HIGH-PERFORMANCE SERVICE-ORIENTED COMPUTING

A DISSERTATION
SUBMITTED TO THE DEPARTMENT OF ELECTRICAL ENGINEERING
AND THE COMMITTEE ON GRADUATE STUDIES
OF STANFORD UNIVERSITY
IN PARTIAL FULFILLMENT OF THE REQUIREMENTS
FOR THE DEGREE OF
DOCTOR OF PHILOSOPHY

Nicholas McDonald
June 2016

This dissertation is online at: http://purl.stanford.edu/gb403yz8060

I certify that I have read this dissertation and that, in my opinion, it is fully adequate in scope and quality as a dissertation for the degree of Doctor of Philosophy.

Bill Dally, Primary Adviser

I certify that I have read this dissertation and that, in my opinion, it is fully adequate in scope and quality as a dissertation for the degree of Doctor of Philosophy.

John Ousterhout

I certify that I have read this dissertation and that, in my opinion, it is fully adequate in scope and quality as a dissertation for the degree of Doctor of Philosophy.

Al Davis

Approved for the Stanford University Committee on Graduate Studies.

Patricia J. Gumport, Vice Provost for Graduate Education

This signature page was generated electronically upon submission of this dissertation in electronic format. An original signed hard copy of the signature page is on file in University Archives.

Abstract

This dissertation presents Sikker, a highly-scalable high-performance distributed system architecture for secure service-oriented computing. Sikker includes a novel service-oriented application model upon which security and isolation policies are derived and enforced. The workhorse of Sikker is a custom network interface controller, called the Network Management Unit (NMU), that enforces Sikker's security and isolation policies while providing high-performance network access.

Sikker's application model satisfies the complex interactions of modern large-scale distributed applications. Experimental results show that even when implemented on very large clusters, the NMU adds a negligible message latency of 41 ns under realistic workloads and 91 ns at the 99.99^{th} percentile of worst-case access patterns. Analysis shows that the NMU can support many hundreds of Gbps of bandwidth with common VLSI technologies while imposing zero overhead on the CPU.

Integrated into Sikker and the NMU is a novel service-oriented, distributed rate-control algorithm, called Sender-Enforced Token and Rate Exchange (SE-TRE), that is able to regulate service-to-service aggregate rates while imposing zero latency overhead at the 99.99^{th} percentile, less than 0.3% bandwidth overhead, and zero overhead on the CPU.

Sikker's service-oriented security and isolation methodology removes high overheads imposed by current systems. Sikker allows distributed applications to operate in an environment with fine-grained security and isolation while experiencing supercomputer-like network performance.

Acknowledgements

I would like to specially thank two advisors in my academic career that have had enormous impact on my success. My bachelor's and master's advisor, Professor Al Davis, provided me with much inspiration and numerous opportunities. Without his positive encouragement I would have settled on a less exciting and unrewarding career path that would have not taken me to Stanford to pursue a PhD. My doctoral advisor, Professor William Dally, taught me the art of academic research and impressed on me the desire to approach fundamental limitations to technology. Throughout my time at Stanford I benefited from his numerous insights and his ability to instantaneously filter out my bad ideas. Professors Christos Kozyrakis, Mendel Rosenblum, and John Ousterhout have all guided my research in significant ways and helped guide me towards my goals. I would like to thank all the members of the CVA research group for being great friends and colleagues.

I would like thank my mother, Tricia McDonald, for her endless support of me and tirelessly putting up with me for many decades. She continuously showed patience and understanding for my childhood desire to destruct all the electronic devices in our home with many failed attempts to piece them back together. The amount of sacrifice my mother has made for me is endless.

Last but certainly not least, I would like to thank my amazing wife, Kara McDonald, who has supported me through thick and thin. Kara has been my driving force of confidence even when I myself did not believe I could achieve my objectives. In addition to her continuous love and patience, Kara brought our two beautiful daughters into this world. She is an extraordinary wife, mother, and friend. I could not and would not have done this without her. Just as my friend Song Han once pointed out, "I am a happy man!"

Contents

List of Tables

List of Figures

Chapter 1

Introduction

The number and variety of applications and services running in data centers, cloud computing facilities, and supercomputers has driven the need for a secure computing platform with an intricate network isolation and security policy. Traditionally, supercomputers focused on performance at the expense of internal network security while data centers and cloud computing facilities focused on cost efficiency, flexibility, and TCP/IP compatibility all at the expense of performance. In spite of their historical differences, the requirements of these computing domains are beginning to converge. With increased application complexity, data centers and cloud computing facilities require higher network bandwidth and predictably low latency. As supercomputers become more cost sensitive and are simultaneously utilized by many clients, they require a higher level of application isolation and security. The advent of cloud based supercomputing [5, 6] brings these domains even closer by merging them onto the same network.

Operating within a single administrative domain allows distributed systems to consider the network a trusted entity and safely rely on the features it provides. Supercomputers use this ideology to achieve ultimate performance, however, they maintain minimal security and isolation mechanisms to achieve their performance goals. In contrast, cloud computing facilities achieve high levels of security and isolation at the expense of much lower performance. In theory, a single administrative domain *can* provide simultaneous performance, security, and isolation as these are not fundamentally in opposition. The unfortunate truth is that modern network technologies have not provided distributed systems that are capable

of supercomputer like network performance while simultaneously providing robust application security and isolation. As a result, system designers and application developers are forced into making trade offs between performance and security. This leaves deficiencies in the system which makes application development harder and yields performance limiting overheads.

This dissertation presents a new distributed system architecture called Sikker, that includes an explicit security and isolation policy. The goal of this system is to provide the highest level of network performance while enforcing the highest level of application security and isolation required by the complex interactions of modern large scale applications. Sikker formally defines a distributed application as a collection of distributed services with well defined interaction policies. Sikker utilizes specially architected network interface controllers (NICs), called Network Management Units (NMUs), to enforce application security and isolation policies while providing efficient network access.

1.1 Service-Oriented Computing

The size of modern distributed applications spans from a few processes to potentially millions of processes. Specifically, web based applications have ubiquitously adopted a service-oriented architecture (SOA) in which the many processes of an application are grouped by similarity into collections called *services*. A service is a collection of processes developed and executed for the purpose of implementing a subset of an application's functionality. Applications can be comprised of one or more services, often tens or hundreds, and services are often shared between many applications.

Figure 1.1 shows a simplified diagram of six services interacting to fulfill the functionality of a user facing blogging application with viewing and editing capabilities. Each service has a defined application programming interface (API) that it exposes to provide functionality to other services. Even though a modern data center might contain thousands of services, each service generally communicates with a small subset of the total services in order to fulfill its designed functionality. Furthermore, it is common for a service to use only a portion of another service's API.

The Organization for the Advancement of Structured Information Standards (OASIS)

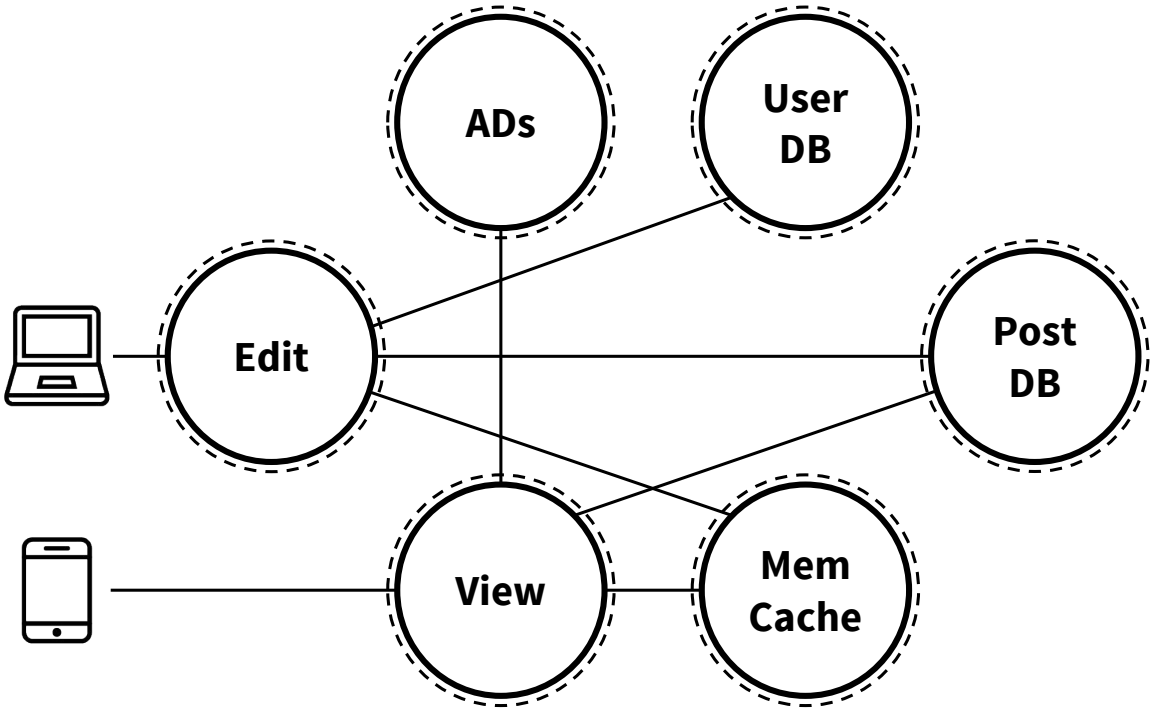


Figure 1.1: High-level service-oriented connectivity

[7], a nonprofit consortium that drives the development, convergence, and adoption of open standards, has published a formal definition for a "Service-Oriented Architecture" as follows [8]:

> Service-Oriented Architecture is a paradigm for organizing and utilizing distributed capabilities that may be under the control of different ownership domains. It provides a uniform means to offer, discover, interact with and use capabilities to produce desired effects consistent with measurable preconditions and expectations.

The ubiquity of SOAs is evidence of their numerous advantages relative to other distributed programming paradigms. By nature of the architecture, core logic elements of an application are separated and are managed separately both during development and operation. SOAs enable easy integration of services developed by third parties because each service can be viewed as a black box and is often distributed along with client libraries that developers can integrate into their own code base. These libraries enable a high level

of abstraction, support system specific optimization, and provide a means to support many intercommunicating programming languages. Although not required by definition, SOAs often make use of client-server protocols which are widely known to reduce the complexity of distributed systems [9]. These protocols are structured as request-response interactions and can be implemented synchronously or asynchronously.

In contrast to SOAs, the distributed shared memory (DSM) [10] paradigm keeps a coherent model of the memory address space across all machines. While this approach can provide high performance data access on small systems, the additional complexity (i.e., distributed memory coherence) implemented by the system severely limits system scalability. Paradigms such as partitioned global address space (PGAS), relax the coherency constraint of DSM to exploit locality and increase system scalability. When faults occur in shared memory paradigms like DSM and PGAS, they often break the entire system either at the hardware level or application level. For this reason, many scientific computing workloads use check-pointing where they revert to known good states upon detection of faults. In contrast, SOAs attempt to isolate faults within a service. In this regime, services are responsible for handling faults within themselves and provide a durable interface to their clients. Instead of creating complex systems with higher cost and lower fault frequency, modern data centers using SOAs expect frequent faults and have built their systems to handle these faults gracefully.

Figure 1.2 is a diagram created by Twitter [1] showing only a small portion of their architecture to illustrate the operation of their protocol agnostic communication system called Finagle. This diagram shows a few services interacting to create the *Twitter* application. Similarly, Figure 1.3 is a diagram created by Netflix [2] illustrating their architecture on Amazon's cloud computing platform. For both of these designs, there exist several services custom written for the application, as well as several services written by third parties.

Figure 1.4 shows a service interaction diagram (occasionally referred to as "the wheel of doom") created by Hailo [3] listing interactions of the 146 services that comprise their application. This diagram highlights one such service which utilizes 12 other services to fulfill its function. Figures 1.2, 1.3, and 1.4 all show that when designing an application at a high level, application developers divide the application's functionality into services with well-defined APIs to achieve modularity.

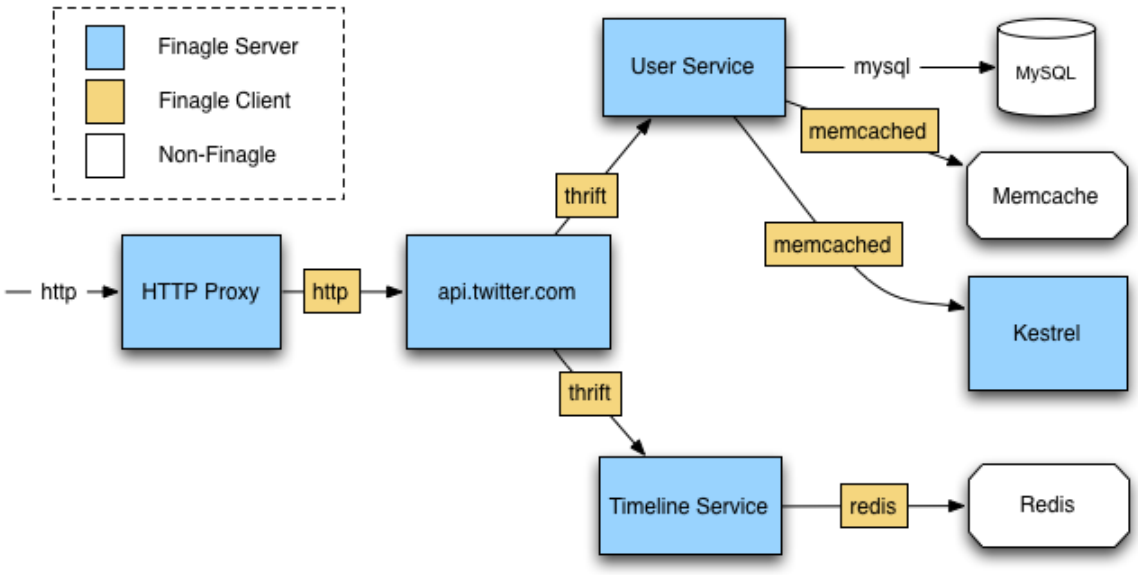


Figure 1.2: Twitter's Finagle RPC system [1].

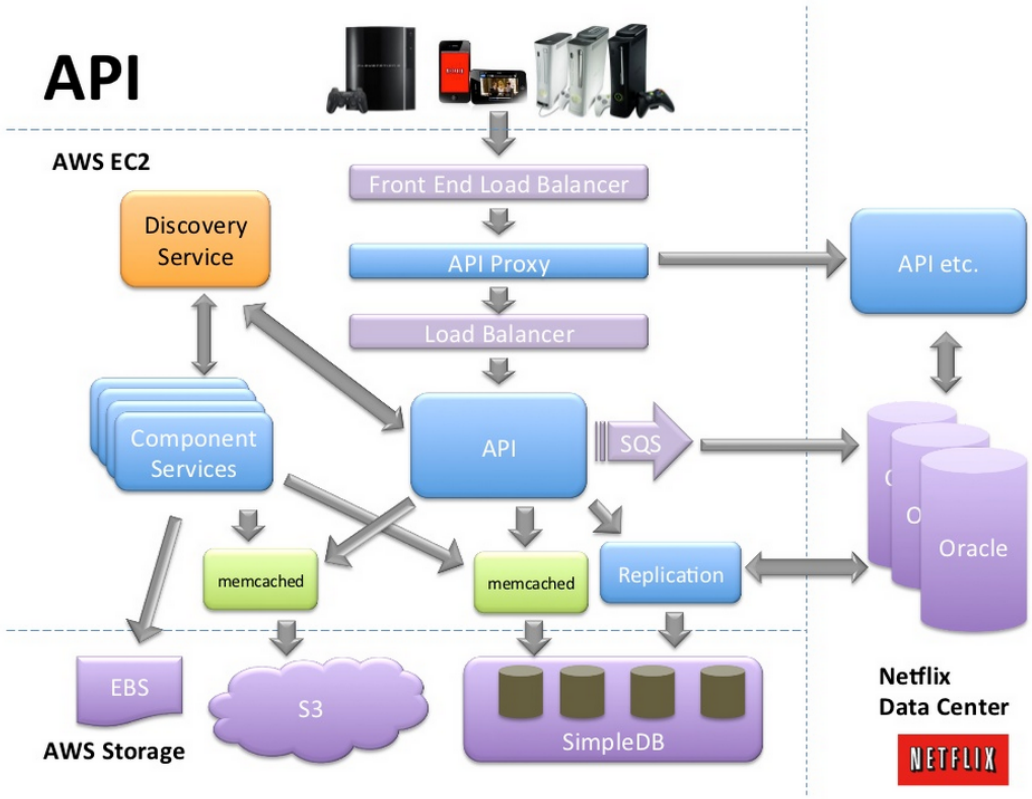


Figure 1.3: Netflix's architecture on AWS [2].

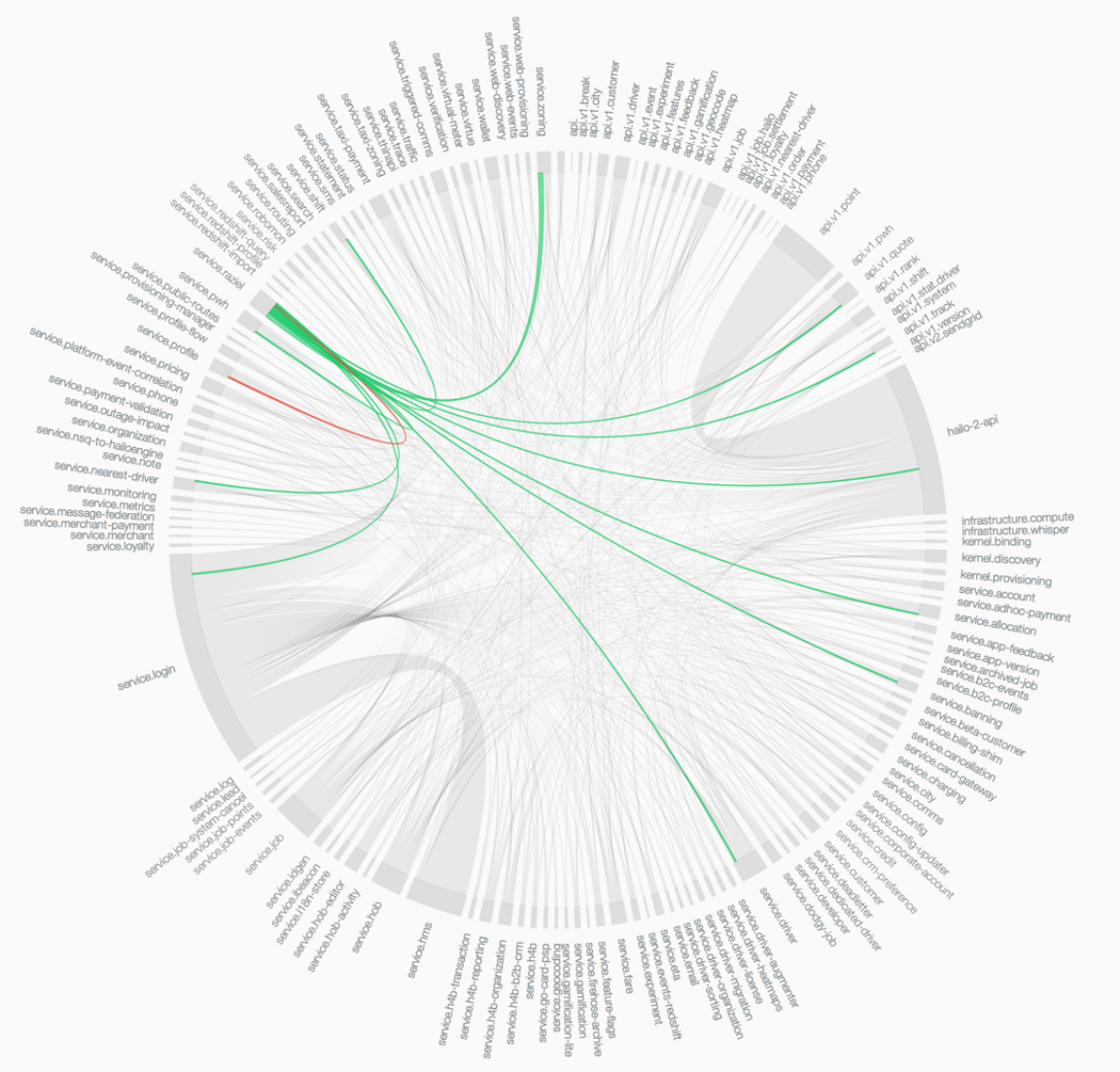


Figure 1.4: Hailo's service interactions [3].

1.2 High-Performance Interconnects

The highest level of network performance available today is found in supercomputing interconnection networks such as Cray Cascade [11] and Gemini [12], IBM Blue Gene/Q [13] and PERCS [14], and Mellanox InfiniBand [15]. These interconnects achieve high bandwidth and predictably low latency while incurring minimal CPU overhead. For example, InfiniBand networks manufactured by Mellanox Technologies achieve round-trip times on the order of 2 µs and bandwidths as high as 100 Gbps [15]. The Cray Cascade system scales the 92,544 nodes, achieves unidirectional latencies as low as 500 ns, and provides 93.6 Gbps of global bisection bandwidth per node [11]. In order to achieve the goal of high network performance, this dissertation defines metrics of performance relative to the highest performing interconnection networks.

One of the major strategies that supercomputers use to achieve high performance is allowing applications to bypass the operating system and interact with the network interface directly. This is called *OS-bypass*. All major high-performance computing fabrics have taken this approach, such as those offered by Cray, IBM, Mellanox, Myricom, Quadrics, etc. On top of better network performance, OS-bypass provides lower CPU overhead as the kernel is freed of the task of managing network interface sharing. CPU overhead can be further reduced by offloading network transport protocols to the network interface.

OS-bypass has one major ramification that limits its ability to be useful in traditional schemes for implementing security and isolation. Bypassing the kernel (or hypervisor) removes its ability to monitor, modify, rate limit, or block outgoing network traffic in an effort to provide sender-side security and isolation features. This is commonly performed in network virtualization software like VMware NSX [16] and Open vSwitch [17]. The work of this dissertation embraces this ramification of OS-bypass and utilizes it as an advantage.

Achieving high network performance, both in terms of high bandwidth and predictably low latency, is crucial to the advancement of future computing technologies. Speaking about the topic in their article titled “It’s Time for Low Latency”, Rumble et al. [18] say:

> Lower latency will simplify application development, increase web application scalability, and enable new kinds of data-intensive applications that are not possible today.

High-performance interconnection networks that enable high bandwidth and predictably low latency already exist, however, the technologies found within these systems are not adequately usable for the computing models ubiquitously used in modern data centers and cloud computing environments. Until now, it has been assumed that there exists a fundamental trade-off between security and performance. This dissertation dispels this belief and proposes a new distributed system architecture that provides both security and performance with zero overhead on the CPU.

1.3 Contributions

This dissertation makes the following contributions:

- A service-oriented application model, called Sikker, is proposed that fits with modern large-scale applications and increases the scalability of access control lists (ACLs) by many orders of magnitude. This is the first work to present a service-oriented network architecture that yields process-oriented authentication.
- A new network interface architecture, called the Network Management Unit (NMU), is proposed that enforces Sikker security policies. The NMU increases message latency by approximately 50 ns, is able to handle hundreds of Gbps of bandwidth, and imposes zero overhead on the CPU.
- A new distributed rate-control algorithm, called Sender-Enforced Token and Rate Exchange (SE-TRE), is proposed that provides strict service-oriented rate control. SE-TRE has zero latency overhead at the 99.99^{th} percentile, less than 0.3% bandwidth overhead, and as an algorithm that can be implemented in the NMU imposes zero overhead on the CPU.

1.4 Dissertation Outline

The remainder of this dissertation is as follows. Chapter 2 provides some background into what the current state-of-practice and state-of-research are including their focus and directions. Chapter 3 describes the motivation behind this dissertation and explains the areas

where prior work is deficient and/or non-existent. Chapter 4 defines and describes Sikker as a new distributed system architecture for efficient service-oriented computing. As an abstract system architecture, this Chapter describes Sikker's functionality and model, not its implementation. Chapter 5 presents potential rate-control algorithms for implementation within Sikker. Chapter 6 describes the architecture of a novel network interface controller, called the Network Management Unit (NMU), that is a Sikker enforcement agent as it upholds the policies laid out by Sikker. Chapter 7 provides an evaluation of the access-control efficiency of Sikker and the NMU. Similarly, Chapter 8 provides an evaluation of the rate-control efficiency of Sikker and the NMU. Chapter 9 describes several optimizations and improvements that can be implemented under the service-oriented programming model of Sikker. Finally, Chapter 10 concludes this dissertation.

Chapter 2

Related Work

2.1 Supercomputing

For the sake of performance, modern supercomputers employ minimal security and isolation mechanisms. For isolation, some fabrics use coarse-grained network partitioning schemes that are efficient at completely isolating applications from each other but they don't provide a mechanism for controlled interaction between applications. This is especially problematic if the system offers shared services, such as a distributed file system (e.g., Lustre [19]).

Some high-performance interconnects, namely InfiniBand, employ mechanisms for secret key verification where the receiving network interface is able to drop packets that do not present the proper access key that corresponds to the requested resource [20]. While this scheme provides a mechanism for coarse-grained security, it does not provide network isolation nor does it provide fine-grained security to cover the application's security requirements. As a result, the endpoints are susceptible to malicious and accidental denial-of-service attacks and they still have to implement the required fine-grained security checks in software.

Current research in the space of supercomputer multi-tenancy focuses on resource utilization and fairness and makes little effort to provide security and isolation in the face of malicious behavior. These proposals [21, 22, 23, 24, 25], while succeeding in their defined goals, do not provide secure supercomputing systems in the presence of multi-tenancy.

Furthermore, none of these proposals provide a scalable architecture on which large-scale service-oriented applications can be built. Even with the advancements of these proposals, supercomputers are still only useful for environments where security and isolation is not a requirement which would imply implicit trust between all users.

2.2 Cloud Computing

In contrast to supercomputers, cloud computing facilities (e.g., Amazon Web Services [26], Microsoft Azure [27], Google Cloud Platform [28], Heroku [29], Joyent [30]) are faced with the most malicious of tenants. These facilities run applications from many thousands of customers simultaneously, some as small as one virtual machine and others large enough to utilize thousands of servers. These facilities must provide the highest level of security and isolation in order to protect their clients from each other. Furthermore, these facilities often have large sharable services that get used by their tenants for storage, caching, messaging, load balancing, etc. These services also need to be protected from client abuse.

Network isolation mechanisms found in modern cloud computing facilities are network partitioning schemes both physical (e.g., VLAN [31]) and virtual (e.g., VXLAN [32], NVGRE [33]). These partitioning schemes are successful at completely isolating applications from each other, but just like the partitioning schemes found in supercomputers, they don't provide a mechanism for controlled interaction between partitions. In efforts to bridge partitions, network virtualization software like OpenStack Neutron [34] and VMware NSX [16] create virtualized switches (e.g., Open vSwitch [17]) and routers that use network-oriented primitives as a mechanism for access control.

Current research in the space of cloud computing multi-tenancy uses hypervisor-based pre-network processing to implement various types of security and isolation. While these proposals [35, 36, 37, 38, 39, 40] achieve their desired goals of providing fair network resource sharing, they significantly increase message latency and CPU utilization and *still* don't provide fine-grained security and isolation. These proposals are often developed and tested on network bandwidths an order of magnitude lower than the bandwidths achieved on supercomputers (10 Gbps vs 100 Gbps) and may not be feasible at supercomputer bandwidths.

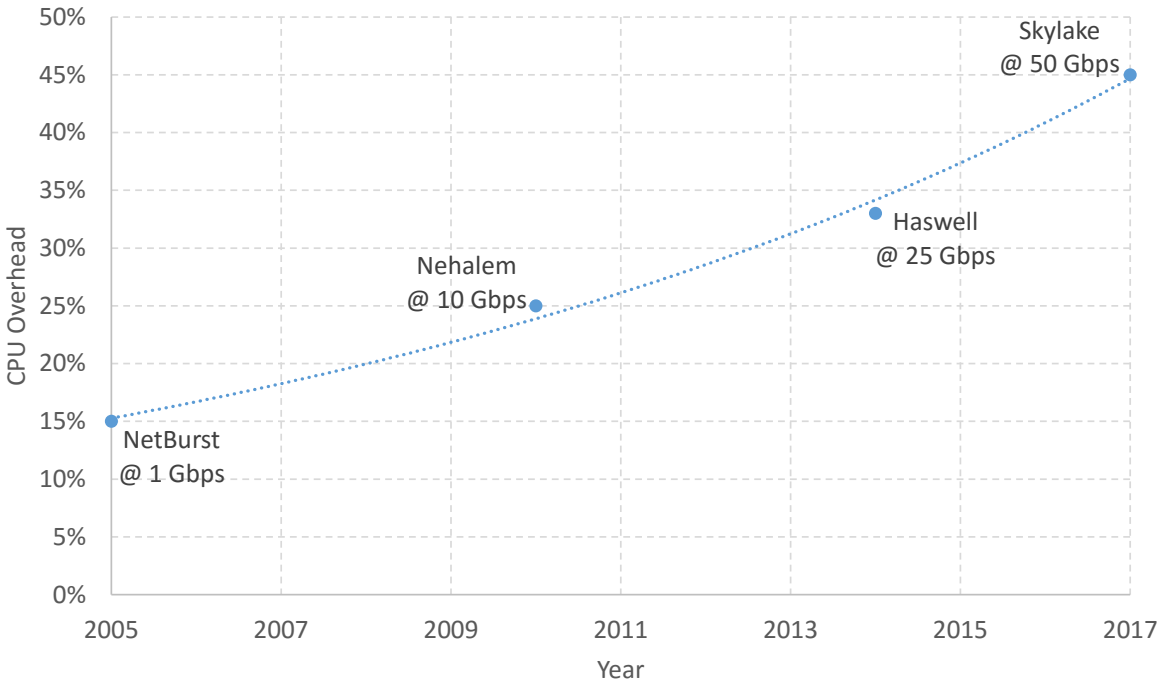


Figure 2.1: Network-related CPU overhead in cloud computing [4].

The combination of higher bandwidth requirements in the data center and the plateau of CPU performance is causing the cost of virtual switching to outrun the capabilities of the CPUs on which it executes. A recent study [4] (shown in Figure 2.1) shows that in 2005 a Xeon-class server with 1 Gbps Ethernet dedicated about 15% of its cycles to networking overhead. By 2010, with Nehalem Xeons, 10 Gbps Ethernet, and the move to virtual switching the overhead rose to 25%. According to the study, the overhead of Haswell Xeons matched with 25 Gbps is 33% and the overhead of future Skylake Xeons matched with 50 Gbps will be 45%. In terms of cost efficiency, this exponential overhead growth is clearly unacceptable.

It is well known that cloud computing environments impose high network overheads and unpredictable performance on their clients [41, 42]. While it is not the claim of this dissertation that all of these poor results are related to security and isolation, it is evident that modern network virtualization and hypervisor-based security cause significant overheads. A recent study [43] shows that two virtual machines communicating on the same

host should expect 25-75 µs of round-trip latency. Similarly, a virtual machine communicating with a native operating system connected to the same 10 Gbps physical switch should expect 35-75 µs of round-trip latency. The latency is significantly worse if the communication is forced to tunnel through an intermediate host containing a virtual router in order to cross the boundary between virtualized network partitions, as is done in OpenStack Neutron [34].

Chapter 3

Motivation

In order to overcome the deficiencies of modern systems this dissertation calls for the design of a system that is highly optimized for large-scale service-oriented computing architectures. This is divided into two main topics: access control and rate control.

3.1 Access Control

An examination of the source code of a particular service reveals the implicit interaction privileges it desires with other services. In most cases, the code expressing the desired interactions does not contain IP addresses or TCP port numbers, but instead contains service names, process identifiers, permission domains, and/or API commands. For example, from the Twitter architecture, shown in Figure 1.2, the code might reveal the *Timeline Service* desiring to communicate with the *Redis* service using its process #6 and using API command *Get*. Services written at Google use the "Borg name service" (BNS) that translates high-level service-oriented identifiers into IP address and TCP port pairs to be used in their ubiquitous RPC system [44].

The implicit service-level privileges expressed in the source code present the ideal level at which permissions should be enforced as these privileges are derived from the applications themselves and represent the actual intent of the interacting services on the network.

Protocol	**Source**		**Destination**	
	Address	*Port*	*Address*	*Port*
TCP	192.168.1.3	54321	10.0.2.10	123
TCP	192.168.1.3	43215	10.0.2.10	456
TCP	192.168.1.3	43215	10.0.3.10	456

Table 3.1: A network-oriented ACL (NACL) entry.

As will be described in this section, the available security and isolation techniques in modern data centers use multiple layers of indirection (e.g., DNS, IP-to-MAC translation) before permissions can be checked and enforced. This creates high operational complexity and presents many opportunities for misconfiguration. Furthermore, these systems lose information about the original intent of the application, and thus cannot enforce the intended permissions. The lack of inherent identity authenticity within the network forces developers to use authentication mechanisms (e.g., cryptographic authentication) that incur high CPU overhead and are unable to properly guard against denial-of-service attacks due to the lack of isolation. This section describes how current systems work and presents a proposal for a better solution.

To moderate network access, modern network isolation mechanisms use access control lists (ACLs). In the abstract form, an ACL is a list of entries each containing identifiers corresponding to a communication mechanism and represent a permissions whitelist. For access to be granted, each communication must match an entry in the ACL. The most common type of ACL entry is derived from TCP/IP network standards. This network-oriented style of ACL will further be referred to as an NACL. Table 3.1 shows an example of NACL entries, commonly represented as a list of *5-tuples*. The first entry states that a packet will be accepted by the network if the protocol is TCP and it is being sent from 192.168.1.3 port 54321 to 10.0.2.10 port 123. The other entries vary the IP addresses and/or the ports. Portions of a NACL can be masked out so that only a portion of the entry must be matched in order for a packet to be accepted by the network.

A comparison between the NACL whitelisting mechanism and the implicit privileges desired by services exposes the deficiencies of using any ACL system based on network-centric identifiers such as protocols, network addresses, or TCP/UDP ports. One important thing to notice is that the source entity is referenced by an IP address and optionally a port.

Source	**Destination**		
Service	*Service*	*Processes*	*Domains*
TimelineService	Redis	6, 17, 32	Get, Set

Table 3.2: A service-oriented ACL (SACL) entry.

For this system to work as desired, the system must know with absolute confidence that the source entity is the only entity with access to that address/port combination and that it is unable to use any other combination. This is hard to ensure because the notion of an IP address is very fluid. While it is commonly tied to one network interface port, modern operating systems allow a single machine to have many network interfaces, network interfaces can have more than one IP address, and/or multiple network interfaces can share one or more IP addresses. There is no definitive way to determine the source entity based solely on the source IP address. Another issue is the use of UDP and TCP ports, which are abstract identifiers shared among all the processes on a given machine. Tying the permissions to ports requires the source and destination to keep numerous open sockets proportional to the number of permission domains required by the application.

ACL whitelisting has the right intent with its approach to security and isolation because of its inherent implementation of the principle of least privilege [45] and its ability to prevent denial-of-service attacks by filtering invalid traffic *before* it enters the network. However, the layers of indirection and loss of original intent imposed by using network-centric ACLs yields security and isolation deficiencies for modern service-oriented applications.

In order to design a better system, this dissertation proposes creating an access control scheme based directly on the implicit privileges desired by each service. The ACL entries of this scheme exactly express the communication interactions of services and their APIs. This service-oriented style of ACL will further be referred to as a SACL. As shown in Table 3.2, the entry is a nested data structure that stores two lists for each source and destination service pair. The first list specifies the set of destination processes within the destination service that the source service is allowed to communicate with. Similarly, the second list specifies the set of destination permission domains within the destination service that the source service has access to. Because the lists are held separately this creates an orthogonality between process access and permission domain access. In this example,

repeated from the Twitter example shown in Figure 1.2, the *TimelineService* is able to access the *Redis* service using process #6 and using the *Get* permission domain, among others. SACLs make reasoning about network permissions much easier and don't tie the permission system to any underlying transport protocol or addressing scheme. It simply enforces permissions in their natural habitat, the application layer.

A tremendous amount of security and isolation benefits are available to the endpoints if the following system-level requirements are upheld for the SACL methodology:

SACL Requirements:

S.1 The network is a trusted entity under a single administrative domain.

S.2 The network is able to derive the identity of a process and it is impossible for a process to falsify its identity.

S.3 The source is able to specify the destination as a service, process, and permission domain.

S.4 Messages sent are only received by the specified destination.

S.5 The source service and process identifiers are sent with each message to the destination.

With these requirements upheld, the system inherently implements source and destination authentication by which all received messages explicitly state the source entity's identification and are only delivered to the specified destination. Combined, source and destination authentication remove the need for complex authentication software in the application layer. Furthermore, senders don't need to use name servers to discover physical addressing for desired destinations as they only need to specify the destination by its virtual identity (i.e., service ID, process ID, and domain ID) and the network will deliver the message to the proper physical location.

3.2 Rate Control

Restricting which entities *have access to* particular resources is an essential component for security and isolation. Another essential component is restricting the *amount of access* an entity has with a particular resource. When applied to service-oriented architectures, this equates to limiting the amount of communication one service has with another. Figure 3.1 shows an example of 6 services that interact to provide the functionality of two user facing applications, similar to Figure 1.1. Also shown in Figure 3.1 is the desire to limit the rate at which the *Editor* service and *Viewer* service are able to utilize the *Post DB* service. In this figure, and in the rest of this dissertation, network bandwidth is the rate-based resource that will be controlled, however, the observations, theories, and practices presented can equally and easily be applied to any other rate-based resource (e.g., requests per second, operations per second, etc.).

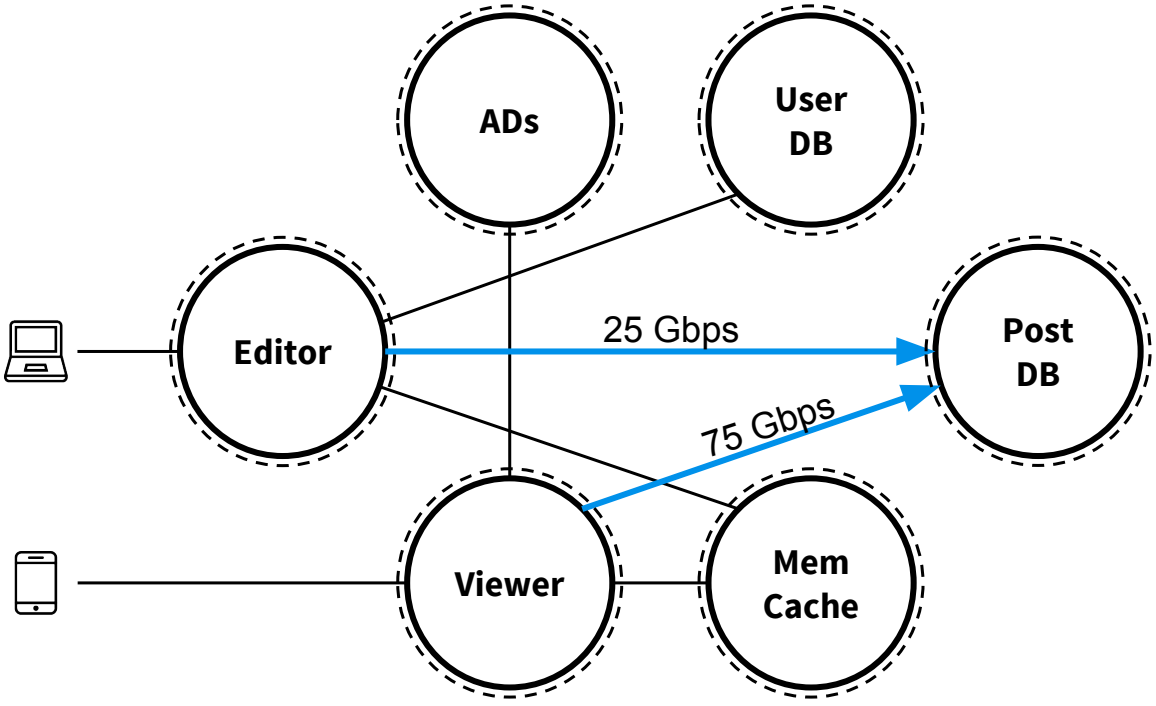


Figure 3.1: High-level service-oriented rate limits

The complex interactions of many distributed systems, predominantly cloud computing, require precise control over the amount of inter-service communication. Service-oriented rate control is a desirable feature for numerous reasons. First, even though services can be designed to satisfy the requests of many clients they have finite capabilities which must not be exceeded. In order to ensure that their services aren't oversubscribed by malicious or faulty clients, service providers can use rate-control mechanisms to limit the amount of communication from each client. Second, because the physical placement of processes of the various client services might be unregulated or unknown, it is undesirable to give priority to those client processes that happen to be closer to the destination. It is desirable to give each service its *fair share* of bandwidth. Third, in cloud computing environments the definition of *fair share* typically comes with a price tag. Cloud computing service providers often price their services in performance brackets and they don't want customers receiving more bandwidth than they paid for. Service-oriented rate control can ensure that clients only receive what they pay for.

Due to the lack of service orientation in today's systems, rate-control mechanisms get applied directly to processes, containers, or virtual machines. The inability to adapt to the demands of a distributed service results in either over-provisioning, which leaves the receiving service at higher risk, or under-provisioning, which leaves the sending service without enough network resources to complete its task. Instead of applying rate limits at the process level, this dissertation makes the proposal to apply rate limits at the service level to fit with the architecture of modern large-scale applications. This work measures the service level rate as the aggregate bandwidth being sent from all processes in the source service to all processes in the destination service. As each process is executing along its own sequence of operations, which might be driven by events external to the service, the rate required by an individual process may vary over time.

Distributed rate control is a difficult problem to solve because both entities (source and destination services) are highly distributed entities potentially consisting of thousands of processes each. Figure 3.2 shows a diagram of two services each containing three processes. As shown, each process within the source service has a unique path through the network for each process in the destination service. The total number of unique paths for a pair of services is equal to the product of the number of processes in each service. Due to

this potentially very large number of unique paths, using a feedback methodology as means to monitor then enforce rate control is infeasible.

Distributed rate control is also difficult to solve because the processes of a service often have a non-uniform usage of another service and programs go through phases where usage of the destination service varies over time. Consider the case where Service #1 is given 30 units of aggregate bandwidth to communicate with Service #2. A naive distributed rate-control algorithm might provision each source process with a fixed and equal allocation of the aggregate rate limit, which would be 10 units of bandwidth per process. This naive algorithm places too many limitations on the types of applications that can run under this scheme as it only allows applications where all processes can complete their tasks under their fixed rate allocation. Assume Process A wants to use 20 units of bandwidth while Processes B and C only want 5. After some short period of time Process A lowers its usage down to 5 units of bandwidth and Process C increases its usage to 20. This sequence is acceptable because the aggregate rate limit is not violated during these periods of time and during the transitions. This rate usage pattern makes any fixed allocation scheme (equal or not) unsatisfactory as it can not adapt to the changing behavior of the program which in turn starves the processes of the bandwidth they require to complete their tasks.

In this dissertation a novel service-oriented rate-control algorithm is proposed, called Sender-Enforced Token and Rate Exchange (SE-TRE), that enforces rate control directly in the network interface of the sending processes. The network interfaces collaboratively exchange control information to dynamically adapt to the continuously changing rate usage of the processes within a service. The class of rate-control algorithms presented in this dissertation only provide the upper limit of the rate used by one distributed entity (e.g., source service) communicating with another distributed entity (e.g., destination service). The aggregate rate limit is only designed to protect the destination entity. It is not the intent to provide minimum rate guarantees (e.g., service level agreements), ensure that the network links can handle the rate limit specified, or provide process-level load balancing. These rate-control algorithms control the high-level interaction between the two entities and assume the physical infrastructure was appropriately designed to provide the necessary link-level bandwidths.

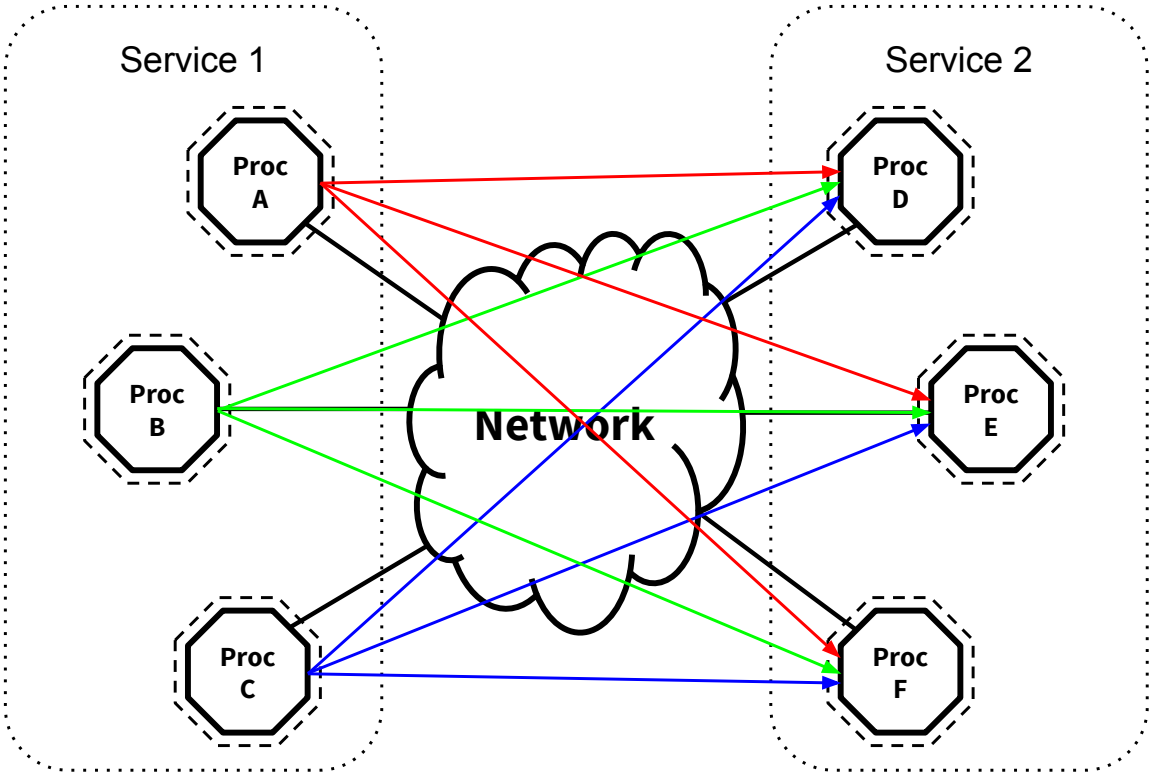


Figure 3.2: Two services each with three processes. Also shown are unique paths that exists from every source process to every destination process.

Chapter 4

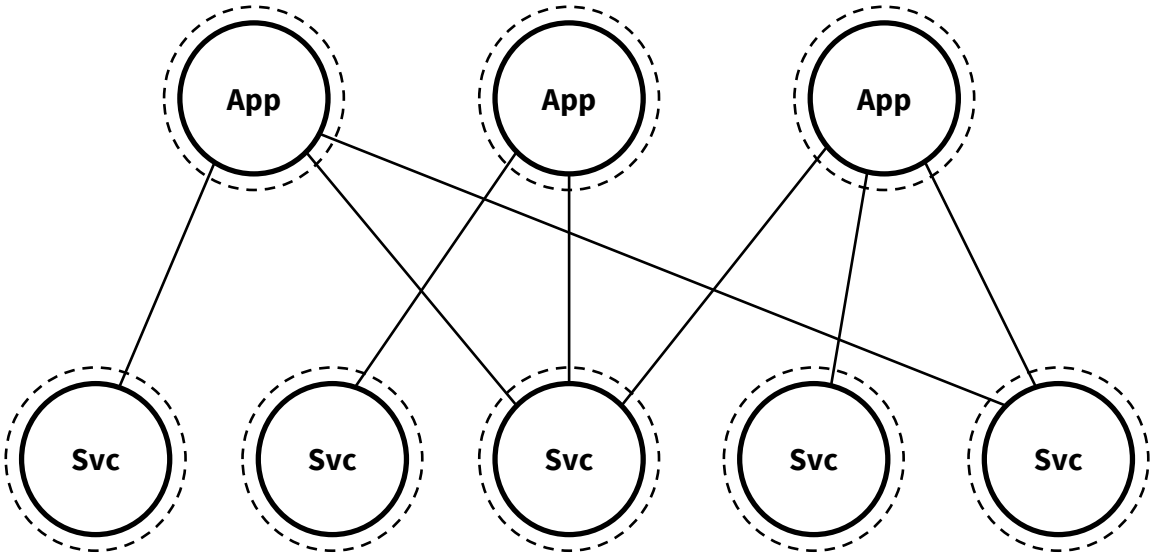
Sikker

With the insights gained in Chapter 3, a new distributed system architecture is proposed, called *Sikker*[1], that formally defines the structure of distributed applications and the interactions within applications. Sikker is strictly a service-oriented architecture and makes no attempt to justify the boundaries of *applications*. As a service-oriented architecture, Sikker designates the *service* as the fundamental building block of distributed applications (shown in Figure 4.1a).

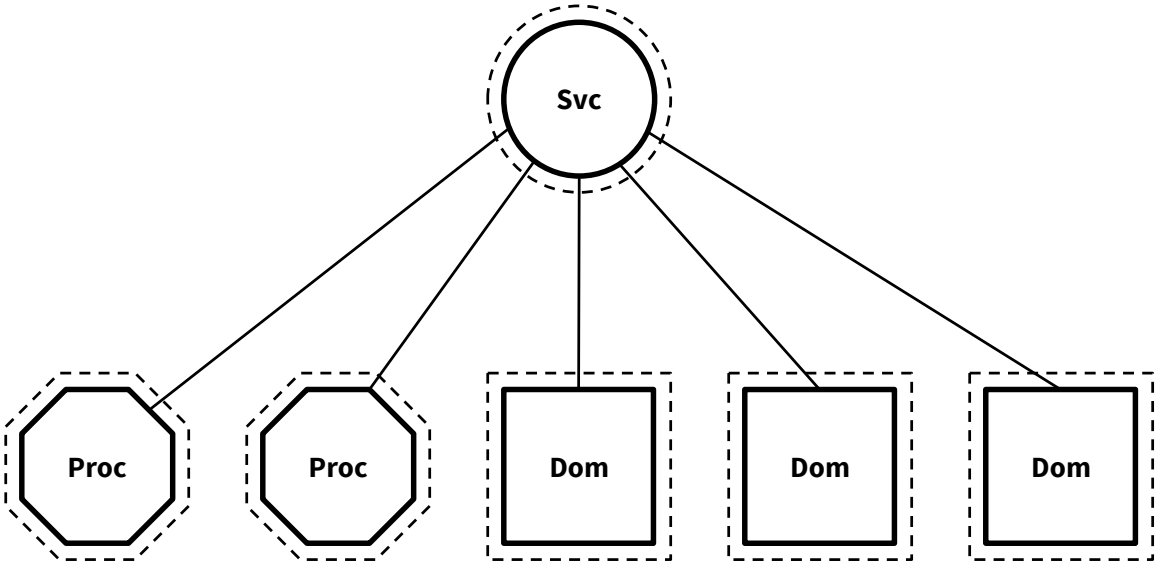
4.1 Application Model

Under the Sikker application model, services are formally composed of a set of *processes* and a set of *domains* (shown in Figure 4.1b). Sikker represents each service with a unique identifier. The processes of a service are the execution units that implement the API of the service. A Sikker "process" can be an OS process, software container, virtual machine, etc. Each process within a service is assigned an identifier unique to the service. The domains of a service are a set of service-specific permission domains that the service uses to define access control regions. Permission domains are useful for defining boundaries around API functionality, data structures, or hybrid combinations. Each domain within a service is assigned an identifier unique to the service. Each service has its own domain number space, thus, two services using the same domain identifier is acceptable.

[1] "sikker" is a Danish word for safe and secure



(a) *Applications* are composed of *Services*. Services can be shared by multiple applications.



(b) Services are composed of *Processes* and *Domains*.

Figure 4.1: *Services* are formally composed of a set of *processes* and a set of *domains*.

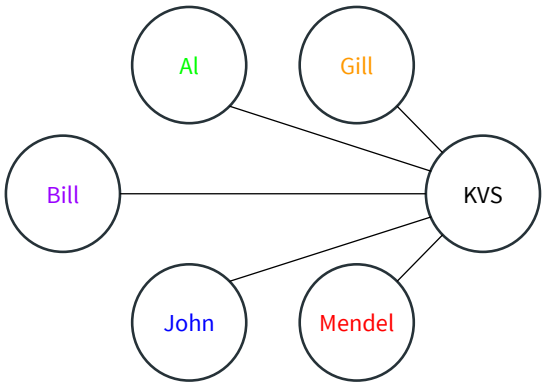


Table	Key	Value
Mammals	pet	dog
Engineering	department	Electrical
Locations	school	California
Companies	internship	Google
Sports	best	wakeboarding
Engineering	tool	oscilloscope
Sports	boring	baseball
Locations	born	Utah
Mammals	fastest	cheetah
Companies	career	HPE

Figure 4.2: An example system where 5 services interact with a tabulated key-value store service. Sikker domains can be defined for this service in a variety of ways.

To understand the usage of Sikker domains, consider a simple key-value storage service that exposes functionality to perform data manipulation in the form of API commands *Get*, *Set*, and *Delete*. Each API command requires the specification of a logical table identifier under which the key-value mapping will be held. Figure 4.2 shows an example system where 5 services desire to utilize the key-value store (KVS) service. This figure shows the internal data structure held within the service which is a single mapping table with logical table divisions. There are many ways in which Sikker domains can be applied to this service. A few options are listed as follows with example domain names:

1. For absolute simplicity, a single domain could be applied to the entire service. Once given access to this domain, a client may issue any API command on any logical table. (Ex: All)

2. One domain could be assigned to each API command. This yields 3 total domains. To issue an API command, a client must have been given access to the corresponding domain. The API command can be used on any logical table. (Ex: Get, Set, etc.)

3. One domain could be assigned to each logical table. This yields 5 total domains. To access a logical table, a client must have been given access to the corresponding domain. Any API command can be used on the logical table. (Ex: Mammals, Engineering, etc.)

4. One domain could be assigned to each API command of every table. This yields 15 domains. To access a logical table using one specific API command, the client must have been given access to the corresponding domain. This domain can only be used for one specific API command on one specific logical table. (Ex: CompaniesGet, LocationsDelete, etc.)

5. The API commands could be put into groups and domains could be applied to the groups and on specific tables. For example, the 3 API commands could be classified as *Read* and *Write* Operations. This yields 10 domains. To access a logical table using one specific API command, the client must have been given access to the corresponding domain where the API command exists within the specific API command group. This domain can only be used for the specific API commands within the group on one specific logical table. (Ex: SportsRead, MammalsWrite, etc.)

These schemes are all acceptable in Sikker but yield different granularities at which access control is defined and enforced. For example, methodology #1 yields no ability to divvy out permissions of the different API commands and logical tables to individual clients. It presents a binary or all-or-nothing access control model. Methodology #4 precisely allows the service to specify which clients have access to specific API commands on specific logical tables. For example, service Bill could be given permission to use the Get and Set commands on logical table Mammals but not the Delete command. Service John could be given permission to use the Get command on all logical tables and blocked from issuing any other command.

4.2 Addressing and Authentication

All communication in Sikker is fully source and destination authenticated. Source authentication means that the receiver knows the identity of the sender. Destination authentication

means the sender is guaranteed that only the specified destination is able to receive the message. Similar to other networks, processes in Sikker reside at physical locations specified by physical addresses. However, in Sikker, processes are referenced by virtual addresses that specify both the service and the process. When a process desires to send a message on the network, it does not specify its own identity as the source. Instead, Sikker derives its identity, consisting of both service and process, and attaches it to the message.

When specifying a destination for a message, the source process specifies the destination by three things: a service, a process within the service, and a domain within the service. Combined, the source and destination specifications are attached to every message transmitted on the network. Sikker guarantees that the message will only be delivered to the specified destination. Receiving processes are able to inspect the source specification in the message to explicitly know the source's identity.

Under the Sikker security model, processes need not be concerned about physical addressing in the network. Processes only use service-oriented virtual network addresses when referencing each other. Sikker performs the virtual-to-physical translations needed for transmission on the network. Name servers are therefore not needed in Sikker. Through Sikker, each process has the high-level identification for the destinations it will be communicating with.

4.3 Fixed Permissions

Each process within a service inherits all the permissions of the service to which it belongs. In order for a process to be able to transmit a message to a specific destination, the service of the sending process must have permission to access the specified process and domain within the specified destination service. Sikker performs permission checks before messages enter the network and for every message. Because the interaction policies of modern large-scale distributed systems are constantly in flux, Sikker allows processes and domains to be added and removed from services dynamically during runtime. When a new process is created, it inherits all the permissions of the service to which it belongs. Any time the permissions of a given service change, the change is reflected in all processes of the service.

Figure 4.3 shows a Sikker system that is performing access control at the injection point

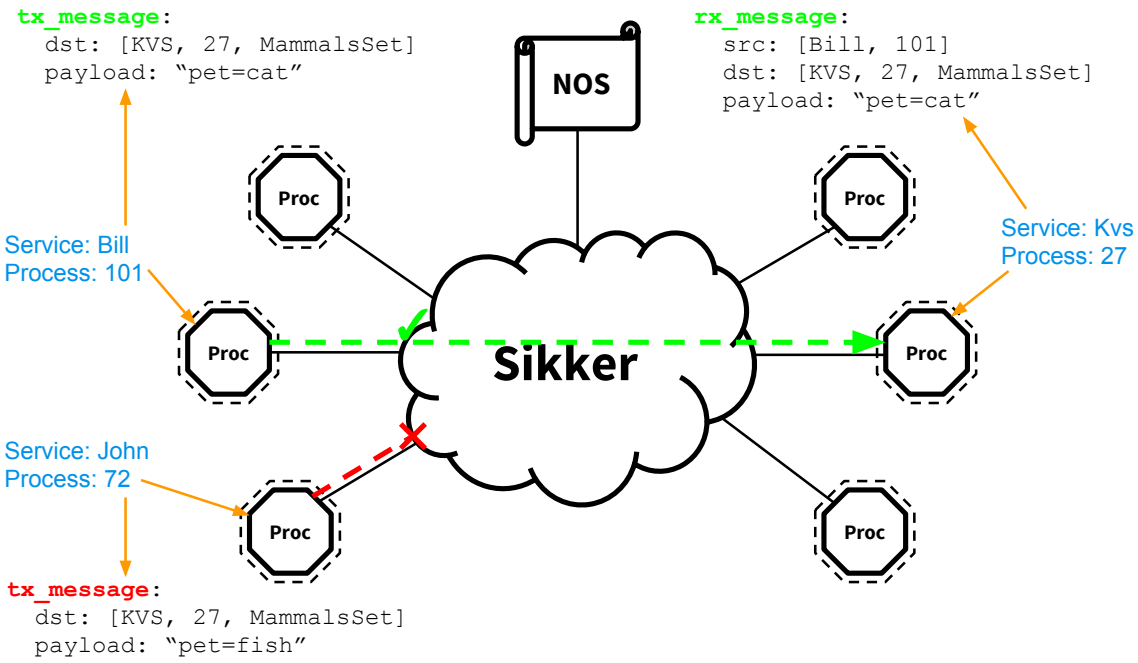


Figure 4.3: A Sikker system performing sender-enforced access control.

of the network. This figure highlights methodology #4 from the key-value store example in Figure 4.2. In this figure, process #101 from the Bill service is granted access to set the key-value mapping for "pet=cat" in the *Mammals* logical table of the KVS service. During this transaction there exists source and destination authentication. The KVS service is guaranteed that the Bill service had the proper permission to perform the transaction. The KVS service did not have to perform any authentication or permissions checking itself. Because of this, the KVS service can process the request immediately without wasting any CPU cycles guarding itself from the network. Also in the figure is process #72 of the John service attempting to perform a similar request. Due to a lack of permissions Sikker stops this request before it enters the network. The KVS service is unaware of and is unaffected by this attempt.

Figure 4.4 is an example service interaction graph under the Sikker application model. This diagram shows three services, each with a few processes and a few domains. Solid lines connect services to their corresponding processes and domains and connects processes to their corresponding hosts. As shown, and widely used in practice, processes from the

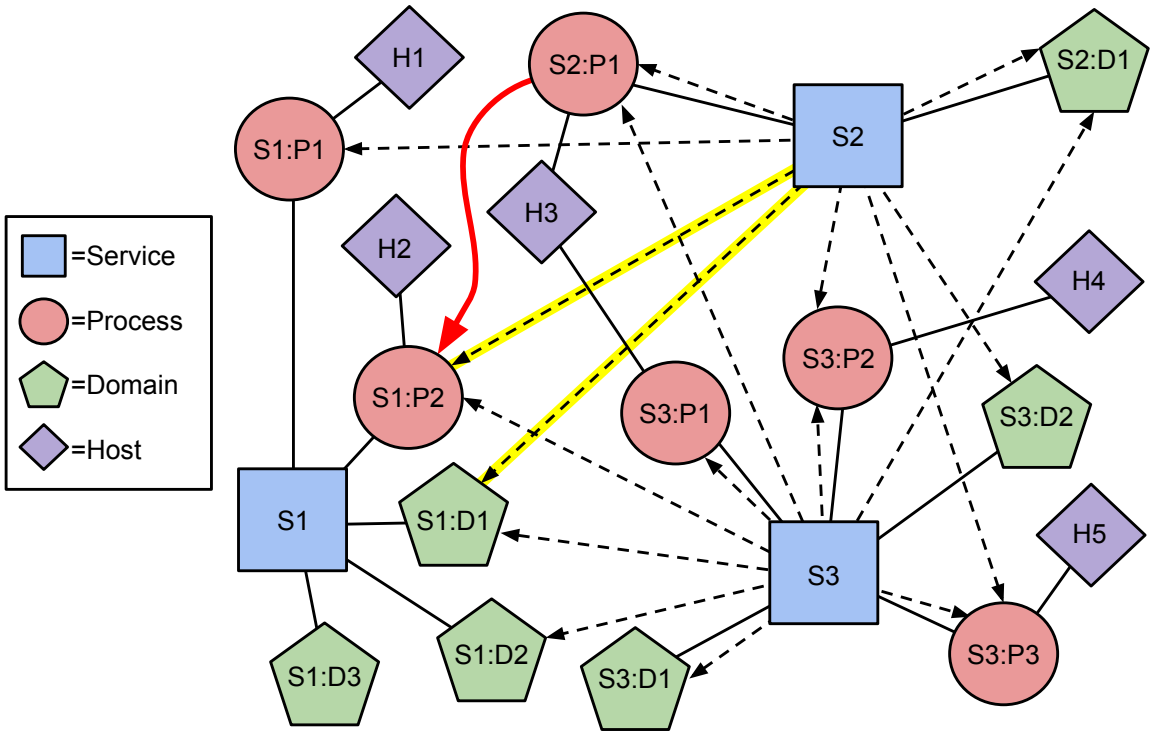


Figure 4.4: An example service interaction graph. Solid edges represent assignment and dashed edges represent permissions. Highlighted is one successful permission set being used.

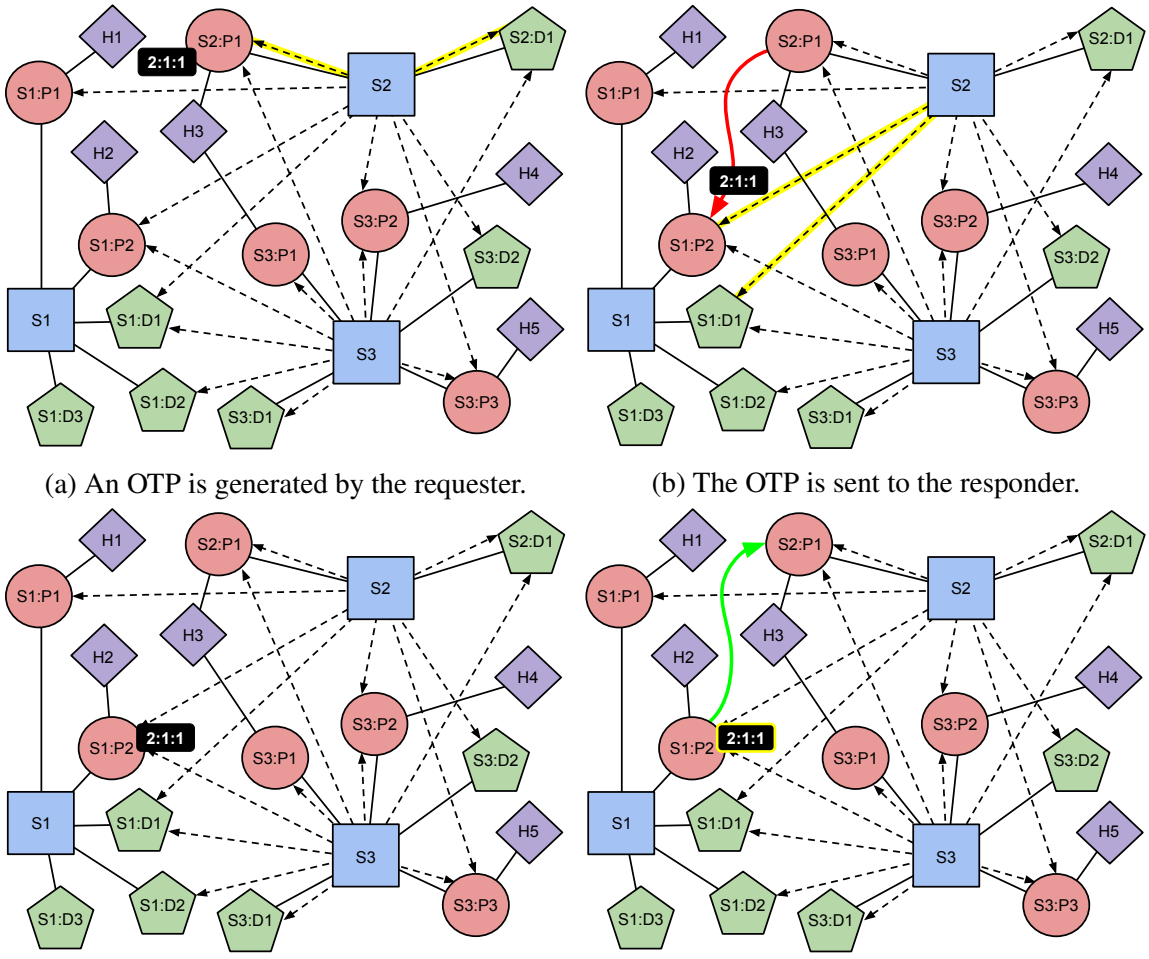
same service and/or different services may overlap on the same host. Dashed lines show the permissions given to services. These lines originate at a service and end at a process or a domain. Highlighted in the diagram is a successful usage of a permission set where Service 2 Process 1 (S2,P1) sends a message to Service 1 Process 2 using the Domain 1 (S1,P2,D1).

4.4 One Time Permissions

The use of request-response protocols is ubiquitous in service-oriented applications. In this environment, many services only become active when they receive requests from other services. This master/slave interaction is achieved via request-response protocols. Cloud computing providers often provide services like this with many features to increase the productivity of their tenants. These services (e.g., Amazon S3[46], Google BigTable[47], Microsoft Azure Search[48]) can be very large and provide functionality to many thousands of clients.

To increase scalability and to fit better with large-scale request-response driven multi-tenant systems, Sikker contains a mechanism for one-time-permissions (OTPs). An OTP is a permission generated by one process and given to another process to be used only once. An OTP specifies a service, process, and domain as a destination and can only be created using the permissions that the creating process already has. When a process receives an OTP from another process, it is stored by Sikker in a temporary storage area until it gets used by the process, at which time Sikker automatically deletes the permission. Because an OTP fully specifies the destination, the process using it specifies the OTP by its unique ID instead of specifying the destination as a service, process, and domain. Only the process that received the OTP can use it. OTPs cannot be shared across the processes in a service.

For an example of using OTPs, consider Service 1 in Figure 4.4 which has no permissions assigned to it, thus, cannot send messages on the network. Assume its API specifies that users of the service must give it an OTP with each request. Now assume that Service 2 Process 1 (S2,P1) wishes to send a request to Service 1 Process 2 Domain 1 (S1,P2,D1). When it formulates its request, it generates an OTP (shown in Figure 4.5a) that specifies itself (S2,P1) with Domain 1 as the recipient (S2,P1,D1). The message will be sent as usual



(a) An OTP is generated by the requester.

(b) The OTP is sent to the responder.

(c) The OTP is received by the responder.

(d) The OTP is used by the responder.

Figure 4.5: The 4 stages of generating, sending, receiving, and using an OTP.

and the OTP will be sent with it (shown in Figure 4.5b). (S1,P2) will receive the OTP with the request (shown in Figure 4.5c) and when the response is ready to be sent, it simply uses the OTP to send it (shown in Figure 4.5d). After the response is sent, Sikker deletes the OTP.

For a real-world example, consider the case of the Hailo application as shown in Figure 1.4. The service with the largest number of service-to-service connections is the "login" service. This service performs the login functionality for users using the Hailo service. Nearly all other services access this service. As such, using fixed permissions would require the service to hold permissions for nearly all other services. As a slave-oriented service,

login functionality becomes active only when another service needs to log a user in or verify that a user has already logged in. Instead of using fixed permissions, this service could just require all client services to provide an OTP along with each login request. In this way, the service doesn't need to hold any fixed permissions.

Another interesting example of using OTPs is allowing one service to act on behalf of another service. This is called a 3-way OTP. Given the same example as before, assume that (S2,P1) wants the response to be sent to (S3,P3,D2) instead of itself. Because it has the proper permissions, it is able to create the OTP with this recipient. The effect is that (S2,P1) sends the request to (S1,P2,D1), then (S1,P2) sends the response to (S3,P3,D2). Continuing from the Hailo example above, 3-way OTPs could be used to reduce messaging latency by service A using a OTP on the "login" service to send a user's status to service B that is expecting the data. In this case, it reduces the number of network transactions from 3 to 2 for service A sending the user's log in information to service B. This increases performance in terms of latency, bandwidth, and CPU utilization.

4.5 Rate Control

Sikker provides a mechanism upon which service-to-service rate limits can be enforced. Rate limits in Sikker specify a source service and a destination service. The enforced rate limit specifies the maximum amount of bandwidth that can be sent by the source service to the destination service. The rate limit is independent of the size of both the source and destination services. When a rate limit is enabled, Sikker guarantees that the specified aggregate rate will not be exceeded. If the source service attempts to exceed its designated rate, further messages will be denied access to the network until the aggregate rate limit has dropped below to specified limit. As each particular process of a service has a varying amount of rate usage to the destination, the rate each process experiences is a dynamic value that can increase or decrease upon demand so long as the aggregate rate limit is not exceeded. When rate limits are not specified, Sikker does not impede traffic being sent on the network and the maximum bandwidth between the pair of services is only limited by the underlying network infrastructure.

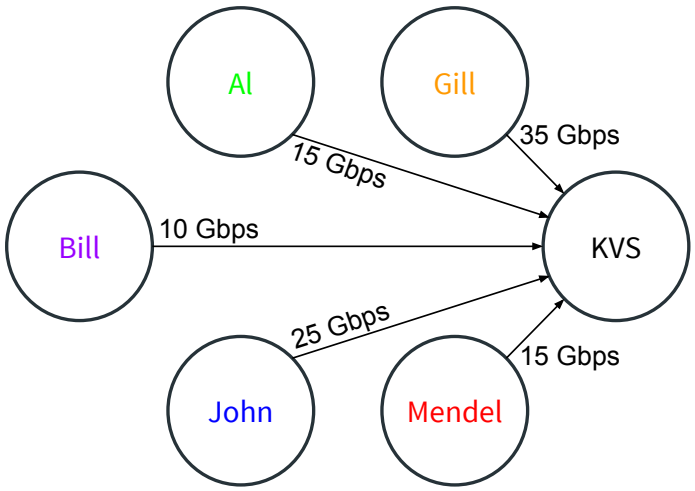


Figure 4.6: Example of assigning unidirectional service-level rate limits.

Figure 4.6 shows an example of assigning rate limits to services. The example is repeated from the key-value store example shown in Figure 4.2. It could be the case that the KVS service only has 100 Gbps of total processing power. Using Sikker's service-oriented rate control it can be guaranteed that this isn't exceeded. Another scenario might be that the KVS service has plenty of processing power enough to cover far more than the sum of all client allocations, however the service desires to limit its clients' access rate based on the amount of money being paid monthly by the clients. Sikker rate limits can be used for this as well. In most cases, it makes sense to apply rate limits only in the request direction for request-response protocols, however, there is no restriction against bidirectional rate limiting.

4.6 Network Operating System

As a system designed for a single administrative domain, Sikker requires the existence of a network operating system or NOS (sometimes referred to as a cluster coordinator, cluster scheduler, or cluster manager) to act as a trusted system-wide governor. The NOS creates the services running on the network, establishes their permissions, and distributes the proper permissions to the proper entities in the system. The NOS is responsible for interacting with the cluster users (i.e., those who run services on the cluster) via a secure

externally accessible user interface. The specific placement, implementation, and fault tolerability of the NOS is beyond the scope of this work as nearly all large-scale cluster coordinators have already solved these issues [44, 49, 50].

When a user desires to start a new service they must first define the service in terms of processes and domains. While the service, its processes, and its domains are all represented with abstract identifiers, process definitions must also specify a program to be run (e.g., the program binary) and how to run it (e.g., program arguments, working directory, environment variables, etc.).

If the service will be providing functionality to other services, the user must create a set of access *portals* for the service. Each portal contains a subset of the total processes and domains within the service that an eligible other service will be able to use. For example, consider the key-value store described in Section 4.1 and Figure 4.2. The user that started the KVS service could have created a password protected portal, called AccessForBill, containing processes 3, 6, and 9 and permission domains CompaniesGet and LocationsDelete (domain methodology #4).

The next step is to determine which of the other services the starting service will require functionality from. The user might explicitly know this information (presumably because they or their colleagues started those services) or they may need to retrieve this information from the NOS. For instance, a user starting several services designed to operate with each other will know the services IDs of all the services that comprise their application. In contrast, a user starting a service in a cloud computing environment will need to get the service identification of a storage service (e.g., Amazon S3[46]) provided by the system operator (e.g., Amazon Web Services[26]).

After having the service identifiers, the user must specify which portals it desires to utilize on the other services and, if required, present valid credentials to use those portals. Continuing the previous example, assume the KVS service defined the AccessForBill portal and was started. Later, the user starting the Bill service identifies that it will require interaction with the KVS service and provides the NOS with AccessForBill and the proper password. When the Bill service is started it will contain fixed permissions to the KVS service using processes 3, 6, and 9 and permission domains CompaniesGet and LocationsDelete. The KVS service ID and the IDs of the processes and domains within the portal

can be directly given to the processes of the Bill service via program arguments.

There are instances where the domains of a service need to be created dynamically as the service runs. For example, a storage service might create a set of domains for each client service utilizing the storage functionality. Alternatively, a storage service might create a new set of domains for a new data set that will be stored within the service. To accomplish this, the NOS allows users to define a set of policies upon which the NOS decides when and how to create or remove domains for their service while it runs. When the domains of a service are modified, the service itself receives a message from the NOS informing it of the change so that it knows how to act accordingly.

4.7 Connectivity Model

For the sake of comparison, a connectivity model for large-scale distributed applications is used to model systems of any size. For a given number of host machines, this model builds a connectivity graph with services, processes, and domains based on configurable parameters. The parameters of this connectivity model are shown in Table 4.1. As an example, consider a system comprised of 131,072 (i.e., 2^{17}) hosts. Under this configuration each host has 16 processes that use the NMU, thus, there are over 2 million processes in the system using Sikker. Since there are 512 processes per service, there are 4,096 total services, each having 256 domains. Each service connects with 819 other services (20% of 4,096) and each service connection is comprised of 333 processes (65% of 512) and 64 domains (25% of 256).

Processes per NMU	16
Processes per service	512
Domains per service	256
Service coverage	20%
Process coverage	65%
Domain coverage	25%

Table 4.1: Connectivity parameters for the service interaction model.

This is a rough estimation of the combination of workloads from data centers, cloud computing, and supercomputing. In cloud computing environments, there are several very

big services but the vast majority of services are small. Small services come from small clients, thus, the inter-process connectivity they require is minimal. The big services that satisfy the requirements of many clients can use the OTP mechanism described in Section 4.4, thus, they will not need fixed permissions for communicating with their clients.

Large singly-operated data centers (e.g., Facebook) more closely approach this connectivity model as they employ many large services. The majority of modern large-scale web services fit within approximately 1,000 processes, however, they only require connection with approximately 10-20 other services.

Supercomputers have very little connectivity between services, however, the services themselves can consume enormous portions of the system. Besides services densely connecting with themselves, scientific supercomputing workloads don't exhibit system-wide dense connectivity.

4.8 Scalability

This section evaluates the scalability of SACLs under the Sikker methodology. In general, the amount of state needed to represent a set of permissions can be expressed as

$$E = A \times R \tag{4.1}$$

where E is the total number of ACL entries, A is the number of agents holding permissions, and R is the number of resources being accessed by each agent. The NACL methodology is compared to the SACL methodology with the following symbols:

s_t : Total services

p_s : Processes per service

d_s : Domains per service

s_a : Services accessed by each service

p_a : Processes per service accessed by each service

d_a : Domains per service accessed by each service

p_h : Processes per host

SACLs have two scalability advantages over NACLs. First, SACLs apply permissions directly to services instead of processes. Second, SACLs provide orthogonality between the access to processes and the access to domains. The amount of ACL entries needed in the NOS is first evaluated. For NACLs the number of permission holding agents is equal to the total number of processes in the system. Because NACLs have no knowledge of services, they assume each process has its own domain set. The resulting expression is:

$$N_{nacl} = \underbrace{s_t \times p_s}_{A} \times \underbrace{s_a \times p_a \times d_a}_{R} \tag{4.2}$$

where N is the number of ACL entries in the NOS. In contrast, the expression for SACLs is:

$$N_{sacl} = \underbrace{s_t}_{A} \times \underbrace{s_a \times (p_a + d_a)}_{R} \tag{4.3}$$

In Figure 4.7 the left Y-axis and the solid lines show a comparison between NACLs and SACLs for the storage requirements of the NOS using the connectivity model from Section 4.7. This shows that SACLs maintain savings of well over 4 orders of magnitude compared to NACLs. For example, if each ACL entry consumes 4 bytes, and the system size is 131,072 hosts, NACLs require 146 TB of storage while SACLs only require 5.33 GB.

The amount of storage needed on each host scales differently than the storage required in the NOS. For both NACLs and SACLs, the number of permission holding agents is the number of resident processes. The resulting expression for NACLs is:

$$H_{nacl} = \underbrace{p_h}_{A} \times \underbrace{s_a \times p_a \times d_a}_{R} \tag{4.4}$$

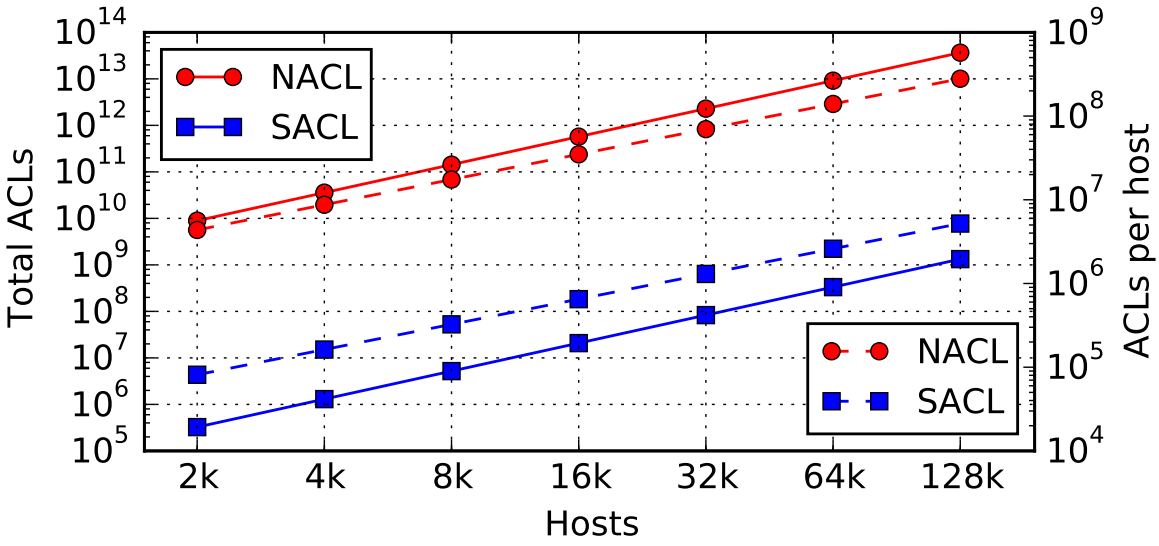


Figure 4.7: Scalability comparison between NACLs and SACLs. The left Y-axis and solid lines show the storage requirements on the NOS. The right Y-axis and dashed lines show the storage requirements at each host.

where H is the number of ACL entries on each host. In contrast, the expression for SACLs is:

$$H_{sacl} = \underbrace{p_h}_{A} \times \underbrace{s_a \times (p_a + d_a)}_{R} \tag{4.5}$$

In Figure 4.7 the right Y-axis and the dashed lines show a comparison between NACLs and SACLs for the storage requirements at each host. This shows that SACLs maintain savings of over 2 orders of magnitude compared to NACLs. For example, if each ACL entry consumes 4 bytes, and the system size is 131,072 hosts, NACLs requires 1.12 GB of storage while SACLs only require 20.8 MB.

4.9 Summary

Sikker's security model is more straight forward than other approaches because the policies on which it is established are derived directly from the applications themselves, instead of being tied to specific network transport mechanisms. Sikker enforces security and isolation

at a much finer granularity than current systems, provides inherent source and destination authentication, implements the principle of least privilege [51], and makes reasoning about service-oriented permissions easier.

Sikker's sender-enforced isolation mechanism removes the ability for denial-of-service attacks between services that don't have permission to communicate. This isolation mechanism creates a productive programming environment for developers since they can assume that all permission checks were performed at the sender. In this environment, developers are able to spend less time protecting their applications from the network and more time developing core application logic.

The scalability benefits of Sikker's service-oriented permission scheme yields significant system wide savings as well as savings at each endpoint. For a NOS holding the permissions of an entire system, the difference between 146 TB and 5.33 GB allows the permissions data to reside in a single DIMM of DRAM rather than multiple racks worth of DRAM. Besides the obvious cost savings, this also presents enormous performance benefits while operating on the data set. Managing a large-scale system with NACLs is rightfully considered a "Big Data" application. In contrast, managing the same system with SACLs can be done on an average laptop.

The amount of storage savings SACLs provide at each endpoint creates a great opportunity for performance gain. While NACLs require 1.12 GB of storage at each point, SACLs require only 20.8 MB which is within the storage capability of on-chip SRAM. Being able to use only on-chip SRAM at each endpoint significantly reduces system cost and increases performance.

Chapter 5

Rate Control Algorithms

This chapter describes the mechanics and operation of six different rate-control schemes which will be evaluated as potential candidate algorithms for use in Sikker. The basis of these algorithms was chosen to explore the space of distributed rate control with respect to high-performance computing. This topic has not been previously covered because, prior to Sikker, environments that support high-performance computing have not had formally defined distributed entities that share permissions. The new computing model has yielded the need for an efficient distributed rate control algorithm to regulate the communication between services.

Rate control is provided using token buckets [52] implemented in the system in various locations depending on the algorithm. In their basic form, token buckets (as shown in Figure 5.1), have a rate R_t at which the *bucket* of size S_b is filled with tokens. Once full, additional tokens are thrown away. When an outgoing packet desires to be sent it must wait until there are enough tokens in the bucket to cover its size. When the packet is sent, the corresponding number of tokens are consumed. It is possible for a token to represent a bit, byte, phit, flit, or packet. This dissertation designates a token to represent one phit, which is the amount of data handled in a single clock cycle [53]. Sikker defines the service-to-service rate as the aggregate rate between all source processes to all destination processes. The rate limit applied to the aggregate usage is denoted as R_a.

The six algorithms presented use variables which control their operation and efficiency. Table 5.1 lists the global variables as well as the variables used in the algorithms. Many of

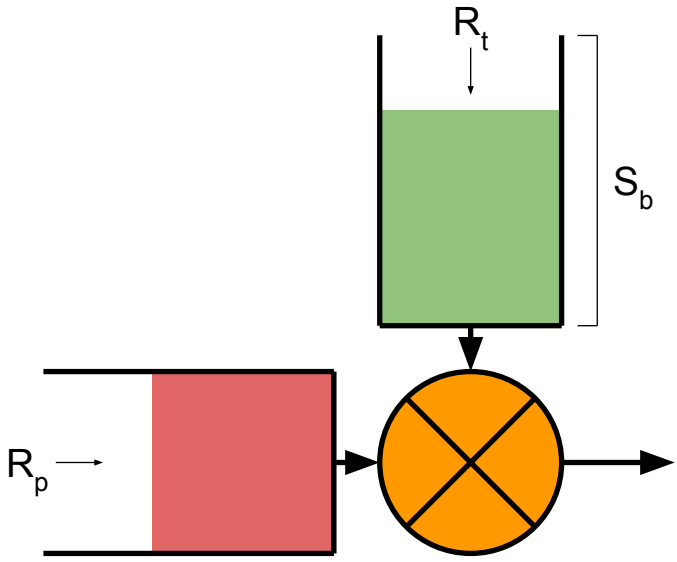


Figure 5.1: A token bucket with size S_b being filled at rate R_t.

the variables are shared across algorithms.

5.1 Nothing Enforced (NE)

The *Nothing Enforced* algorithm (shown in Figure 5.2) is simply to do nothing. It doesn't provide any protection for the destination service, thus it does not produce correct behaviour. It allows all source processes to send to the destination processes at whatever rate they desire. While this algorithm is obviously a bad choice, it does however have zero overhead which makes it a good comparison point.

5.2 Relay Enforced (RE)

The *Relay Enforced* algorithm (shown in Figure 5.3) appoints N_r intermediate entities as relay devices through which all traffic is tunneled flowing from the source service to the destination service. The minimum number of relays is the aggregate rate limit R_a divided

Symbol	Description	Algorithm(s)
S_b	Maximum number of tokens per token bucket	-
R_t	Rate at which a token bucket fills	-
R_a	Maximum aggregate rate limit	-
N_r	Number of relays employed	RE
R_r	Rate capability of an individual sender or relay	RE
N_o	Number of outstanding requests per relay	RE
Th_{low}	Low token bucket threshold	SE-TE, SE-RE, SE-TRE
N_{peers}	Number of peers for parallel requests	SE-TE, SE-RE, SE-TRE
F_t	Token ask factor	SE-TE, SE-TRE
N_t	Current number of tokens in the bucket	SE-TE, SE-RE, SE-TRE
T_{ask}	Number of tokens asked for per peer	SE-TE, SE-TRE
Th_t	Token giveaway threshold	SE-TE, SE-TRE
T_{give}	Number of tokens to give to requesting peer	SE-TE, SE-TRE
F_r	Rate ask factor	SE-RE, SE-TRE
R_{ask}	Amount of rate asked for per peer	SE-RE, SE-TRE
Th_r	Rate giveaway threshold	SE-RE, SE-TRE
F_{mr}	Maximum rate giveaway factor	SE-RE, SE-TRE
R_{give}	Amount of rate to give to requesting peer	SE-RE, SE-TRE

Table 5.1: Rate-control variables

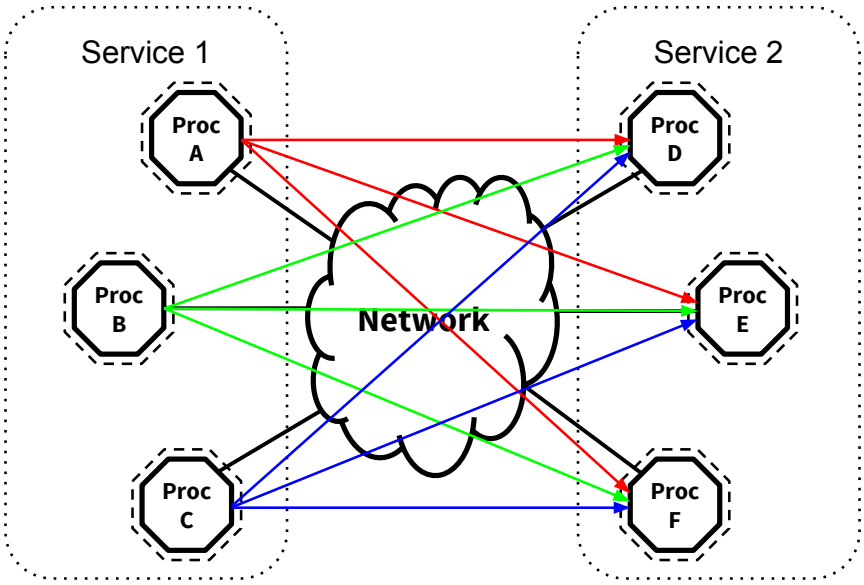


Figure 5.2: Nothing Enforced (NE) rate-control algorithm.

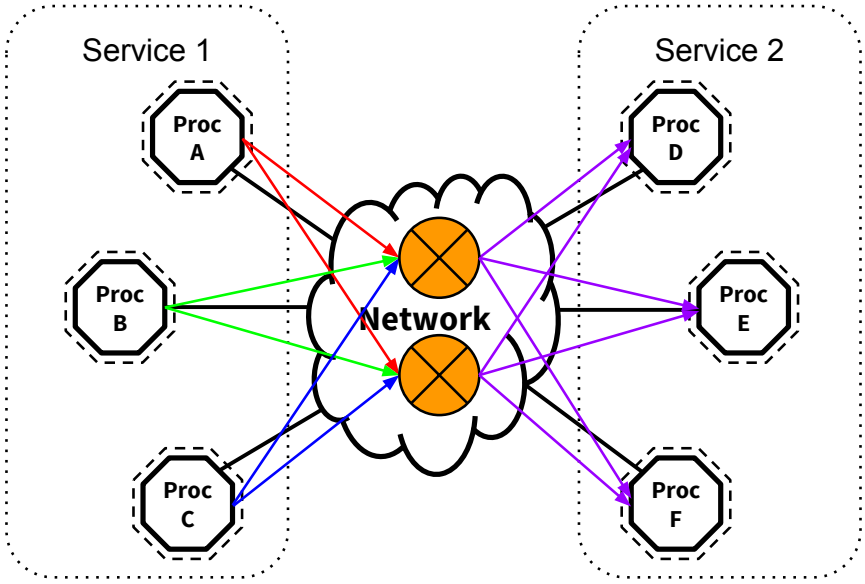


Figure 5.3: Relay Enforced (RE) rate-control algorithm.

by the rate capability of a single relay R_r, rounded up.

$$N_r = ceiling(R_a/R_r) \tag{5.1}$$

For example, if the rate limit is set to 80 Gbps and each relay has a 25 Gbps capability, a minimum of 4 relays is required. There is no logical maximum number of relays that can be employed, although physical limitations may exist.

When sending messages to the destination service, the relay that tunnels the traffic is chosen with a uniform random distribution. Each relay is configured to rate limit the traffic it receives at a fixed and equal allocation:

$$R_t = R_a/N_r \tag{5.2}$$

Each relay device uses a token bucket configured at the fixed rate R_t. The random selection between source and relay provides correct behavior even when the various source processes are sending at different rates. From the example above, the rate of each of the 4 relays would be configured to limit the traffic to 20 Gbps. If not already handled in

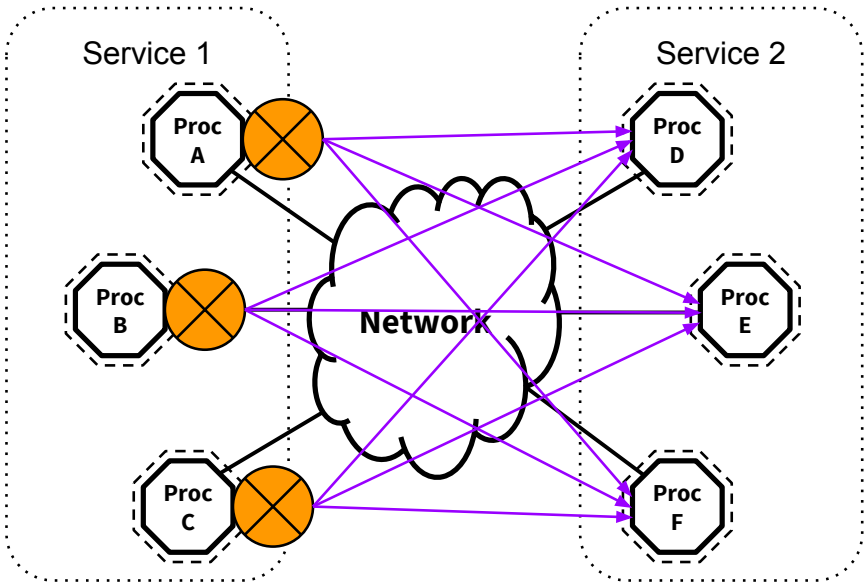


Figure 5.4: Sender-Enforced - Fixed Allocation (SE-FA) rate-control algorithm.

the networking infrastructure, a simple credit-based flow-control [53] scheme can be employed from source processes to relay devices to prevent the relays' queues from getting overwhelmed. The credit scheme limits the number of outstanding messages from each sender to N_o.

This algorithm provides correct behaviour as it is impossible for the senders to communicate at a rate higher than the configured aggregate rate limit. However, this algorithm induces a bandwidth overhead of at least 100% as each packet is relayed through an intermediate device requiring two network transactions. Queuing latencies will also exist in the relay devices when multiple senders randomly choose the same relay device. Statistically, this affects the average latency very little, however the tail latency suffers severely.

5.3 Sender-Enforced - Fixed Allocation (SE-FA)

For all *Sender-Enforced* algorithms one token bucket is co-located with each process in the source service.

Using a *Fixed Allocation* (shown in Figure 5.4) gives each token bucket a fixed portion of the aggregate rate and the sum of all token bucket rates is equal to the aggregate rate

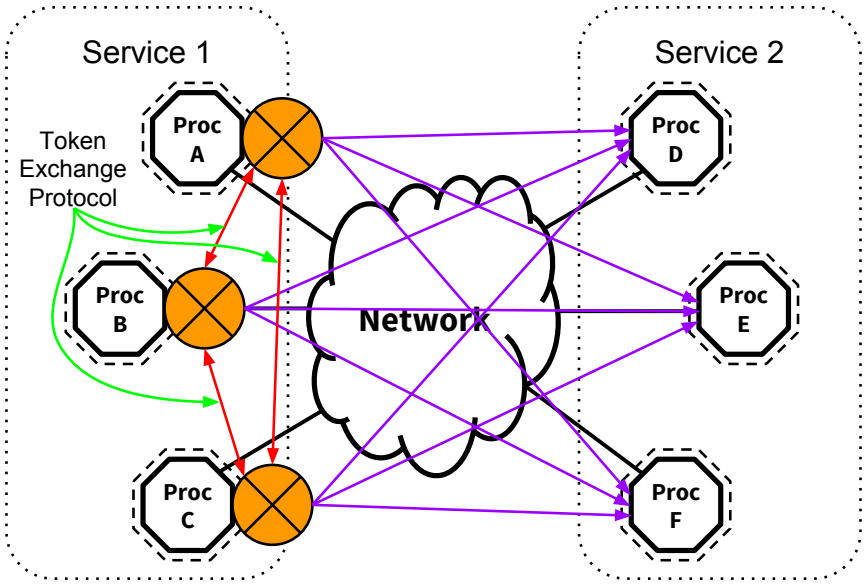


Figure 5.5: Sender-Enforced - Token Exchange (SE-TE) rate-control algorithm.

limit. These rates can be distributed equally or unequally.

This algorithm provides correct behavior as the aggregate rate limit cannot be violated. It also has zero bandwidth overhead as messages flow directly from source to destination and there is no auxiliary control messages being sent. When each source process sends at a rate that is less than the rate at which its corresponding token bucket fills, this algorithm provides zero latency overhead. However, any process that tries to send at a rate higher than its token bucket fills will see an infinite amount of latency because its messages will wait for an increasingly longer time and it will never recover. This is the case even when the desired rate doesn't cause the aggregate rate to exceed the limit.

5.4 Sender-Enforced - Token Exchange (SE-TE)

The *Token Exchange* algorithm (shown in Figure 5.5) augments the baseline *Fixed Allocation* algorithm by allowing each token bucket to ask its peer token buckets for tokens during times of need. To accomplish this, each token bucket has a threshold Th_{low} that triggers when the amount of tokens gets too low. In this scenario, the token bucket chooses N_{peers} random peers from which it will ask for additional tokens. For each request, the

token bucket asks for tokens using the following equation:

$$T_{ask} = ((S_b - N_t)/N_{peers}) \times F_t \tag{5.3}$$

where F_t is the token ask factor and N_t is the current amount of tokens in the bucket. The token ask factor creates an opportunity to vary the algorithm to be more greedy or more conservative. A token ask factor of greater than 1.0 means the algorithm may receive more tokens than it is able to handle, thus wasting tokens. However, it may be found that statistically, this greedy approach is beneficial. For example, assuming $F_t = 1.0$ and $N_{peers} = 3$, if the bucket size is 500 but there only 200 tokens in the bucket, each request would ask for 100 tokens. When $F_t = 1.0$ a token bucket asks for exactly enough tokens to fill its bucket.

When a token bucket receives a request from one of its peers asking for tokens, it decides whether it wants to give tokens away or not, and if so, how many to give. Each token bucket has a threshold Th_t which is the minimum amount of tokens needed before it is willing to give away any of its tokens. If this threshold is exceeded, it is willing to give away all of its tokens above the threshold. It gives tokens according to the following equation:

$$T_{give} = min(T_{ask}, max(0, N_t - Th_t) \tag{5.4}$$

This algorithm provides correct behaviour. Just like the previous algorithm, this has zero overhead if the processes stay within their allotted rate. However, unlike the previous algorithm, this algorithm is able to adapt to non-uniform rate usage by the various processes. In this case, it employs token exchange to retrieve the necessary tokens to send its packets. As long as the aggregate rate of the processes does not violate the aggregate rate limit, this algorithm can, in theory, have zero latency overhead if the token exchange is able to adapt fast enough. However, bandwidth overhead is incurred by the token exchange requests and responses when non-uniformity exists.

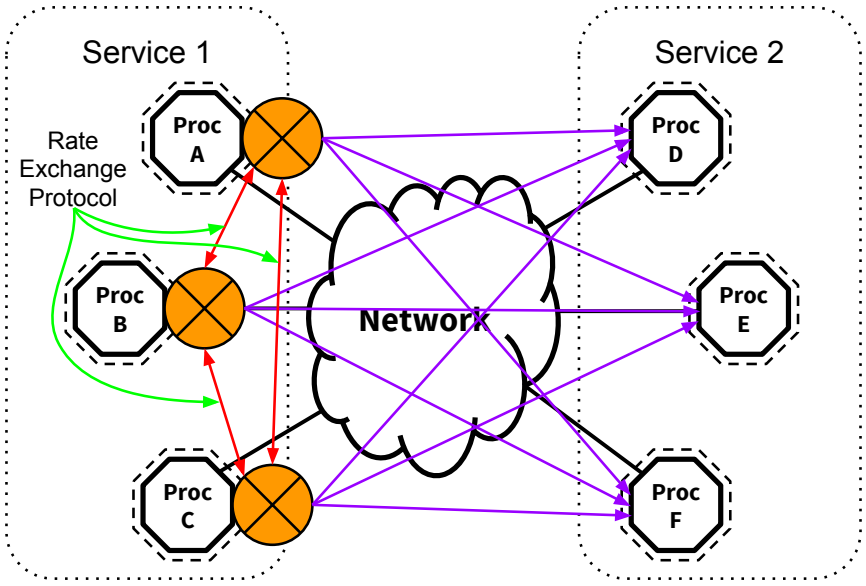


Figure 5.6: Sender-Enforced - Rate Exchange (SE-RE) rate-control algorithm.

5.5 Sender-Enforced - Rate Exchange (SE-RE)

The *Rate Exchange* algorithm (shown in Figure 5.6) is similar to the *Token Exchange* algorithm but attempts to make exchanging more permanent. The key insight is that non-uniformity is expected to persist throughout the life of the application and uniformity is very rare. *Token Exchange* induces bandwidth overhead during times of non-uniformity, which means *Token Exchange* is expected to have a constant bandwidth overhead. Instead of exchanging tokens, the *Rate Exchange* algorithm uses a similar technique to exchange rate. Rate exchanging operates similar to token exchanging using the following equation:

$$R_{ask} = ((1.0 - R_t)/N_{peers}) \times F_r \tag{5.5}$$

where F_r is the rate ask factor. For example, assuming $F_r = 1.0$, $N_{peers} = 3$, and $R_T = 0.4$, each request would ask for 0.2 rate. When $F_r = 1.0$ a token bucket asks for exactly enough rate to get to 100% injection rate.

When a token bucket receives a request from one of its peers asking for rate, it decides whether it wants to give rate away or not, and if so, how much to give. Each token bucket has a threshold Th_r which is the minimum amount of tokens needed before it is willing to

give away any of its rate. If this threshold is exceeded, it is willing to give away some of its rate based on a factor F_{mr}, where $0.0 < F_{mr} < 1.0$. It gives rate according to the following equation:

$$\begin{aligned} R_{give} &= min(R_{ask}, R_t \times F_{mr}) \\ R_t &= R_t - R_{give} \end{aligned} \tag{5.6}$$

After rate is given away, the token bucket fills at the new rate R_t, which is now slower than is was prior to the request. When rate is retrieved from peers, the token bucket fills at a new faster rate.

This algorithm provides correct behaviour and also has zero overhead if the processes stay within their allotted rate. During times of non-uniformity the amount of bandwidth overhead is proportional to the derivative of the rates. In other words, exchanging rates allows the algorithm to adapt to non-uniformity by putting the rate where it needs to go. Bandwidth overhead is only incurred during times of change. Latency overhead might be incurred if the algorithm isn't able to adapt fast enough to the rate of change. Receiving additional rate from a peer doesn't immediately allow blocked packets to proceed. They must still wait until the proper number of tokens have been generated.

5.6 Sender-Enforced - Token and Rate Exchange (SE-TRE)

The *Token and Rate Exchange* algorithm (shown in Figure 5.7) is simply the combination of the *Token Exchange* algorithm and the *Rate Exchange* algorithm. Each exchange request asks for tokens and/or rate. Similarly, responses can contain tokens and/or rate. The two exchange protocols remain logically independent, but use the same network requests and responses. The insight into the usefulness of this algorithm is that token exchange makes the right short term decision and rate exchange makes the right long term decision. This hybrid algorithm yields the fast recovery of token exchanging and the dynamic non-uniform adaptation of rate exchanging.

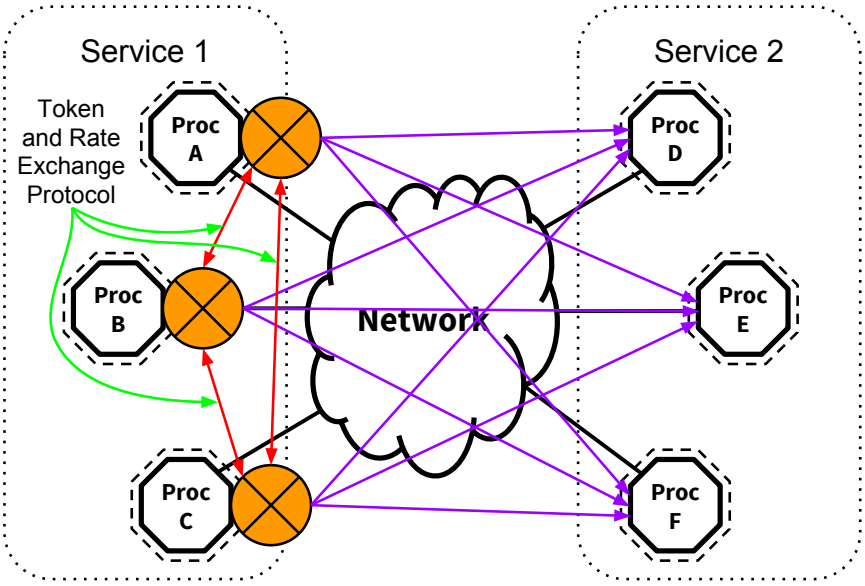


Figure 5.7: Sender-Enforced - Token and Rate Exchange (SE-TRE) rate-control algorithm.

Chapter 6

Network Management Unit

The workhorse of Sikker is a device called the Network Management Unit (NMU). This chapter describes the architecture and functionality of the NMU. The NMU provides each process with high-performance network access while implementing the Sikker security and isolation model, described in Chapter 4. The name *Network Management Unit* is used to resemble the name of the Memory Management Unit (MMU) used in all modern CPU architectures. MMUs are hardware devices that sit between the processor and its corresponding memory system in effort to implement efficient process-level isolation. MMUs are governed by the host operating system using tables that hold mappings between process identifiers and their memory-oriented permissions. In a similar vein, the NMU sits between a host and the network. It is a hardware device that provides efficient service-level security and isolation on the network. It is governed by the network operating system and contains tables that hold mappings between services and their network-oriented permissions.

6.1 Architecture

This section describes the architecture of the NMU which is a new network interface controller (NIC). As the entry point to the network (shown in Figure 6.1), the NMU is able to enforce access control at the sender for every network transaction. A high-level architectural diagram of the NMU is shown in Figure 6.2 which shows that the NMU is an extension to the standard NIC architecture with the following requirements:

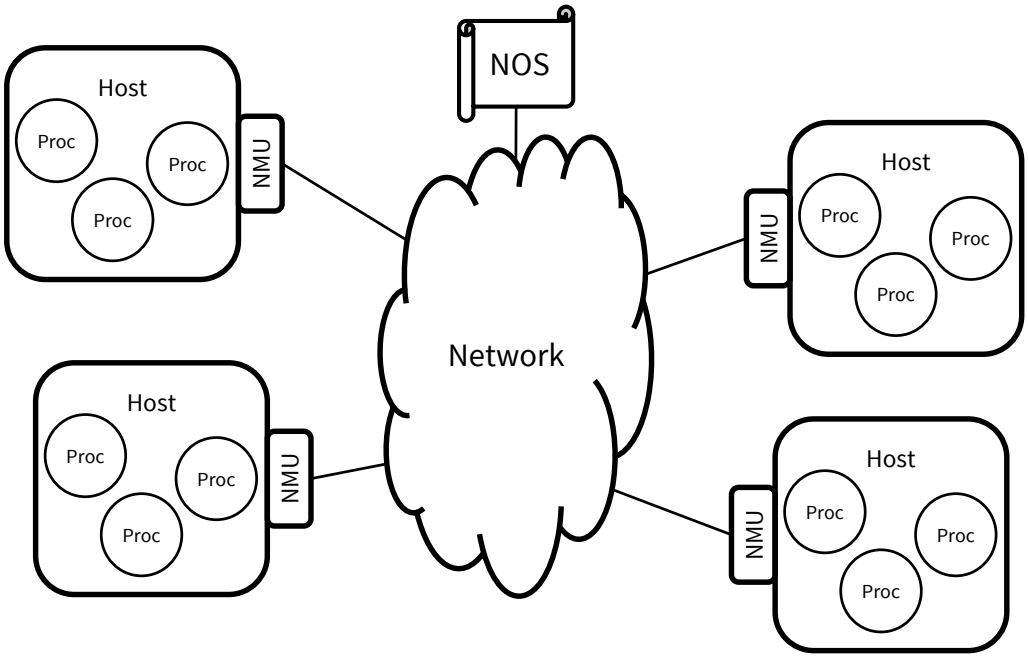


Figure 6.1: Hosts connect processes to the network via NMUs.

NMU Requirements:

N.1 A method for efficient high-performance interaction between resident processes and the network.

N.2 A method of deriving the identity of local processes using the network.

N.3 A method for storing Sikker permissions relevant to the resident processes.

N.4 A method for regulating network access based on Sikker permissions.

6.1.1 Authenticated OS-Bypass

To implement high-performance network access, from requirement **N.1**, the NMU implements OS-bypass. As with most other OS-bypass implementations, the NMU allows a process and the NMU to read and write from each others memory space directly without the assistance of the kernel. The NMU's OS-bypass implementation has one major difference compared to other implementations, namely, it uses the memory-mapped interface to

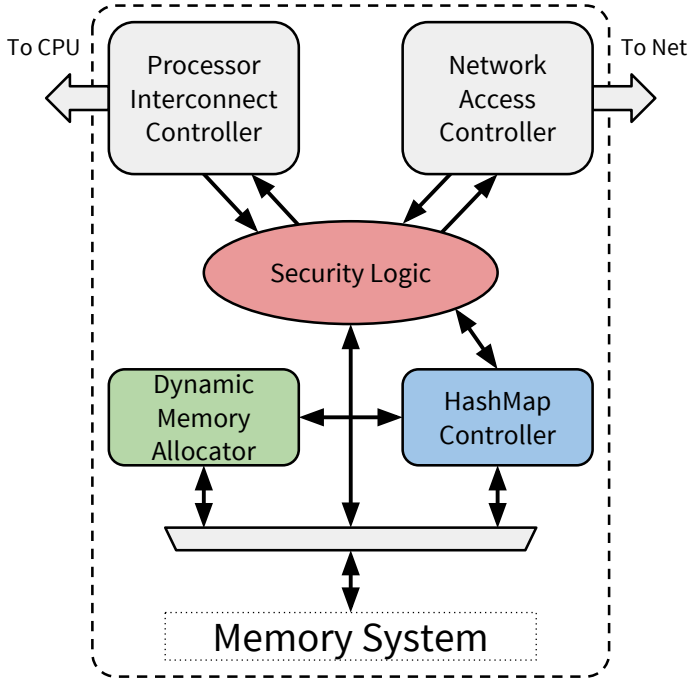


Figure 6.2: The NMU architectural diagram.

derive the identity of a communicating process, which fulfills requirement **N.2**. This process is shown in Figure 6.3. The NMU contains many virtual register sets, upon which, the various processes are able to interact with the NMU. This corresponds to a large physical address space mapped to the NMU. When a new networked process is started, the NMU gives the host's operating system the base address of the register set that the process will use. The NMU contains an internal table that maps register set addresses to process identities, in terms of service ID and process ID. After the process is started, the register set is mapped into the process's address space via the memory management unit (MMU) and the process is only able to use this register set for interaction with the NMU. The process never tells the NMU its identity, instead, the NMU derives its identity from the memory address used for communication with the NMU.

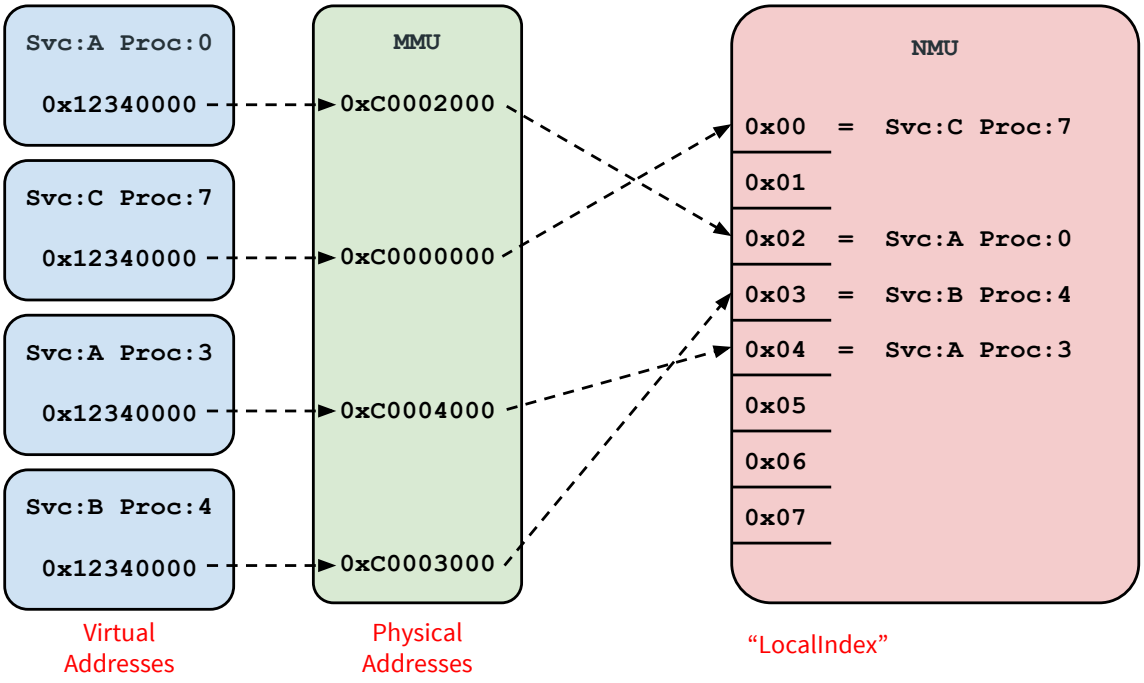


Figure 6.3: The interaction between 4 process, the MMU, and the NMU. The NMU maps virtual register set addresses to Sikker's high-level service-oriented process identifiers.

```
IndexMap: (LocalService, LocalProcess) → LocalIndex
InfoMap: LocalIndex → (LocalService, LocalProcess, OtpNextKey,
                       PermissionMap,OtpMap,PeerSet)
  PermissionMap: RemoteService → (ProcessMap,DomainSet, RateFSM)
    ProcessMap: RemoteProcess → Address
    DomainSet: RemoteDomain
  OtpMap: OtpKey → (RequesterService, RequesterProcess,
                    RecipientService, RecipientProcess,
                    RecipientDomain, RecipientAddress)
  PeerSet: PeerAddress
```

Figure 6.4: The NMU's internal nested hash maps data structures.

6.1.2 Nested Hash Map Accelerator

The internal data structures of the NMU have been crafted such that all variable sized data is represented as nested hash maps[1]. Furthermore, the hash mappings and value placements have been optimized to keep the hash maps as small as possible in effort to produce predictably low search times. The elements of the NMU's internal data structures are listed in nested form in Figure 6.4. These data structures are the NMU's fulfillment of requirement **N.3** and comprise all the information needed for the resident processes to communicate on the network. Because the NMU is governed by the NOS and not the host's operating system, the NMU's memory subsystem is inaccessible by the host's operating system[2].

To implement the NMU's internal data structures efficiently, the NMU architecture has been designed as a data structure accelerator specifically for searching and managing nested hash maps. As shown in Figure 6.2, the top-level architecture of the NMU consists of three main elements above that of a standard NIC architecture: security logic, hash map controller, and dynamic memory allocator. The combination of these logic blocks facilitates the management of its internal data structures, the nested hash maps.

Attached to the memory system of the NMU is the dynamic memory allocator which

[1] Hash sets are considered the same as hash maps. A hash set is simply a hash map with a zero sized value.

[2] It is possible to use the same memory system as the host processor if the NMU uses digital signatures to verify that the information has not been tampered with.

is a hardware implementation of a coalescing segregated fit free list allocator. This allocator design has a good performance to memory utilization ratio across a wide variety of access patterns [54]. Furthermore, the allocator is not in the critical path of the NMU's operation as dynamic memory allocation is only performed when the hash mappings are modified. The allocator allows both the security logic and the hash map controller to create, resize, and free dynamically sized blocks of memory. The hash map controller is a hardware implementation of a linear probed open addressing (a.k.a. closed hashed) [55] hash map controller. This particular hash map controller is used because it is extremely cache friendly. It connects to the dynamic memory allocator and directly to the memory system. Since the hash map controller handles all hash map operations, the security logic simply issues a set of operations for each NMU function.

6.1.3 Permissions Enforcement

The NMU's task is to efficiently check the permissions of every outgoing message before it enters the network. For each potential message being sent on the network, the security logic issues commands to the hash map controller, which traverses the nested data structures to ensure that proper permissions exist. If proper permissions do exist, the security logic translates the virtual service-oriented network address, consisting of a destination service, process, and domain, into a physical network address directly from the entries in the hash maps. The message is then given to the network access controller to be sent on the network. When proper permissions do not exist, the security logic rejects transmission of the message and flags the process with an error code in its corresponding register set. This functionality fulfills requirement **N.4**.

6.1.4 Management

The NOS manages every NMU in the system using an in-band management protocol which gives the NOS the ability to send messages over the network directly to the NMUs. The management protocol allows the NOS to modify the contents of the NMU's internal data structures explicitly giving the NOS control over the permissions that are placed within each NMU.

The NMU is able to distinguish control messages from the NOS with explicit authentication as the service ID of the NOS uses a reserved value. Before a new networked process is started on a host, the NOS configures the NMU with all the permissions needed by that process. Similarly, the NOS is also responsible for removing the permissions of a process when the process ends.

6.2 Operation

This section walks through the operations the NMU performs and its traversal and management of the data structures shown in Figure 6.4. "`Local`" variables refer to entities resident on the NMU and "`Remote`" variables refer to entities that exist on other NMUs. It is possible to send messages between processes resident on the same NMU, however, the "`Local`" and "`Remote`" distinctions will still be used. When using OTPs, the process that generates the OTP is defined as the *requester*, the process that receives the OTP as the *responder*, and the process that receives the message that was sent using the OTP as the *recipient*. Thus, the *requester* sends an OTP and request message to the *responder* and the *responder* uses the OTP to send a response message to the *recipient*. For two-way request-response protocols, the *requester* and *recipient* are the same.

6.2.1 Send

To initiate a standard message send operation, the source process gives the NMU the `RemoteService`, `RemoteProcess`, and `RemoteDomain` of the destination. The NMU derives the sender's `LocalIndex` which is a simple bit selection from the physical memory address used by the process to communicate with the NMU. Next, the `LocalIndex` is used as the key for an `InfoMap` lookup which yields, among other things, the `PermissionMap`. The NMU then uses the `RemoteService` to perform a `PermissionMap` lookup which yields the `ProcessMap` and `DomainSet` corresponding to the `RemoteService`. The NMU now checks that the `RemoteProcess` exists within the `ProcessMap` and the `RemoteDomain` within the `DomainSet`. If both lookups are successful, the `Address` that was returned by

the ProcessMap lookup is used as the destination physical network address of the message. The message header will contain LocalService and LocalProcess as the message's source and the RemoteService, RemoteProcess, and RemoteDomain as the message's destination. If any lookup during this procedure fails, the NMU will not send the message and will set an error flag in the process's register set.

6.2.2 Receive

When the destination NMU receives the message the destination service, process, and domain have now become the LocalService, LocalProcess, and LocalDomain. Using the LocalService and LocalProcess, the NMU performs an IndexMap lookup which yields the corresponding process's LocalIndex and tells the NMU which register set the message should be placed in.

6.2.3 Send with OTP

When the requester desires to generate and send a message with an attached OTP, on top of specifying the responder as the destination of the message, it must also specify the recipient. The NMU uses the same permission check procedure as in Section 6.2.1 except now it performs two PermissionMap, ProcessMap, DomainSet lookup sequences, one for the responder and one for the recipient. Upon successful lookups, the NMU sends the message just like it did in Section 6.2.1 except that the message header also contains the recipient's information as the OTP.

6.2.4 Receive with OTP

When the responder's NMU receives the message containing the OTP it starts as usual by performing an IndexMap lookup yielding the LocalIndex. It also performs an InfoMap lookup to retrieve the OtpNextKey and OtpMap. The OtpNextKey and the received message are now placed in the corresponding process's register set. The NMU performs a hash map insertion into the OtpMap which maps the OtpNextKey to the OTP information given in the message. The NMU then advances OtpNextKey to the next key and writes it into the

proper memory location.

6.2.5 Send using OTP

When the responder is ready to send the response message using the OTP, it does not specify the destination in terms of service, process, and domain. Instead, the process gives the NMU the `OtpKey` it was given during the receive operation. The NMU uses the process's corresponding `LocalIndex` to retrieve its `OtpMap` from the `InfoMap`. The NMU then uses the `OtpKey` to perform an `OtpMap` removal operation to retrieve and remove the OTP, which consists of the requester's information as well as the recipient's information. The recipient's information is used as the message destination and the requester's information is also added to the message header so the recipient knows where the message sequence originated from. Since the OTP was removed from the `OtpMap` during this procedure, the OTP cannot be used again.

6.2.6 Rate Control

After a message send operation passes access-control checks, it then proceeds to the rate-control mechanism. If the message is being sent with an OTP or if a rate limit is not specified for the destination service, the message will bypass the rate-control mechanism and will be sent on the network immediately. During the send operation, the source process's `PeerSet` is pulled from the `InfoMap` as well as the `RateFSM` for the corresponding destination service. The `PeerSet` contains the addresses of a set of other processes within the source service. When the rate-control algorithm needs to exchange control information with other processes in the service, it uses the `PeerSet` to find peer process addresses. The `RateFSM` holds all the values needed to implement the distributed rate-control algorithm being used. Chapter 5 covered a set of candidate algorithms in detail and Chapter 8 evaluates these algorithms for use in the NMU.

Chapter 7

Access Control Evaluation

This chapter provides an evaluation of the access-control functionality defined by Sikker as implemented by the NMU. Since the NMU can be viewed as an extension to the standard NIC architecture, its performance is quantified by measuring the additional overhead incurred by performing its operations in terms of latency and bandwidth. Since the NMU performs at least one permission check[1] for every message being sent on the network, it is extremely critical that this is completed in a timely manner without limiting bandwidth and without inducing CPU overhead.

7.1 Methodology

This section describes the methodology used to explore potential NMU configurations as well as the creation of synthetic workloads that stress test the NMU's permissions checking procedure.

7.1.1 Simulation

The logic of the NMU can be attached to any memory system and the performance of the NMU widely depends on the structure and size of the memory system chosen. To explore

[1] As mentioned in Section 4.4, the NMU performs a single permission check for regular messages and two permission checks for messages that also generate an OTP.

the design space of the NMU, a custom simulator, called *SikkerSim*, was developed and used to measure the NMU's performance while performing permissions checks. At the top level of SikkerSim is an implementation of the core logic of a NOS that manages the permissions of all the NMUs on a network. It does this by creating a permission connectivity graph as shown in Figure 4.4 and connects a simulated NMU on each simulated host. For each simulated NMU, SikkerSim models the internal logic elements of the NMU as well as various memory system architectures under design consideration. SikkerSim can be used to model potential NMU memory systems spanning from single SRAMs to multi-stage cache hierarchies connected to DRAM. CACTI 6.5 (32nm process technology) [56] and DRAMSim2 (DDR3 SDRAM) [57] are used in connection with SikkerSim to produce accurate timing results for each case.

Even though Section 4.8 shows that SACLs allow endpoints to use pure SRAM storage, the modeling and simulations presented in this chapter model the main memory as off-chip DRAM to remain agnostic to system size and connectivity density. For the sake of performance analysis, an average memory system design was chosen that yields high performance while not incurring excessive cost. This design attaches the NMU logic to a memory system containing two levels of cache and a DRAM main memory. The first cache level (L1) is an 8-way set associative 32 kiB cache. The second cache level (L2) is a 16-way set associative 4 MiB cache. Unlike standard microprocessor cache hierarchies, the NMU operates directly on physical memory addresses and considers all memory as "data". The NMU doesn't need an MMU, TLB, or instruction cache, thus, the NMU's physical interface to the L1 cache is fast and lightweight.

7.1.2 Permission Access Patterns

The data structures of the NMU present abundant spatial locality to the memory system, and depending on the permission access pattern, significant temporal locality can also exist. SikkerSim contains a configurable synthetic permission access pattern that is placed on each simulated NMU. For every permissions check, the access pattern selects a source process as the sender and a destination process as the receiver. The access pattern also selects a domain to be used that is defined within the destination service.

The worst-case access pattern is a uniform random selection across the source and destination possibilities. In this pattern, each permissions check randomly selects a resident process as the source, then randomly selects the destination service, process, and domain from the corresponding source service's permissions. This pattern exhibits no temporal locality in the NMU's memory system.

The best case access pattern is repeatedly choosing the same source and destination. This pattern exhibits full temporal locality in the memory system. While this pattern is unrealistic for long durations, it is realistic for very short durations. A slight variant of this pattern would be repeatedly accessing the same destination service, while switching destination process and/or domain. Similarly, the same source process might be repeatedly accessing the network but choosing a new destination each time.

Since both the worst and best case access patterns are somewhat realistic, the synthetic permission access pattern in SikkerSim was designed to reflect two common attributes that control temporal locality in a realistic way.

Repeated Access - The first attribute configures the amount of repeatability at each step of the selection process for the source and destination. There are several aspects that make this realistic in practice. For instance, it is common for a process using the network to interact several times with the network before another process has the chance to or chooses to. This can be caused by CPU thread scheduling or application-level network bursting. Also, it is common for a process to send multiple back-to-back messages to the same destination service or even the same destination service and process and/or service and domain. The result is a higher level of temporal locality simply due to repeated accesses in a particular selection group.

Hot Spots - The second attribute configures the selection distribution when the synthetic permission access pattern chooses a new source and destination. This is used to model hot spots in network traffic. For instance, an high-throughput application using an SQL database will often use an in-memory caching service to reduce the load on the SQL database. For this example, the in-memory cache is a hot spot as it is accessed with higher frequency than the SQL database. To model this behavior, the selection process is allowed to choose using a uniform random distribution or a Gaussian random distribution. The uniform random distribution models network traffic that is irregular and unpredictable while

the Gaussian random distribution models network traffic that contains hot spots both in terms of the source and destination with all its components.

Using these controllable attributes, SikkerSim's synthetic permission access pattern was used to create four access patterns that are used to benchmark the performance of the NMU. They are as follows:

- **Uniform Random (UR):** All selections are from a uniform random distribution.
- **Uniform Repeated Random (URR):** Same as UR, except that portions of the selection are re-used a configurable number of times.
- **Gaussian Random (GR):** All selections are from a Gaussian random distribution.
- **Gaussian Repeated Random (GRR):** Same as GR, except that portions of the selection are re-used a configurable number of times.

7.2 Results

This section presents the performance results of the NMU in terms of latency and bandwidth overhead. SikkerSim was used to simulate systems spanning from a few thousand hosts up to systems of well over 100,000 hosts. All simulations use the connectivity model presented in Section 4.7 to represent the service-oriented connectivity density in the system.

7.2.1 Latency

This section analyzes the latency incurred in the NMU for checking permissions. Figure 7.1 shows the mean, 99^{th} percentile, and 99.99^{th} percentile latency response of permission checks for each of the four permission access patterns described in Section 7.1.2. As expected, the UR and GRR patterns represent the worst and best patterns, however, the mean of the UR pattern is only up to 25% worse than the GRR pattern and both curves flatten out by 32,768 hosts. Even under extreme conditions, the NMU adds negligible latency overhead to network transactions. On a large system with over 2 million networked processes

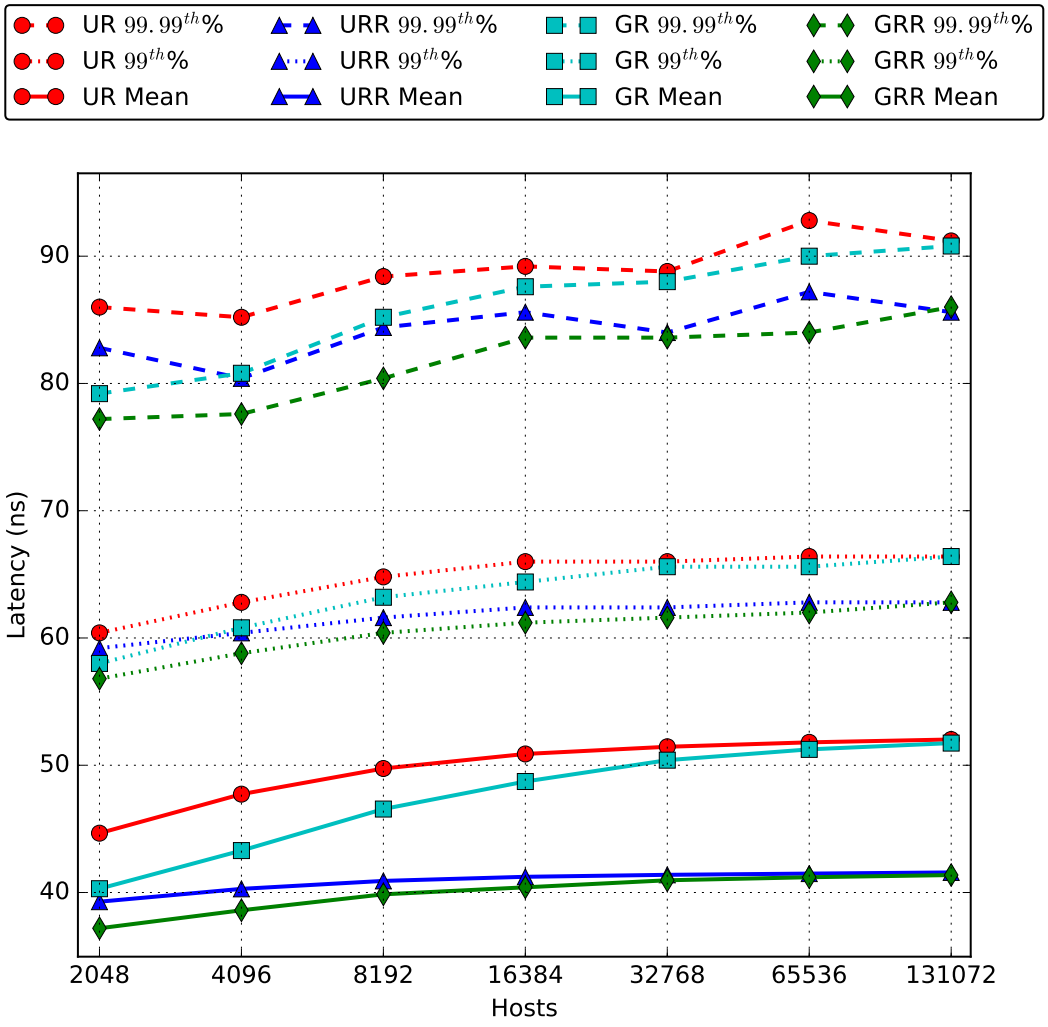


Figure 7.1: Mean, 99^{th} percentile, and 99.99^{th} percentile latency of all four access patterns. Solid lines are mean latency, dotted lines are 99^{th} percentile latency, and dashed lines are 99.99^{th} percentile latency.

(131,072 hosts), the mean latency of a realistic access pattern (GRR) is only 41 ns and the 99.99^{th} percentile latency of the worst-case access pattern (UR) is only 91 ns. Relative to the standard permissions checking process, using OTPs incurs the same latency overhead with negligible differences. The main difference is that generating an OTP during a message send operation requires two permission checks, however, these can run in parallel.

7.2.2 Bandwidth

While predictably low latency is an important metric of performance, bandwidth is also an important metric in high-performance computing. Table 7.1 shows the throughput of a single NMU logic engine in terms of millions of permission checks per second (Mcps). The ability to translate checks per second to bytes per second requires an understanding of the average message size within the system. This value varies widely depending on the applications running within the system. For example, a study of Microsoft's data centers shows the average packet size to be 850 bytes [58] while a study of Facebook's data centers shows the average packet size to be 200 bytes [59]. Table 7.1 shows the translation to bandwidth for both of these values. This shows that a single NMU logic engine on a very large cluster (131,072 hosts) with a realistic permission access pattern (GRR) can process over 24 million permission checks per second. The worst case access pattern only degrades the throughput by 21%.

	UR	**GRR**
Permission checks per second	19.23 Mcps	24.39 Mcps
Bandwidth (850 byte packets)	130.77 Gbps	165.85 Gbps
Bandwidth (200 byte packets)	30.77 Gbps	39.02 Gbps

Table 7.1: Throughput performance of a single NMU logic engine. *Mcps* is million permission checks per second. *Gbps* is gigabits per second.

The amount of bandwidth induced by the NMU on its main memory is very small. The logic engine of the NMU is designed with simplicity to have only a single outstanding memory request. Due to the NMU's efficient organization of its nested data structures and the performance attributes of well-tuned open addressed hash maps, each permission check averages only a single main memory operation. Since each operation transfers an entire

cache line of 64 bytes, when the NMU is running at 24.39 Mcps, the bandwidth induced on the main memory is approximately 1.756 GB/s. In comparison, a typical DRAM memory channel is capable of approximately 25 GB/s.

Because the complexity of the NMU is abstracted away by its internal data structures, the complexity of adding multiple logic engines to the NMU is fairly trivial. Furthermore, the majority of the operations performed in the NMU are read-only operations, which are highly parallelizable. For the operations that require writes (i.e., OTPs), distributing data structure ownership across multiple engines and using hash-based message steering to the corresponding engine allows near lock-free parallelization. With relatively little effort, an NMU can be built with many logic engines. Multiple logic engines don't impose a bandwidth bottleneck on the NMU's main memory as it would take over 14 logic engines to saturate a single 25 GB/s memory channel and there's no reason multiple memory channels couldn't be employed. Based on the results in Table 7.1 and degrading performance by 10% to account for potential lock contention, an NMU with 8 logic engines is able to process 138 - 176 million permissions checks per second which yields over 1 Tbps for Microsoft packet sizes. This only requires 14 GB/s of DRAM memory bandwidth.

7.2.3 Security

The NMU implements all the security and isolation features of Sikker as discussed in Chapter 4. This includes source and destination authentication, virtual-to-physical network address translation, sender-enforced service-oriented permission checks, and permissions management.

The Sikker application model uses individual endpoint machines to host the processes of the various services (hence the name *host*). As such, Sikker relies on the host's operating system to provide process-level isolation between the processes resident on that host. In general, Sikker assumes that the various host operating systems within the network are unreliable. For this reason, the NMU was designed to be explicitly controlled by the NOS rather than individual host operating systems. For the same reason, the NMU has its own memory system that is inaccessible by the host's operating system.

In the event that a host's operating system is exploited by a resident process, the process

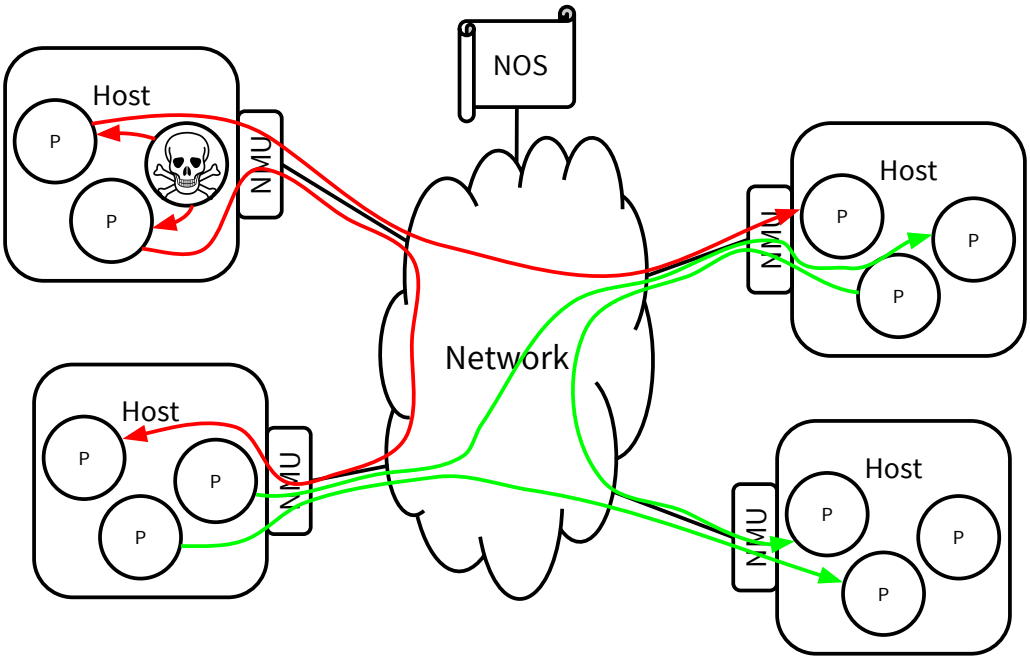


Figure 7.2: The NMU contains an exploited host to the permissions that exists within that NMU. Red lines show potentially affected permissions. Green lines show un-affected permissions.

might be able to assume any of the permissions that have been given to *all* processes on that host, as shown in Figure 7.2. This is a large improvement over current systems that utilize the host operating systems for security (e.g., hypervisor-based security and isolation). In those systems, an exploited operating system might be given access to anything in the entire network, not just the permissions resident on that host. In Sikker, if a host's operating system cannot be deemed reliable enough provide process-level isolation, it is recommended to co-locate processes only where an attack would not prove detrimental if one resident process gained access to another resident process's permissions. In many scenarios this is already the case due to the difficulty of providing machine-level performance isolation.

7.3 Summary

As an extension to the common NIC architecture, the NMU can only increase the overhead of network transactions relative to a system without any security and isolation features. However, due to Sikker's service-oriented permission scheme and the NMU's efficient implementation, the overhead of the NMU is negligible. If all round trip network transactions took only 2 µs, the NMU imposed overhead would only increase latency by 4%. More realistic round trips of 10 µs that include potential queuing delays and/or endpoint processing time brings this overhead down to 0.8%. Furthermore, operations that induce this negligible overhead cover the entire access-control needs of the application.

Today's security and isolation schemes impose at minimum many tens of microseconds of latency overhead and high CPU overhead. These systems are unable to completely satisfy the security and isolation requirements of modern large-scale applications thus forcing the applications to attempt to solve these issues themselves. In contrast, Sikker and the NMU provide a secure and efficient computing environment with essentially no overhead.

Chapter 8

Rate Control Evaluation

This chapter evaluates the six rate-control algorithms presented in Chapter 5 for use in Sikker. While any of the algorithms could be implemented in the NMU, it is desirable to minimize the amount of overhead incurred by the rate-control feature specified by Sikker's service-oriented application model. For these algorithms, overhead is measured in terms of additional network latency incurred as well as any additional bandwidth consumed beyond the original payloads being sent from the source service to the destination service.

8.1 Methodology

A custom simulator, called RateSim, was developed to explore the proposed distributed rate-control algorithms while simulating the interactions of services on a network. The simulator models many processes, each being a part of a service, and the network. The network modeling allows each process to send at 100% injection rate bandwidth and receive with infinite bandwidth. This type of optimistic network modeling isolates any network congestion from congestion that could be caused by a rate-control algorithm under investigation. Even though the modeled processes can receive messages with infinite bandwidth, serialization latency is modeled as each process can only send at line rate and queuing congestion occurs when a process has more than one packet in its queue to send.

The algorithm proposed in this dissertation is designed to make high-performance interconnection networks usable for service-oriented applications as ubiquitously deployed in

data centers and cloud computing environments. As such, the methodology for evaluating distributed rate-control algorithms is based around the performance of high-performance computing systems. For all simulations, the link bandwidth of each process is 100 Gbps and the network latency is 500 ns, similar to a modern supercomputer [11]. The simulation assumes a 1 GHz clock cycle which yields a phit size of 100 bits. The minimum data packet size is 5 phits (≈64 bytes) and the maximum size is 82 phits (≈1024 bytes).

Each process within the source service has a designated rate at which it desires to send packets to the destination service. For each packet sent, the simulation generates a randomly sized packet between 5 and 82 phits and randomly selects the destination process within the destination service. All latencies are measured from the time a packet was created to the time in which the process within the destination service receives the packet entirely. Bandwidth overhead is measured as the amount of bandwidth being received on the network by entities other than the destination processes. For the algorithms presented in Chapter 5, this is the source processes and relay devices.

For all simulations, the source service contains 1000 processes and the destination service contains 750 processes. The rate limit is set to an aggregate rate of 500 phits per cycle, which equates to an average of 50% injection rate for each source process. The traffic pattern used for simulation, shown in Figure 8.1, is a benign traffic pattern as the aggregate rate is always 40% or 400 phits per cycle. It is meant to stress test the various corner cases of the presented rate-control algorithms. On cycle 1 all processes in the source service start sending at 40%. On cycle 10,000 senders 1 through 500 increase their rate to 80% while senders 501 through 1000 turn off. During this period the aggregate rate is still 40%. On cycle 50,000 the two groups of senders swap configurations with senders 1 through 500 turning off and senders 501 through 1000 sending at 80%. On cycle 90,000 all senders again send at 40% rate until cycle 130,000 when the simulation ends.

The time periods from cycle 10,000 to 50,000, 50,000 to 90,000, and 90,000 to 130,000 will be referred to as “period 1”, “period 2”, and “period 3”, respectively. The interval between all periods is 40,000 cycles, or 40 µs. While the frequency at which many applications change usage behaviour is a lower than this, the 40 µs interval is used as a harsh stress-test for the rate-control algorithms under investigation.

When utilizing token buckets for rate control, it is advantageous to minimize the size of

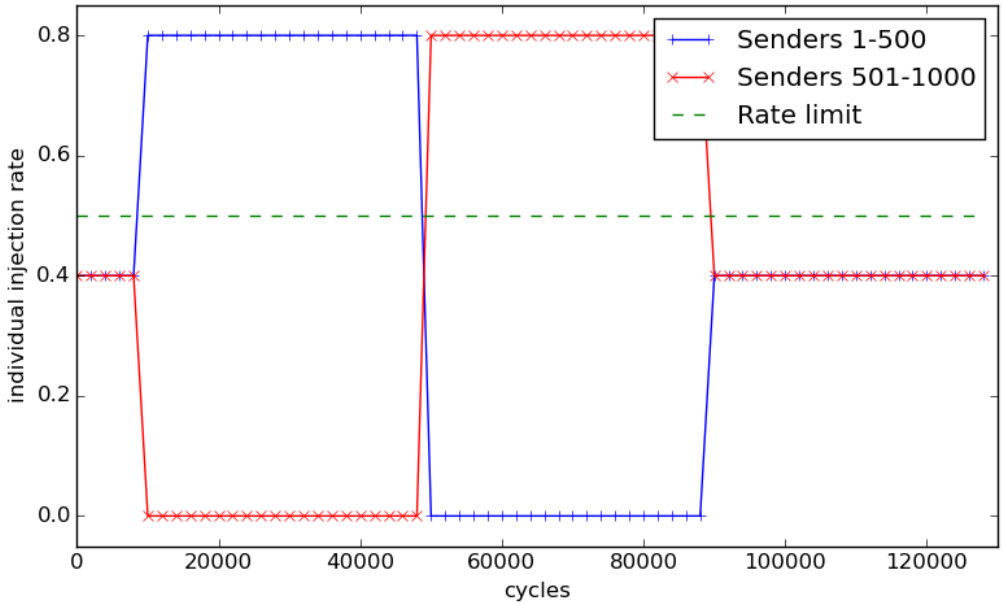


Figure 8.1: Stress testing traffic pattern for rate-control simulation.

the token buckets used. The size of the token bucket represents the largest burst size that can be sent to the destination. Because this work explores distributed rate-control algorithms that utilize multiple token buckets, the maximum aggregate burst size T_{bs} is

$$T_{bs} = S_b \times N_{tb} \tag{8.1}$$

where N_{tb} is the number of token buckets being utilized by an algorithm. The results in Section 8.2 represent simulations using token buckets of size 2000. An algorithm using 1000 token buckets each with 2000 token slots can yield up to 2 million outstanding tokens. If each token represents 1 phit (100 bits) and the clock frequency is 1 GHz, each token bucket can hold up to 25k bytes worth of tokens which it could be sent at full speed (100 Gbps) for 2 µs. If all 1000 token buckets simultaneously sent a maximum size burst the aggregate size would be 25M bytes.

The large number of parameters in the algorithms presents a challenge in representing the *optimal* configuration for each algorithm. To tackle the issue, a massively parallel simulation was performed to simulate nearly all practical settings of each algorithm. The

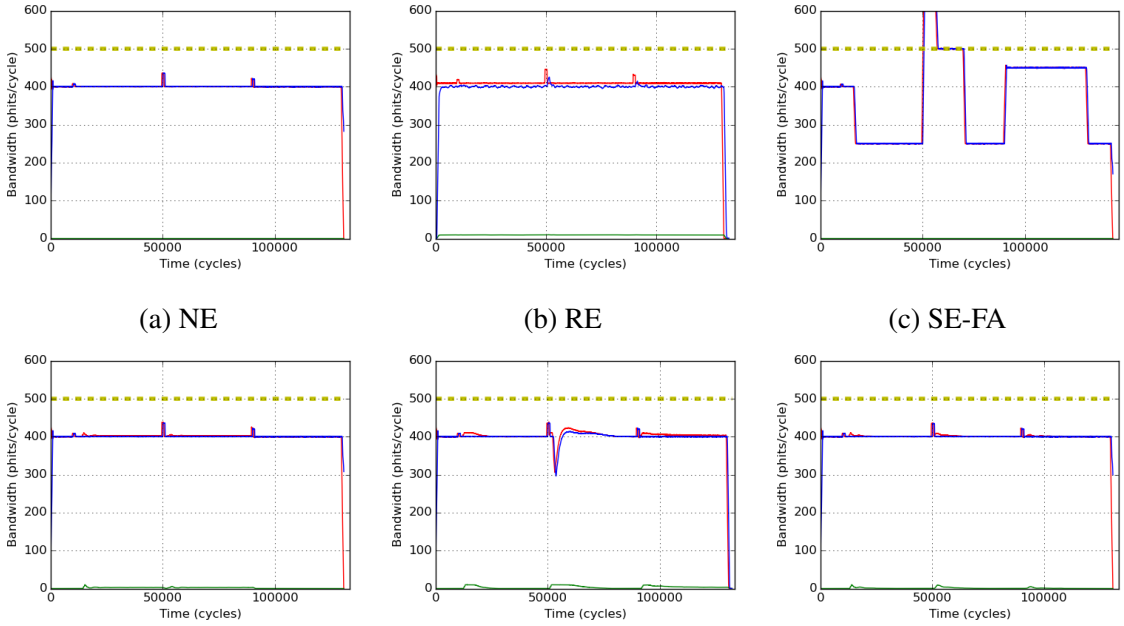
results in Section 8.2 evaluate the configurations that yield the highest performance for each algorithm assuming system designers have properly tuned the algorithms to their system. The discussion in Section 8.3 gives insight to the values of some of the parameters.

When utilizing token buckets for rate control, many parameters express a trade-off between bandwidth and latency where a more aggressive setting can lower latency by using more bandwidth. To find the *optimal* parameter settings for a set of simulation runs, each simulation result is assigned a penalty score based on a weighted sum of the bandwidth overhead and latency overhead. In the penalty calculation, latency overhead is defined as the worse case 99.99^{th} percentile across the 3 time periods. Bandwidth overhead similarly takes the worse case of the 3 time periods except the 3^{rd} time period has a 2x weight. The reason for this is that during this time period all senders are sending at the same rate, thus, in theory there should be no extra bandwidth needed. The latency overhead and bandwidth overhead are summed after the bandwidth overhead is weighted by 75x. This effectively states that 1 phit per cycle (or 100 Gbps) of bandwidth is considered equal in cost to 75 ns of 99.99^{th} percentile latency. This coefficient is subjective and system designers should choose its value carefully based on their specific network constraints. For a set of simulation runs, the penalty of each result is computed and the simulation run that produces the lowest penalty is deemed *optimal*.

8.2 Results

Figure 8.2 shows the bandwidth usage of the six rate-control algorithms. All algorithms besides the SE-FA algorithm are able to supply the senders with enough bandwidth. At the start of each time period when the application changes its behavior there is a small disruption in the smooth aggregate bandwidth usage. This is a result of some senders starting their transactions while the queues of their peers are still draining.

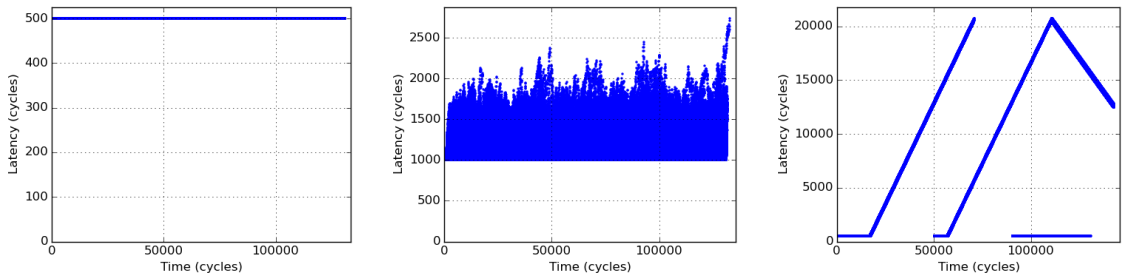
During times of non-uniform use among the senders, the SE-FA algorithm is unable to provide enough bandwidth and consequently the latency is effectively infinite. This infinite latency can be seen in Figure 8.3 which shows a latency scatter plot for each of the algorithms. As shown, the NE, SE-TE, and SE-TRE algorithms all yield good latency



(a) NE (b) RE (c) SE-FA

(d) SE-TE (e) SE-RE (f) SE-TRE

Figure 8.2: Bandwidth usage of the six rate-control algorithms. The yellow dashed line shows the aggregate rate limit of 500 phits per cycle. The blue line shows bandwidth received by the receiving service. The red line shows bandwidth sent by sending service. The green line shows bandwidth received by sending service.



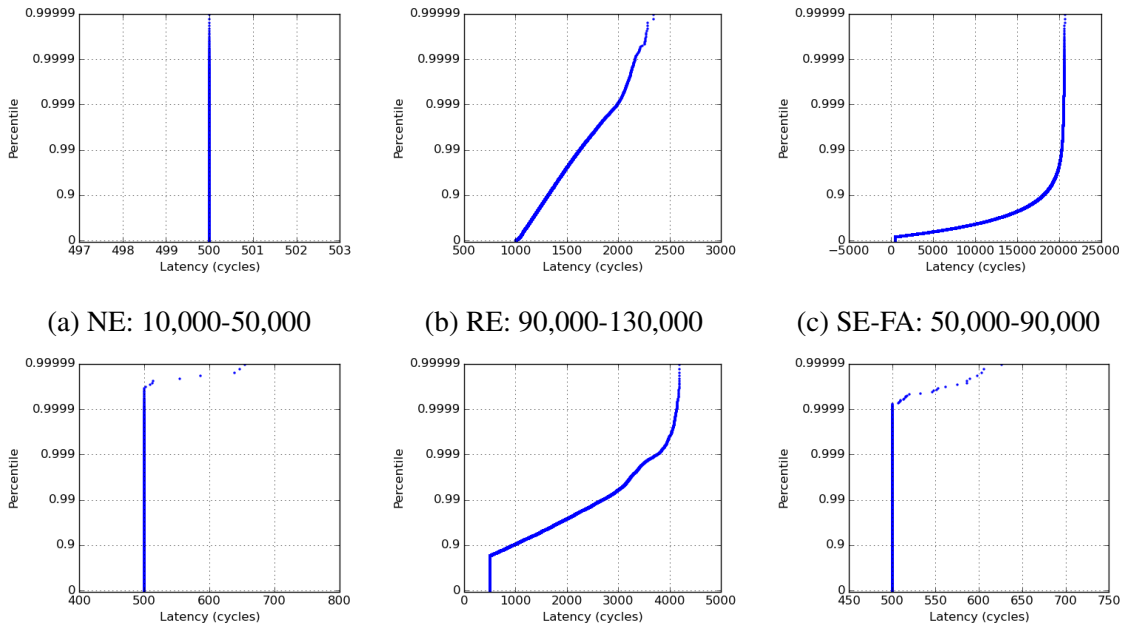
(a) NE (b) RE (c) SE-FA

(d) SE-TE (e) SE-RE (f) SE-TRE

Figure 8.3: End-to-end latency of the six rate-control algorithms.

responses. The minimum latency of the RE algorithm is twice as large as all other algorithms (1000 ns vs. 500 ns) because each packet must traverse the network twice. For each algorithm, Figure 8.4 shows a percentile plot of the worst-case latency across the 3 time periods. Figures 8.3 and 8.4 show that the RE algorithm induces a significant amount of queuing delay which severely effects its tail latency response. The SE-RE algorithm also yields a very poor latency response in the second time period which is caused by the algorithm adapting too slow to the harsh transition made by the senders abruptly changing their rate usage by 80%. The senders are able to exchange the necessary rates but having the right rate is only good in the long term as the senders still need to wait for the new rate to generate tokens for the currently outstanding packets.

The bandwidth response, shown in Figure 8.5, shows that the NE and SE-FA algorithms have zero bandwidth overhead as expected. The RE algorithm results in over 100% bandwidth overhead because of the 2x network traversals and the extra protocol needed to maintain the queues in the relays. As expected, the SE-TE algorithm has zero bandwidth



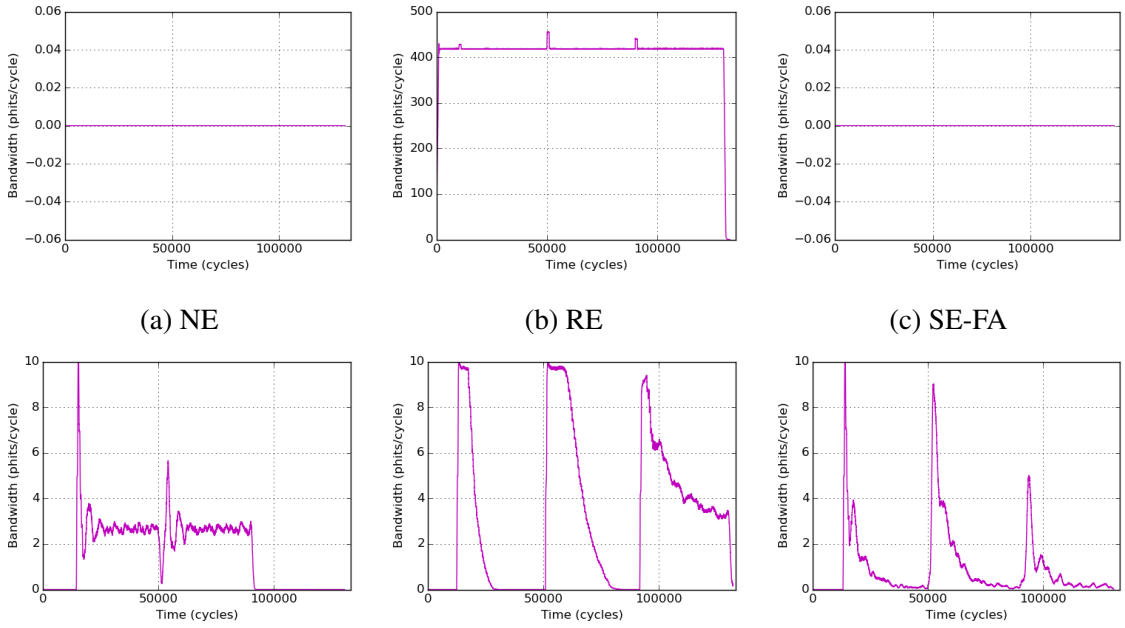
(a) NE: 10,000-50,000 (b) RE: 90,000-130,000 (c) SE-FA: 50,000-90,000

(d) SE-TE: 10,000-50,000 (e) SE-RE: 50,000-90,000 (f) SE-TRE: 50,000-90,000

Figure 8.4: Worst-case latency percentiles of the six rate-control algorithms.

overhead when all senders are sending below their fixed allocation and has a constant bandwidth overhead during times of non-uniform sending rate. The SE-RE algorithm results in bandwidth spikes at each transition point with the spike magnitude proportional to the number of peers chosen in the rate exchange. The spike decays to zero very slowly because exchanging rate doesn't immediately yield the required tokens. Similarly, the SE-TRE algorithm results in bandwidth spikes at each transition point with the same magnitude, however the overhead quickly decays to zero because the senders exchange enough rate to cover their differences and also exchange enough tokens to cover the needs of the currently outstanding packets.

As mentioned in Section 8.1, the penalty scoring function takes into account the 99.99^{th} percentile latency and bandwidth overhead. In Figure 8.4 notice that the SE-TE and SE-TRE algorithms do not yield perfect latency responses, however, their 99.99^{th} percentile latency is perfect. Because the penalty function optimizes the case for the 99.99^{th} percentile latency, the search chose the configurations that have the smallest amount of bandwidth overhead while having a perfect 99.99^{th} percentile latency.



(a) NE (b) RE (c) SE-FA

(d) SE-TE (e) SE-RE (f) SE-TRE

Figure 8.5: Bandwidth overhead of the six rate-control algorithms.

The aggregate results for the six rate-control algorithms are shown in Table 8.1. As shown, of all algorithms that produce correct behavior the SE-TRE has the best results. The second best algorithm is the SE-TE algorithm. The bandwidth overhead of the SE-TE algorithm is constant and its value is proportional to the amount of non-uniformity among the source processes. The bandwidth overhead of the SE-TRE algorithm is dependent on the frequency at which the application changes its behaviour. Table 8.1 shows the results for 25 kHz (or 40 µs intervals). If the frequency of transition happened to be 1 kHz (or 1 ms intervals), the SE-TE algorithm would still result in 262 Gbps of bandwidth overhead, however the SE-TRE algorithm would produce approximately 6 Gbps of bandwidth overhead.

Algorithm	Section	Correct	99.99^{th}% Latency	Bandwidth Overhead
NE	5.1	No	500 ns	0 Gbps
RE	5.2	Yes	2168 ns	41978 Gbps
SE-FA	5.3	Yes	∞	0 Gbps
SE-TE	5.4	Yes	500 ns	262 Gbps
SE-RE	5.5	Yes	4143 ns	463 Gbps
SE-TRE	5.6	Yes	500 ns	142 Gbps

Table 8.1: Rate-control evaluation results of the six algorithms.

8.3 Discussion

The results in Section 8.2 were achieved using massive amounts of simulation to find the optimal configurations of each algorithm. This section discusses the effects of some of the dominant variables as they are varied.

8.3.1 Token Bucket Sizing

As discussed in Section 8.1, the sizing of token buckets creates a trade-off between the efficiency and the strength of a rate-control algorithm. A smaller token bucket reduces the amount of traffic that the source service can burst to the destination service. It is desirable to have this be as small as possible. However, the token buckets need to be large enough such that the rate-control algorithm can adapt to the changes in application behavior before it runs out of resources and incurs high latency. The minimum size of the token bucket is proportional to the network round trip between the peer token buckets. The simulations presented in Section 8.2 used a token bucket of 2000 tokens (2000 phits or 25k bytes). This represents 2 network round trips given the fixed 500 ns (500 cycles) unidirectional network latency. This yields a result where the algorithm is able to serially ask 2 peer token buckets for additional resources. It is important to allow serial requests because the random selection process sometimes yields scenarios where none of the randomly selected peers have excess resources. In this case, the token bucket needs to ask another set of peers for resources before it runs out of tokens.

Figure 8.6 and Table 8.2 show the latency percentile and bandwidth overhead of the

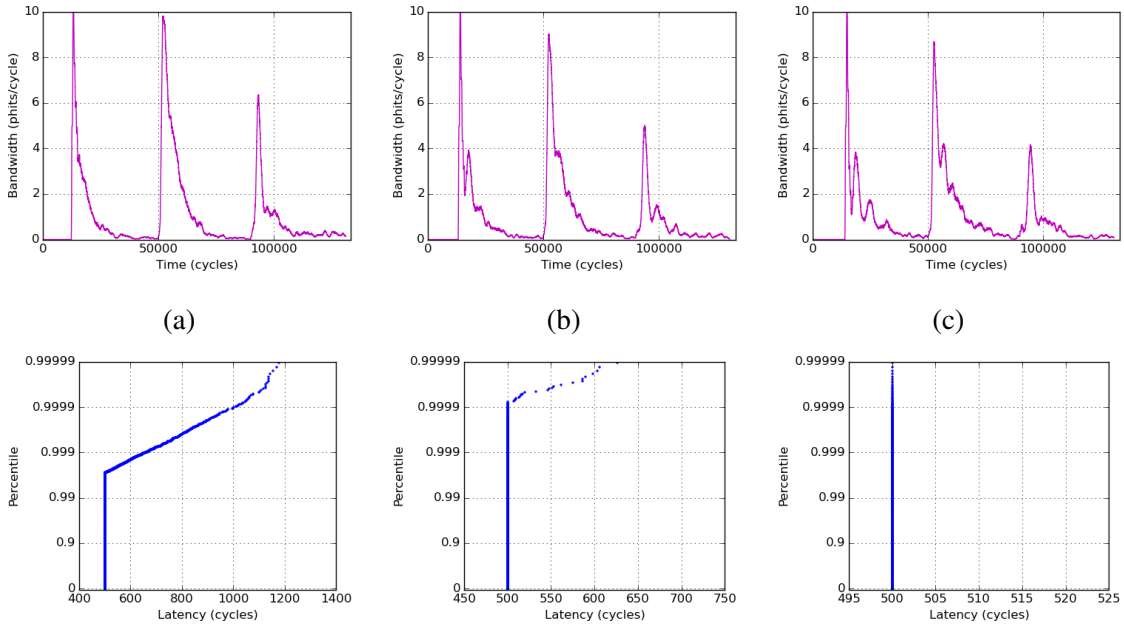


Figure 8.6: A comparison of the SE-TRE algorithm implemented with different token bucket sizes. (a) and (d) use 1500 tokens, (b) and (e) use 2000 tokens, and (c) and (f) use 2500 tokens. (a), (b), and (c) show bandwidth overheads and (d), (e), and (f) show worst-case latency percentiles.

Token Bucket	**99.99th% Latency**	**Bandwidth Overhead**	**Maximum Burst Size**
1500	1010 ns	159 Gbps	18.7M bytes
2000	500 ns	142 Gbps	25M bytes
2500	500 ns	140 Gbps	31M bytes

Table 8.2: SE-TRE performance when varying the token bucket size.

SE-TRE algorithm using a token bucket size of 1500, 2000, and 2500. The rest of the parameters remain the same as configured for the simulations in Section 8.2. As shown, larger token bucket implementations yield higher performance, both in terms of lower latency and lower bandwidth overhead, however, the bandwidth overhead is not significantly reduced. Using token buckets of size 1500 yields an individual burst size of 18.7k bytes and a aggregate burst size from 1000 processes of 18.7M bytes. Using token buckets of size 2500 yields an individual burst size of 31k bytes and a 1000 process aggregate burst of 31M bytes. Given the increase in performance and the small increase of burst size, it is most likely the best choice to find the smallest token bucket size that overcomes all latency overheads for an expected workload. For the setup in these simulations, token buckets of size 2500 perfectly overcome all latency overhead. Token buckets of size 2000 were chosen for the analysis because it is the minimal amount of tokens that yields perfect 99.99th percentile latency.

8.3.2 Greed and Generosity

When tuning the *Sender-Enforced* rate-control algorithms, the values for Th_{low}, F_t, F_r, Th_t, Th_r, and F_{mr} present several interesting trade-offs (see Table 5.1 for variable definitions). From the perspective of an individual token bucket, these variable express when requesting additional resources from peers should begin, how many resources should be asked for, when to give resources to requesting peers, and how many resources should be given. The *Token Exchange* protocol is a short term optimization and the *Rate Exchange* protocol is a long term optimization. As such, optimal tuning of these independent protocols exhibit much different behavior. From the many simulations performed during the search for optimal settings, it was learned that exchanging tokens should be done generously. The requester should start soon and the responder should give as many as possible. In contrast, it was learned that exchanging rate should be done very cautiously. The requester should still start early but the responder should be very careful not to give away any of its rate unless it is sure it will not need it soon. Specifically, the SE-TRE algorithm optimized for token buckets of size 2000 gives away tokens when its token bucket is only 35% full but waits until it is 90% full before it gives away any of its rate.

8.4 Summary

This chapter has shown that service-oriented distributed rate control is feasible and can operate with low overhead. The ability to achieve this efficiently relies on the service-oriented programming model of Sikker. The Sender-Enforced Token and Rate Exchange (SE-TRE) algorithm has been shown to provide extremely low latency and bandwidth overhead even when put under pressure by constantly changing distributed applications. The high efficiency of the SE-TRE algorithm empowers the use of token buckets of minimal size while also having enough resources for efficient distributed rate control. This algorithm is implemented as a reactionary state machine that only executes when a message is being sent. When combined with efficient access control by the NMU, SE-TRE is able to provide a secure and isolated computing environment for distributed applications incurring minimal latency and bandwidth overhead and zero CPU overhead.

Chapter 9

Optimizations and Improvements

This chapter discusses several options for optimization and improvement for Sikker and/or the NMU. None of these are required for Sikker to operate as designed, however they pose potential productivity and/or performance gains under certain environments. As a service-oriented computing model, Sikker can provide optimizations and improvements for many bottlenecks related to expressing application-specific communication patterns both in hardware and in software. A small subset of these issues is discussed in this chapter.

9.1 Contiguous Process Placement

The way in which the NOS (i.e., cluster coordinator) schedules services on the network has the potential of providing Sikker with an extraordinary optimization opportunity. The Sikker application model, as described in Section 4, defines the structure of a service to be a set of processes and domains. Each process of a service gets placed at a specific location on the network residing on a host. In efforts to remain agnostic to the underlying network infrastructure, Sikker places no constraints on the physical network addressing scheme of the network nor the placement of processes. The data structures in the NMU, as shown in Figure 6.4, hold a unique address value for every destination process the source process has been given access to.

Group	Base Address	Length	Start ID	End ID
1	9345	2,500	0	2,499
2	4392	1,500	2,500	3,999
3	101	4,000	4,000	7,999
4	23000	2,000	8,000	9,999

Table 9.1: A set of ranges specifying the physical network addresses of the processes in a Sikker service.

Many large-scale supercomputing fabrics use the network interface to perform fast address translation. In order to perform fast address translation for a particular job[1], the physical addresses of the various processes must be found using an optimized mechanism. For example, the network interfaces used in Cray supercomputing systems allow small jobs to have random process placement while large jobs are constrained to have a contiguous process placement in the physical network address space. Given a particular process identifier within the job, a contiguous process placement yields an address translation as a simple arithmetic operation (i.e., addition of base address with the node identifier). For example, if processes 0 through 99 are located at physical network addresses 5000 through 5099, network addresses are simply the process ID added to 5000.

In Sikker, if the processes of a service are located in a consecutive order the NMU is still going to hold each and every value uniquely in its data structures. An optimization to Sikker which overcomes this potentially wasteful behavior is to allow the processes of a service to be expressed as a set of contiguous regions in the physical address space. For example, a service with a total of 10,000 processes divided into 4 unequal groups could be expressed as shown in Table 9.1.

In the most dense scenario, a service could have all of its processes contiguously placed. With this optimization the number of values held in the *ProcessMap* (shown in Figure 6.4) would no longer be the number of processes but would become a single range specifier. If all services were scheduled in this fashion there would exist a large reduction in the state needed in the NMU. Assuming the same connectivity density values as described in the connectivity model described in Section 4.7, the reduction of state would be on the order of

[1] A "job" in a supercomputing environment is similar to a service in Sikker however jobs rarely intercommunicate.

2 to 3 orders of magnitude as the reduction is proportional to the number of processes per service. In terms of NMU performance, this not only reduces the memory size requirements but also significantly increases performance due to more data fitting in the cache. Scientific computing environments often schedule jobs much larger than the services found within data centers and cloud computing environments. This poses an even larger opportunity of optimization for Sikker as the number of processes per service can be many tens of thousands up to millions.

9.2 End-to-End Zero Copy

Zero copy is a technique that aims to send messages from one side of the network to another without making unneeded copies while transferring data between the application and the network interface [60, 61]. This technique was first introduced as a mechanism to remove the copying the kernel performed while interacting between the application and the network interface. Zero-copy was extended to work in OS-bypass environments such that the message being sent or received could be directly pulled from or placed into the application's memory space without explicit copies being made in the network interface. Zero copy has been successful in reducing latency and providing better memory bandwidth utilization.

One major problem with zero copy still remains, namely, the application has to construct the message before it is sent and the message has to be dissected when received on the other end. In nearly all cases, this entails a copy being made on both sides of the communication. Take for example a simple key-value store that holds a hash map data structure. A client has issued a *Get* operation in which it desires to retrieve the value mapped to one of the keys. A common thing for the key-value store to do is to access the hash map and copy out the key and value into a buffer that will become the message sent on the network. The standard zero copy mechanism allows no more copies to be made until the buffer arrives back to the client. Upon reception by the client, the client dissects the message into its parts containing the key and value. In order for the values to be placed inside the client's data structures another set of copy operations is needed. In summary, zero copy does allow for application-to-application message transfers without copying, but it doesn't solve the problem the two applications have that the network message is constructed in a single

buffer that must be constructed at the source and dissected at the destination [2].

Sikker explicitly defines domains as a boundary of application-specific permission domains. These domains might often be defined around API commands and/or data structures. An extension to Sikker that provides a true end-to-end zero copy solution is a system called *Send and Receive Templates*. The templates system has many similarities with gather-scatter lists found in common DMA engines, except that it works in connection with Sikker's service-oriented computing model and are specified on a per-domain basis. This domain-oriented template system provides a complete end-to-end zero-copy solution that places message data right where the application needs it and frees the processor from performing wasteful copying. The operation of send and receive templates will be described individually as follows.

9.2.1 Send Templates

Send Templates overcome much of the copying needed when an application is creating a message to be sent on the network based from data found within its internal data structures. To send a message the process first creates a *send template* that contains information about the layout of a message, but does not contain any message data. The template's first element is a count that specifies the number of remaining elements. All remaining elements describe a type of data, of which there are three types: immediate, fixed size buffer, and variable sized buffer. Immediate and fixed size buffer entries also specify a length while variable sized buffer entries do not. The process then creates a *send list* that contains the message's information. The list's first element is a state indicator that is used to flag the process of send completion and must be marked *active* before using. All remaining elements are immediate values, fixed size buffer pointers, and variable sized buffer pointers with their corresponding lengths. Before sending a message, the process writes the location of the send template and send list into its NMU send control register. To send the message, the process writes the desired destination service, process, and domain into the NMU send control register, then triggers a write-only "send" register. After checking permissions, the

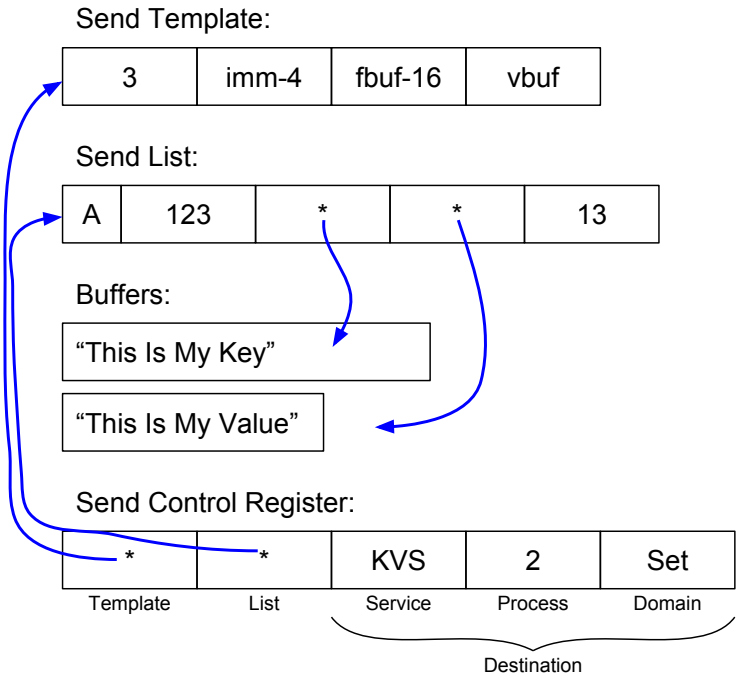
[2] Some applications actually desire the ability to copy data from an incoming message into their data structures in a particular way. For example, RAMCloud [62] contiguously appends data to a log data structure after the request has been processed.

NMU transfers the data directly from the process's memory space to the network. After the message has been received at the endpoint, the NMU informs the process of successful delivery via marking the send list as *inactive* and optionally via an interrupt.

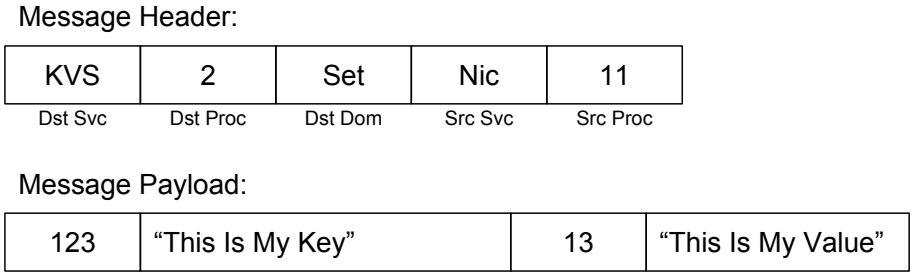
To illustrate the operation of send templates, let's assume process 11 from the *Nic* service is sending a key-value set request to the *KVS* key-value store service. The message will be sent to process 2 within the KVS service using the *Set* domain. The KVS service contains many logical tables of key-value maps identified by 32-bit identifiers (similar to the example in Section 4.1). Keys are fixed sized 16 byte character arrays and values are variable sized character arrays up to 96 bytes. The Nic service desires to map key "This Is My Key" to value "This Is My Value" in table 123. The process creates a send template with 3 type elements being a 4-byte immediate, a 16-byte fixed sized buffer, and a variable sized buffer. The process then creates an active send list that contains the 4-byte table identifier, memory address of the 16-byte key buffer, memory address of the 13-byte value buffer, and length of value buffer. The process writes the memory location of the send template and the send list into the send control register as well as the desired destination service, process, and domain. These data structures are shown in Figure 9.1a. The message is assembled and sent after the process triggers the send action by writing to the "send" register. The assembled message is shown in Figure 9.1b. After the message has been received at the destination, the NMU marks the send list as inactive.

9.2.2 Receive Templates

The receiving procedure of the NMU utilizes *receive templates* and *receive lists* which are similar to the send templates and send lists described in Section 9.2.1. The only structural difference is that receive templates contain a maximum size for variable sized buffers, whereas send templates do not. This is necessary because the process must allocate the buffers before they are used. Receive templates are specified on a per-domain basis and are held within a data structure called the *receive map* held by the NMU for each resident process. The receive map maps a single domain to a single receive template, however, for each receive template there is a fixed depth queue wherein references to receive lists are stored. The receive list queue size is allocated at setup time, but the entries are added and



(a) Data structures used during a message send with templates.



(b) The message assembled by the send template.

Figure 9.1: Send templates.

removed during runtime.

Before receiving messages, a process must register a receive template for each domain and one or more receive lists for each receive template. All receive lists are marked *ready* as they are linked with the receive list queue. When a message is received, the NMU inspects the message header for the destination service, process, and domain then it performs a lookup into the receive map for the receive template and next available receive list. Using the information in the receive template and receive list, the NMU places the data directly into receive list and buffers specified by the receive list. After having placed the message, the NMU marks the receive list as *used*, unlinks it from the receive list queue, and optionally interrupts the process. When another message is received in the same domain, the next available receive list will be used. It is the responsibility of the process to guarantee that enough receive lists are available for each receive template as messages arrive.

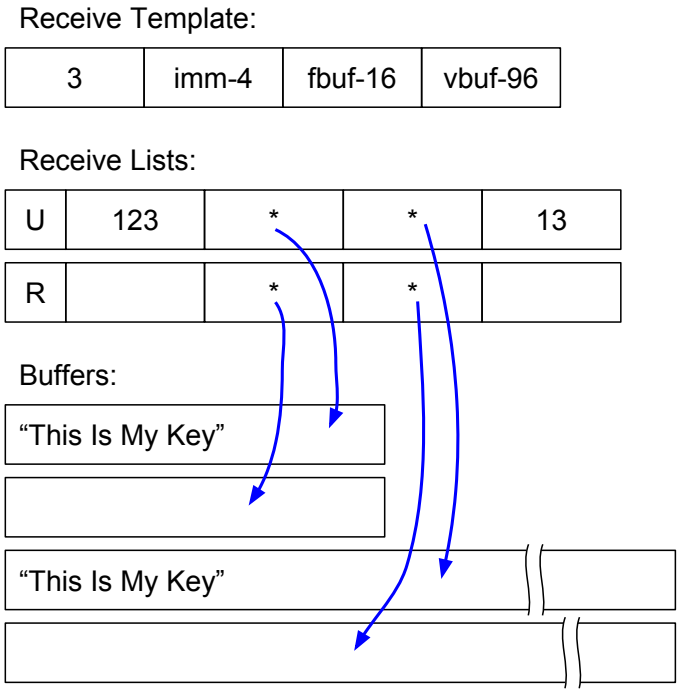


Figure 9.2: Data structures used during a message receive with templates.

To illustrate the operation of receive templates, continue from the example in Section 9.2.1 and view the perspective of the KVS service as it receives the message previously

sent by the Nic service. In preparation to receiving messages for domain *Set*, process 2 in the KVS service creates a receive template similar to the send template used by process 11 in the Nic service and registers it with the *Set* domain in the NMU. This process also creates two receive lists by creating two sets of buffers and registers the two receive lists with the NMU. When the message arrives at the NMU, the NMU inspects the message header (shown in Figure 9.1b) and determines that the destination is process 2 of the KVS service using the *Set* domain. A lookup into the process's receive map results in the designated receive template and the next available receive list. Using the receive template and receive list, the NMU places the message data in the receive list and corresponding buffers. The NMU then marks the receive list as used, removes it from the receive list queue, and optionally interrupts the process. Figure 9.2 shows the state of the receive data structures after having received the message sent by the Nic service.

9.3 Buffered Demux

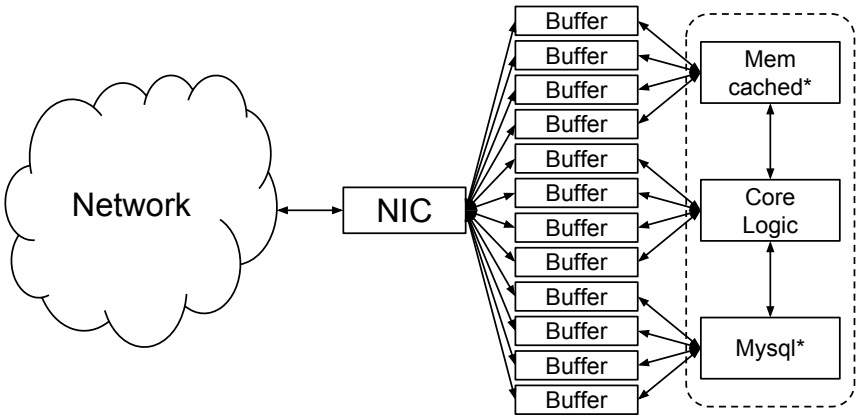
Environments where OS-bypass is enabled have shown to dramatically improve network performance and decrease CPU utilization required for the network communication of the application. However, OS-bypass presents a programming challenge in that it provides a single pipe into the network whereas traditional socket-oriented networking provides many multiplexed pipes into the network. The loss of multiple logical pipes into the network generally limits the types of workloads that can be run in these environments. Some application structures specifically require multiple logically independent network pipes to enable library modularity, efficient multithreading, and programmer productivity. For example, nearly all web-programming frameworks (e.g., NodeJS, Django, Ruby on Rails) allow the use of client libraries where the vendors of software packages also provide an interface library that decouples the package's functionality from the communication with the software. For instance, a NodeJS application developer can simply include the memcached client library into their code in order utilize a set of memcached servers on the system. The developer need not understand the memcached wire protocols as they only need to call the functions in the client library and the client library will handle the communication on the network to perform those functions. A requirement for this to work is that the client

library is able to establish independent communication flows as needed. These must be independent as there is often many client libraries running in parallel to fulfill the desired functionality of the application.

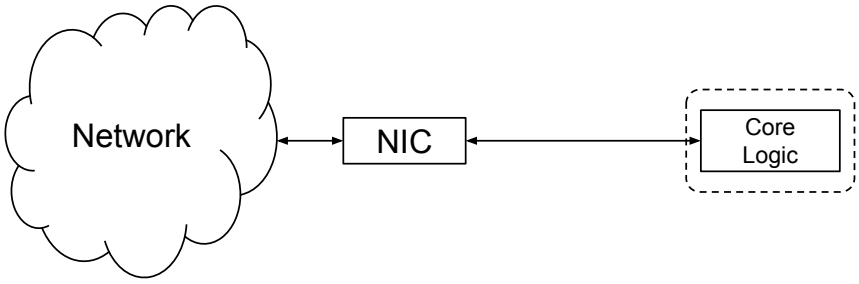
Traditional TCP/IP networking managed by the host's operating system allows many logical independent flows of communication through the use of sockets which are dedicated buffers for each flow. InfiniBand has a similar technique called queue pairs, except that queue pairs support OS-bypass. In both cases of traditional TCP/IP and InfiniBand queue pairs, the state overhead and buffering requirements to implement these independent flows of communication scales proportional to the number of independent flows. When matched with large applications, a large number of required buffers (shown in Figure 9.3a) can saturate the endpoint's resources or present significant performance bottlenecks to the application. The service-oriented communication nature of Sikker allows for a scalable alternative to per-flow buffering strategies. This mechanism is called a *Buffered Demux*.

In the most optimized case, the process interacts directly with the hardware registers of the network interface (i.e., the NMU). In this scenario there is no software layers between the process and the network as the process is able to directly control the network interface. For many services in Sikker this is possible and also productive since the network addressing model represents high-level application interactions, not network interactions. For example, a very optimized in-memory key-value store doesn't need multiple independent flows of communication because all it needs to do is repeatedly pull the next message off the network, process it, then send the response. This type of interface is shown in Figure 9.3b.

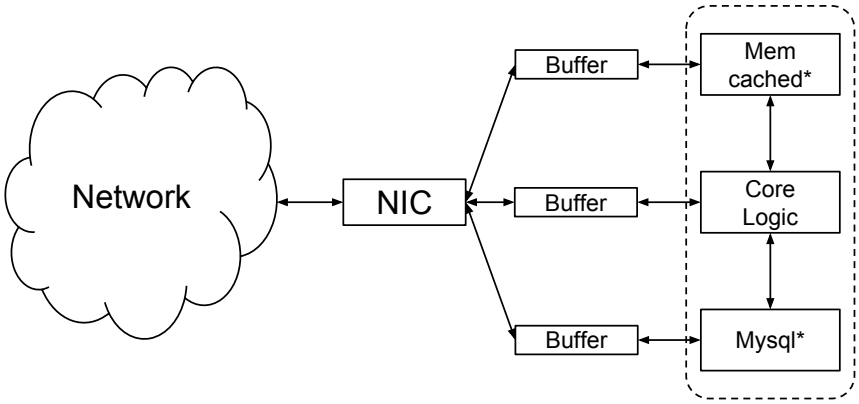
For the reasons discussed above, there are many scenarios where direct message processing isn't desirable or even feasible and messages must be buffered before they are processed. In these cases, there are often many application-specific ways in which buffering can be optimized. The Buffered Demux is a software layer that allows the developer to define a custom buffering strategy to enable the exact amount of flow independence needed for their particular application. To the programmer, the NMU receive interface looks like a FIFO. The Buffered Demux is made to connect to this FIFO interface and expose many FIFO-like interfaces, one for each programmer-defined independent flow category. An example of this is shown in Figure 9.3c. Programmers can define these flow categories



(a) Traditional per-flow buffering.



(b) Direct network access without buffering.



(c) Application-specific message buffering.

Figure 9.3: Three different schemes of network message buffering.

directly for their client libraries, for each individual permission domain, or for any other application-specific reason.

The Buffered Demux works by inspecting each message as it arrives and forwarding it to the proper FIFO based on an application-specific selection function. The main logic of the application defines this behavior as a function and passes it to the Buffered Demux during setup. This function takes a message and returns a buffer ID for the message and the Buffered Demux handles message forwarding. Each module of the application is given one or more buffer IDs in which it should look for messages to be processed. In this way, each module sees an independent receive path into the application. Of course this is not as efficient as complete ownership of the network interface, but it is a great performance optimization over per-flow buffering and still allows for full flexibility and modularity.

9.4 NMU Placement

The placement of the NMU within the system can have a significant effect on its performance. The network access latency of a network interface located on a common peripheral bus (e.g., PCI Express [63]) is bounded by the latency of the bus [64]. Moving the network interface closer to the processor by locating it on a coherent processor interconnect (e.g., Intel QuickPath Interconnect [65]) allows the network interface to be accessed faster and have quicker access to system memory. Locating the network interface on the same die as the processor (e.g., IBM PowerEN [66]) would yield even lower access latencies as the network interface can be accessed as fast as other on-chip devices. Some system designers have taken the opposite approach and have moved the network interface into the network by integrating it with a network router (e.g., Cray Gemini [12] and Cascade[11]). While this doesn't produce low network interface access latencies, the end-to-end latency is reduced because two network hops have been removed. These trade-offs must considered when determining the optimal placement of the NMU.

Chapter 10

Conclusion

While the computing requirements of modern data centers, cloud computing facilities, and supercomputers are beginning to converge, there exists an enormous divide between the performance capabilities of supercomputers and the flexibility and productivity of data centers and cloud computing facilities. Today, system designers and application developers are forced into making significant trade-offs between performance, security, and isolation. The rapid rise of service-oriented computing calls for a new distributed system architecture that is able to achieve supercomputer-like network performance while providing a secure environment for today's large-scale applications.

This dissertation has introduced a new distributed system architecture, called Sikker, with an explicit security and isolation model designed for large-scale distributed applications. This model formally defines services as a collection of processes and permissions domains as a distributed entity. In contrast to today's systems, Sikker applies and enforces permissions at the service level yielding many orders of magnitude of scalability over current process-oriented models while providing an inherent implementation of the principle of least privilege as well as source and destination authentication. The permission definitions in Sikker are designed to express the complete access-control needs of the application such that the application no longer needs to be concerned with protecting itself from the network. Sikker's service-oriented access-control methodology is an intuitive and effective alternative to network-derived access-control systems as it was derived directly from the interactions and structure of modern large-scale applications.

The Network Management Unit (NMU) was presented as a network interface controller architecture that implements the sender-enforced permissions scheme of Sikker, both in terms of access control and rate control. The NMU provides extremely low overhead network access to processes communicating on the network. The efficient design of the NMU allows it to perform permission checks in about 50 ns and supports hundreds of Gbps of bandwidth while imposing zero overhead on the CPU. A novel distributed dynamically-adapting rate-control algorithm was presented, called Sender-Enforced Token and Rate Exchange (SE-TRE), that yields extremely low overhead while providing precise control of service-to-service communication rate limits. Even when being stress tested with an application that dramatically changes its behavior every 40 µs, SE-TRE imposes zero latency overhead at the 99.99^{th} percentile latency, less than 0.3% bandwidth overhead, and zero overhead on the CPU.

Sikker and the NMU enable a new generation of distributed systems performing like supercomputers while operating with inherent service-oriented security and isolation. This new generation of computing supports large-scale multi-tenant computing platforms where system architects and application developers are able to access remote data quickly, spend less time writing tedious and error-pone security checks, and spend more time developing core application logic.

Bibliography

[1] Twitter. Finagle: A protocol-agnostic rpc system. [Online]. Available: https://blog.twitter.com/2011/finagle-a-protocol-agnostic-rpc-system

[2] Netflix. Netflix cloud architecture. [Online]. Available: http://www.slideshare.net/adrianco/netflix-velocity-conference-2011

[3] M. Heath. A journey into microservices: Dealing with complexity. [Online]. Available: https://sudo.hailoapp.com/services/2015/03/09/journey-into-a-microservice-world-part-3/

[4] T. Morgan. Broadcom goes end to end with 25g ethernet. [Online]. Available: http://www.nextplatform.com/2015/07/27/broadcom-goes-end-to-end-with-25g-ethernet/

[5] Amazon. High performance computing. [Online]. Available: https://aws.amazon.com/hpc/

[6] J. Jackson. (2014, July) Ibm aims to disrupt supercomputing market with cloud enticements. [Online]. Available: http://www.pcworld.com/article/2457580/ibm-aims-to-disrupt-supercomputing-market-with-cloud-enticements.html

[7] OASIS. Organization for the advancement of structured information standards. [Online]. Available: https://www.oasis-open.org/

[8] C. M. MacKenzie, K. Laskey, F. McCabe, P. F. Brown, R. Metz, and B. A. Hamilton, "Reference model for service oriented architecture 1.0," *OASIS Standard*, vol. 12, 2006.

[9] A. S. Tanenbaum and M. Van Steen, *Distributed systems*. Prentice-Hall, 2007.

[10] J. L. Hennessy and D. A. Patterson, *Computer architecture: a quantitative approach*. Elsevier, 2011.

[11] G. Faanes, A. Bataineh, D. Roweth, E. Froese, B. Alverson, T. Johnson, J. Kopnick, M. Higgins, and J. Reinhard, "Cray cascade: a scalable hpc system based on a dragonfly network," in *Proceedings of the International Conference on High Performance Computing, Networking, Storage and Analysis*. IEEE Computer Society Press, 2012, p. 103.

[12] R. Alverson, D. Roweth, and L. Kaplan, "The gemini system interconnect," in *2010 18th IEEE Symposium on High Performance Interconnects*. IEEE, 2010, pp. 83–87.

[13] D. Chen, N. A. Eisley, P. Heidelberger, R. M. Senger, Y. Sugawara, S. Kumar, V. Salapura, D. L. Satterfield, B. Steinmacher-Burow, and J. J. Parker, "The ibm blue gene/q interconnection network and message unit," in *High Performance Computing, Networking, Storage and Analysis (SC), 2011 International Conference for*. IEEE, 2011, pp. 1–10.

[14] B. Arimilli, R. Arimilli, V. Chung, S. Clark, W. Denzel, B. Drerup, T. Hoefler, J. Joyner, J. Lewis, J. Li, N. Ni, and R. Rajamony, "The percs high-performance interconnect," in *High Performance Interconnects (HOTI), 2010 IEEE 18th Annual Symposium on*. IEEE, 2010, pp. 75–82.

[15] Mellanox Technologies. (2015) Infiniband performance. [Online]. Available: http://www.mellanox.com

[16] VMware. Nsx. [Online]. Available: http://www.vmware.com/products/nsx

[17] B. Pfaff, J. Pettit, K. Amidon, M. Casado, T. Koponen, and S. Shenker, "Extending networking into the virtualization layer." in *Hotnets*, 2009.

[18] S. M. Rumble, D. Ongaro, R. Stutsman, M. Rosenblum, and J. K. Ousterhout, "It's time for low latency." in *HotOS*, vol. 13, 2011, pp. 11–11.

[19] P. J. Braam, "The lustre storage architecture," 2004.

[20] Infiniband Trade Association, "Infiniband architecture specification," 2000.

[21] O. Sarood, A. Langer, A. Gupta, and L. Kale, "Maximizing throughput of overprovisioned hpc data centers under a strict power budget," in *Proceedings of the International Conference for High Performance Computing, Networking, Storage and Analysis*. IEEE Press, 2014, pp. 807–818.

[22] M. C. Kurt and G. Agrawal, "Disc: a domain-interaction based programming model with support for heterogeneous execution," in *Proceedings of the International Conference for High Performance Computing, Networking, Storage and Analysis*. IEEE Press, 2014, pp. 869–880.

[23] H. Liu and B. He, "Reciprocal resource fairness: Towards cooperative multiple-resource fair sharing in iaas clouds," in *Proceedings of the International Conference for High Performance Computing, Networking, Storage and Analysis*. IEEE Press, 2014, pp. 970–981.

[24] X. Yang, Z. Zhou, S. Wallace, Z. Lan, W. Tang, S. Coghlan, and M. E. Papka, "Integrating dynamic pricing of electricity into energy aware scheduling for hpc systems," in *Proceedings of the International Conference on High Performance Computing, Networking, Storage and Analysis*. ACM, 2013, p. 60.

[25] A. D. Breslow, A. Tiwari, M. Schulz, L. Carrington, L. Tang, and J. Mars, "Enabling fair pricing on hpc systems with node sharing," in *Proceedings of the International Conference on High Performance Computing, Networking, Storage and Analysis*. ACM, 2013, p. 37.

[26] Amazon. Amazon web services (aws). [Online]. Available: http://aws.amazon.com

[27] Microsoft. Azure: Microsoft's cloud platform. [Online]. Available: http://azure.microsoft.com

[28] Google. Google cloud platform. [Online]. Available: http://cloud.google.com

[29] Salesforce. Heroku. [Online]. Available: http://www.heroku.com

[30] Joyent. High-performance cloud computing. [Online]. Available: http://www.joyent.com

[31] V. Rajaravivarma, "Virtual local area network technology and applications," in *Southeastern Symposium on System Theory*. IEEE Computer Society, 1997, pp. 49–49.

[32] M. Mahalingam *et al.*, "Vxlan: A framework for overlaying virtualized layer 2 networks over layer 3 networks (draft-mahalingam-dutt-dcops-vxlan-02. txt)," *VXLAN: A Framework for Overlaying Virtualized Layer*, vol. 2, 2012.

[33] M. Sridharan, K. Duda, I. Ganga, A. Greenberg, G. Lin, M. Pearson, P. Thaler, C. Tumuluri, N. Venkataramiah, and Y. Wang, "Nvgre: Network virtualization using generic routing encapsulation," *IETF draft*, 2011.

[34] OpenStack Foundation. Openstack neutron. [Online]. Available: https://wiki.openstack.org/wiki/Neutron

[35] M. Kallahalla, M. Uysal, R. Swaminathan, D. E. Lowell, M. Wray, T. Christian, N. Edwards, C. I. Dalton, and F. Gittler, "Softudc: A software-based data center for utility computing," *Computer*, no. 11, pp. 38–46, 2004.

[36] C. Guo, G. Lu, H. J. Wang, S. Yang, C. Kong, P. Sun, W. Wu, and Y. Zhang, "Secondnet: a data center network virtualization architecture with bandwidth guarantees," in *Proceedings of the 6th International Conference*. ACM, 2010, p. 15.

[37] P. Soares, J. Santos, N. Tolia, D. Guedes, and Y. Turner, "Gatekeeper: Distributed rate control for virtualized datacenters," *Technical Report HP-2010-151, HP Labs*, 2010.

[38] A. Shieh, S. Kandula, A. G. Greenberg, C. Kim, and B. Saha, "Sharing the data center network." in *NSDI*, 2011.

[39] H. Ballani, P. Costa, T. Karagiannis, and A. Rowstron, "Towards predictable datacenter networks," *ACM SIGCOMM Computer Communication Review*, vol. 41, no. 4, pp. 242–253, 2011.

[40] L. Popa, G. Kumar, M. Chowdhury, A. Krishnamurthy, S. Ratnasamy, and I. Stoica, "Faircloud: sharing the network in cloud computing," in *Proceedings of the ACM SIGCOMM 2012 conference on Applications, technologies, architectures, and protocols for computer communication*. ACM, 2012, pp. 187–198.

[41] F. Xu, F. Liu, H. Jin, and A. Vasilakos, "Managing performance overhead of virtual machines in cloud computing: A survey, state of the art, and future directions," *Proceedings of the IEEE*, vol. 102, no. 1, pp. 11–31, Jan 2014.

[42] J. Ciancutti. (2010, December) 5 lessons we've learned using aws. [Online]. Available: http://techblog.netflix.com/2010/12/5-lessons-weve-learned-using-aws.html

[43] VMware. Network i/o latency on vsphere 5, performance study. [Online]. Available: http://www.vmware.com/files/pdf/techpaper/network-io-latency-perf-vsphere5.pdf

[44] A. Verma, L. Pedrosa, M. Korupolu, D. Oppenheimer, E. Tune, and J. Wilkes, "Large-scale cluster management at google with borg," in *Proceedings of the Tenth European Conference on Computer Systems*. ACM, 2015, p. 18.

[45] J. H. Saltzer, "Protection and the control of information sharing in multics," *Communications of the ACM*, vol. 17, no. 7, pp. 388–402, 1974.

[46] Amazon. Simple storage service (s3). [Online]. Available: https://aws.amazon.com/s3/

[47] F. Chang, J. Dean, S. Ghemawat, W. C. Hsieh, D. A. Wallach, M. Burrows, T. Chandra, A. Fikes, and R. E. Gruber, "Bigtable: A distributed storage system for structured data," *ACM Transactions on Computer Systems (TOCS)*, vol. 26, no. 2, p. 4, 2008.

[48] Microsoft. Azure search - search-as-a-service for web and mobile app development. [Online]. Available: https://azure.microsoft.com/en-us/services/search/

[49] B. Hindman, A. Konwinski, M. Zaharia, A. Ghodsi, A. D. Joseph, R. H. Katz, S. Shenker, and I. Stoica, "Mesos: A platform for fine-grained resource sharing in the data center." in *NSDI*, vol. 11, 2011, pp. 22–22.

[50] M. Schwarzkopf, A. Konwinski, M. Abd-El-Malek, and J. Wilkes, "Omega: flexible, scalable schedulers for large compute clusters," in *Proceedings of the 8th ACM European Conference on Computer Systems*. ACM, 2013, pp. 351–364.

[51] P. J. Denning, "Fault tolerant operating systems," *ACM Computing Surveys (CSUR)*, vol. 8, no. 4, pp. 359–389, 1976.

[52] J. Turner, "New directions in communications(or which way to the information age?)," *IEEE communications Magazine*, vol. 24, no. 10, pp. 8–15, 1986.

[53] W. J. Dally and B. P. Towles, *Principles and practices of interconnection networks*. Elsevier, 2004.

[54] R. Bryant and O. David Richard, *Computer systems: a programmer's perspective*. Prentice Hall, 2003.

[55] A. M. Tenenbaum, *Data structures using C*. Pearson Education India, 1990.

[56] N. Muralimanohar, R. Balasubramonian, and N. P. Jouppi, "Cacti 6.0: A tool to model large caches," *HP Laboratories*, 2009.

[57] P. Rosenfeld, E. Cooper-Balis, and B. Jacob, "Dramsim2: A cycle accurate memory system simulator," *Computer Architecture Letters*, vol. 10, no. 1, pp. 16–19, 2011.

[58] T. Benson, A. Anand, A. Akella, and M. Zhang, "Understanding data center traffic characteristics," *ACM SIGCOMM Computer Communication Review*, vol. 40, no. 1, pp. 92–99, 2010.

[59] A. Roy, H. Zeng, J. Bagga, G. Porter, and A. C. Snoeren, "Inside the social network's (datacenter) network," in *Proceedings of the 2015 ACM Conference on Special Interest Group on Data Communication*. ACM, 2015, pp. 123–137.

[60] P. Shivam, P. Wyckoff, and D. Panda, "Emp: zero-copy os-bypass nic-driven gigabit ethernet message passing," in *Supercomputing, ACM/IEEE 2001 Conference*. IEEE, 2001, pp. 49–49.

[61] M. N. Thadani and Y. A. Khalidi, *An efficient zero-copy I/O framework for UNIX*. Citeseer, 1995.

[62] J. Ousterhout, A. Gopalan, A. Gupta, A. Kejriwal, C. Lee, B. Montazeri, D. Ongaro, S. J. Park, H. Qin, M. Rosenblum *et al.*, "The ramcloud storage system," *ACM Transactions on Computer Systems (TOCS)*, vol. 33, no. 3, p. 7, 2015.

[63] D. Anderson, T. Shanley, and R. Budruk, *PCI express system architecture*. Addison-Wesley Professional, 2004.

[64] M. Flajslik and M. Rosenblum, "Network interface design for low latency request-response protocols," in *Presented as part of the 2013 USENIX Annual Technical Conference (USENIX ATC 13)*, 2013, pp. 333–346.

[65] N. Kurd, J. Douglas, P. Mosalikanti, and R. Kumar, "Next generation intel® microarchitecture (nehalem) clocking architecture," in *VLSI Circuits, 2008 IEEE Symposium on*. IEEE, 2008, pp. 62–63.

[66] J. D. Brown, S. Woodward, B. M. Bass, and C. L. Johnson, "Ibm power edge of network processor: A wire-speed system on a chip," *Micro, IEEE*, vol. 31, no. 2, pp. 76–85, 2011.