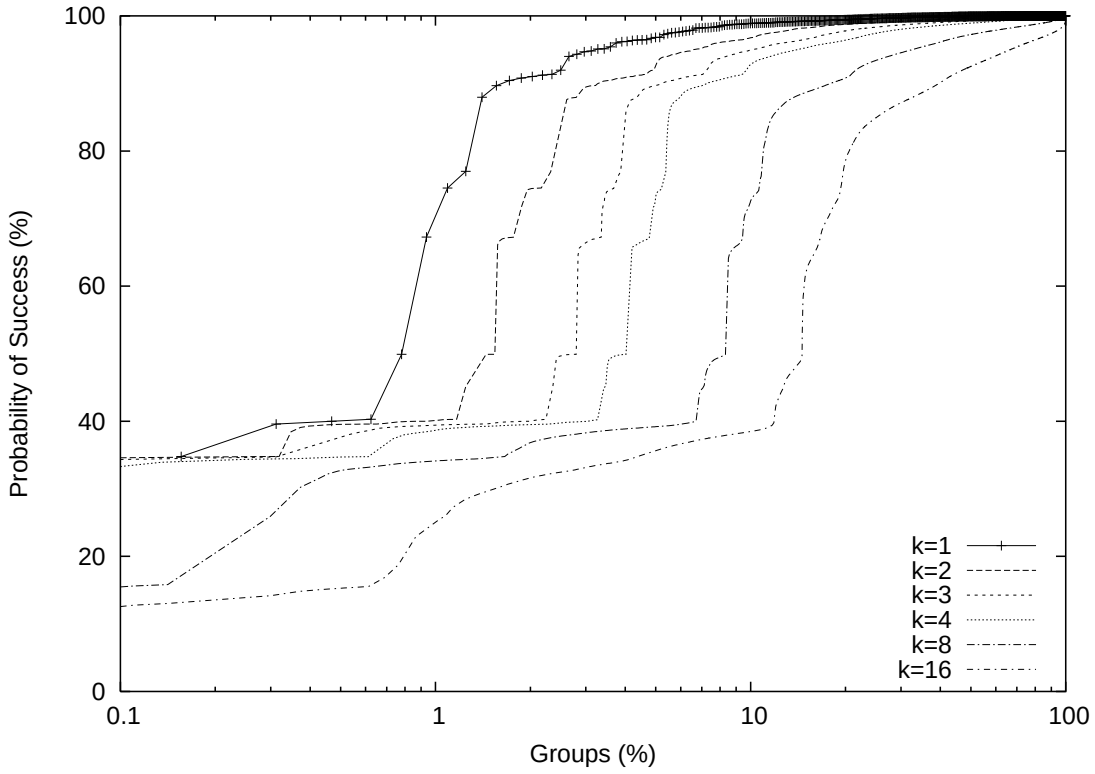


Figure 2.11: Estimated probability of uninterrupted port-predictions for various home network topologies

ure 2.11. The figure shows success rates for different k's where $k = 1$ represents one host behind a NAT while $k = 8$ represents eight hosts behind the same NAT. We find that in the one-host case, more than 95% of the port-predictions are successful for 96% of the networks simulated. In the median case, 99.92% of the predictions are correct. As the number of internal hosts increases, the chance of success drops. The mean probability of success drops from 98.9% in the one-host case to 97.7% in the two-host case to 96.7% and 95.3% for three and four internal hosts respectively.

We combine these results for different k with the distribution of internal hosts in US households. Table 2.4.1 lists the number of computers in US house-

Table 2.10: Number of computers owned in US households according to Mintel International Group Limited.

Hosts	Fraction of US households
One	61%
Two	27%
Three	8%
Four+	4%

holds based on a market survey conducted by Mintel International Group Limited [74]. While home networks with more than one host are extremely likely to use NATs, networks with one host may not. The same survey also finds that 82% of homes with high-speed Internet connections use wireless routers that by default act as NATs. We compute a weighted sum of the mean success rates of port-prediction attempts for $k = 1, 2, 3, 4$ weighed by the proportion of households using NATs with as many internal hosts. The simulation results from above already incorporate the fact that not all computers in a home are powered on all the time, therefore we do not compensate for that here. The weighted sum thus calculated is 98% , which represents our best estimate for the likelihood of an uninterrupted port-prediction attempt in the Internet today. This does not represent the probability of a successful TCP setup between peers, since additional NAT effects need to be taken into account, as discussed in Section 2.5.

2.4.2 Problems

Port prediction has several corner cases where it can fail. In Figure 2.12, if A uses STUNT server T to predict an address and port when trying to establish a connection to C, it would end up learning NAT M's external address instead of NAT

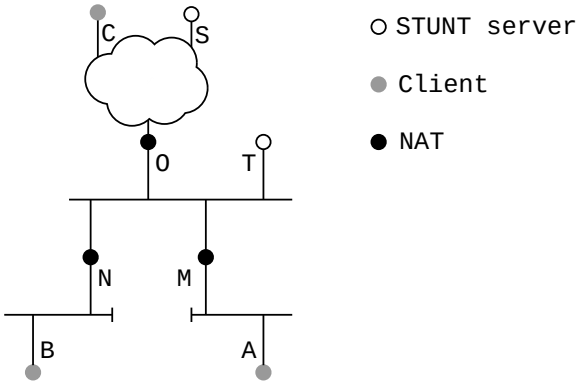


Figure 2.12: Problematic scenarios for port-prediction.

O's external address. Port prediction requires that the line of NATs between the client and STUNT server be the same as that between the client and the most external NAT of the endpoint it wishes to connect with. Therefore A somehow needs to discover S in order to connect to C. If, however, A wishes to communicate with B and both use STUNT server S then their port prediction attempts can interfere with each other, preventing either from correctly predicting the port. In addition, even if the port is predicted correctly, both A and B will end up using O's external address. This scenario is called a *hairpin translation* since A's SYN addressed to B's predicted address and port (O's external address in this case) will be delivered to O, which needs to send it back out on an *internal* interface. Not all NATs handle hairpin translations correctly and we estimate this erroneous behavior to be as high as 70.1% based on tests performed by the STUNT test client.

The port-prediction technique described earlier does not handle NB:Connection NATs, since it assigns sequential connections randomly. In [11] the authors propose an interesting technique for handling such cases that uses the birthday paradox to cut down on the number of guesses before a collision is found. The technique initiates 439 connections such that the guessed port will match one of

them with 95% probability. Unfortunately, we find that some NATs, like Netgear, limit the total number of pending connection attempts to 1000, causing this approach to quickly stifle the NAT. Fortunately, very few NATs demonstrate NB:Connection$_{\not R}$ behavior, mitigating the problem.

2.5 TCP Establishment

In this section, we estimate the success of the various NAT traversal approaches, as well as report our experience with peer-to-peer TCP establishment for a small wide-area testbed. The success of TCP NAT-traversal approaches depends on the behavior of all NATs between the two hosts, as well as the activity of other hosts behind the NATs. Section 2.3 analyzes a variety of NATs in isolation while Section 2.4 analyzes competing network activity and its effect on port-prediction. Combining the results from these sections we can quantitatively estimate the success of each NAT traversal approach.

We make the following assumptions about the deployment of the TCP-traversal approaches. We assume that STUNT servers are deployed widely enough to ensure that for each pair of hosts, there is a STUNT server that meets the port-prediction requirements and can spoof packets appearing to come from the mapped address and port of each host. We assume that host network stacks can be fixed so all software issues at the ends are resolved. Lastly, since we lack data to model the scenarios presented in Section 2.4.2, we assume the contribution from such scenarios to be negligible. As a result of these assumptions, our estimates may be optimistic.

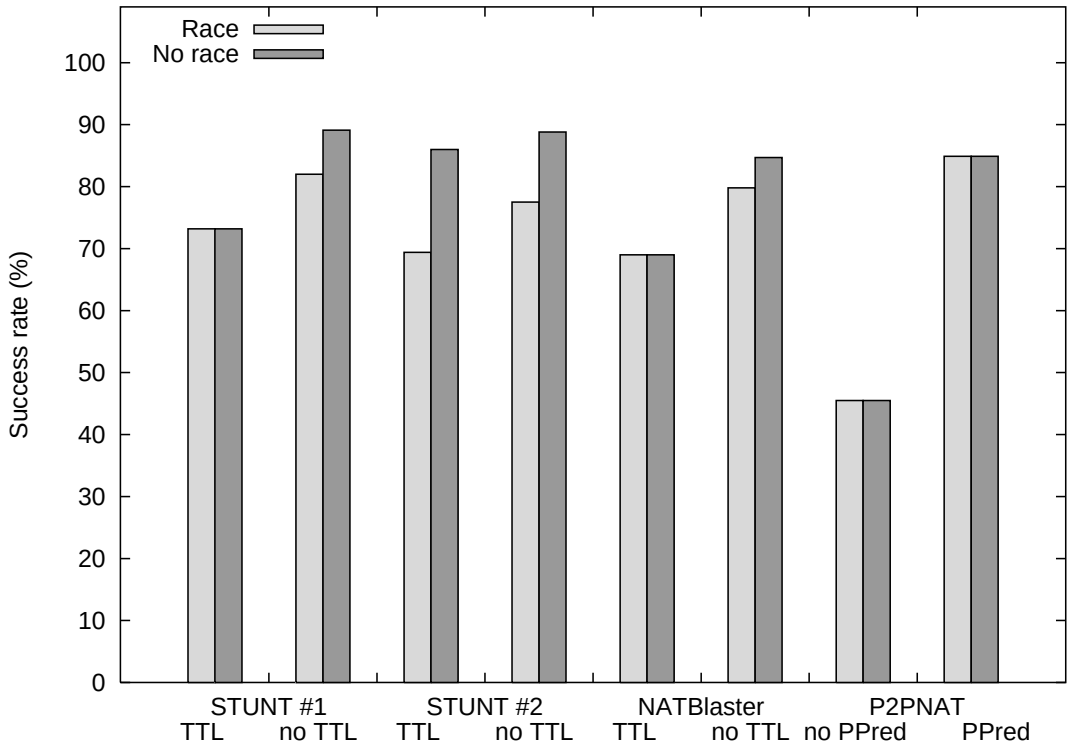


Figure 2.13: Estimated peer-to-peer TCP NAT traversal success rates of various approaches. With prevailing race conditions (Race) the success rate is lower than when races are resolved for the best outcome (No race).

Peer-to-peer TCP establishment depends on the NATs at both end. An endpoint with an unpredictable NAT may still be able to establish a connection if the other endpoint's NAT is predictable but not if it is unpredictable. We estimate TCP connectivity in the wild by considering all pairs of NAT behavior observed in practice. Figure 2.13 plots the estimated success rate of various TCP NAT traversal approaches. We plot the two STUNT approaches (#1 and #2), and NATBlaster and P2PNAT as proposed in [48, 11, 28]. In addition, we plot modified versions of STUNT #1 and #2 and NATBlaster approaches that do not use low TTLs. We also plot a modified version of the P2PNAT approach that

uses port-prediction. There is a race condition between the SYN packets in some of these approaches that leads to spurious packets for certain NAT-pairs. The light-gray bars represent the success rate when each end has an equal chance of winning the race; this corresponds to simultaneous invocation of the approach on both ends. The dark-gray bar represents the success rate when the race is broken in favor of a successful connection; this corresponds to two separate invocations of the approach where for the first invocation, one end is initiated slightly before the other while for the second invocation, the order is reversed. The attempt is declared successful if either invocation succeeds in establishing a TCP connection.

As shown in the graph, the original approaches proposed succeed between 45.5% to 73.2% of the time for P2PNAT and STUNT #1 respectively. Breaking the race condition in the original STUNT #2 approach by trying it once from each end boosts its success to 86.0%. Similarly, adding port-prediction to the P2PNAT approach allows it to handle symmetric NATs, increasing its success rate to 84.9%. Surprisingly, modifying the original approaches to not use low TTLs benefits all of them by ~5%! Breaking the race conditions thus introduced yields the best success rates of 89.1% and 88.8% for the two modified STUNT approaches and 84.7% for the modified NATBlaster approach.

The unexpected benefits to *not* using low-TTL SYNs are explained as follows. A large fraction of NATs silently drop the first SYN packet (Section 2.3.2) and only a small fraction of NATs filter inbound SYN packets after the outbound SYN packet (Table 2.3.2). Consequently in a large number of cases, the modified approaches end up triggering TCP simultaneous open even though they do not intend to. The small penalty they pay for NATs that generate a TCP

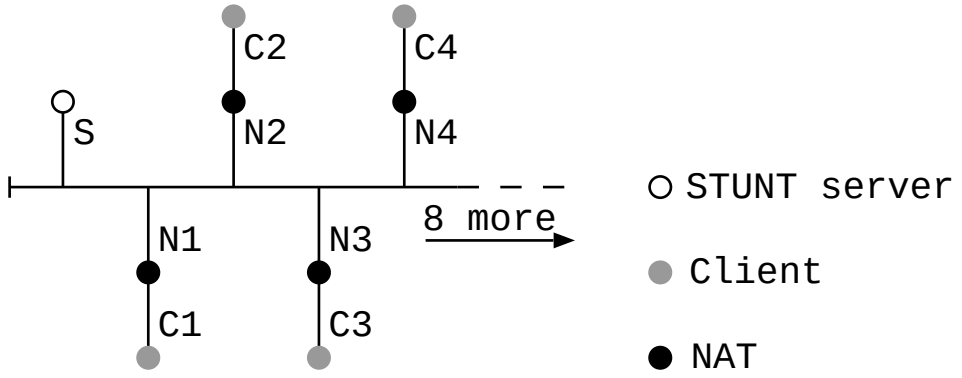
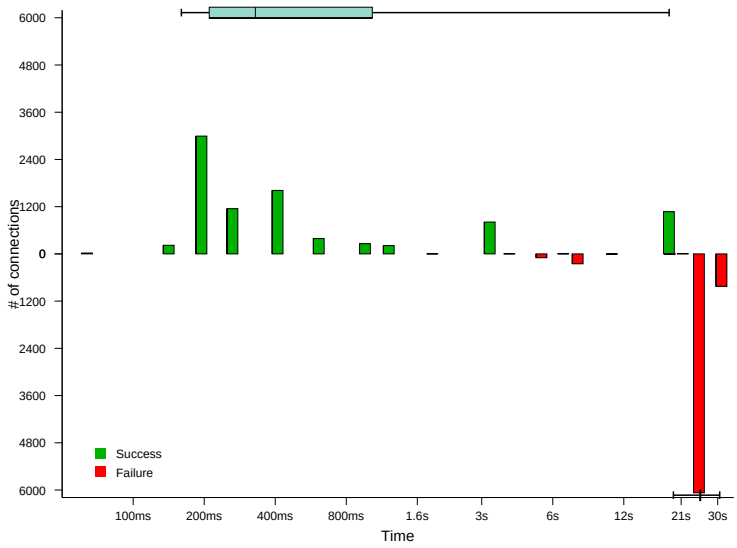


Figure 2.14: Network of 12 clients for peer-to-peer TCP NAT-traversal tests.

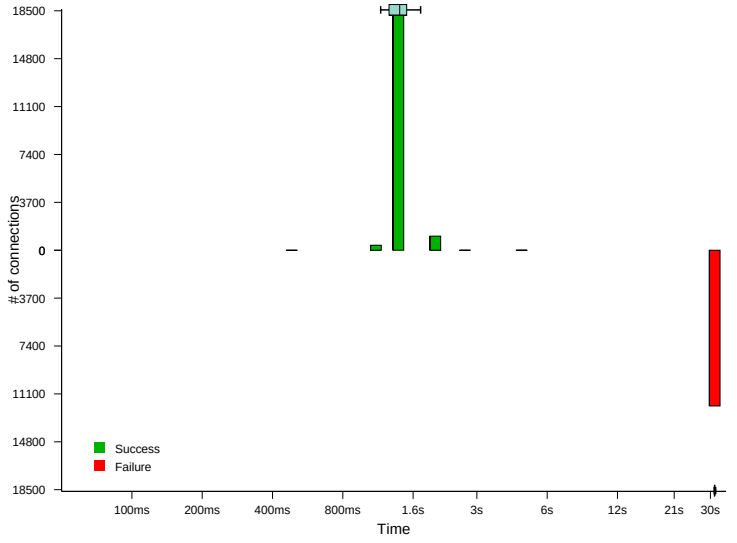
RST response is more than compensated for by the successful TCP simultaneous opens. This advantage is eroded if more NATs respond with TCP RST packets or if the endhost's operating system does not support TCP simultaneous open.

2.5.1 Implementation

We implemented the above approaches in a peer-to-peer program written in C. The program was run on 12 windows clients connected in a LAN as shown in Figure 2.14 as well as a slightly larger set of 20 clients connected over a WAN. Each client randomly picks another client and attempts to establish TCP with it. While all the approaches work as advertised, we limit this discussion to STUNT #2 without low-TTL and P2PNAT with port-prediction. This is because these two approaches perform as well as STUNT #1 and NATBlaster, and are additionally easier to deploy globally; in contrast, the STUNT #1 approach requires a broad deployment of servers that can spoof arbitrary packets while the NATBlaster approach requires RAW socket functionality that has been removed following security concerns in Windows XP.



(a) P2PNAT



(b) STUNT #2

Figure 2.15: Semi-log plot of time taken to successfully establish a connection or report a failure.

Figure 2.15 shows a semi-log plot of the time taken by each of the approaches to establish a connection or to report a failure in a low-latency network. The time distribution of successful connections (plotted above the x-axis) varies widely for the P2PNAT approach, while that of the second STUNT approach is very

consistent. This is because the P2PNAT approach repeatedly initiates a connection until one of them succeeds or a timeout occurs, while the second STUNT approach only initiates one connection. From the graph in Figure 2.15(a), a fair number of connections do not succeed until 21 seconds into the connection attempt, thus requiring a large timeout to achieve the estimated success rate determined earlier. In several cases, a dangerous side-effect of such a large timeout is observed when port-prediction fails and a peer's NAT responds with TCP RST packets. This causes the P2PNAT approach to generate a SYN-flood as the endhost repeatedly retries the connection until the timeout expires. Admittedly, this problem does not exist in the original P2PNAT approach, since it does not advocate port-prediction.

Overall, we find that the second STUNT approach is the most robust approach in establishing peer-to-peer TCP connections in the Internet today. It succeeds 88% of the time and does not require spoofing, RAW sockets, superuser privileges or TCP simultaneous open support. We also find that it is the simplest to implement out of all the approaches and works on both Linux and Windows. We encourage application developers to adapt this approach to provide TCP NAT-traversal in their peer-to-peer applications that currently cannot establish TCP between NATed peers.

2.6 Related Work

NAT traversal is an idea that has long existed since it was first proposed for UDP by Dan Kegel, and used for P2P gaming, in the late 90's. His basic approach was standardized and published in [100]. The UDP approach has resulted in several

working documents and Internet drafts that classify UDP behavior of NATs [56] and propose standardization of the same [7]. The area of TCP NAT-traversal without explicit control of the NAT, however, is fairly new. Several approaches have been analyzed in this chapter [48, 24, 11, 28] and have been demonstrated to traverse NATs in the current Internet. In [28], the authors present a similar study where the authors test a small subset of UDP and TCP NAT characteristics for a number of NATs. We present the first broad study of NAT behavior and peer-to-peer TCP establishment for a comprehensive set of TCP NAT traversal techniques over a wide range of commercial NAT products.

Protocols such as UPnP [73] and MIDCOM [111] allow endhost applications to explicitly control the NAT in order to facilitate peer-to-peer connections. The downside of this type of approach is that they require that these protocols exist and are enabled in the NAT. An application developer cannot depend on either of these, and so for the time being they are not an attractive option.

Another endhost approaches to TCP connectivity includes TURN [97], where TCP data is proxied by a third party. This third party represents a potential network bottleneck. Teredo [54] allows IPv6 packets to traverse IPv4 NATs by tunneling IPv6 over UDP. Here, TCP runs natively in the sense that it is layered directly above IPv6. Teredo has been implemented in the Windows OS, but to our knowledge is not widely used.

2.7 Conclusion and Future Work

This chapter presents the first measurement study of a comprehensive set of NAT characteristics as they pertain to TCP. While this study shows that there

are a significant number of problems with TCP traversal of current NATs, it nevertheless gives us much to be optimistic about. Even with existing NATs, which have not been designed with TCP NAT traversal in mind, we are able to show a 88% average success rate for TCP connection establishment with NATs in the wild, and a 100% success rate for pairs of certain common types of NAT boxes. These numbers are especially encouraging given that only a couple years ago, it was widely assumed that TCP NAT traversal was simply not possible. While a failure rate of 11% is not acceptable for many applications, the users of those applications at least have the option of buying NAT boxes that allow TCP traversal (e.g. Linksys). Additionally, NAT vendors have the option of designing their NAT boxes to be more TCP friendly; to this end, RFC 5283 [45] lays out guidelines for NAT vendors.

This study is limited in several respects. First, we did not test all existing NAT boxes. For the most part our study was limited to NAT boxes available in North America. Second, our field test is too small and biased to be able to accurately predict success rates broadly. Using market data helps, but even this data was limited in scope and in detail. Third, we tested only for home NATs. Port prediction in enterprise networks for "delta" type NATs will succeed less often, and it would be useful to measure this. Fourth, although TCP NAT traversal techniques apply broadly to firewalls, we did not test firewalls outside the context of NAT. Nor did we test IPv4-IPv6 translation gateways. Finally, like most measurement studies, this is a snapshot in time. Indeed, during the course of this project, new versions of the same NAT product exhibited different behavior from previous versions. Moving forwards, we hope that our TCP NAT traversal test suite can continue to be used to broaden our knowledge of NAT and firewall characteristics as well as to track trends in NAT products and their deployment.

CHAPTER 3

SHUTUP: REDUCING UNWANTED TRAFFIC

An ongoing problem in the Internet is that of large scale DoS attacks launched from botnets, worms, and port scanning. A wide variety of systems have been deployed or proposed that address one or more of these types of unwanted traffic, including [137, 136, 2, 61, 6, 69] that require in-network infrastructure help to filter DoS traffic, and [132, 131, 105] for worm detection and mitigation. By requiring infrastructure deep in the network, these approaches are forced to forfeit the benefits of an E2E deployment model [103] that has enabled the edge of the Internet to evolve much faster than the middle. It is no surprise, perhaps, that DoS attacks remain commonplace, and the Internet remains vulnerable to flash worms.

More recently, the authors of [4], propose a new approach of filtering DoS traffic at the source endhost; however, to do so securely, the authors rely on a clean-slate redesign of the network layer that adds cryptographic accountability to each packet. While we like this idea of filtering at the offending endhost, requiring a clean-slate Internet Protocol is far too heavy-weight. We believe DoS attacks can be mitigated using E2E techniques with far more light-weight mechanisms.

In this chapter, we explore a new point in the solution design space. In particular, we propose a pure end-to-end *ShutUp Service* that does not require any changes to the protocol stack, nor require any in-network infrastructure. In essence, a host that receives packets can tell the sending host to rate-limit or stop sending packets to it. This is enforced at the sending host by a tamper-

resistant enforcement agent, for instance implemented in the NIC device or in a protected virtual machine.

This ShutUp service is used by the receiver of packets (the *recipient*) to manage network and resource DoS attacks. The ShutUp is used as a capability-based rate-control system. After an initial short grace period during which the sender (*initiator*) can send at full speed, the enforcement agent at the initiator imposes a low rate unless the recipient explicitly allows a higher rate. The recipient may request a smaller or zero rate at any time, for instance in response to increased volume.

If the agent receives a significant number of ShutUp signals, it responds by slowing the rate at which connections to new hosts can be initiated. This slows down scanning worms, and port scanning (collectively called *scanning attacks* in this paper). In the case of port scanning, the slow down reduces the severity of the problem. In the case of scanning worms, the slow down buys precious time for other mechanisms (e.g. human response) to kick in.

The ShutUp service is implemented through the simple request-challenge handshake shown in Figure 3.1. The recipient sends a ShutUp message in response to an unwanted flow F. The enforcement agent at the initiator validates that indeed the recipient sent the ShutUp by sending a challenge message with a nonce to the recipient that the recipient must acknowledge. While the design of the ShutUp service requires no infrastructure support, our design intentionally allows it to operate between any pair of systems on the physical path of packets between initiator and recipient. This maximizes the deployment options, allowing intervention by the middle where appropriate. The main rationales for this E2E approach are as follows.

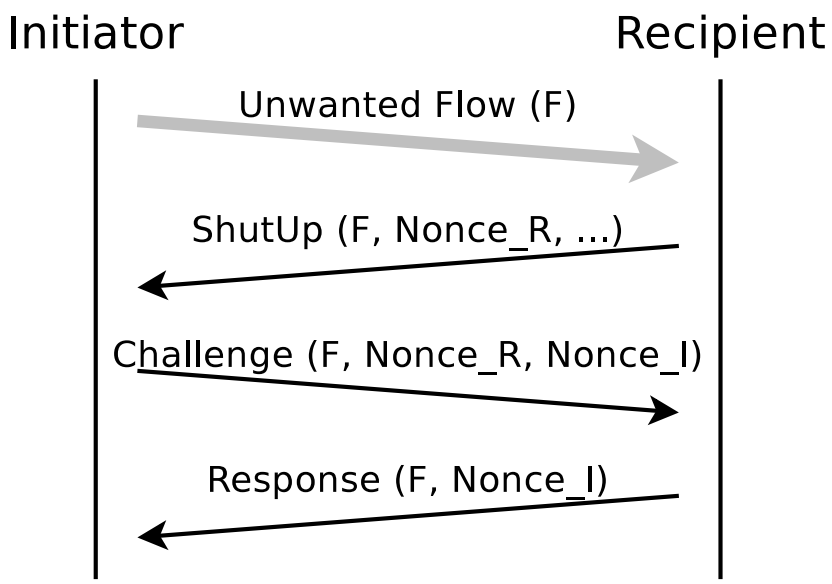


Figure 3.1: Basic ShutUp operation: Recipient NM sends SHUTUP for unwanted data. Initiator SM validates ShutUp was sent by purported recipient before blocking the flow at the source. Nonces protect against spoofing.

- By putting enforcement at the misbehaving host, we maximize the scenarios whereby the ShutUp service can be effective. For instance, hosts within an enterprise or datacenter would be protected from each other.
- By allowing recipient endhosts to issue ShutUp requests, we maximize the potential of integrating with external mechanisms to detect unwanted packets, such as those based on application-specific knowledge (e.g. [92]). At the same time, by allowing firewalls or even routers on the data path to issue ShutUp requests, we can exploit the broader knowledge a firewall has due to its vantage point.
- The ShutUp service requires no new changes to existing protocols or to infrastructure equipment. While we do not want to pretend that globally deploying ShutUp (or any other defense system) is easy, the deployment model for ShutUp is intriguing because it could be accomplished

with buy-in from a relatively small number of organizations in one or a few industry segments: OS, firewall, or antivirus vendors. Neither router vendors nor ISPs need be involved.

The deployment model for the ShutUp service is two-pronged. The first deployment model is purely E2E: the ShutUp service is enabled in hosts by default when they are sold by the vendor or when software is upgraded. This model leverages well-intentioned users [106] and lay users that do not wish to participate in DDoS attacks but who are technically incapable of securing their host from becoming part of a botnet. ShutUp could additionally be enabled by default in off-the-shelf customer NAT devices. The key advantage of this E2E deployment model is the small number of vendors buy-ins needed to deploy ShutUp. The top OS vendor, virtualization vendor, anti-virus vendor, or top-two NAT vendors, can each independently reach over 50% of hosts [39, 37, 120].

Enterprises, of course, can enable ShutUp internally without any vendor buy-in. By doing so, enterprises can protect themselves from internal flash worm breakouts that can otherwise cause millions of dollars worth of damage [77].

The second deployment model is for ISPs to have ShutUp enforced at the edges of their network. Doing so expedites deployment in networks where the upgrade cycle of hosts is too slow for the E2E deployment model to gain momentum.

Globally, however, we can expect to see some (hopefully small) percentage of hosts without ShutUp enforcement. These would be hosts that have either physically disabled or opted-out from the ShutUp service, and are on ISP net-

works without ShutUp enforcement. Opting out certainly allows individuals to launch attacks from their own hosts, but they are unable to enlist the help of botnets, thus mitigating the attack by several orders of magnitude.

Although the ShutUp service borrows from recent work in DoS defense systems that exploit endhost support, it is unique in several ways. Argyraki and Cheriton [6] propose the use of a handshake similar to ShutUp, but require enforcement devices in the network near the attacker, and a control device on the attacker-side of the bottleneck resource. By contrast, ShutUp is a pure E2E mechanism. In particular, ShutUp avoids the need for a control device outside of the bottleneck resource through a capability mechanism. To our knowledge, Shaw is the first to propose putting enforcement at the endhost [106]. The approach has some limitations, in particular with respect to source address spoofing and the scoping of ShutUp requests, and in any event only outlines the approach and does not experiment. It is fair to characterize ShutUp as a deeper exploration of the same vision. AIP [4] is a new network layer architecture with two-level self-certifying addresses. Among the many uses of this architecture, it can defend against DoS through an E2E handshake with enforcement at the endhost (putting the mechanism in a tamper-proof NIC card). Notably AIP still requires infrastructure support, to detect source address spoofing by attackers. ShutUp works with legacy network layers, does not require cryptographic mechanisms, and prevents source address spoofing at the tamper-resistant driver in the attacking endhost.

This chapter makes the following three contributions. *First, we present the first design and implementation of a pure E2E ShutUp service.* The ShutUp protocol itself is quite simple, allowing us to confidently reason about its security prop-

erties and the correctness of the implementation. The enforcement module, for instance, is implemented in less than 200 lines of Python code. This small footprint also maximizes the deployment options. For instance, the ShutUp protocol may be implemented in a NIC or on small wireless devices. It requires no encryption or key distribution, allowing us to avoid the associated complexity and hazards. *Second, we analyze the ShutUp service's effectiveness as a DoS prevention mechanism.* We show, for instance, that a host with 10Mbps of access bandwidth can completely stop a 10Gbps attack from ~10,000 ShutUp-enabled hosts behind broadband links. *Finally, we analyze the ShutUp service's effectiveness as a defense against scanning attacks.* Of particular interest here is the trade-off between effectiveness (how quickly scanning attacks are discovered and how much they are slowed down) and false positives (identifying legitimate activity as scanning attacks). For example, we show that ShutUp slows down scanning worms to the same degree as existing approaches while reducing false positives by two orders of magnitude.

3.1 ShutUp Details

This section starts with an overview of the ShutUp components, followed by a detailed description of its operation.

3.1.1 ShutUp Components

There are two components in ShutUp, *ShutUp modules* (SM) and *notification modules* (SM). As illustrated in Figure 3.2, an SM is deployed at the initiator, out

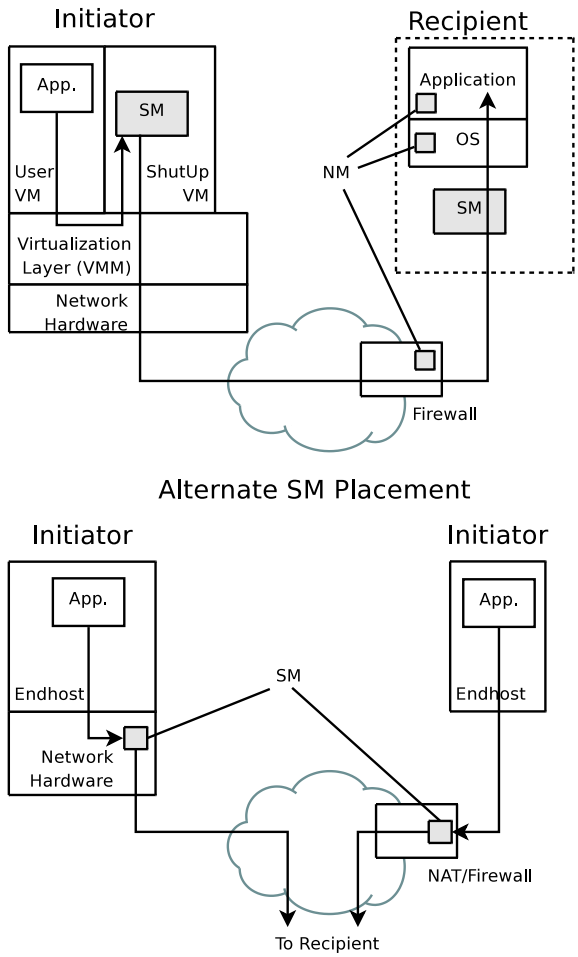


Figure 3.2: **ShutUp Module (SM)** Option 1: The SM runs in a separate tamperproof VM. Network traffic to and from the user VM is routed through the SM by the VMM. Option 2: The SM runs in the Network Interface Card. Option 3: The SM runs in the user's NAT/firewall. The **Notification Module (NM)** runs at multiple layers: in the recipient application, endhost OS and recipient firewall. The initiator and recipient endhost setup is symmetric (abbreviated for clarity).

of reach of untrusted components, from where the SM can exert direct control over inbound and outbound traffic. NMs are deployed at the recipient at one or more levels (application, endhost OS, etc.) from where they can analyze inbound traffic at multiple layers. The recipient NMs are responsible for identifying unwanted flows and sending ShutUp messages, and the initiator SM is

Table 3.1: ShutUp primitives and message contents

ShutUp Primitive
SHUTUP(FLOW, APPNAME, NONCE$_R$, TTL, ID$_{NM}$)
THROTTLE(FLOW, APPNAME, NONCE$_R$, TTL, RATE, ID$_{NM}$)
Recipient NM directs initiator SM to ShutUp or throttle a flow (5-tuple; wildcards allowed).
CHALLENGE(FLOW, NONCE$_R$, NONCE$_I$, ID$_{NM}$)
Initiator SM requests confirmation of request
RESPONSE(FLOW, NONCE$_I$)
DISCLAIM(FLOW, NONCE$_I$)
Recipient NM confirms or denies original request
Query Primitive
QUERYAPPNAME(FLOW, NONCE$_I$)
APPNAME(FLOW, APPNAME, NONCE$_I$)
SM queries remote NM for application name (to block scanning attacks before probe is sent)

responsible for enforcing the recipient's decision and for correct operation of the ShutUp protocol. Since an endhost can both initiate and receive flows, NMs and an SM are deployed in each endhost. The SM is the only trusted component in the system; the NM is not a trusted component since it cannot affect flows to other recipients.

3.1.2 Basic Operation

ShutUp offers two primitives (Table 3.1.2). The primary primitive, SHUTUP, is used to block or rate-limit individual flows. The basic ShutUp operation illustrated in Figure 3.1 is fairly straightforward. The recipient NM sends a SHUTUP request (unencrypted) in response to unwanted traffic. The request includes a nonce (NONCE$_R$) for verification purposes. To save state, the nonce may be computed as a cryptographic hash of the flow identifier and a local time-varying

Table 3.2: State maintained by SM

Per L2 Neighbor		
IPs	:	Source IP addresses validated
Per Destination IP		
acknowledged	:	Sent throttle before initial timeout
Per Active Flow		
id	:	5-tuple plus source port offset
rate	:	Rate limit set for flow (possibly 0)
TTL	:	Time rate is in effect
$4 \times$ *timestamp*	:	Time ShutUps and data for the flow were last sent and received
Per Application		
#tokens	:	Number of ShutUps allowed
rate	:	Current replenishment rate
shutups	:	Recipients that have sent ShutUps
whitelist	:	Recipients that never sent ShutUps

secret key. The initiator SM challenges the purported recipient with a second nonce (NONCE_I) and includes the first nonce in the challenge. If NONCE_R is validated, the recipient completes the challenge by returning NONCE_I. Otherwise, the recipient signals an error. The first nonce protects against a spoofed SHUTUP message and replays, while the second nonce protects against a spoofed response, thus preventing attackers not on the physical path between the initiator and recipient from abusing the protocol. Once the ShutUp is validated, the SM blocks (or rate-limits) the unwanted traffic.

3.1.3 SM Operation

The SM: 1) prevents spoofed source address, 2) unilaterally rate-limits flows after an initial grace period unless the NM explicitly allows a higher rate (effectively acting as an E2E capability), 3) enforces compliance with ShutUp re-

Algorithm 3.1: **at SM** ONRECVFROMAPP(P)

```
1: if ISUNVALIDATED(P.IP_src, P.MAC_dst) then
2:     rate-limit                                   ▷ May be spoofed
3: end if
4: if FOREXISTINGFLOW(P) then
5:     F ← GETFLOWSTATE(P)
6:     ADDSRCPORTOFFSET(P, F)                       ▷ Obscure 5-tuple
7:     if ISSHUTUP(P) then                          ▷ App sending ShutUp/Throttle
8:         if RCVDDATASINCESENTSHUTUP(F) then
9:             forward                              ▷ To net
10:        else
11:            drop                                 ▷ Redundant ShutUp
12:        end if
13:    else if FLOWRATELIMITED(F) then
14:        rate-limit                               ▷ Set by recipient
15:    else if FLOWBLOCKED(F) then
16:        drop
17:    else if INITIALTHROTTLETIMEOUT(F.IP_dst) then
18:        rate-limit                               ▷ Recipient under DoS?
19:    else
20:        forward
21:    end if
22: else                                            ▷ New flow
23:    if ISSHUTUP(P) then
24:        drop                                     ▷ No flow to ShutUp
25:    else if not NEWFLOWALLOWED(P) then
26:        drop                                     ▷ For scanning attacks
27:    else
28:        F ← NEWOUTBOUNDFLOW(P)
29:        ADDSRCPORTOFFSET(P, F)
30:        forward
31:    end if
32: end if
33: if not WASDROPPED(P) then
34:    UPDATESENTTIMESTAMPS(F, P)
35: end if
```

Algorithm 3.2: **at SM** OnRecvFromNet(P)

```
1: SetValidated(P.IP_dst, P.MAC_src)                          ▷ Learn IP
2: if ForExistingFlow(P) then
3:     F ← GetFlowState(P)
4:     if IsShutUp(P) then                                    ▷ Got ShutUp/Throttle
5:         if SentDataSinceRcvdShutUp(F) then
6:             SendChallenge(P)
7:         else
8:             drop                                           ▷ Redundant ShutUp
9:         end if
10:    else if IsValidChallengeResponse(P) then
11:        if IsForThrottle(P) then
12:            RateLimitFlow(F, P.Rate, P.TTL)
13:        else
14:            BlockFlow(F, P.TTL)
15:            CheckScanning(P.IP_src, P.App)
16:        end if
17:    else
18:        SubDstPortOffset(P, F)                             ▷ Unobscure 5-tuple
19:        deliver                                            ▷ To app
20:    end if
21: else                                                      ▷ New flow
22:    if IsShutUp(P) then
23:        drop                                               ▷ No flow to ShutUp
24:    else
25:        F ← NewInboundFlow(P)
26:        deliver
27:    end if
28: end if
29: if not WasDropped(P) then
30:    UpdateRcvdTimestamps(F, P)
31: end if
```

quests, and 4) detects and slows scanning attacks. The state maintained at the SM is listed in Table 3.1.3. The SM operations when forwarding a packet from the application to the network and vice versa are listed in pseudo-code in Algorithms 3.1 and 3.2 respectively, with key fragments highlighted in the description below.

Preventing Address Spoofing

A host that can spoof source addresses can launch an attack without allowing ShutUp requests to reach it. To prevent this, the SM must prevent source spoofing. The difficulty in doing this lies in determining what an acceptable source address is. A firewall can be configured with this information. An endhost SM must however determine whether the endhost is authorized to use the present address. It is hard for the SM to make this determination, especially if the endhost statically assigns the address. However, we consider cryptographic solutions [3] unnecessarily complex in this context.

The SM rate-limits sending "unvalidated" packets to each layer-two neighbor, where validation requires receiving a packet for the purported IP address from that L2 neighbor (Algorithm 3.1.1 and 3.2.1). Since the SM does not know which addresses are spoofed and which not, rate-limiting, rather than blocking, is necessary to give an unvalidated address a chance to be validated. If the address is in fact spoofed, the validation does not succeed as long as the spoofing host is not on the same subnet as the recipient, because responses are not routed back to the SM, and the rate-limit is maintained indefinitely. Since virtually all applications send packets in both directions [65], validation is effectively piggy-

backed on application traffic. The approach does not require any infrastructure or changes to the protocol stacks.

We choose to validate the IP address and destination MAC pair rather than just the IP address. Doing so foils two colluding endhosts on the same network attempting to validate a spoofed address. For instance, a colluding L2 neighbor may trigger validation by sending a packet to the endhost's MAC with the destination address set to the spoofed address — the resulting validation whitelists the spoofed address, but for use with that neighbor only. Therefore for the security of the mechanism, the SM must prevent spoofing the source MAC address. But since the SM has exclusive access to the physical network hardware, this is not difficult. (Section 3.2 discusses the case where the colluding host does not have an SM.)

Delivering ShutUp Messages

ShutUp messages contain the 5-tuple (protocol number, and source and destination address and port) of the data flow they refer to. The messages are sent along the datapath encoded as ICMP destination unreachable packets with the 5-tuple contained in the encapsulated payload headers. The reason for this is that NATs and firewalls today already translate and forward ordinary ICMP messages related a flow [110], and so will transparently forward ShutUp messages. Another option would be to use a shim layer between IP and the transport, which we avoid since it modifies the protocol stack, thus breaking existing middleboxes and application firewalls.

Flow Initiation and ShutUps

A general problem with using multiway handshakes in DoS prevention mechanisms is that the validation message is likely to be dropped at the bottleneck link. If the attack is large enough, very few validation messages will get through, and it will take a long time to slow the attack. To deal with this, the SM enforces a rate-limit on flows that must be explicitly lifted by the recipient NM. The rate-limiting operates as follows.

The SM initially allows the application to send at an unlimited rate on new flows, but only for a short time interval. The interval is picked conservatively (10 seconds in our simulations) within which time the recipient must send a THROTTLE message indicating the allowed rate-limit for the flow. If neither a THROTTLE nor a SHUTUP is received, the SM automatically rate-limits the flow (Alg. 3.1.18). When the throttle's TTL expires, the SM again rate-limits the flow (to 10kbps in our simulations). The NM must periodically send throttles to maintain the flow at high speed.

In addition to rate-limiting flow packets, the SM limits both inbound and outbound ShutUp requests. For outbound ShutUp requests sent by the application, the SM ensures that the associated flow exists, and has received data since the last ShutUp was sent (Alg. 3.1.11). This gates ShutUp messages with application traffic, preventing endhosts from abusing ShutUp messages to launch certain attacks. Similarly, the SM ignores inbound ShutUp requests for flows that do not exist and flows on which no data has been sent since the last ShutUp was received. These safeguards require the SM to maintain per-flow state and timestamps.

Legacy recipients: The automatic rate-limiting above does not affect legitimate flows to NM-enabled recipients, but creates tradeoffs for legacy recipients that lack a NM. On the one hand, the rate-limit mitigates DoS attacks on legacy recipients incapable of sending ShutUps. While on the other hand, legitimate flows to legacy recipients that last more than a few seconds are unnecessarily slowed. To avoid this latter issue, the SM disables the rate-limit if it receives *any* packet for the flow from the receiver. But to prevent an attacker from abusing this mechanism, the SM makes it hard for the attacker to spoof a flow packet, for instance by adding a random offset to the source port in the 5-tuple[1] (Alg. 3.1.29); an attacker may attempt to brute force the 5-tuple, but the SM can detect such an attempt. The security implications of this mechanism are considered in Section 3.2.

Slowing Scanning Attacks

ShutUp usually slows the scanning application rather than cordoning off the entire infected endhost, in order to reduce collateral damage to other applications. Identifying the application from just the flow 5-tuple is notoriously hard. Firewalls often resort to complex deep-packet inspection to identify application flows to dynamically selected ports. ShutUp avoids this complexity by having the NM provide the application *name* in the ShutUp message. Doing so is not hard for the NM, since it is deployed in the recipient application or endhost OS and can readily query the application. Since the NM is not trusted in any event, the name reported by the application is not verified.

[1]Like NAT except only the local port for locally initiated flows is modified. Since inbound packets can be unambiguously delivered (Alg. 3.2.18), NAT traversal approaches are not needed.

At its simplest, the SM considers an anomalous rate (per unit time) of ShutUps an indication of a scanning attack, at which point new flow initiations for the application are rate-limited (lines 3.2.15 and 3.1.25). In principle, a number of other options can be used for detecting scanning attacks, for instance based on the fraction of flows that receive ShutUps [131, 105], or more complex introspection of the user application [36]. We base our choice on the simplicity of the approach and the low number of false positives we encountered, as we report in later sections.

The SM algorithm for slowing scanning attacks is as follows. The SM maintains a token bucket for each application. A token is consumed by each ShutUp from a distinct recipient. Flow initiations to recipients are blocked if no tokens are present; however, flows to a dynamic list of previously successfully contacted recipients are not affected. Over time, tokens are replenished at a configured rate up to a maximum value, but persistent scanning decreases this rate at which tokens are replenished. The reasoning behind these design decisions are as follows.

Distinct recipients: Consuming one token for multiple ShutUps from the same recipient limits the potential damage caused by a (malicious) recipient. For this purpose, the SM maintains per-application state consisting of recipients that have previously sent ShutUps.

Blocking by application: While the application name is piggybacked on ShutUp requests as mentioned above, at flow initiation time the packet 5-tuple does not, in general, identify the application. Since the first flow packet can itself exploit a vulnerability [77], this creates a chicken-and-egg scenario where the first flow packet must be blocked if it is for the offending application, but

to determine the application, the flow must be allowed and a ShutUp sought. ShutUp breaks this dependency with the Query primitive (Table 3.1.2), which is a simple request-response exchange whereby the SM queries the NM for the name of the application the flow would reach if it were to be allowed. The query is not needed for well-known ports and is only invoked when the host is believed to be participating in an attack. Legacy recipients without an NM cannot respond to these queries.

New Recipients: To mitigate the impact of false positives, the SM maintains a list of recipients contacted previously that have never sent a ShutUp for the application. While flow initiations to new recipients are rate-limited during a suspected scanning attack, flows to recipients on this list are whitelisted.

Rate limiting flow initiation: A common concern with thresholds is how the threshold is picked. A low static threshold creates false positives, which we determined in our dataset to be bursty, whereas a threshold high enough to accommodate bursts allows sub-threshold scans that does not adequately slow the attacker. For this purpose, ShutUp uses a dynamic threshold: the rate at which new flow initiations are allowed is picked randomly (our implementation uses an exponential distribution around a configured mean), and this rate is halved every time the token-bucket underflows. In effect, the rate-limit allows large but short bursts of ShutUps for legitimate applications, while forcing persistent scans to a much lower rate.

3.1.4 NM Operation

A single recipient NM cannot detect the wide range of unwanted traffic we are interested in (address scans, DoS, intrusions etc.). The design therefore accommodates separate collaborating NMs operating at various layers. Each NM tags ShutUp messages with its ID and responds to challenges intended for it. That said, *how* unwanted traffic is detected, for instance distinguishing between flash crowds and DoS attacks, is beyond the scope of this thesis. We rely on existing or future methods of detecting these attacks at the application, endhost OS, and recipient firewall, including IDS (e.g. [88]), CAPTCHAs [125], and exploit detectors (e.g. [20]) as applicable. Given these detection mechanisms, ShutUp answers the question of what to do once the attack is detected.

Out of the three, the design of the endhost OS NM is non-trivial. If the recipient application is not running, the endhost OS must determine whether or not to send a ShutUp. The choice is not clearcut because sending a ShutUp prevents the source from reattempting the flow, at least until the ShutUp expires. Such ShutUps hinder interactive applications such as web browsing if the web server is momentarily unavailable. Further, if enough endhosts generate false positives, it unduly triggers the scanning defense at the initiator. This affects P2P applications, for example, where stale membership information results in peers attempting to contact a recently running (or crashed) application for some time. To avoid these false positives, applications in ShutUp register an application-specific "linger" time when binding to a port (few minutes for P2P applications, infinity for server applications). The endhost OS does not send ShutUps for an unbound port until the linger period expires. For legacy applications, the OS NM may be configured with a small default linger time.

3.1.5 Protecting the SM

One of the challenges in implementing ShutUp is protecting the SM from being subverted by endhost malware, while at the same time not placing undue constraints on the resources available to it. There are a number of E2E deployment options, for instance, implementing the SM as part of the NIC firmware, in the NAT/firewall router, or as a virtualization layer. In the virtualization scenario, user software and the SM are run in separate virtual machines (VM), and only the ShutUp VM is allowed access to the physical network interfaces. Traffic to and from the user VM is routed through the ShutUp VM (Figure 3.2), where the SM can filter packets as necessary. Designing the SM for the NIC, NAT/firewall, or ISP router near the initiator is, of course, more straightforward.

There are a number of reasons why we believe a VM-based deployment is a good choice. First, desktop virtualization has come of age. VMs are used today to allow users to run games, or business software on otherwise unsupported operating systems [124, 86]. Second, a VM has access to more resources (multiple cores, memory) than a hardware implementation (e.g. NIC). Based on trace-driven simulations, we estimate a typical endhost will require around 256kB of SM state isolated from the user; tamperproof memory in hardware is expensive, whereas a VMM can virtualize main system memory. Third, a VM protected by a VMM provides a trusted platform [35] while being easy to update once deployed. Updating hardware securely requires physical access. In contrast, a VM can be securely upgraded by the VMM (assuming the VMM has not been compromised). Finally, a VM is needed to demultiplex ShutUps for middleboxes. With a hardware or NAT/firewall SM, a ShutUp for a middlebox (webproxy, virtual private network tunnel endpoint) resulting from traffic initi-

ated by a single user would affect the entire middlebox, thus affecting all users behind the middlebox. In a VM, a trusted version of the middlebox service can run alongside the SM to redirect ShutUps to the offending user.

3.1.6 Deployment

Enterprises can deploy NMs and SMs with or without buy-in from hardware and software vendors. The ability to protect enterprise hosts from one another, particularly during internal worm outbreaks, provides incentive to do so.

Buy-in from the major OS vendors would result in rapid endhost NM deployment, potentially through automated software update mechanisms. Deploying the SM securely, however, requires an environment isolated from malware already present in the endhost. Buy-in from the top five PC vendors would result in 49% deployment [38]; buy-in from the top five home router vendors would result in 79% [120]; and, buy-in from the top five anti-virus vendors would result in 91% global deployment [37]. It is therefore conceivable that a concerted effort by 10 or so companies could realize a near global ShutUp deployment in a few years time. At the same time, it isn't necessary for all concerned vendors to buy in. As we show later, even a partial deployment is beneficial.

3.2 Attacking ShutUp

In this section we consider attacks on ShutUp components and mechanisms through which an attacker may attempt to disrupt flow establishment.

Attacker Model: We assume that the attacker has complete physical control over a small number of hosts. In particular, the attacker can disable the SM on these hosts. In addition, the attacker has software control over a much larger number of compromised hosts.

We initially assume the attacker is not on the datapath between the initiator and recipient. An on-path attacker, for instance on a compromised router, can disrupt communication by dropping or modifying packets enroute even in the absence of ShutUp. We later relax this assumption to consider eavesdroppers — on-path attackers that can observe but not modify in-flight packets.

Software Compromise: Vulnerabilities in the SM implementation or virtualization layer could allow an attacker to gain control of the SM through software methods. In case of compromise, the VMM can restore a pristine SM from readonly media, and rely on automatic software update mechanisms to apply all relevant patches. Physical access is required only if the VMM itself is compromised or for updates to the readonly media.

Compromising the NM is less severe. A compromised NM may fail to ShutUp unwanted flows, or worse, ShutUp flows that are not unwanted. In the first case, NMs deployed at other layers can still ShutUp unwanted flows. In the second case, the danger is not so much to the recipient, since such an attacker can block flows passing through the NM even without cooperation from the SM, but rather to the initiator for being falsely implicated of unwanted traffic. However, the attacker must compromise multiple NMs at recipients contacted by the initiator to successfully trigger the scanning defense, mitigating the severity of the attack.

Spoofed Packets: In order to spoof packets at an unlimited rate without disabling the SM, the attacker must be able to complete the address validation process by spoofing the MAC address of the first-hop router, which requires physical access to a second endhost on the same network. With software-only access to a given network, an attacker can at best hijack an address in that network, but since the endhost would receive all packets for the hijacked address, in particular ShutUp messages as well, there is little impact on ShutUp for doing so. Consequently, the number of networks an attacker can spoof packets from is limited by the number of hosts the attacker has physical access to — significantly better than today, where up to 25% of edge ASs allow spoofing [10]. If an attacker compromises a legacy (no SM) host on the subnet, it can fake the MAC address of the router and then create arbitrary addresses on compromised SM-enabled hosts. If the attacker already has such a foothold in a legacy host, however, it can in any event spoof arbitrary addresses.

In order to successfully fake ShutUps, an attacker must be able to eavesdrop packets. This is because, as previously mentioned, all ShutUp messages must be validated before any action is taken by the SM, and the attacker must be able to guess the nonces involved to fake the validation. But since nonces are not encrypted, an eavesdropper can win a validation race. However, since the eavesdropper cannot stop the in-flight challenge, the real recipient will generate a conflicting response. The SM can therefore at least detect the presence of an eavesdropper, although not which of the responses is legitimate. In any event, an eavesdropper can disrupt flows even without ShutUp, so ShutUp does not introduce a qualitatively new attack.

Abusing ShutUp Messages: ShutUp messages are gated by application traffic at the SM. The mechanism prevents an attacker from flooding a victim with ShutUp messages absent an application flow between the two. Even when a flow exists, the number of ShutUp messages that can be sent is upper bound by the number of flow packets. The same mechanism gates reflection attacks consisting of ShutUp challenges if an attacker spoofs a stream of ShutUp messages; the amplification factor for such attacks is at most one.

Avoiding Initial Rate-Limit: An attacker may attempt to disable the initial rate-limit for unacknowledged flows by abusing the mechanism intended for legacy recipients. Doing so requires the attacker to guess the random initiator port, which requires on average 2^{15} guesses. After the first few guesses, the SM can detect the attack and lock in the rate-limit such that only a validated THROTTLE may lift it. Thus an attacker cannot disable the initial rate-limit, nor affect flows to NM-equipped recipients, and can at best target flows to legacy recipients to be rate-limited; the recipient can counter by deploying an NM.

Triggering Scanning Defense: An attacker can trigger the scanning defense at an endhost by convincing the initiator to contact recipients not expecting the flows. For instance, a malicious website may return a webpage with inline image URLs pointing to recipients not running a webserver. Alternatively, a compromised router enroute to multiple destinations may fake ShutUps. While ShutUp does not defend against such attacks, ShutUp limits the impact. First, in the case of a duped application, the scanning defense only applies to the one application and not to other applications running on the endhost. Second, the defense does not affect communication to recipients that have never previously ShutUp the endhost, allowing the application to operate with diminished

reachability. Finally, as the defense is lifted unless the application is persistently scanning, a legitimate application can resume normal operations after cautioning the user against reattempting flows to the problematic destinations.

3.3 Stopping DoS with ShutUp

The ShutUp service can be used to mitigate both application-level as well as network-level DoS attacks. At the application level, the efficacy depends on how quickly and accurately the attack is detected, which is contingent on detection mechanisms external to ShutUp; once detected, the application NM sends ShutUps to the attackers. Defending against network-level DoS, where attackers saturate a bottleneck link, is more involved, because the ShutUp challenge-response mechanism must operate through the same bottleneck.

If the bandwidth of challenges incident at the bottleneck link is B, and aggregate attacker bandwidth is X times the bandwidth of the bottleneck link, then only B/X challenges will cross the bottleneck. To put in numbers, if a small number of attackers, say 5000, command a bandwidth 10 times that of a 100Mbps bottleneck, an incident challenge bandwidth of 10Mbps would ShutUp around 1000 attackers per second assuming 128 byte challenge packets, stopping the attack in 5 seconds.

Relying solely on the challenges getting through the bottleneck suffices for small attacks but not large attacks. This is because the incident challenge bandwidth B required to maintain the same rate of validated ShutUps increases linearly with X. However, since challenges are driven directly by ShutUp requests, B is upper bound by the victim's upload bandwidth, which remains fixed as X

increases. Moreover, B is more realistically a fraction, typically one-tenth, of the bottleneck bandwidth assuming ~1kB attack packets. This is because the victim can, at best, generate one ShutUp per attack packet crossing the bottleneck, each of which results in a challenge. Generating more ShutUps or challenges creates the possibility of amplification attacks. Since B cannot be increased indefinitely, ShutUp decreases X for large DoS attacks.

In large DoS attacks (e.g. $X > 20$), the SM enforced automatic rate-limit after the first few seconds cuts X by several orders of magnitude. The rate-limit alone does not address congestion at the bottleneck, as even with the diminished bandwidth, the attackers can be numerous enough to saturate the bottleneck. The purpose of the rate-limit is to increase the fraction of challenges in the attack traffic, which is possible because B is independent of X as long as the bottleneck is saturated. This allows the challenge-response mechanism to kick in more effectively and stop large DoS attacks.

The above discussion assumes that only the NM at the victim is responding to the attack, however, other NMs may additionally be involved. For instance, ShutUp challenges to an ISP NM upstream of the bottleneck link would succeed. Or in case of an endhost behind the bottleneck colluding with attackers by not sending ShutUps for traffic crossing the bottleneck, a firewall NM between the bottleneck and the colluding endhost may instead send ShutUps to protect other endhosts behind the bottleneck. Overall ShutUp requires an NM to be present somewhere along the attack path to stop a DoS.

Simulation Results: We used *ns-2* to simulate how well ShutUp mitigates DoS attacks. We use a fan-in topology where a varying number of attackers (from 10 to 9600) send traffic to a bottleneck link; the recipient is on the other

side of the bottleneck. The topology models a DoS where attack traffic does not self-interfere except at the bottleneck. In the Internet, attackers sharing a common link other than the bottleneck may cause additional packet loss. Nevertheless, since ShutUp is entirely end-to-end (or edge-to-edge), we expect such AS- and router-level topology to have little impact. To convince ourselves of this, we simulated ShutUp on the router-level topology collected by the Rocketfuel project [108] and a sampling of the AS topology collected by the RouteViews [119] project and found the results to be qualitatively similar; however, these simulations were restricted to small attacks (upto $X = 100$) due to simulator limitations.

The latency between the attacker and the bottleneck is parameterized by end-to-end RTT data collected by the Meridian project [133]. Each attacker sends 1Mbps of attack traffic. The bottleneck link is set to 10Mbps with drop-tail queuing. All other links are set to 10Gbps. We simulate attack traffic from 10Mbps to 9.6Gbps, that is between 1 and 960 times the bottleneck bandwidth.

To determine the contribution of the ShutUp mechanism and automatic rate-limit, we compare the effectiveness of ShutUp with and without the rate-limit. When enabled, the SM automatically rate-limits traffic to 10Kbps after 10 seconds. The choice of these parameters is primarily to explore the scaling properties of the DoS defense mechanisms in ShutUp rather than to guide deployment.

Figure 3.3 plots a representative attack with aggregate bandwidth 240 times the bottleneck link bandwidth. The figure is split in two parts: the top plots the bottleneck link utilization over time, while the bottom plots the number of attackers that have not received a validated ShutUp. We observe three distinct phases during the attack. Phase 1 lasts for 10 seconds where the bottleneck

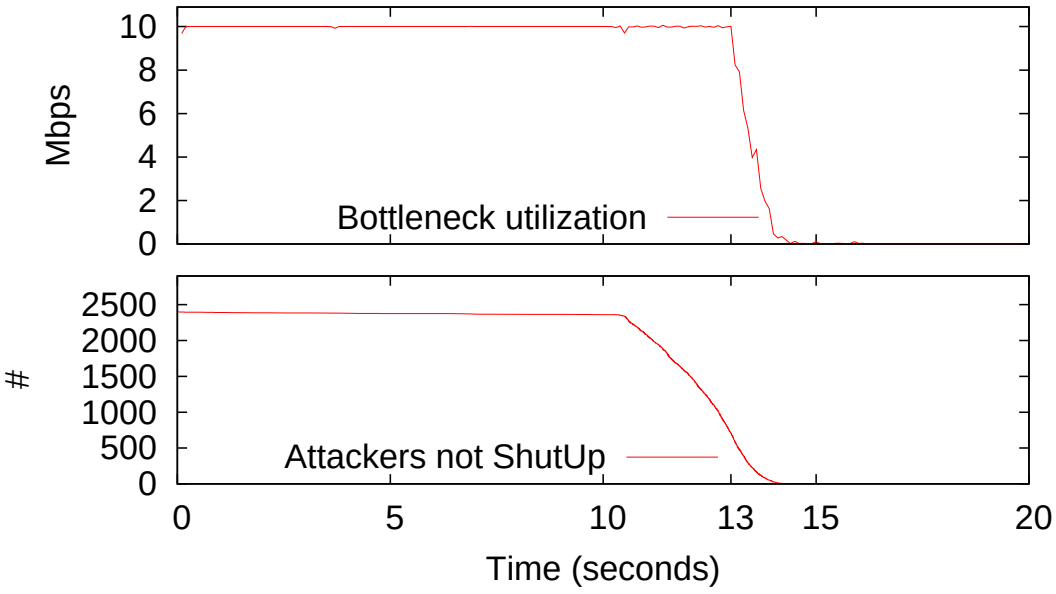


Figure 3.3: The effect of a DoS attack under ShutUp. Aggregate attack traffic is 240 times the bottleneck bandwidth. The SM slows down attackers at 10s, increasing the rate of challenges through the bottleneck.

is saturated and the number of active attackers decreases marginally by 1.6% because few challenges get through. Phase 2 begins at 10 seconds when the automatic rate-limiting kicks in; the active attackers drop rapidly as more challenges succeed, but the aggregate attack traffic, while decreasing, still exceeds the bottleneck. Phase 3 begins at 13 seconds when the aggregate attack traffic equals the bottleneck capacity, at which point all challenges are delivered and the remaining attackers blocked within 1.3 seconds. The figure illustrates that both the ShutUp mechanism and the automatic rate-limit mechanism are necessary to contain large DoS attacks, and together, are sufficient to block the attack in 15 seconds in the above example.

Figure 3.4 plots the time taken to stop a DoS as a function of the attack magnitude. For small attacks, enough challenges get through that the ShutUp mech-

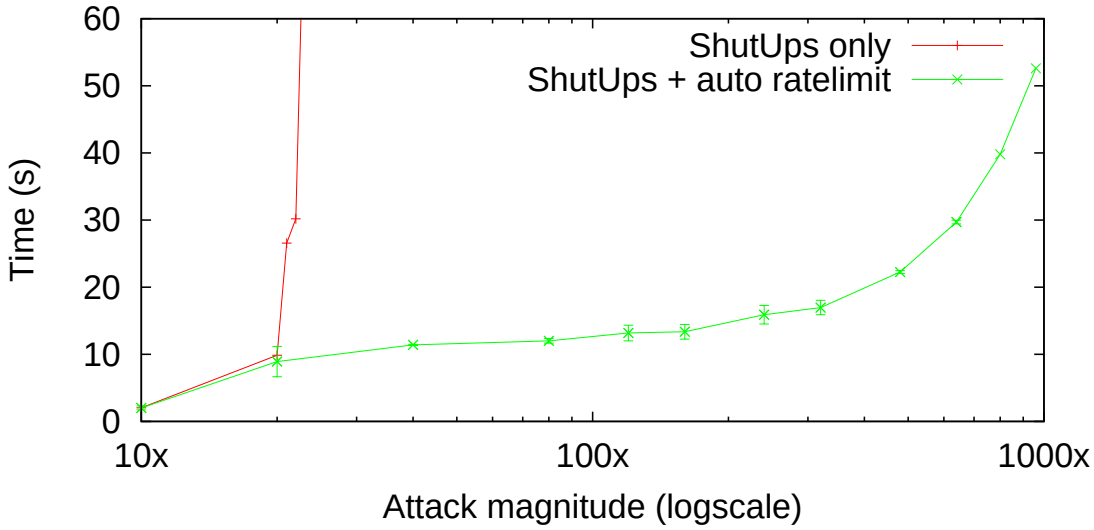


Figure 3.4: Time taken to stop a DoS attack as attack magnitude increases

anism alone can completely stop the attack, but there is a threshold ($X = 20$) beyond which the automatic rate-limiting is needed to keep pace with the attack magnitude. Only when the attack magnitude increases by two orders of magnitude — the same factor as our chosen automatic rate-limit parameter — does the combined approach experience a disproportionate increase. Overall, ShutUp is able to stop attacks ranging widely in their magnitude.

To determine the effectiveness of a partial deployment, we simulated attacks where we varied the fraction of legacy attackers that do not have an SM, and are therefore immune to ShutUps. We simulated ShutUp deployments from 90% to 10% holding the aggregate bandwidth of legacy attackers constant at 90% of the bottleneck link. The results were not surprising: in all cases, ShutUp is able to desaturate the bottleneck in under two seconds by stopping all SM-enabled attackers. ShutUp reduces attack bandwidth by the same fraction as the degree of deployment (i.e. 90% reduction for 90% deployment) — what we might call

"incremental improvability" — the more endhosts in the botnet population that have it, the better the results.

3.4 Slowing Scanning Worms

The scanning defense in ShutUp allows worm outbreaks to be contained (or slowed) using the service. A scanning worm, by its nature, propagates by discovering and infecting vulnerable endhosts. There are several ways how such worms may receive ShutUps. ShutUps may result, for instance, from probes to endhosts not running the vulnerable application, probes to honeynets [107] or network telescopes [76], attack packets to vulnerable endhosts protected by application-level defenses [20], traffic monitored by in-network intrusion detection systems [88] and firewalls.

As mentioned, the SM slows down new flow initiations when a threshold rate of ShutUps is crossed. In contrast, threshold random walk (TRW) based detectors [131, 105] place bounds on the fraction of bad flows to good flows. Both approaches have their pros and cons. TRW can detect an extremely low rate of scanning, but is susceptible to collusion where worm instances can maintain a fixed fraction by establishing "good" flows amongst themselves or with botnet members to absolve an equal number of probes. ShutUp is not affected by such collusion, but does not target sub-threshold scanning; to the extent the randomness and exponential rate-limit in ShutUp compel the worm to scan slowly, ShutUp buys time for other approaches to detect and disinfect compromised hosts. Consequently, we focus on fast scanning worms.

ShutUp ignores worm probes that do not generate a response (not even a ShutUp), for instance probes to non-existent addresses or to hosts behind NATs; interpreting such silent probes as implicit ShutUps would increase the ShutUp frequency and thus reduce the reaction time, but it would also increase false positives (Section 3.5.2). Inside an enterprise, last-hop routers could certainly be configured to send ShutUps for such probes. On the Internet, however, we expect policy will prevent stealthy NATs and firewalls from sending ShutUps for such probes [46]. Fortunately, as we report below, the impact of ignoring these silent probes is minimal for a typical worm attack. We hope to revisit this design decision in the future if the behavior of legitimate applications improves.

Simulation Results: We simulated worm outbreaks in a custom simulator to determine how well ShutUp performs as compared to the TRW-based approach proposed in [131]. We parameterized our simulator with Internet census data collected by the ANT project [51]. As per the data, out of the 2.8 billion allocated unicast public IPv4 addresses, 187 million (6.7%) respond to probes. The remaining addresses are either unused, or used by stealthy hosts. We simulate a Code Red-like worm [78] that scans at a peak rate of ~11 addresses per second, and has a vulnerable population of 359K endhosts. The vulnerable endhosts are distributed uniformly at random among the 2.8B allocated addresses. If the worm probes a vulnerable address, the destination is instantly infected (unless already infected) and begins scanning. Otherwise, if the probe is to the 6.7% addresses that send a response, a ShutUp is simulated, else the probe times out (remaining 93% of the time). At $t = 0$, the attacker infects 1000 vulnerable hosts to initiate the outbreak.

Figure 3.5 plots the number of infected hosts as a function of time. Without

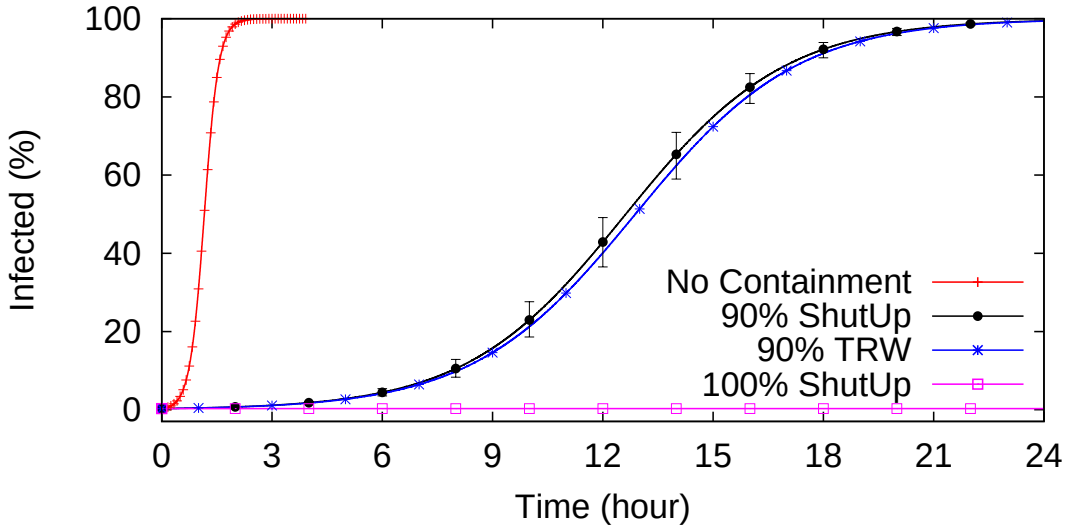


Figure 3.5: Worm outbreak under ShutUp. While a full ShutUp deployment can completely contain scanning worms, a partial deployment slows a worm significantly

containment, the worm infects 95% of the vulnerable population in 1hr 43m. With full ShutUp deployment, the worm is completely contained at 0.3% infections; a full TRW deployment performs identically (not shown). If ShutUp deployment is only 90%, worm propagation is slowed down by an order of magnitude. In 1hr 43m, the worm is able to infect 0.63% versus the 95% without containment, and time taken to infect 20% vulnerable hosts is increased by 8hr 44m. Surprisingly, the difference between ShutUp and TRW, while noticeable, is small even though ShutUp ignores probes to the 93% stealthy/unused addresses while TRW does not. This is because the fraction of responding hosts that are vulnerable is small enough that a sufficient rate of ShutUps is generated to contain the infected host before it discovers a vulnerable host. Consequently, trading off sensitivity to silent probes for fewer false positives is well justified which, as we report later, significantly reduces false positives for ShutUp in comparison to TRW. Overall, a full ShutUp deployment on (or near) endhosts

can completely stop a worm, while a partial deployment can slow it down significantly.

3.5 Evaluation

To evaluate the impact on real endhosts, we evaluated ShutUp using datasets from an enterprise environment, an academic environment, and a home environment.

Implementation: We implemented a proof-of-concept SM in Python. The SM processes connection events generated by Bro [88], infers ShutUps from TCP errors, and for each new outbound flow, passes a verdict whether the flow would have been allowed, rate-limited, or blocked had the SM been running on the endhost during data collection. The implementation is 153 lines long, demonstrating the simplicity of the SM. While our implementation allowed us to rapidly prototype (and refine) the ShutUp design, and provided insights into the impact on legitimate flows, performance and auditability of the SM software stack are important deployment concerns that we believe are best addressed using a non-interpreted language.

Trace Data: Our enterprise dataset consists of a month-long trace of packet headers for all traffic to and from endhosts, primarily employee laptops, for a major corporation. The data was collected *at the endhost*, and thus includes traffic even when the endhost was outside the enterprise. This is the ideal dataset for evaluating an endhost SM. The trace contains little peer-to-peer filesharing traffic as enterprise policy forbid the use of P2P applications. The trace covers 357 users during the first quarter of 2007 with the median trace lasting 26 days.

Processed through Bro, the trace contains about 24.5M TCP flows initiated by the endhosts to 111K addresses.

Our university dataset consists of a 6-day trace of packet headers from endhosts in the computer science department of a major university collected at the border router. The dataset is ideal for evaluating a firewall SM. As with the enterprise, policy discourages the use of P2P filesharing applications. The data contains 5M flows from 1680 IP addresses during a week when the university was in session.

Our home-user dataset consists of an 8-day trace of primarily BitTorrent traffic collected in a home network. It contains 37K flows over which around 5Gb of data was transferred in both directions, which we use to counter the lack of P2P filesharing traffic in the other two datasets.

Methodology: We treat any error responses from the endhost as ShutUps; this typically includes TCP RST packets and non-transient ICMP errors in response to the initial SYN. Flows over which at least one byte of application data is exchanged do not generate ShutUps regardless of how the flow is terminated (e.g. FIN exchange, RST packets, or TCP timeout). Flows where the initial SYN packet does not elicit any response within 3 seconds are treated as unacknowledged flows. We discard other TCP flows (e.g. when a SYN ACK is seen but no data is transferred); such flows comprise less than 0.1% of our data. We do not analyze UDP traffic as we cannot, in general, determine if a flow would have generated a ShutUp. Furthermore, because we do not have packet payloads, our evaluation is limited to ShutUps that an endhost or firewall NM may generate; consequently, application-level errors over a successful TCP flow (e.g. SMTP errors) count as successful flows.

3.5.1 Tuning Parameters

Our choice of system parameters is driven by our data.

IP Spoofing Limit: Users in our enterprise trace use, in the median case, 24 unique source IP addresses over the duration of the trace, with 10% of users using more than 70. The number is significantly higher than that measured for the average Internet user (1 IP address over 2 weeks) [16], which can be explained by the difference in vantage points as [16] measures from the perspective of a content provider. On average, the SM must validate a source IP address and MAC pair every 7 hours.

87% of the time the address is validated within one second, typically by the first application flow. In the worst case, an endhost attempted 1 flow per second for 18 minutes before eliciting any response; in such cases, the endhost OS may be configured to ping the first-hop router shortly after assigning a new IP address to more quickly validate the address. Our implementation rate-limits packets with unvalidated addresses MAC pairs to a burst of at most 60 packets in the first ten minutes, and one packet per minute after that. The rate-limit does not impact 99% of address changes.

Linger period and ShutUp TTL: In order to determine the minimum linger period to be observed by the endhost NM, and the TTL for ShutUp requests, we plot the CDF of the time elapsed until a flow to a destination succeeds after the first encountered failure in Figure 3.6. The y-value represents the fraction of retries that would have succeeded had a failed flow not been incorrectly ShutUp for a given TTL value (x-axis). Curiously, the plot for our enterprise trace is linear on a semi-log scale across six orders of magnitude — a phenomenon well

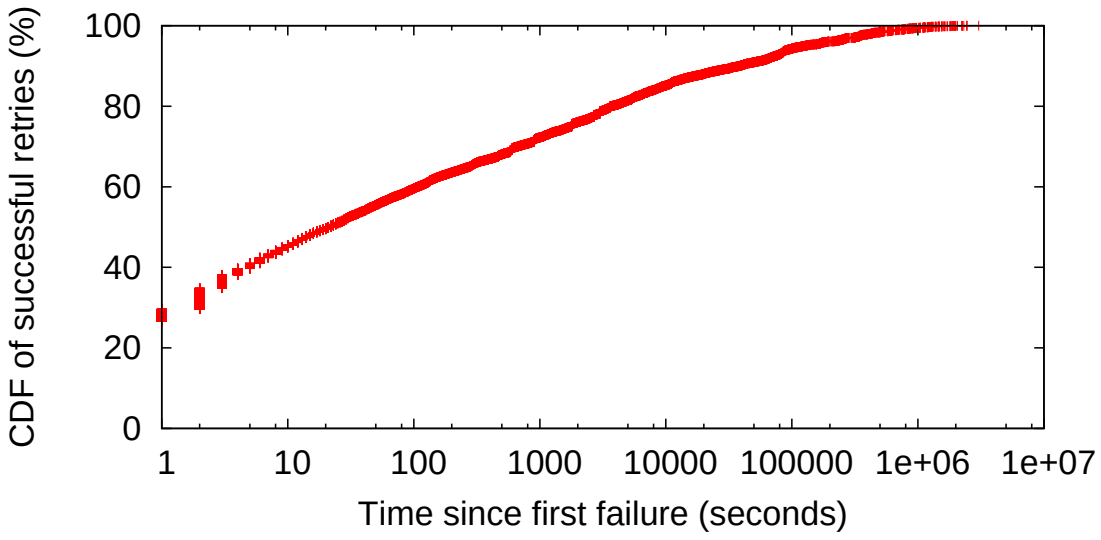


Figure 3.6: Determining the impact of endhost OS ShutUp TTL and linger time on legitimate retries

worth investigating in its own right. In essence, the longer an application has been failing, the longer it is expected to keep failing, whereas the more recent the failure, the more likely it is that a quick retry will succeed, something which we term "failure momentum". A minimum linger time of 2 seconds prevents 35% of retries from being incorrectly blocked. Beyond this, the NM may dynamically pick the TTL in proportion to the length of the outage, however, our implementation uses a static value of 60 seconds.

Scanning threshold: In order to determine a threshold that does not impede legitimate flows while placing a bound on scanning activity, we require an oracle to separate benign traffic from suspicious traffic in our dataset. For this purpose we use the TRW-based scan detection algorithm proposed in [131] to flag potential scanners in our enterprise dataset. We assume the 182 users *not* flagged are least likely to be scanners. For these users, a minimum threshold of 9 ShutUps per minute is necessary to not trigger the scanning defense. Adding a

Table 3.3: Applications generating the most potential false positives for each dataset

Application	*Port(s)*	*Dataset*	*# Hosts*	*# Flows*
ShutUp				
Patch Dist.	63422	Enterprise	9	60
SMTP	25	University	4	171
-	-	Home User	-	-
	Total (all applications):		*7.3%*	*0.02%*
ShutUp (w/ timeout as ShutUp)				
Patch Disc.	63422	Enterprise	119	959
Unknown	3274	University	4	4811
BitTorrent	*	Home User	1	216
	Total (all applications):		*17.6%*	*0.14%*
TRW-based				
Web	80,443	Enterprise	97	15115
SMTP	25	University	3	459750
BitTorrent	*	Home User	1	1078
	Total (all applications):		*8.9%*	*1.7%*

margin for error, we use a threshold of 15 ShutUps per minute before triggering the scanning defense.

State: The state maintained by the SM is dominated by the per application whitelist that is built dynamically. The list can potentially grow unboundedly. The 99 percentile size was 2300 entries for a node (8kB of IPv4 addresses); the maximum was 32K (128kB). At most 437 ShutUps were received in all by a host, of which fewer than 15 were concurrently active in the 95^{th} percentile case. In all, 256kB of state per endhost is sufficient for an endhost SM for our dataset. An SM with limited memory may, however, use a fixed-sized cache for the whitelist, taking care to protect against attackers abusing the cache-replacement strategy.

3.5.2 False Positives

We evaluated ShutUp on all three datasets. For comparison, we also evaluated the TRW-based approach proposed in [131], and a modified version of ShutUp that treats unacknowledged flows as implicit ShutUps. We use well-known port numbers to determine the application name in the enterprise and university trace, while for the home user trace, we tag all flows generated by BitTorrent as such regardless of the port number. For ShutUp, we use the parameters above. For TRW, we primarily use the parameters in [131], which the authors tune using multiple datasets and conclude are applicable in general. We also tested TRW with higher thresholds, and while this reduced the false positives quantitatively, it did not do so qualitatively: the same applications generate the most false positives.

For each approach, we consider flows that the approach decided to block, but which did in fact exchange data during the trace. We manually examined the most egregious endhosts and eliminated blatant scanners based on sequential address or port scans. However, since we lack payload data to automatically disambiguate certain cases, some successful flows may in fact be successful scan attempts; as such, our results identify legitimate applications potentially affected by ShutUp. Table 3.5.2 plots the top applications generating potential false positives for each dataset. Overall, ShutUp results in fewer false positives (0.02% versus 1.7%) for the following reasons:

Enterprise: In the enterprise case, as mentioned previously, the linger mechanism reduces the number of ShutUps generated due to momentary outages. False positives in ShutUp stem from the automated patching application which performs distributed scanning to quickly discover unpatched endhosts; we ex-

pect IT to configure enterprise endhost NMs to not send ShutUps for benign scanning authorized by the enterprise, or filter ShutUps for such applications in the network.

University: The outbound mail-server has a ratio of 70% unsuccessful flows, largely due to undeliverable bounces to spam messages, triggering the scanning defense. The whitelist in ShutUp allows flows to legitimate mail-servers that have never sent a ShutUp to continue unimpeded.

Home User: BitTorrent generates a large number of unacknowledged flows, presumably to other clients behind NATs, but potentially also to clients that have recently left the cloud. Since ShutUp ignores such flows, the scanning defense is not triggered.

3.6 Extensions to ShutUp

In this section we present some extensions to ShutUp that enhance the service provided by the SM. Additionally, we promise cake to those who contact us with ideas for other interesting extensions. In general, extensions to ShutUp present new potential vulnerabilities that need to be considered.

Weighted ShutUps: The SM can give more importance to ShutUps for more severe violations or from authoritative sources. For instance, a ShutUp from an application protected by Vigilante [20], could instantly trigger the scanning defense. Such ShutUps would need to carry proof of their authenticity, such as Vigilante's self-certifying alert that the SM can verify in the ShutUp VM.

Collaborating SMs: The SM can collaborate with nearby SMs to determine the uncleanliness [19] of the local network. The metric may be used to adjust the sensitivity of the SM to better contain worms.

Initiator behavior: The SM can provide information about the initiator's past behavior, such as the diversity of recipients contacted, to new recipients. The NM can compare this to the expected diversity for the application [60] to help determine the legitimacy of the flow.

Wireless Contention: Wireless nodes sharing the broadcast medium with an overly-chatty node could request it to ShutUp. Once the SM verifies that a majority of wireless neighbors concur, it can throttle the application reducing contention.

3.7 Related Work

Receiver oriented communication models are not new. IP multicast [23], i3 [112], and off-by-default [8] all allow the recipient to choose what data it receives. Multicast is not intended for one-to-one communication, i3 requires infrastructure, and off-by-default requires modification to routers and Internet routing. In contrast, ShutUp is end-to-end (or edge-to-edge) and requires no infrastructure.

The work closest to the ShutUp service, either because they use a challenge-response approach or because they propose tamper-proof mechanisms in the attacking endhost [106, 4, 6], are discussed in Section 4.

Many DoS mitigation approaches operate by installing filters along the attack path. The simplest approach is to thin attack traffic using passive and active

filters at the upstream ISP [93]. Taking this idea further, endhosts may install filters in gateway routers, or in routers progressively closer to the source [6, 70, 69]. Combining filtering with explicit authorization, routers upsteam of the bottleneck may be configured to, by default, impede packets not explicitly authorized by the endhost through capability nonces [136, 137]. A second class of approaches [1, 34, 67, 2, 61] dissipate DoS attacks before absorbing the unwanted traffic. One approach is to use a currency, such as bandwidth, to grant access through filters at network choke points [128]. In contrast, ShutUp explores a hybrid design point in between filtering and capabilities, thereby retaining both higher effectiveness and greater resistance to attackers. Furthermore, ShutUp filters DoS traffic at the source (or its firewall), preventing attack traffic from consuming any network resources. Finally, ShutUp, more generally, handles scanning attacks within the same architecture.

Worm containment has been subject of much recent work. In [79], the authors provide guidelines applicable to any worm containment approach. One approach is to rate-limit all flow initiations or to bound the diversity of recipients [132, 41]. Such approaches may negatively impact certain classes of applications such as peer-to-peer applications. Targeted more directly at worms, TRW [58] detects scanning worms from the recipient firewall's perspective. Turning the TRW approach inside-out and operating at a router upstream of the infected endhost, worms may be detected and contained based on an anomalous fraction of failed flows for a particular port number [131, 105]. While the SM can use any combination of these mechanisms, ShutUp extends to applications that use dynamic ports.

Much work has been done in detecting unwanted traffic both in the network and at the recipient endhost that ShutUps complements. In the network, traffic can be classified as portscans [58], worms [62, 104], DoS attacks [55], flash crowds [57], and of dubious origin [22]. Classification can be performed through correlation [116], rule-based matching [63], or entropy in feature sets [66] in, potentially aggregated, flow parameters. At the endhost, application-agnostic exploit detection [20, 107], as well as application-specific detection based on common usage profiles, and deviations therefrom [92, 114] have been proposed. More active methods of detection include CAPTCHAs [80, 59] or tracking user-activity [87] to verify the presence of a human, and puzzle auctions [129] to rate-limit attacks. The NM can directly use these and future approaches to detect unwanted traffic to ShutUp.

3.8 Summary

In this chapter, we propose ShutUp, an end-to-end (or edge-to-edge) service to reduce unwanted traffic in the Internet. We thoroughly explore the design space and determine the small set of mechanisms required to mitigate, and in many cases completely stop, a wide range of unwanted traffic ranging from DoS attacks to scanning worms. ShutUp does not require any additional infrastructure or changes to the protocol stack. Through simulations we find that ShutUp scales well to defend against large scale DoS attacks up to three orders of magnitude in excess of bottleneck links, and can slow scanning worms by an order of magnitude. Using extensive trace data from three environments we establish that ShutUp's impact on legitimate traffic is minimal. Overall, we believe there is a compelling case to be made for containing unwanted traffic at the ends.

That being said, there is still a lot work to be done before ShutUp can be deployed. While ShutUp is easily deployable in enterprises, and globally deployable with buy-in from only a few vendors, it is likely that deployment, at least initially, will be driven by ISPs. Convincing ISPs would require, first, a more thorough analysis of the security implications of deploying an in-network SM, and second, an evaluation of the performance and scaling characteristics of the SM in an ISP setting. In any event, we believe that our biggest challenge is more social than technical, or economical. While the model pushes cost onto the attacker, the cost is actually small. Rather, software and hardware vendors, and ISPs all too often do not even acknowledge their responsibility, far less actively participating, in stemming attacks originating from infected computers or networks under their control. Success requires that we convince vendors and ISPs, or regulatory bodies governing them, that ShutUp is far better than the status quo.

CHAPTER 4

NUTSS: END-MIDDLE-END CONNECTION ESTABLISHMENT

This chapter presents an architecture and protocol, called NUTSS, that satisfies these core EME naming and addressing requirements identified in Chapter 1. Specifically, NUTSS names endpoint applications with user-friendly names, and uses signaling protocols to dynamically and securely do late binding of named endpoints to ephemeral 5-tuple transport flows. Unlike previous architectures [42, 127], transport flows in NUTSS are ephemeral and renegotiated using *both offpath* (an overlay off of the data-path) *and onpath* (on the IP data path) signaling protocols when required. This is in contrast to SIP [99] (offpath only) and RSVP [14] (onpath only), neither of which solves the EME problem.

A simplified NUTSS connection establishment is described as follows. An initiating host transmits a signaling message containing source and destination name, and the name of an application. Using these names as the basis for routing, this message traverses *offpath* policy-aware boxes near both ends, where authentication is done and decisions are made to allow or disallow the connection. Once allowed, ephemeral addresses, ports, and firewall-traversal tokens are conveyed to both ends. Using the learned address as the basis for routing, this information is then used by an *onpath* signaling protocol to establish a data connection through firewalls. The firewall uses the secure tokens as capabilities to allow or disallow flows. It is these tokens that couple the offpath and onpath signaling phases. If the connection breaks, for instance because of mobility or firewall crashing, NUTSS can retry the onpath signaling using the addresses and tokens previously obtained, or failing that, fall back on offpath signaling using the names to re-establish the data flow.

NUTSS does more than satisfy the core EME requirements listed above. By using names as stable unique identifiers, and binding them late to 5-tuple flows as explored in much recent work [81, 112], NUTSS separates identification from network location, thus supporting network mobility and host and site multi-homing. Finally, NUTSS signaling allows endpoints and middleboxes to negotiate the protocol stack used in the data-path. This can be used not only to negotiate transport (UDP, TCP, SCTP, etc.) and security (IPsec, TLS, SSH, etc.), but different network layers as well (IPv6, IPNL [32], TRIAD [42], HIP [81], i3 [112], DoA [127], etc.). The ability to negotiate protocols as well as middleboxes creates a framework for managing multiple network layers created through virtual routers and switches, for instance as proposed for the GENI infrastructure [40]. Indeed, this very flexibility is exploited by NUTSS to provide itself with an incremental deployment path (Section 4.1.6).

Up to this point, we have asserted that NUTSS satisfies contemporary EME requirements without changes to existing network protocols. Indeed, we can make a stronger assertion: that *any* new network protocol benefits tremendously from a name-based signaling approach like NUTSS. This claim flies in the face of recent self-certifying, identity-based architectures [81, 112, 126, 127, 64], which suggest not only that flat identities can serve as the basis of network or content identities, but in some cases go so far as to suggest that there is no need for a single global user-friendly name space [126, 64]. Rather, a wide range of ad hoc mechanisms, such as search engines and HTML links, can be used to "discover" identifiers.

Our difficulty with these architectures derives mainly from the fourth EME requirement—that unwanted packets must be blocked before they reach the ap-

plication, ideally in the network. This requires, among other things, that access control policy (e.g. ACLs) be configured in middleboxes. Today firewall vendors strive to build devices that may be configured using user-friendly names ("BitTorrent", or "ftp"), and that can filter on aggregates such as DNS zones or IP prefixes [17]. Flat identifiers are neither user-friendly nor aggregatable, and therefore are not well-suited to serve as the basis for ACL configuration. This is an issue that the proponents of identity-based approaches have not addressed, in spite of the fact that they recognize middleboxes as being no longer harmful, and incorporate mechanisms to steer packets through them [112, 127, 64]. There must be a globally-understood user-friendly namespace that identifies endpoints (applications, services, users, etc.), as well as a way to bind those names to the addresses, ports, and identifiers of data packets (collectively referred to here as "addressing material").

A key issue, then, is how to bind names to the addressing material. Both TRIAD [42] and IPNL [32], which use DNS names as user-friendly host identifiers, bind those names to network addresses by carrying both names and addresses in data packets. These schemes literally treat names as addresses in the sense that network routers run routing algorithms on the names themselves, and bind these to numerical addresses primarily as an optimization for the address lookup function. Both name-routed and address-routed packets follow the data path (in other words, are routed onpath).

While neither TRIAD nor IPNL sought to solve the middlebox problem, one can imagine extending them to do so, for instance by extending their host names with user, application, and service names, and by authenticating those extended names. Even so, we find onpath approaches to be less attractive than offpath

approaches that use overlays to do name-based routing. Onpath approaches are both overly constraining and overly intrusive. They are constraining in that they force the name-based access control policy decision to be made onpath. They are intrusive in that they force all routers to participate in a name-based routing algorithm that, in the case of TRIAD, may scale poorly, or in the case of IPNL, requires a DNS lookup at packet forwarding time.

An overlay approach to name-based routing, by contrast, allows the access control policy decision to be made anywhere in the Internet. In particular, it allows access control to be widely replicated and therefore more resilient to flash crowds or DoS attacks [90]. DNS, of course, is a name-based routing overlay, and certainly much of its success may be attributed to the fact that it is decoupled from onpath routing and is therefore easier to deploy. The problem with DNS in the EME context is that it is not at all designed to do access control. DNS is not aware of who is making a DNS query, and is typically not aware of the purpose of the query (i.e. which application the query is for). Indeed, current use of dynamic DNS [123] reveals private location information about mobile users, making it possible for instance to follow their travel itineraries [47]. Merely confirming the existence of a valid name to an unauthorized user can be considered a breach of privacy defined as contextual integrity [83].

Another widely deployed name-based routing overlay is SIP [99], which is used for media (voice or video) flow establishment. For the purposes of EME requirements, SIP is at least better than DNS in that it carries the identity of both endpoints and allows them to be authenticated. Furthermore, SIP enables a powerful set of features, including mobility, rich naming of users and endpoints, discovery, the ability to negotiate different protocols, independence from

underlying transport, and the creation of infrastructure to support it all. Nevertheless, SIP itself is not designed to couple the offpath access control policy decision with onpath access control enforcement. Industry has tried to address this shortcoming in two ways. One is to implement SIP in the firewall itself [18]. This approach does not work in all cases, because the name-routed signaling path may differ from the address-routed data path. For instance, consider a dual-homed site with combined firewall/SIP servers $F1$ and $F2$. The signaling path may traverse $F1$, which authorizes the flow and allows access for the associated addressing material. The subsequent data path, however, may traverse $F2$, which has not authorized the flow.

The other way is to define a protocol that allows the SIP server to coordinate with the firewall [72, 115]. This approach suffers from a similar problem which may be solved in a brute-force fashion by having the SIP server enable a given flow in all possible firewalls that the data flow may traverse. While in the common case (a dual-homed site) this may be reasonable if inefficient, it becomes unworkable in scenarios where there are many firewalls. For instance, a widely replicated distributed firewall addressed as an IP anycast group might have hundreds or thousands of firewalls [30].

The key contribution of this chapter is the design of NUTSS, a protocol that satisfies the core EME requirements through the novel combination of dual signaling—the explicit coupling of offpath name-routed signaling with onpath address-routed signaling to establish ephemeral 5-tuple flows. It is this novel coupling that overcomes the disconnect between name-based routing and IP routing that plagues previous approaches. NUTSS works with existing data protocol stacks (IPv4 or IPv6), and includes an incremental deployment

path that initially requires no changes to NAT boxes. As with other architectures that separate location from identity, NUTSS facilitates mobility and multihoming. Besides describing the design of NUTSS, this chapter presents a proof-of-concept implementation and deployment of NUTSS and examines whether SIP [99] is appropriate as the offpath signaling protocol for NUTSS.

4.1 NUTSS Architecture

This section starts with a brief overview of the NUTSS architecture, followed by a detailed description of NUTSS.

4.1.1 NUTSS Overview

In NUTSS, named *endpoints* may be applications or services, and may be associated with individual users or endhosts. The names are user-friendly, long-term stable, and location-independent. When an endpoint application wishes to establish a data flow with another endpoint, it opens a NUTSS socket using the names only (and not IP addresses) as endpoint identifiers. This triggers an end-to-end name-based signaling exchange that authenticates the endpoints and establishes the state necessary to transmit a 5-tuple (source and destination IP address, source and destination port, and IP protocol) *data flow* end-to-end via a series of middleboxes, including NATs and firewalls. In addition to the 5-tuple parameters and NAT mappings normally required by flows, this state also includes authorization tokens needed to traverse middleboxes that do access control.

There are two components in NUTSS, *P-boxes* and *M-boxes* (for policy-box and middlebox respectively). P-boxes and M-boxes are deployed in the network as well as in endhosts. Networks that enforce policies, such as access control or steering policies, must deploy P-boxes and M-boxes. P-boxes form an overlay over which name-routed signaling messages are carried end-to-end. Data flows (or just *flows* for short) do not traverse P-boxes. Flows do, on the other hand, traverse M-boxes, either because the M-box is deployed on the IP path between endpoints (as with a firewall), or because the signaling has negotiated to steer a flow through an M-box (for instance, an anonymizer). P-boxes make policy decisions about richly-named flows: whether to allow or disallow them, whether to steer them through M-boxes, whether to require encryption, and so on. M-boxes enforce the policy decisions made by an associated P-box.

Signaling messages may traverse P-boxes or M-boxes. They traverse P-boxes, routed by name, when no IP address is known for the destination, or when the security tokens needed to traverse M-boxes have not been obtained. Signaling through P-boxes is referred to as *name-routed* signaling. Otherwise, signaling messages naturally traverse M-boxes, routed by the destination IP address obtained during name-routed signaling (called *address-routed* signaling). Because a name-routed P-box overlay path always exists between endpoints, even for endpoints behind NAT boxes, there is always a way to signal another endpoint to establish a flow (policy permitting).

There is a bidirectional coupling that exists between name-routed and address-routed signaling, which exists by virtue of shared information (keys and addresses) between P-boxes and their associated M-boxes. This coupling is necessary to overcome the unavoidable lack of coordination between name-

Table 4.1: NUTSS API for establishing flows and controlling access

Parameters
E : ($user$, $domain$, $service$) - Endpoint name
A : $address$ - Network address to reach endpoint
P : $port$ - Transport port for data flow
τ : ($token$, $nexthop$) - address-routing state
ρ : (E_P, A_P) - Referral to P-Box

Name-routed messages (sent to P-Box)
REGISTER(E, A)
Register a name-route (wildcards OK).
FLOWNEGOTIATE(E_{src}, E_{dst}, A_{src}, $\tau_{1...n}$)
Use name-routed signaling to negotiate address-routed path.
P-Boxes add τ_i, and modify A_x to effective address $A_{x'}$

Address-routed messages (sent through M-Box)
ρ = FLOWINIT(A_{self}, $A_{peer'}$, P_{self}, $\tau_{1...n}$)
Use address-routed signaling to initialize data path.
An M-Box may refer to additional P-Boxes to contact
M-Boxes modify P_x to effective port $P_{x'}$
ρ = SEND(A_{self} : P_{self}, $A_{peer'}$: $P_{peer'}$, $data$)
Send data packet

Access Control (sent to P-Box)
DISALLOW(E_{dst}, E_{src})
ALLOW(E_{dst}, E_{src})
Add/remove filters for destination (wildcards OK).

based overlay routing and IP-based address routing. Specifically, P-boxes convey secure tokens to endpoints during name-routed signaling, which are then carried in address-routed signaling to traverse M-boxes. If an unapproved flow is attempted through an M-box, the M-box *refers* the sending endpoint to a P-box that may authorize the flow.

4.1.2 Naming and Access Control

Endpoint names in NUTSS are (*user*, *domain*, *service*) 3-tuples. The *user* is a user-friendly identifier that is not globally unique (e.g., bob). The *domain* is a globally-unique, user-friendly, hierarchical DNS name (e.g., acme.org). Together the user and domain identify the *principal* that is considered to own the endpoint; the user may be NULL, in which case the domain effectively identifies a machine. The *service* is a globally-unique, user-friendly identifier for the service provided by the endpoint (e.g. ftpd for an FTP-server). Names are independent of network location.

Access control policy is defined in terms of names. Wildcards are permitted in policy definitions. A wildcard service name $*$ matches all services run by a particular principal (e.g. (*bob*, *acme.org*, $*$)), while a wildcard user name matches all principals in that domain. Because, as domains are organized hierarchically, a wildcard prefix in the domain name matches all subdomains below that domain (e.g. ($*$, $*$.*cs.acme.org*, $*$)).

NUTSS relies on existing mechanisms to authenticate endpoint identities. Standard protocols, such as public-key signatures or challenge-response protocols (e.g. DIAMETER [15]), over the name-routed path are used to authenticate principals. Similarly, services can be authenticated if the necessary hardware and software support is present at endpoints. For instance, [101] proposes an architecture that leverages trusted hardware [117] to measure the integrity of the software-stack. Therefore, authentication is not further addressed in this chapter.

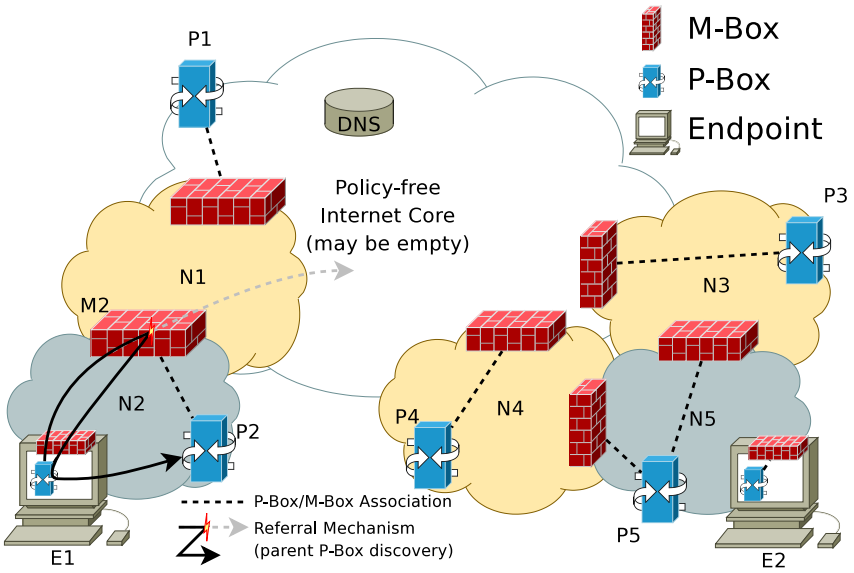


Figure 4.1: Network topology and referral mechanism. Network $N5$ is multi-homed.

4.1.3 Name-routed Signaling

We now discuss how NUTSS creates a name-routed path between endpoints. Our goal in creating this path is to incorporate the policy of networks on the data path between the endpoints. As mentioned, this is accomplished through policy-aware P-Boxes that, by design, form a name-routing tree[1] (rooted at the DNS). Endpoints form the leaves of the tree such that the path between two endpoints along the tree passes through P-Boxes designated by networks topologically between the two endpoints.

Network Topology

NUTSS models the Internet topology as policy-aware edge networks connected by a policy-free core (Figure 4.1). The policy-free core (or just *core* for short)

[1]In the presence of multi-homed networks, this is a directed acyclic graph

is defined as the set of interconnected networks that do not assert middlebox policies and so do not deploy P-Boxes. This model reflects the current Internet: networks with firewalls today correspond to (policy-aware) edge networks, and networks without firewalls correspond to the (policy-free) core. Edge networks may comprise smaller networks that are separate administrative entities. Each network designates one logical P-Box (potentially multiple physical instances), which may be located either inside or outside that network (e.g. in the figure, network $N2$ designates P-Box $P2$ and $N1$ designated $P1$). A P-Box for a network not connected directly to the core has a *parent P-Box*. The parent P-Box is the P-Box for an adjacent network through which the former connects to the core ($P1$ is $P2$'s parent). A P-Box for a multi-homed network ($P5$) has multiple parents ($P3, P4$).

The network administrator associates M-Boxes with the P-Box for the network. M-Boxes are typically, though not always, deployed at the network boundary (e.g. $M2$). P-Boxes use standard cryptographic mechanisms (shared symmetric keys, or public keys) to pass confidential messages to the M-Box via untrusted third-parties. To facilitate deploying many M-Boxes, a P-Box need not know the addresses of the M-Boxes (except for M-Boxes that must be explicitly addressed e.g. NATs). M-Boxes, on the other hand, are all configured with the name and address of their associated P-Box. The P-Box and M-Box may optionally be co-located in the same physical package.

Endhosts have a resident P-Box and M-Box (Figure 4.1). NUTSS primitives (Table 4.1.1) are initially sent by endpoints to their local in-host P-Box and M-Box, and from there, to other P-Boxes and M-Boxes.

NUTSS assumes the presence of the DNS (or a similar name-resolution service) in the core for name-based routing across domains. For each domain, the DNS contains the addresses of one or more *contact P-Boxes* for that domain. The contact P-Box is the outermost P-Box through which the endpoints in that domain can be reached. For example, in Figure 4.2, endpoints from acme.org typically register with $P2$; the DNS administrator for acme.org lists $P1$ as the contact P-Box for his domain as $P1$ can reach those endpoints through its child $P2$. Contact P-boxes must be globally addressable.

Note that the core may in fact be empty, as long as contact P-boxes can exchange packets between each other. The rationale for exploiting DNS in the core is similar to that of IPNL [32]: it allows NUTSS to scale globally without changing the Internet core and without requiring new name-based routing protocols as do TRIAD [42] and DONA [64].

Discovery

A P-Box discovers its parent P-Box through the M-Box referral mechanism mentioned earlier. The child P-Box (e.g. $E1$'s in-host P-Box) sends an address-routed message to a public address. The message contains any authorization tokens needed to clear the M-Boxes for the originating network/host (generated by the P-Box itself), but does not contain authorization tokens for the parent network ($N2$); the parent network's M-Box ($M2$) therefore blocks the message and responds with the name and address of the parent P-Box ($P2$). An advantage of using normal address-routed messages for P-Box discovery is that if P-Boxes and M-Boxes are added (even mid-flow), for instance if a site becomes multihomed, they can be discovered via normal operation of the protocol.

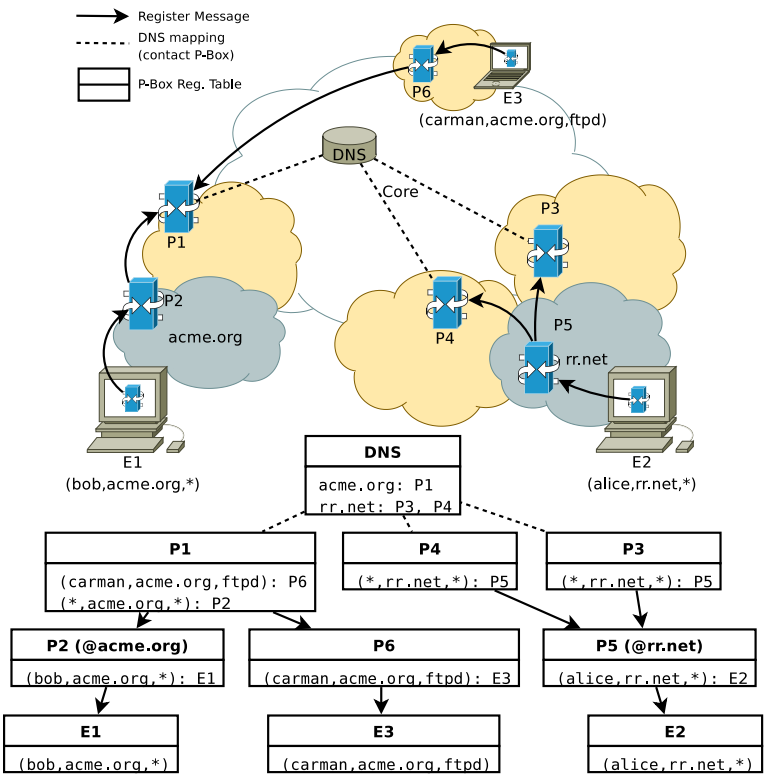


Figure 4.2: Endpoint registration, and name-routing state created. Network $N5$ is multi-homed. Endpoint $E3$ is roaming.

Name-Route Creation

The REGISTER message, sent by endpoints through P-Boxes of networks connecting it to the core, creates a (reverse) route from the DNS to the endpoint through P-Boxes designated by the middle. The process is described as follows: endpoint E with network address A that wishes to accept flows sends the REGISTER(E, A) message to the local P-Box (Figure 4.2). When a P-Box receives a REGISTER message (Algorithm 4.1), it adds the mapping to its local *registration table* (assuming the endpoint is authenticated and authorized to register with that P-Box). If the P-Box has any parent P-Boxes, the P-Box propagates a mapping between the endpoint's name and the P-Box's own address to all the parents. This process is repeated recursively until the REGISTER reaches the core.

Algorithm 4.1: PROCESSREIGSTER(E,A)

Require: E is endpoint name (EU,ED,ES)
Require: A is next-hop address to E
Require: E has been authenticated, can be reached through A, and is authorized to register as per local policy.
Ensure: Name-routed path from contact P-Box for ED to E exists

```
1: UPDATEREGISTRATIONTABLE(E,A)
2: AL ← GETLOCALADDRESS()
3: FWDTO ← GETPARENTPBOXADDRESSES()
4: if ISEMPTY(FWDTO) then
5:     FWDTO ← GETCONTACTPBOXADDRESSESFOR(ED)
6:     if CONTAINS(FWDTO, AL) then
7:         return
8:     end if
9: end if
10: MSG ← new REGISTER(E,AL)
11: for all AP in FWDTO do
12:     SENDTO(MSG, AP)
13: end for
```

For instance, in Figure 4.2, $E1$'s registration is forwarded by his in-host P-Box to $P2$ then to $P1$, $E2$'s registration to $P5$ then to both $P3, P4$, and $E3$'s registration to $P6$. Now, if the outermost P-Box is a contact P-Box registered for E's domain, then the registration process terminates as the reverse route from the DNS to the endpoint is complete (e.g. for $E1, E2$). Otherwise, to complete the route the message is forwarded one last hop to the contact P-Boxes for that domain (e.g. for $E3$); this second case is typically encountered by roaming endpoints.

As an optimization, wildcards in REGISTER messages are used to register default routes. A principal can register a default route for all services owned to point to his primary endhost, while a domain (or sub-domain) administrator can register a default route for all endpoints in that domain (or sub-domain) to go through a P-Box he administers. During name-based routing, the most

specific registration is used, that is, a route for the endpoint is preferred over a route for the principal, which is preferred over a route for the longest matching domain portion.

Access Control

Flow requests may be rejected by P-Boxes in the network in one of two ways. First, the lack of a registration for a given service or principal will cause a P-Box to reject a flow request for that service or principal. Second, an endpoint or P-Box administrator may specify that flow requests for registered names be additionally filtered by the name of the requester, either as a whitelist or a blacklist.

These filters are installed in much the same way as name-routes. An endpoint E_{dst} that wishes to disallow flow requests from E_{src} sends the DISALLOW(E_{dst}, E_{src}) message to the local P-Box; wildcards can be used in either endpoint name to broaden the scope of the filter. A P-Box administrator may likewise do a DISALLOW(E_{dst}, E_{src}) at its P-Box. Either way, P-Boxes may forward the filter up the name-routing tree (as with REGISTER messages), but unlike REGISTER messages, the filter message need not bubble up all the way to the top. The filter should nevertheless go beyond the local (in-host) P-Box to allow for in-network filtering. How to resolve conflicting filters is a matter of local policy.

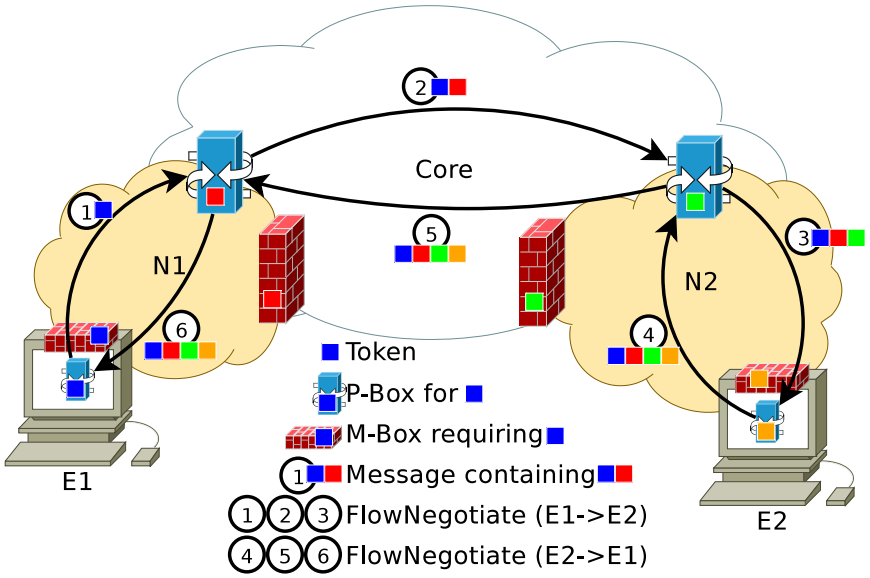


Figure 4.3: Flow negotiation over name-routed signaling.

Name-Routing

Name-routing is performed over the tree-overlay created by P-Boxes and endpoints in the registration process. An endpoint E_{src} that wishes to initiate a flow with E_{dst} sends a FLOWNEGOTIATE $(E_{src}, E_{dst}, A_{src}, [\,])$ message to its local P-Box. E_{src} and E_{dst} are the endpoint names (no wildcards allowed), and A_{src} is the network address of the initiator. The P-Box authorizes the flow based on installed filters and local network policy. If authorized, the P-Box forwards the message towards the destination as illustrated in Algorithm 4.2: if the local registration table has an entry matching E_{dst}, the message is forwarded to the associated address. If no matching entry exists and the P-Box has a parent P-Box, the message is forwarded to the parent. If no parent P-Box exists (outermost P-Box), the message is forwarded to a contact P-Box for the destination domain. Local policy may be consulted to pick one or more of many candidate P-Boxes to forward to (e.g. for multi-homed networks).

Algorithm 4.2: PROCESSFLOWNEGOTIATE(ES,ED,AS,T)

Require: ES is source endpoint
Require: ED is destination endpoint (EDU,EDD,EDS)
Require: AS is effective source address
Require: T is address-routing state $\{\tau_{1...n}\}$
Require: ES is authenticated and authorized to contact ED
Ensure: Endpoints acquire address-routing information needed

```
1: if DISALLOWEDBYFILTER(ED,ES) then
2:     return false
3: end if
4: if EXISTSINREGISTRATIONTABLE(ED) then
5:     FWDTO ← REGISTEREDADDRESS(ED)
6: else if HAVEPARENTPBOX() then
7:     FWDTO ← SELECTPARENTPBOXADDRESS()
8: else
9:     FWDTO ← SELECTCONTACTPBOXADDRESSFOR(EDD)
10: end if
11: TOK ← CREATEAUTHTOKEN()
12: if BEHINDNAT(AS) or EXPLICITMBOX() then
13:     AS' ← GETMBOXEXTERNALADDRESS()
14: else
15:     AS' ← AS
16: end if
17: T' ← T ∪ {(TOK,AS)}
18: MSG ← new FLOWNEGOTIATE(ES,ED,AS',T')
19: SENDTO(MSG, FWDTO)
```

Before forwarding the FLOWNEGOTIATE, the P-Box modifies it by adding τ_i : (*token*, *nexthop*), which is the state needed by endpoints and M-Boxes to initialize the address-routed path. τ contains an authorization *token*, which is a nonce signed by the P-Box. If the A_{src} advertised by the endpoint is behind a NAT M-Box, or if the M-Box terminates the address-routed flow (e.g. application level M-Boxes that must be explicitly addressed), the P-Box replaces A_{src} with the address of the M-Box — this is the address that the remote endpoint should send packets to. In such cases, the M-Box will, however, eventually need the original A_{src} for address-routing of processed packets; for this purpose, the P-

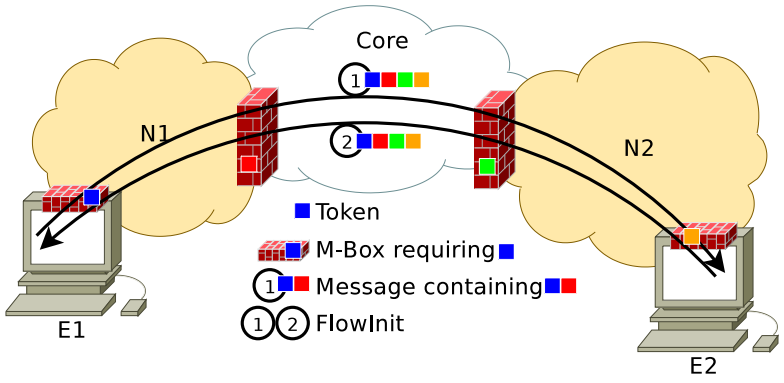


Figure 4.4: Flow initialization over address-routed signaling (performed after flow negotiation in Figure 4.3).

Box uses the *nexthop* field in τ to communicate A_{src} to the M-Box. This addition of tokens is illustrated in Figure 4.3 where each P-Box enroute adds a token required by its M-Box.

When the destination receives the FLOWNEGOTIATE, it learns the effective address of the initiator and a set of tokens $\tau_{1...n}$ that it needs to initialize its data path. The destination name-routes its own address (A_{dst}) and the acquired tokens $\tau_{1...n}$ back to the initiator in a FLOWNEGOTIATE message, which allows the initiator to learn the destination's effective address and tokens.

4.1.4 Address-routed Messages

Endpoints use the peer address and $\tau_{1...n}$ acquired over name-routed signaling to initialize the address-routed path. The initialization installs any necessary per-flow state in M-Boxes enroute. The initialization process is described as follows: both endpoints address-route a FLOWINIT($A_{self}, A_{peer'}, P_{self}, \tau_{1...n}$) message to the remote endpoint; the message is sent to the peer's effective address $A_{peer'}$

Algorithm 4.3: PROCESSPACKET(P)

Require: P is an address-routed packet
Ensure: Only authorized flow packets can pass

```
 1: if FOREXISTINGFLOW(P) or FORMYPBOX(P) then
 2:     FORWARDPACKET(P)
 3:     return
 4: end if
 5: if PACKETISFLOWINIT(P) then
 6:     for all τ_i in P do
 7:         if ISVALIDAUTHTOKENFORME(τ_i) then
 8:             if IAMANAT(P) then
 9:                 FWDTO ← GETNEXTHOPIN(τ_i)
10:                 CREATENATSTATE(P, FWDTO)
11:             end if
12:             FORWARDPACKET(P)
13:             return
14:         end if
15:     end for
16: end if
17: RESPONDWITHREFERRAL(P)
```

over IP from the local source address A_{self}. P_{self} is the local transport port allocated for the flow, and $\tau_{1...n}$ are the tokens accumulated in the FLOWNEGOTIATE. The message is naturally routed through M-Boxes for networks on the IP-path between the endpoints as shown in Figure 4.4.

At each M-Box, the message is checked for the presence of a τ_i with a valid authorization token for that M-Box. If found, the message is forwarded to the next-hop as per normal IP routing. If an M-Boxes requires additional state to forward the message (e.g. NATs), the M-Box initializes this state from the *nexthop* field in τ_i. Port-translating NAT M-Boxes also translate the advertised port P_{self} for outbound messages; this allows the remote endpoint to learn the effective port to use. Once both endpoints have sent FLOWINIT messages, application data can flow along the address-routed path.

As mentioned earlier, if an M-Box receives a message without a valid authorization token, the M-Box responds with a REFERRAL message for its associated P-Box (Algorithm 4.3). The only exception is a message sent to the associated P-Box, as the P-Box must by default be reachable from both inside and outside that network to route new name-routed messages.

Note that M-Boxes, in general, are not explicitly addressed. This is needed so that IP routers retain the ability to route around network failures (particularly around a failed M-Box). If an M-Box fails, the IP route may fail over to another M-Box in the same network; the second M-Box generates a referral for the first data packet routed through it (due to lack of flow state). In such cases, the endpoint attempts to re-initialize the address-routed flow through the new M-Box with the tokens it used to initialize the first M-Box; this is likely to succeed and data flow can resume immediately. In cases where the IP route fails over to a different network altogether (with potentially different flow policies), the original set of tokens is insufficient and the endpoint must renegotiate the flow over name-routed signaling through the referred P-Box before reinitializing the new address-routed path.

4.1.5 Security Considerations

P-Boxes, M-Boxes, referrals, tokens, names and name-routed messages are new elements for attackers to attack, and through them, attack flow establishment. We now discuss how the architecture defends against these new attacks.

NUTSS brings Akamai-like protection to all endpoints. NUTSS allows for massive replication of P-Boxes and M-Boxes by being flexible about (and dy-

namically discovering) where they are placed in the network. Furthermore, the NUTSS token mechanism can be co-opted by approaches such as TVA [137], to provide capability-based DDoS protection to endhosts. While this approach is similar to that taken by Akamai [1], NUTSS operates at the network layer and need not rely a single large proxy provider. NUTSS assumes the presence of external DDoS protection mechanisms [70, 6, 8, 136, 137, 61, 2, 53] to protect P-Boxes and M-Boxes at the IP level. Other than that, standard defenses (crypto puzzles [130], CAPTCHAs [125], etc.) delivered over the name-routed path apply against resource exhaustion attacks.

We assume that standard authentication protocols on the name-routed path are used by P-Boxes and endpoints to establish trust in each other. P-Box to P-Box communication may be secured with keys exchanged out-of-band when possible (e.g., when establishing customer-provider relationships, or stored in DNS). NUTSS does not mandate the mechanism for establishing trust. As today, trust can be established through reputation-based "webs-of-trust" [139], mutually trusted certificate authorities [122], trusted hardware [117], trust in domains that have good security practices through [15], and so on as per individual preference.

Another target for attack is the authorization token used to couple the name-route to the address-route. An eavesdropper may attempt to use the token generated for legitimate endpoints. A small alteration in how tokens are handled protects tokens against eavesdroppers. The token is never sent in the clear: P-Boxes append three copies of the token in τ, one encrypted for each endpoint, and one encrypted for the M-Box. Endpoints sign FLOWINIT messages with their copy and include the encrypted M-Box copy within. M-Boxes decrypt the

token and use it to verify the signature to establish that the endpoint sending the packet possesses the token.

A malicious P-Box (or M-Box) can, at worst, deny service to an endpoint behind it. Note, however, that a malicious P-Box not on the name-routed path between the endpoint and its contact P-Box cannot fake a registration, nor can a malicious P-Box redirect flows to malicious endpoints; authentication protocols along the name-routed path prevent it. Malicious M-Boxes may attempt to redirect FLOWINIT messages to an alternate (malicious) destination, however, without access to the tokens possessed by the intended destination, the alternate destination cannot complete the initialization process in the reverse direction. The only time an address-routed path can be diverted without authorization tokens is if *every* M-Box between the two endpoints is compromised — including, in particular, the in-host M-Box of the non-malicious endpoint.

A malicious endpoint may attempt to abuse its network privileges; the middle can, in response, contain such endpoints at the cost of additional name-routed messages. For instance, an endpoint can attempt to replay legitimately acquired tokens to initialize paths to multiple destinations only one of which is explicitly authorized. This is possible because, *by default*, tokens are bound to named flows and not to ephemeral addresses (to allow for some mobility); P-Boxes may however choose to bind the token to the addresses from which the token can be used, limit the time or number of data bytes the token is valid for, or in extreme cases, make the token single-use by requiring M-Boxes to notify the P-Box of each use. The cost of restricting tokens to granularities finer than flows is additional name-routed signaling each time the address-route breaks trivially (e.g. M-Box reboots).

4.1.6 Incremental Deployment

We now describe how the NUTSS architecture can be realized in three incremental phases. The goal of the deployment strategy is to create incentives for applications and networks to adopt NUTSS while keeping costs low.

In the first phase, only endpoint applications are made NUTSS-aware; this involves performing name-routed and address-routed signaling during connection establishment but does not require any changes to networks. A third party provides a public P-Box that the application can use. Endpoints benefit from architectural support for mobility, ability to traverse legacy NATs (the "killer-app" use-case of NUTSS as described in the next section), and end-to-end (but not end-middle-end) access control. In Section 4.3, we report on our implementation and deployment of this first phase.

In the second phase, the middle is gradually made name-aware. This is accomplished by individual networks deploying a P-Box. Endpoints behind these networks are configured to use the P-Box (in the same way that DNS resolvers are configured today i.e. through DHCP). The need for configuration is temporary until networks deploy M-Boxes in the third phase allowing the referral mechanism to operate. Networks benefit by gaining insight into, and weak access control over, flows carried by the network.

In the third and final phase, networks replace legacy middleboxes with NUTSS-aware M-Boxes. M-Boxes allow networks to enforce access control policies, and control network use in multi-homed settings. The need for legacy NAT traversal and P-Box configuration introduced in the first two deployment phases is eliminated. If the network still has some legacy (non-NUTSS-aware)

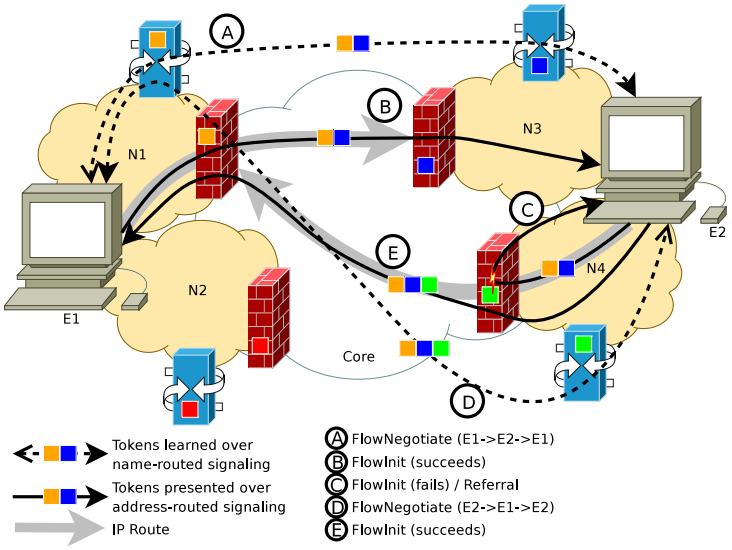


Figure 4.5: Asymmetric routing example. $E1$ and $E2$ are multi-homed. All M-Boxes perform NAT. IP routing is asymmetric.

endpoints that were not upgraded in the first phase, the M-Boxes are made aware of them so the M-Boxes can allow them through.

4.1.7 An Example: Asymmetric Routing through Firewalls

We end this section with an example that demonstrates the need to couple name-routed and address-routed signaling, and describes how existing approaches fail in this case. The example involves a scenario, shown in Figure 4.5, that may easily arise with site multi-homing. In this example endpoints E_1 and E_2 wish to communicate. Both endpoints are multi-homed; E_1 connects to the Internet through networks N_1 and N_2, and E_2 connects through N_3 and N_4. Each network N_i operates a NAT M-Box (M_i with external address A_{M_i}) and an associated P-Box (P_i). Inside the multi-homed networks, IP routing results in asymmetric paths — packets from E_1 to A_{M_3} and A_{M_4} are routed through N_1 and N_2

Table 4.2: Message-flow for asymmetric routing example.

#	From	To	Message
1.	E_1	P_1	FLOWNEGOTIATE($E_1, E_2, A_{E_1}, []$)
2.	P_1	P_3	FLOWNEGOTIATE($E_1, E_2, A_{M_1}, [\tau_1]$)
3.	P_3	E_2	FLOWNEGOTIATE($E_1, E_2, A_{M_1}, [\tau_1, \tau_3]$)
4.	E_2	P_3	FLOWNEGOTIATE($E_2, E_1, A_{E_2}, [\tau_1, \tau_3]$)
5.	P_3	P_1	FLOWNEGOTIATE($E_2, E_1, A_{M_3}, [\tau_1, \tau_3]$)
6.	P_1	E_1	FLOWNEGOTIATE($E_2, E_1, A_{M_3}, [\tau_1, \tau_3]$)
7.	E_1	M_1	FLOWINIT($A_{E_1}, A_{M_3}, P_{E_1}, [\tau_1, \tau_3]$)
8.	M_1	M_3	FLOWINIT($A_{M_1}, A_{M_3}, P_{M_1}, [\tau_1, \tau_3]$)
9.	M_3	E_2	FLOWINIT($A_{M_1}, A_{E_2}, P_{M_1}, [\tau_1, \tau_3]$)
10.	E_2	M_4	FLOWINIT($A_{E_2}, A_{M_1}, P_{E_2}, [\tau_1, \tau_3]$)
11.	M_4	E_2	REFERRAL(P_4, A_{P_4})
12.	E_2	P_4	FLOWNEGOTIATE($E_2, E_1, A_{E_2}, [\tau_1, \tau_3]$)
13.	P_4	P_1	FLOWNEGOTIATE($E_2, E_1, A_{M_4}, [\tau_1, \tau_3, \tau_4]$)
14.	P_1	E_1	FLOWNEGOTIATE($E_2, E_1, A_{M_4}, [\tau_1, \tau_3, \tau_4]$)
15.	E_1	P_1	FLOWNEGOTIATE($E_1, E_2, A_{E_1}, [\tau_1, \tau_3, \tau_4]$)
16.	P_1	P_4	FLOWNEGOTIATE($E_1, E_2, A_{M_1}, [\tau_1, \tau_3, \tau_4]$)
17.	P_4	E_2	FLOWNEGOTIATE($E_1, E_2, A_{M_1}, [\tau_1, \tau_3, \tau_4]$)
18.	E_2	M_4	FLOWINIT($A_{E_2}, A_{M_1}, P_{E_2}, [\tau_1, \tau_3, \tau_4]$)
19.	M_4	M_1	FLOWINIT($A_{M_4}, A_{M_1}, P_{M_4}, [\tau_1, \tau_3, \tau_4]$)
20.	M_1	E_1	FLOWINIT($A_{M_4}, A_{E_1}, P_{M_4}, [\tau_1, \tau_3, \tau_4]$)

respectively, while packets from E_2 to A_{M_1} and A_{M_2} are routed through N_4 and N_3.

NUTSS establishes an end-middle-end path as follows (Table 4.1.7). After registration state is created, E_1's FLOWNEGOTIATE is exchanged with E_2 through P_1 and P_3 (say). In the process E_1 learns A_{M_3} and E_2 learns A_{M_1} as the other side's effective address, along with the tokens needed (messages #1–6 in the table, arrow $\vec{A}$ in the figure). E_1's FLOWINIT to E_2 succeeds (#7–9, $\vec{B}$), however, E_2's FLOWINIT, IP routed through M_4, fails due to the lack of the necessary token resulting in a referral to P_4 (#10–11, $\vec{C}$). E_2 resumes name-routed negotiation through P_4, and both endpoints acquire tokens for M_4 (#12–17, $\vec{D}$). E_2 successfully re-attempts the FLOWINIT with the newly acquired tokens (#18–20,

$\vec{E}$). As a side-effect, E_1 learns A_{M_4} as an alternate effective address for E_2 that can be used as a failover (once initialized).

In comparison, existing approaches fail to establish a path. As one might expect, any approach that relies solely on address-routed signaling (e.g. TCP/IP, HIP [81]) simply cannot signal through the facing NATs due to both endpoints having private addresses. Relaying application data through public proxies (e.g. i3 [112]) is suboptimal as public proxies are potential network bottlenecks. Approaches that use name-routed signaling before address-routed signaling (e.g. DONA [64], i3+DoA [127], SIP+STUN [94, 100]) but do not strongly couple the two fail to recover when the name-routed path does not coincide with the address-routed path (i.e. unexpectedly encountering M_4 above).

Note that the default path discovered by NUTSS is asymmetric owing to the underlying asymmetric IP routing. If this asymmetry is undesirable, the P-Box can use explicit M-Box addressing whereby P_3 changes the address advertised in the FLOWNEGOTIATE (#3) from A_{M_1} to A_{M_3} (and stores A_{M_1} in τ_3); E_2 learns A_{M_3} as the effective address for E_1. E_2's FLOWINIT (#10) in this case is addressed to A_{M_3} instead of A_{M_1}. The message is address-routed to M_3, which validates τ_3 and NATs the message to A_{M_1}, which in turn NATs the message to E_1 completing the initialization. The resulting path is symmetric despite the underlying asymmetry.

4.2 Using and Extending NUTSS

This section supplies a number of scenarios that serve to elucidate the operation of NUTSS. Some of these scenarios require minor extensions to the basic

architecture described in the previous section. While we should note that each of these scenarios may be handled by one or another existing technology, taken together they demonstrate the breadth of NUTSS and its ability to serve as the foundation for a wide variety of important Internet features.

4.2.1 Mobility

Mobility in NUTSS follows naturally from the basic connection establishment mechanism. A mobile endpoint registers with the P-Box at the new network address. Once registration state is installed in the intermediate P-Boxes, FLOWNEGOTIATE messages are routed to the new location. An added option to the REGISTER message can be used to explicitly expunge the previous registration state if it is no longer valid. Data transfer for already established flows is suspended while the endpoint is disconnected. Upon rejoining, the endpoint attempts to reinitialize the suspended flow from the new address using the existing tokens; if the initialization succeeds, for instance mobility inside the same domain, data flow can be resumed immediately. Otherwise, data flow is resumed after the name-based path is reestablished, the flow renegotiated, and the address-routed path reinitialized with new tokens.

4.2.2 Legacy NAT Traversal

Endpoints use name-based routing as a generic signaling mechanism to conduct legacy NAT traversal as proposed in [46]. In the presence of legacy M-Boxes without an associated P-Box, endpoints use a configured third party P-

Box service on the Internet. Endpoints advertise their public address and port in FLOWNEGOTIATE messages. To learn their public address and port, endpoints use a public service like STUN [100]. While key architectural components (tokens, referrals etc.) are not used in this particular case, legacy NAT traversal is a killer-app for endpoints; being able to support legacy NAT traversal creates incentives for endpoints to implement NUTSS, thus bootstrapping deployment.

4.2.3 Endpoint-Imposed Middleboxes

The NUTSS architecture as discussed focuses on the ability of the middle to impose middleboxes. Endpoints too, can impose middleboxes on a per-flow basis. We outline one method as follows. The initiating endpoint imposes a middlebox (e.g., anonymizing proxy) by sending the FLOWNEGOTIATE to the middlebox and having it proxy both name-routed and address-routed messages. The endpoint accepting a flow imposes a middlebox (e.g., virus scanner) by returning the address of the M-Box instead of its own address in the FLOWNEGOTIATE; in addition, the endpoint appends a τ_{dst} that contains its own name. The initiator initializes the address-routed path to the intended middlebox. The middlebox recovers the name of the intended destination from τ_{dst}, negotiates the path to that destination and proxies processed data. Endpoints chain multiple middleboxes by adding a τ_i for each link.

4.2.4 Application-Level Anycast

Multiple endpoints REGISTER different addresses for the same name; P-Boxes store all address mappings in their local registration table and choose one to forward name-routed messages to. The choice can be based on local network policy, or policy specified by each endpoint at registration time (e.g. round-robin, primary-backup and so on). This is in contrast to i3 (where the middle cannot specify policy), and Oasis [34] (where the policy specified by the middle can be sidestepped at flow establishment).

While the approach above works for cases where a single FLOWNEGOTIATE can acquire all the tokens needed, a small modification is needed if multiple FLOWNEGOTIATEs are needed to acquire all tokens as there is no guarantee that subsequent messages will be name-routed to the same instance. To rectify this lack of affinity, we add an additional *instance* component to the name making it a 4-tuple. The *instance* field uniquely identifies an endpoint in a group of anycast endpoints. The instance name may be picked by the application to be user-friendly. For instance, an application may detect other instances and use the hostname to differentiate itself, or if appropriate may ask the user to name each instance, e.g. home and work. REGISTER messages contain the *instance* as part of the name. An application can elect to send a FLOWNEGOTIATE to a specific instance (set in E_{dst}), or to any instance (instance name of $*$); P-Boxes use the destination instance name to route to a matching endpoint. An endpoint, however, must always include its own instance name in FLOWNEGOTIATE messages that it generates so the other endpoint can learn its unicast name.

4.2.5 Negotiating Multicast

NUTSS can be used to support several forms of multicast. The basic idea is to use the 4-tuple names that define a group of endpoints defined in the previous section (4.2.4) for multicast instead of anycast by transmitting messages to all members rather than only one. There are several possible variants, depending on how large the multicast group is, and whether IP multicast is available. For instance, for small-scale multicast, endpoints could simply establish point-to-point flows with all other members, and individually replicate packets onto all flows. If IP multicast is available, then similar to SIP, the IP multicast address may be signaled through P-boxes, and members may join that multicast group. Otherwise, the rendezvous point and group name for an application multicast protocol ([52, 121], among many others) may be conveyed through P-boxes, and endpoints can join the appropriate application multicast group. Finally, P-Boxes and M-Boxes can participate in carving out IP multicast islands in overlay-based approaches [138].

4.2.6 Default-Off

NUTSS can be used to implement the default-off paradigm [8], where Internet hosts cannot communicate with other hosts without the recipient first explicitly authorizing the connection, without requiring changes to the public IP routing core; this is accomplished by disallowing name-routed messages between all endpoints by default. Endpoints must explicitly enable specific flows with ALLOW messages. A common concern is how non-Internet-savvy users make use of this paradigm without the default quickly regressing to default-on. We

believe that the application must ultimately involve itself in selecting an appropriate policy. For example, remote-desktop-like applications can elect to be default-off, while BitTorrent-like applications can be default-on. Over time, one could well imagine applications evolving to be slightly more sophisticated. For example, a given BitTorrent endpoint could use the name of a given torrent as its instance name (Section 4.2.4) and indicate to the P-Box to filter on that.

4.2.7 Protocol Negotiation

FLOWNEGOTIATE messages can be used to negotiate the entire protocol stack with involvement from the middle. A general protocol negotiation mechanism would enhance the evolvability of the Internet, and could be used to manage multiple network layers created through virtual routers and switches, for instance as proposed for the GENI infrastructure [40]. In addition to addresses and tokens, endpoints would advertise supported protocol stacks including available transports, network layers, security protocols, tunneling protocols and so on, and how to layer them. For instance, a web-browser may advertise: 1) HTTP-TCP-IPv4, 2) HTTP-TCP-IPsec-IPv6, 3) HTTP-TLS-SCTP-IPv4, etc. P-Boxes remove advertised stacks if the network cannot support it (e.g. #3 if the M-Box does not support SCTP) or if the stack violates network policy (e.g. #1 if policy requires TLS or IPsec).

4.2.8 Optimizations

One of the main concerns with NUTSS is the added latency required for establishing data flows. Here we discuss three optimizations that may alleviate this concern. The first, is to piggyback application data onto signaling messages in order to expedite initial data transfer. With appropriate changes to the networking stack in endhost OS's, this piggybacking could conceivably include protocol handshakes such as HIP, IPsec, TCP, and HTTP, potentially resulting in an overall reduction in data exchange latency as compared with today's protocol operation.

The second optimization is combining FLOWNEGOTIATE and FLOWINIT when the P-Box and M-Box are co-located (likely for small networks) to initialize the data path sooner. The FLOWINIT in such cases may contain incomplete information regarding tokens and the remote address. The P-Box fills in the token and uses it to initialize the M-Box flow state. Note that the embedded FLOWINIT is piggybacked with the FLOWNEGOTIATE along the name-route so the remote address is not needed for address-routing; however, if the remote addresses is needed for per-flow state in the M-Box, the P-Box waits until the FLOWNEGOTIATE in the reverse direction before initializing the M-Box.

A third optimization couples NUTSS with Self-Certifying Identifiers (SC-ID) in the protocol stack, for instance HIP [81], in order to eliminate the need for additional signaling during IP-path changing events like mobility or middlebox failover. The idea is to include the SC-ID in FLOWINIT messages, and to transmit multiple FLOWINIT messages in parallel to M-boxes. In this way, failover M-boxes (for instance) will have pre-authorized the SC-ID, and can forward data packets immediately upon receiving them. Indeed, this approach can be used

Table 4.3: Mapping from socket operations to NUTSS primitives, and NUTSS primitives to SIP messages used.

NUTSS name: (*user*, *domain*, *service*, *instance*)
SIP URI encoding: user@domain;srv=service;uuid=instance

Socket API	NUTSS Primitive	SIP Counterpart
`nbind`	REGISTER	REGISTER
`nsetpolicy`	ALLOW/DISALLOW	re-REGISTER (w/ CPL)
`nconnect`	FLOWNEGOTIATE	INVITE
`naccept`	FLOWNEGOTIATE	200 OK
`nsend/nrecv`	FLOWINIT (one-time)	-
`nclose`	-	BYE

to establish multiple parallel data flows through the network, for instance to increase throughput.

4.3 Implementation

To test the feasibility of NUTSS, we implemented a library that adds NUTSS support to endpoints, and implemented an M-Box used for legacy NAT traversal, which we deployed on Planetlab. While the implementation did not uncover any unexpected issues, it did help us iron out the design. Using off-the-shelf software and existing infrastructure, our implementation enables EME communication (including name-based connection establishment, legacy NAT traversal, mobility, default-off behavior and application-level anycast) in many cases requiring little to no modifications to existing applications.

Our implementation uses SIP [99] for name-routing. While other name-routed signaling protocols (e.g. Jabber/XMPP [102]) may be used, we chose SIP because of its maturity and support in commercial hardware. At the same time, we can assess what aspects of SIP are most important for NUTSS.

P-Boxes (SER) and Access Control (CPL): We chose to base name-routed components on off-the-shelf commercial software in order to facilitate the second phase of deployment (upgrading networks with support for name-routing). P-Boxes in our implementation are (as yet) unmodified SIP Express Router (SER) [27] proxies. Policy definitions (for ALLOW/DISALLOW messages and domain policy) are compiled (manually, at present, using CPLEd [26]) into the Call Processing Language (CPL) [68], a declarative language used for user-specified VoIP policy and call routing.

Name-routed messages in NUTSS are encoded as SIP messages (Table 4.3 lists the mapping). Source and destination endpoint names are encoded in SIP header fields (From:, To:), and advertised addresses and tokens are encoded in the body of the SIP message. The messages are (optionally) signed using S/MIME certificates [91]. Address-routed messages are normal TCP/IP messages; the library inserts the FLOWINIT message into the 5-tuple data flow in front of application data.

M-Boxes: While our implementation supports legacy NATs, we implemented a NUTSS-aware M-Box that performs TURN-like [97] relaying of application data to assist endpoints behind particularly poorly behaved NATs [46] to communicate through them. To allow for an unmodified SER proxy, our M-Box includes a shim P-Box that generates tokens for the M-Box; the SER proxy coupled with our shim in series perform the coupling between name-routing and address-routing. The token itself is a 32-bit nonce, one copy of which is sent to the endpoint and another exchanged in-memory between the shim P-Box and M-Box that is used for validating the impending data path.

Endpoints: Endpoint support is implemented as a userspace library for

Linux and Windows applications. The library consists of roughly 10K lines of C code and relies on a number of external libraries including eXosip2 [5] for SIP support and OpenSSL [85] for data security. Our library has two interfaces. The first interface offers NUTSS equivalents of the socket API including an `AF_NUTSS` socket address family, and a `sockaddr_ns` structure that encodes the user, domain and application, and optionally, the instance name, as strings. In order to use this interface, applications must be modified accordingly; however, as the API is similar to BSD sockets, only minor modifications are needed—mostly confined to populating the address structure with names instead of IP addresses and ports.

The second interface to our library accommodates unmodified existing application binaries. This interface is available only for Linux applications. The library is pre-loaded into memory by the Linux dynamic loader's LD_PRELOAD functionality. This allows our library to transparently hijack libc socket calls in the application and redirect them to NUTSS equivalents. The local endpoint name is configured into environment variables by the user. The user also enters specially encoded hostnames into the legacy application. When the legacy application performs a `gethostbyname` call for the encoded hostname, the call is intercepted by our library, which decodes the NUTSS name and creates a mapping between the identifier and a fake IP address returned to the application. When the application later initiates a `connect` to the fake IP address, the library intercepts and initiates an `nconnect` to the associated name. Calls to other legacy BSD socket functions are handled similarly.

In order to encourage adoption, the NUTSS library transparently performs NAT traversal. After exchanging addresses and ports over name-routed

signaling if the direct TCP connections (in both directions), and TCP hole-punching [46] fail, endpoints negotiate the use of a public relay (the TURN-like M-Box described earlier). M-Boxes are deployed on Planetlab hosts. The associated P-Box can be contacted through sip.nutss.net, which routes to the shim P-Box in a randomly selected M-Box. Endpoints acquire a token and transport address for the M-Box. Both endpoints connect to the M-Box and initialize the flow by sending a FLOWINIT message over the connection; the M-Box verifies the token and uses it to pair up the two connections.

As the M-Boxes in our Planetlab deployment do not lie on the IP data path between endpoints, we have not gathered sufficient experience with the referral mechanism.

We have successfully run a number of applications using both the legacy and non-legacy interfaces to our library while transparently incorporating endpoint policy, authentication, legacy NAT traversal and endpoint mobility. Our library works with client-server applications written to our API, as well as with many unmodified legacy applications (e.g. iperf, VNC, GNOME GUI desktop). The library is available for public download at nutss.net.

4.3.1 Findings

SIP Lessons Learned: We were surprised to find that although SIP was originally conceived for a very different purpose, it can be used to implement name-routing in NUTSS (with one minor modification to SIP). Admittedly SIP is rather heavy weight for the purpose and we would prefer to use a leaner protocol. Nevertheless, given that SIP is deployed widely today and enjoys sig-

nificant mindshare, there is a compelling case to be made for using a subset of it.

One aspect of SIP that requires special workarounds in our implementation is the lack of support for nested domains. A single REGISTER message only creates a single registration at the local P-Box and not the chain of registrations in the P-Boxes in front of the endpoint as required by NUTSS. While this limitation is not a concern in the first phase of deployment where a public P-Box is used, in the second phase it affects networks not connected directly to the core. A temporary brute-force workaround is for endpoints to explicitly create registrations for each link of the chain; however, this is not always possible due to firewall policy. A more permanent solution is to modify SIP with support for nested domains, and accordingly modify our SER proxy to forward the registrations to the parent P-Box.

Latency: Since ours is a proof-of-concept implementation of the NUTSS architecture, performance numbers are of little relevance as they relate only to our (perhaps simple) access control policy. Nevertheless, some brief comments on performance are worth making. We found that there is little added latency in establishing connections (less than 15ms) with P-Boxes deployed on the same network segment as the endpoints. This is because signaling in our particular setting added one name-routed round-trip (FLOWNEGOTIATE) and one address-routed round-trip (FLOWINIT). Quite predictably, when two nearby Planetlab nodes on the west coast use our public P-Box service deployed at Cornell, the connection establishment latency shoots up to 100–200ms due to name-routed messages having to make four coast-to-coast trips before data can flow. The optimization suggested in Section 4.2.8 where data is piggybacked in signal-

ing messages should allow initial data bytes to be exchanged in half that time while the address-routed path is established in the background. Our P-Box (SER proxy) can route approximately 1200 name-routed messages per second on contemporary hardware (~1050 with challenge-response authentication enabled). A single such P-Box can handle moderate sized sites, but load-balancers will be needed for large sites.

In real-world settings, interaction with multi-homing, complex access control policy, mid-flow reconfigurations, and mobility will make signaling more heavy weight.

4.4 Related Work

Several other Internet architectures have been proposed that move away from 5-tuple addressing. TRIAD [42], IPNL [32], HIP [81], SHIM6 [84] and i3 [112] route datagrams based on URLs, FQDNs, host keys, hashes and flat identifiers respectively; these approaches advocate end-only control and require protocol-stack modifications at endhosts and middleboxes. NUTSS advocates control shared by both the end and the middle and uses a separate name-routed signaling phase that is strongly coupled to existing address-routed stacks. GMPLS [71], which doesn't involve endpoints, uses IP for "name"-routed signaling to negotiate the layer-2 path. Selnet [118], Plutarch [21], AVES [82], Metanet [135], SIP [99], UIA [29], and DONA [64] all involve the middle in resolving endpoint names to realm-specific addresses and routes. In order to provide complete end-middle-end connectivity, however, we believe the middle must play a yet larger

role in blocking unwanted flows, and in additionally instantiating the address-routed path.

When it comes to blocking unwanted flows, one weakness of the E2E security model as mentioned is that not everyone who has a stake in security is empowered to provide that security. In the E2E model, the middle has little say in what flows are allowed and must rely completely on the endpoints for the protection of the network itself. In select scenarios, in an enterprise for example, the IT department can enforce this control over the ends through software update and configuration management tools like Marimba [12]. In other cases, such as with DoA [127], endpoints can explicitly invoke security services provided by the middle. Such solutions, however, do not protect against malicious or compromised endpoints that may preempt the IT department's control and abuse the network.

An alternate solution is where the middle exerts direct control on flows with the help of a middlebox on the address-routed path. While middleboxes protect the network against uncooperative endpoints, they face the aforementioned problems which we repeat here: firewalls must infer malice based largely on the 5-tuple and costly deep packet inspection, D-WARD [75] infers malice based on deviations from a "normal traffic model", NATs protect only against drive-by intrusions from the outside, and VPNs cannot authenticate remote endpoints that are not VPN members. Ultimately, such middle-only approaches that cannot explicitly negotiate the intent of the endpoint rely on heuristics, which potentially block non-malicious flows.

When it comes to establishing the address-routed path, protocols such as UPnP [73] and Midcom [111] allow the endpoint to create state in the middle.

A limitation in these approaches, however, is that the middle cannot participate in flow negotiation either to enforce network policy or to indicate whether the address-routed path and protocol stack chosen by the endpoints is even possible. A second limitation is that a thorough read of [73] and [111], unlike this thesis, does not yield any easter eggs, redeemable for dinners, to keep one interested in reading it further. Session Border Controllers [50] combine name-routing and address-routing in one box, but do so without endpoint knowledge or consent creating authentication and authorization hurdles.

4.5 Summary

In this chapter, we propose NUTSS, a name-based "end-middle-end" approach to establishing network flows through middleboxes like NATs and firewalls that takes into account policies of the ends and the middle. NUTSS is unique in that it couples both offpath and onpath signaling in order to overcome the weaknesses of either. NUTSS has an incentivized incremental deployment path, and enables a number of important features, including mobility, multi-homing, NAT traversal, negotiation of different network layers, multicast, and anycast.

Although this chapter shows NUTSS to be a promising approach, the devil is in the details. Our partial proof-of-concept implementation notwithstanding, the most important next steps are to gain experience with NUTSS and to do more security analysis. Towards this end, we hope that the NAT traversal features of our implementation may serve as a "killer app" to drive deployment and experimentation.

CHAPTER 5

IMPACT

STUNT, ShutUp, and NUTSS, presented in the preceding three chapters, are pragmatic, albeit research systems designed to solve real-world problems. Perhaps the hardest problem in systems research is actually making the jump from a research system to a real-life deployment. In the real-world, the solution must coexist with wide-range of other systems that may compete with, complement, or even impede the proposed solution. The scenarios in which the system will be deployed may well be countless and unimaginable. And above all, the solution must be aligned with the goals of vendors most likely to adopt it, taking into account legacy considerations, and future prospects. It is no surprise then that few research systems manage to transition into the real-world.

There are two bodies — the Internet Engineering Task Force (IETF) and the Internet Research Task Force (IRTF) — that serve as vehicles for impact in the Internet. The IETF is a technical forum for primarily networking equipment vendors, application developers, and researchers concerned with standardizing (or making changes to) Internet protocols. The IRTF, with a membership that significantly overlaps with that of the IETF, takes a more research-oriented long-term view of changes to the Internet. Both forums are valuable resources for researchers wishing to transfer their research into the real-world.

In this chapter we chronicle our efforts within the IETF and the IRTF, and present our experiences. We were exceptionally successful in transferring the lessons learned from designing STUNT into the design of future NATs, and in standardizing the STUNT technique itself. In the case of NUTSS, however, we were unable to solicit the participation needed from a wider community of

contributors to create a formal specification, implementation, and testbeds with which to gather experience with real applications.

5.1 BEHAVE-TCP: Standardizing NAT TCP Behavior

In Chapter 2 we found that NATs in the wild exhibit a wide range of behaviors that ultimately complicate the process of TCP traversal. The lack of NAT behavior standards is partly to blame for this. Vendors independently decided on their NAT design based on their respective understanding of the security implications, application transparency, and implementation effort. Taking into account these constraints, we found that while there is no one single NAT design that addresses all vendor concerns, just two designs (one geared towards security and the other towards transparency) are sufficient for reaching a practical compromise between vendors and application designers.

Working with vendors and application developers in IETF's BEHAVE working group, we developed a set of ten design guidelines for NAT vendors [45]. In it we note that Endpoint-Independent Mapping behavior greatly simplifies NAT traversal by removing the need for port-prediction. Second, we note that the security properties of the NAT depend on the Filtering behavior, rather than the Mapping behavior that most vendors had previously incorrectly assumed. Further, we provide guidelines that make NATs less fragile by mandating minimum durations for various inactivity timeouts, and describe the proper handling of tricky situations, such as hairpinning, that vendors have foundered on in the past.

Our biggest learning experience was the art of reaching a consensus. Many

of our original proposals, drafted from the researcher perspective, were rejected outright by vendors for non-technical reasons; in one case, the objection was that the vendor would be unable to competitively market a product that could be *perceived* as being less secure. The approach to reaching consensus that served us well was selecting a second proposal that, while non-optimal from the research perspective, still had significant merits, and was more appealing to parties that had objections to the first proposal. Picking a suitable alternate proposal was in itself a research exercise; it would need to satisfy our original goals (i.e., simplifying NAT traversal) while interoperating with the first proposal and meeting vendor constraints. This significantly lengthened the consensus process. Ultimately, however, it was time well spent as we were able to avoid alienating parties who would otherwise have balked at the document.

An example of such a compromise is the proposed handling of unsolicited inbound SYN packets by the NAT. To facilitate NAT traversal, application developers would prefer NATs to silently drop the packet[1]. NAT vendors would also prefer to stealthily drop packets so as to avoid being detected by address scans. However, silently dropping packets goes counter to the requirements of other IETF efforts, because it complicates problem diagnosis. Joe Touch proposed the alternate solution of NATs delaying the error response (by several seconds) instead of silently dropping the packet, which was an acceptable solution for application developers and other IETF groups, but not for all vendors. Consensus was reached by recommending vendors to delay errors, while allowing vendors to silently drop packets at their discretion. Since both approaches simplify NAT traversal for application developers, we met our goal without alienating any vendors or other IETF groups.

[1]See [45] for details.

In the year since the document has been published there have been reports of several vendors that have declared compliance with the requirements of the document. Additionally, the document has been adopted for standardizing TCP and SCTP behavior of IPv6 firewalls [134].

5.2 ICE-TCP: TCP NAT Traversal

Standardizing the STUNT TCP NAT traversal technique was simpler for two reasons. First, it is a new protocol that is not constrained by backwards compatibility. And second, being similar to STUN UDP traversal, it can reuse much of the protocols developed for it with minor modifications. Discovering the TCP Mapping for STUNT required updating the original STUN protocol [100] to operate over TCP [98]. Coordination between the endpoint is performed over the ICE [95] signaling protocol originally designed for STUN. To this end, ICE-TCP [96] extends the ICE protocol for use with STUNT. ICE and ICE-TCP remain drafts as of this writing because the attention of the BEHAVE working group has been refocused on the more urgent issue of standardizing IPv4-IPv6 NAT behavior.

5.3 EMERG: End-Middle-End Research Group

Our work on NUTSS led to the creation of the End-Middle-End Research Group (EMERG) in the IRTF. EMERG was chartered to explore the problem-space presented by today's Internet and come up with a new Internet architecture that incorporates both the end and the middle. In [31] we proposed the design of an

EME signaling protocol. Despite significant interest, however, EMERG lacked the direct contributions and participation of the community needed to arrive at a formal specification, and initial implementation through which to gather actual experience. While the EMERG has been suspended due to lack of participation as of this writing, we are confident that the principles behind EME are sound, and remain hopeful that this area will receive the necessary attention from researchers and vendors in the future.

CHAPTER 6

SUMMARY AND THE ROAD FORWARD

The Internet architecture is showing its age. The research community is presently engaged in a spirited debate about how best to address the shortcomings and prepare for the future. Some favor a clean-slate redesign of the Internet, while others favor a more evolutionary approach. In this thesis we have argued that in the context of Internet connection establishment, evolutionary solutions, which do not require any changes to the currently deployed Internet routing core, are not only possible, they are practical, and from a pragmatic standpoint, perhaps even necessary.

The root cause of connection establishment problems is the implicit assumption that the path between the communicating endpoints is open. Developments such as NATs and firewalls, which necessarily violate this assumption, have rendered applications fragile. Meanwhile networks that have strived to uphold the assumption have done so at the cost of being vulnerable to DoS and worm attacks. We have argued that it is neither reasonable, nor desirable, to guarantee an open path between endpoints. As a result, connection establishment must be rearchitected.

In this thesis we have proposed using explicit signaling between the endpoints and the middle for connection establishment. Our proposal derives from our experience in solving connection establishment through NAT, firewalls, and middleboxes that takes into account policy of all stakeholders in a robust yet incrementally deployable manner. In our experience, signaling also allows architectures for mitigating DoS attacks and worm outbreaks to additionally take advantage of end-to-end deployment options. Through these examples we have

presented an existence proof of the value of signaling in incrementally evolving Internet connection establishment to cater to present day needs.

We empirically determine the features of a signaling primitive suited to the Internet. Explicit signaling is necessary to bridge the wide information gap between the endpoints and the middle created by the original Internet architecture, although ancillary implicit signaling can serve to simplify protocol design. Deployment options are greatly increased by using offpath signaling over a separate control plane to "open" the data plane, however ultimately onpath signaling is needed to control the data plane during the lifetime of the connection. Furthermore, the data plane leverages the existing (essentially unmodified) Internet in both hardware and software. Doing so requires preserving the independence of the data plane, which is accomplished by requiring very weak coupling between it and the control plane.

There is a tremendous amount of work still to be done. Foremost among these is to write a complete specification of the signaling primitive, implement that spec, and start gaining experience with real applications. Much of the process for this must include a wider community of contributors, perhaps through an IETF or IRTF working group. As always, the devil is in the details, and our partial proof-of-concept designs notwithstanding, there are many details to be worked out. Nevertheless, we believe that the basic approach of using explicit signaling and combining offpath and onpath signaling through weak coupling is compelling.

Having said that, our experience with the IETF and IRTF has been mixed. On the one hand we have seen some ideas generate tremendous enthusiasm, vigorous debate, and highly productive community participation. While on the

other hand vested interests, Internet religion, or nonchalance have conspired to derail the process for other ideas. If such impediments make incremental evolution of the Internet hard, one wonders what hope clean-slate solutions have of reversing four decades of momentum. Indeed there is much to be gained by reflecting on why only a small fraction of research in flagship conferences manages to make the transition from research to practice.

The researchers' arbitrary understanding of "incrementally deployable" or "practical" deserves some blame in the matter. Throughout this thesis we have used these terms without ever precisely defining them. An ex post facto definition would be "partial gain for partial deployment", or "scalable, robust, cheap, secure, powerful, ..." — all important technical challenges. But ultimately, technology is only half the equation; the other half is business incentives. The disconnect between research and practice lies all too often in not even acknowledging, far less addressing, this issue of business incentives.

6.1 The *Real* Challenge in Evolving the Internet

Internet technology is implemented by entrepreneurs, start-ups, and established companies that are all driven by the bottom line. There is little, if any, involvement of the government or philanthropic organizations that have more altruistic goals. Creating impact then requires solutions with a business model for deployment. In the ideal case, a practical solution will naturally lead to tangible profits. In other cases, however, despite technical practicality, profits may be contingent on some amount of customer altruism. The real challenge in evolving the Internet is to incorporate this insight as part of the original problem statement.

Keeping business interests at the back of one's mind during the research stage would subtly bias design decisions in favor of creating impact. Who would build the product? How much would R&D cost? Would they be able to market the product? How many customers would they get, and how much would the customers be willing to pay? Would the customers buy the product because it lowers their operating costs, or would their operating costs increase? Could the customer opt for a cheaper solution that works "well-enough"? Is there a business case to be made for the proposed solution? Clearly these questions cannot be answered with any degree of accuracy during the research phase. However, they force a more nuanced look at the question of deployment incentives.

To summarize, the Internet has long passed its tipping point — billions of dollars are at stake for companies maintaining and extending it, as well as companies conducting business over it. A purely technological perspective on evolving the Internet is not sufficient. As researchers, we must rise to the challenge of tackling not only the technological issues, but doing so in a way that is aligned with the business interests of those involved. A promising research methodology for the future is to design abstractions, primitives, architectures, and systems, that are explicitly aligned with business interests. The end product of research conducted thus would stand the best chance to truly evolve the Internet.

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